



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Configuring SIP Trunking Using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 8.1 – Issue 1.0**

### **Abstract**

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Contact Center VoIP Inbound SIP Trunk Service and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500 v2 Release 8.1 Essential Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, SIP, digital, and analog endpoints.

The Verizon Business IP Contact Center VoIP Inbound offer referenced within these Application Notes enables a business to receive inbound toll free calls via standards-based SIP trunks, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

**IP Office Release 8.1 has not been independently certified by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.**

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted in the Avaya Solution & Interoperability Test Lab, utilizing a Verizon Business Private IP (PIP) circuit connection to the Verizon Business IP Contact Center service.

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## Introduction

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Contact Center (Verizon Business IPCC) VoIP Inbound Service and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500 v2 Release 8.1 Essential Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, SIP, digital, and analog endpoints.

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Customers using Avaya IP Office with the Verizon Business IPCC service are able to receive inbound toll-free calls from the PSTN via the SIP protocol. The converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI.

Verizon Business IPCC service can be delivered to the customer premise via either a Private IP (PIP) or Internet Dedicated Access (IDA) IP network terminations. Although the configuration documented in these Application Notes used Verizon's IPCC service terminated via a PIP network connection, the solution validated in this document applies also to IP Contact Center services delivered via IDA service terminations.

For more information on the Verizon Business IPCC service, visit <http://www.verizonbusiness.com/Products/communications/contact-center/>

## General Test Approach and Test Results

The Avaya IP Office location was connected to the Verizon Business IPCC service, as depicted in **Figure 1**. Avaya IP Office was configured to use the commercially available IP Toll Free VoIP Inbound solution. This allowed Avaya IP Office to receive inbound toll-free calls from the PSTN via the SIP protocol.

### 1.1. Interoperability Compliance Testing

The testing included executing the test cases detailed in Reference [VZ-Test-Plan], which contains the Verizon IPCC Interoperability Lab Test Plan. To summarize, the testing included the following successful SIP trunk interoperability compliance testing:

- SIP OPTIONS monitoring of the health of the SIP trunk was verified. Verizon Business IPCC and IP Office can both monitor health using SIP OPTIONS.
- Proper recovery from induced failure conditions such as IP Office reboots, and long and short duration IP network outages between Verizon and IP Office
- Incoming calls from the PSTN were routed to the toll-free numbers assigned by Verizon Business to the Avaya IP Office location. These incoming calls arrived via the SIP Line configured in Section 5.4 and were answered by Avaya H.323 telephones, Avaya SIP

telephones, Avaya digital telephones, analog telephones, Avaya IP Office Softphone, and Avaya IP Office Voicemail Pro.

- Proper disconnect when either party hangs up an active call
- Proper disconnect when the PSTN caller abandons (i.e., hangs up) a toll-free call before the IP Office party has answered.
- Proper SIP 486 response and busy tone heard by the caller when a PSTN user calls a toll-free number directed to a busy IP Office user, an IP Office user with Do-not-disturb active, or an IP Office user that is logged out (i.e., assuming no redirection is configured for these conditions). Similarly, busy tone is heard when a PSTN user calls a toll-free number whose “SIP URI Max Calls per Channel” has been reached (see Section 5.4). Similarly, busy tone is heard when a PSTN user calls a toll-free number directed to a hunt group whose queue is “full” (i.e. if no redirection is configured for hunt group busy conditions, see Section 5.5.4).
- Proper termination of an inbound IP Toll Free call left in a ringing state for a relatively long duration
- The display of caller ID on display-equipped Avaya IP Office telephones was verified. The IP Office capability to use the caller ID received from Verizon to look up and display a name from a configurable directory was also exercised successfully.
- Privacy requests for inbound toll-free calls from the PSTN were verified. That is, when privacy is requested by a PSTN caller (e.g., dialing \*67 from a mobile phone), the inbound toll-free call can be successfully completed to an IP Office telephone user while presenting a “WITHHELD” or anonymous display to an IP Office user (i.e., rather than the caller’s telephone number).
- Inbound toll-free long holding time call stability (See Section 2.2. Although long SIP sessions are not refreshed, the media paths remain connected.)
- IP Office complies with RFC 3261 SIP Methods
- IP Office can use UDP for SIP transport with Verizon IPCC
- IP Office can use a configured UDP port for SIP signaling with Verizon IPCC
- IP Office accepts the full SIP headers sent by Verizon IPCC
- IP Office sends SIP 180 RINGING (no SDP in 180) for inbound calls and ring back tone is heard by the caller.
- IP Office does not return a SIP 302 to Verizon IPCC
- Telephony features such as hold and resume, transfer of toll-free calls to other IP Office users, and conference of toll-free calls.
- Incoming voice calls using the G.729A and G.711 ULAW codecs, and proper protocol procedures related to media
- DTMF transmission using RFC 2833. Successful IP Office Voicemail Pro menu navigation for incoming toll-free calls. Successful use of IP Office Mobile Call Control, where DTMF sequences can be performed remotely using the SIP Line.
- Incoming toll-free calls directed to the Hunt Groups configured in Section 5.5.4 were verified. Incoming calls could be queued, queued with priority, and be answered by members of the hunt group as members become available.

- Outgoing calls from the Avaya IP Office location to the PSTN were routed via a SIP Line to the Verizon Business IP Trunk service described in reference [VZBIPT-IPO81]. As detailed in reference [VZBIPT-IPO81], these outgoing PSTN calls can be originated from Avaya H.323 telephones, Avaya SIP telephones, Avaya digital telephones, analog endpoints, and Avaya IP Office Softphone. The display of caller ID on display-equipped PSTN telephones was verified. In the context of inbound toll-free calls using Verizon Business IPCC, inbound toll-free calls arriving via the SIP Line configured in Section 5.4 could be forwarded or twinned out the Verizon Business IP Trunk service SIP Line. Inbound toll-free calls from the Verizon Business IPCC SIP Line could also trigger mobile callback calls that use the Verizon Business IP Trunk service SIP Line.
- Call Forwarding of Verizon toll-free calls to PSTN destinations via the Verizon Business IP Trunk service documented in reference [VZBIPT-IPO81], presenting true calling party information to the mobile phone. See Section 2.2 for additional information.
- Mobile twinning of Verizon toll-free calls to a mobile phone via the Verizon Business IP Trunk service documented in reference [VZBIPT-IPO81], presenting true calling party information to the mobile phone.
- Inbound mobile call control, mapping a Verizon toll-free number to the mobile call control feature, as shown in Section 5.6. That is, a configured mobile twinning PSTN caller may dial a Verizon toll-free number, receive dial tone from IP Office, and place calls using IP Office, as if the user were calling from their IP Office telephone. Calls to the same toll-free number from calling numbers that are not configured in IP Office for mobile call control receive busy tone.
- DiffServ markings for Avaya IP Office SIP signaling and RTP media consistent with network capability for optimum routing of VoIP

## 1.2. Test Results

Interoperability testing of the sample configuration was completed with successful results as described in Section 2.1. The following observations may be noteworthy:

1. The Verizon Business IPCC service does not support fax.
2. When a call is put on hold by an IP Office user, there is no indication sent to Verizon via SIP messaging. This is transparent to the users on the call.
3. Although the Verizon Business IPCC service supports transfer using the SIP REFER method and IP Office supports sending REFER, IP Office will not send REFER to Verizon in the verified configuration.
4. The SIP protocol allows sessions to be refreshed for calls that remain active for some time. In the tested configuration, neither Verizon nor IP Office send SIP re-INVITE or UPDATE messages to refresh a session. In the tested configuration, this is transparent to the users that are party to the call in that the media paths remain established.
5. When a user on the PSTN hangs up an active call, Verizon Business IPCC will send an INVITE with SDP containing 0.0.0.0 before sending the BYE to clear the call. IP Office

processes the INVITE with SDP containing 0.0.0.0 as a request to hold the call, and then processes the BYE to disconnect the call. If the IP Office user is still listening after the PSTN user hangs up, the IP Office user may very briefly hear music on hold from IP Office before the BYE is processed and the call appearance is idled.

6. IP Office does not support the receipt of an initial INVITE that does not contain SDP. Therefore, IP Office does not support the Verizon Business IPCC “enhanced transfer” service, which sends an initial INVITE without SDP to the transfer-to site of an enhanced transfer.

## **1.3. Support**

### **1.3.1. Avaya**

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

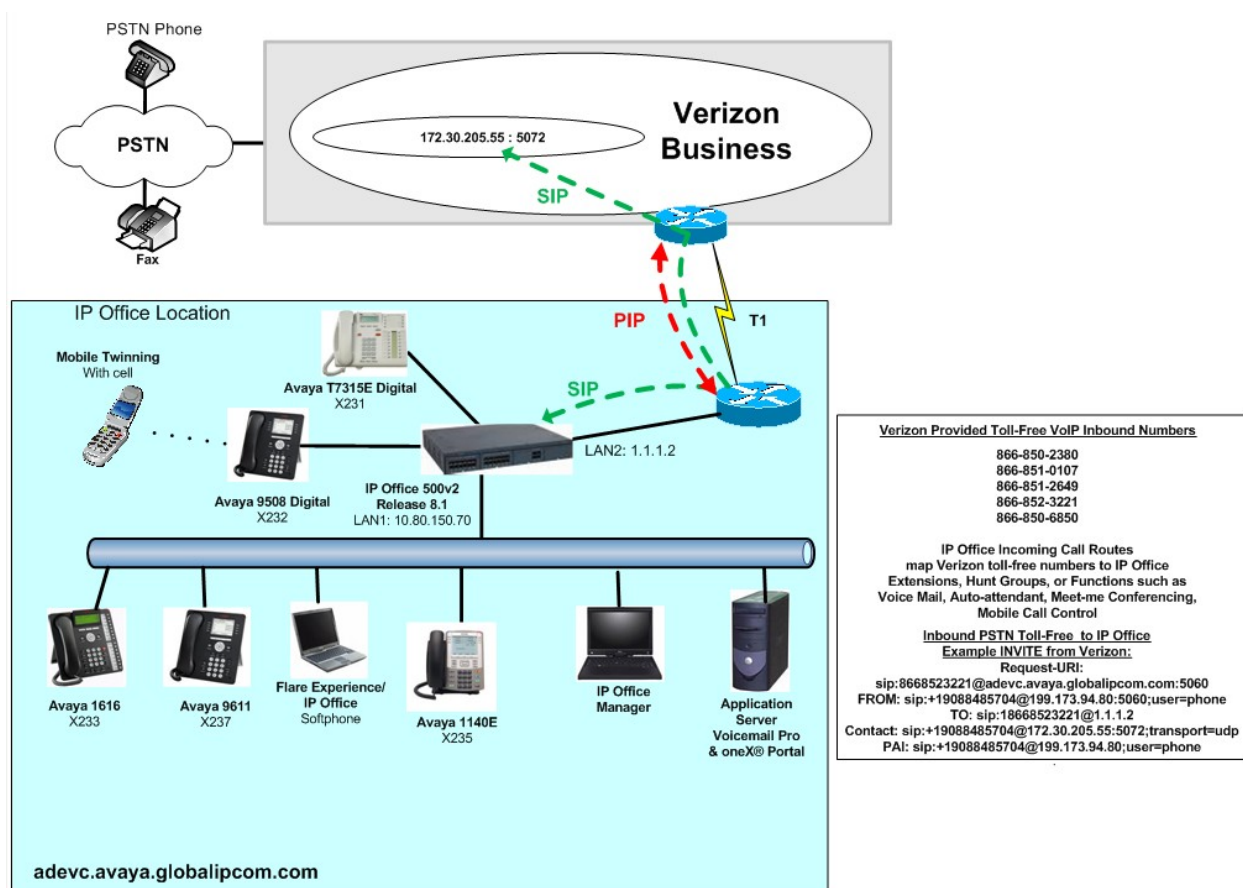
### **1.3.2. Verizon**

For technical support on Verizon Business IPCC service, visit online support at <http://www.verizonbusiness.com/us/customer/>

## Reference Configuration

**Figure 1** illustrates an example Avaya IP Office solution connected to the Verizon Business IPCC SIP Trunk service. The Avaya equipment is located on a private IP subnet. An enterprise edge router provides access to the Verizon Business network via a Verizon Business T1 circuit. This circuit is provisioned for the Verizon Business Private IP (PIP) service. Reference [VZBIPT-IPO81] illustrates IP Office interoperability with the Verizon Business IP Trunk service. In the verification testing associated with these Application Notes, both the Verizon IP Trunk service and the Verizon Business IPCC service were accessible via the same PIP connection.

In the sample configuration, IP Office receives traffic from the Verizon Business IPCC service on port 5060 and sends traffic to port 5072, using UDP for network transport, as required by the Verizon Business IPCC service. Verizon provided five toll-free numbers associated with the IP Contact Center service. These toll-free numbers were mapped to IP Office destinations via Incoming Call Routes as summarized in **Table 1**. The Avaya IP Office environment domain known to Verizon was *adevc.avaya.globalipcom.com*.



**Figure 1: Avaya IP Office with Verizon IP Contact Center Service**

**Table 1** shows an example mapping of toll-free numbers to IP Office users, groups, or functions. The associated IP Office configuration is shown in Section 5. Since the quantity of toll-free numbers was limited in the test configuration relative to the desired test coverage, the same toll-free number was routed to different IP Office destinations (i.e., IP Office configuration changes were made to the Incoming Call Route destination as needed between successive tests).

<b>Verizon Provided Toll-Free Number</b>	<b>Configured Avaya IP Office Destination(s)</b>	<b>Notes</b>
866-851-0107	x235	Avaya 1140E
866-850-2380	x232, x241, x234	Digital Telephone with Mobile Twinning and Mobile Call Control permission. Also used to test analog telephone and Avaya Flare Experience capabilities.
866-851-2649	x233, x237	Avaya 1616 Telephone, Avaya 9611 Telephone
866-850-6850	Voicemail Collect on Voicemail Pro	Allow external callers to access voice mail toll-free
866-850-6850	Inbound Mobile Call Control	Allow toll-free calls from pre-configured twinning numbers to access mobile call control
866-850-6850	Conference Bridge on Voicemail Pro	Allow external callers to access conference bridge toll-free
866-852-3221 (any caller)	“401 Sales” Hunt Group (with default priority)	Hunt Group with queuing
866-852-3221 (specific callers)	“400 Overdue Account” Hunt Group	Show IP Office destination selection based on caller ID
866-852-3221 (specific priority callers)	“401 Sales” Hunt Group (with High Priority)	Show IP Office priority queuing based on caller ID

**Table 1: Example Verizon Toll Free Number to IP Office Destination Mappings**



## Equipment and Software Validated

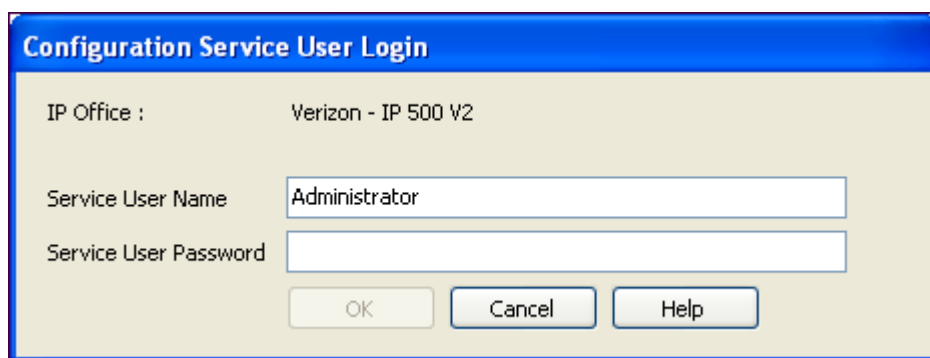
Table 2 shows the equipment and software used in the sample configuration.

Equipment	Software
Avaya IP Office 500 v2	Release 8.1 (65)
Avaya IP Office Manager	Release 10.1 (65)
Avaya Application Server	8.1.20-3
Avaya 2500 Analog Telephone	N/A
Avaya 9508 Digital Telephone	N/A
Avaya T7315E Digital Telephone	N/A
Avaya 1616 IP Telephone (H.323)	Release 1.302B
Avaya 9611 IP Telephone (H.323)	Release 6.2209
Avaya 1140E SIP	04.03.12
Avaya IP Office Softphone	Release 3.2.3.20 64770
Avaya Flare Experience	1.1.0.5

Table 2: Equipment and Software Tested

## Avaya IP Office Configuration

IP Office is configured via the IP Office Manager program. For more information on IP Office Manager, consult reference [2]. From the IP Office Manager PC, select **Start → Programs → IP Office → Manager** to launch the Manager application. A screen that includes the following in the center may be displayed:



The screenshot shows a dialog box titled "Configuration Service User Login". It contains the following fields and controls:

- IP Office :** Verizon - IP 500 V2
- Service User Name:** Administrator
- Service User Password:** (empty text box)
- Buttons:** OK, Cancel, Help


























Log in with the appropriate configuration credentials. The appearance of the IP Office Manager can be customized using the **View** menu (not shown). In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side.

## 1.4. Physical, Network, and Security Configuration

This section describes attributes of the sample configuration, but is not meant to be prescriptive. Consult reference [1] for more information on the topics in this section.

In the sample configuration, looking at the IP Office 500 from left to right, the first module is a TCM 8 Digital Station Module. This module supports BCM / Norstar T-Series and M-Series telephones. The second module is a COMBO6210/ATM4 module. This module 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.711, G729A and G.723 with 64ms echo cancellation. G.722 is supported by IP Office Release 8.0 and higher. The “Combo” card 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. Referring to **Figure 1**, the Avaya T7315E telephone with extension 231 is connected to port 1 of the TCM8 module, and the Avaya 9508 telephone with extension 232 is connected to port 1 of the “Combo” card. The analog extension or fax machine is connected to the “Combo” card on port 7

The following screen shows the modules in the IP Office used in the sample configuration. To access such a screen, select **Control Unit** in the Navigation pane. The modules appear in the Group pane. 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.

IP Offices	Control Unit	 IP 500 V2																												
<div><ul style="list-style-type: none"><li> BOOTP (6)</li><li> Operator (3)</li><li> Verizon<ul style="list-style-type: none"><li> System (1)</li><li> Line (6)</li><li> Control Unit (3)</li><li> Extension (23)</li><li> User (24)</li><li> HuntGroup (3)</li><li> Short Code (67)</li><li> Service (0)</li><li> RAS (1)</li><li> Incoming Call Route (4)</li><li> ... (2)</li></ul></li></ul></div>	<table><tr><th>Dev No.</th><th>Dev Type</th><th>Version</th></tr><tr><td> 1</td><td>IP 500 V2</td><td>8.1 (65)</td></tr><tr><td> 2</td><td>TCM8</td><td>8.1 (65)</td></tr><tr><td> 3</td><td>COMBO6210/ATM4</td><td>8.1 (65)</td></tr></table>	Dev No.	Dev Type	Version	 1	IP 500 V2	8.1 (65)	 2	TCM8	8.1 (65)	 3	COMBO6210/ATM4	8.1 (65)	<table><tr><th colspan="2">Unit</th></tr><tr><td>Device Number</td><td>1</td></tr><tr><td>Unit Type</td><td>IP 500 V2</td></tr><tr><td>Version</td><td>8.1 (65)</td></tr><tr><td>Serial Number</td><td>00e007058e33</td></tr><tr><td>Unit IP Address</td><td>10.80.150.70</td></tr><tr><td>Interconnect Number</td><td>0</td></tr><tr><td>Module Number</td><td>Control Unit</td></tr></table>	Unit		Device Number	1	Unit Type	IP 500 V2	Version	8.1 (65)	Serial Number	00e007058e33	Unit IP Address	10.80.150.70	Interconnect Number	0	Module Number	Control Unit
Dev No.	Dev Type	Version																												
 1	IP 500 V2	8.1 (65)																												
 2	TCM8	8.1 (65)																												
 3	COMBO6210/ATM4	8.1 (65)																												
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Device Number	1																													
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Version	8.1 (65)																													
Serial Number	00e007058e33																													
Unit IP Address	10.80.150.70																													
Interconnect Number	0																													
Module Number	Control Unit																													

In the sample configuration, the IP Office LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is 10.80.150.1. To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New**. To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the Details pane with the relevant route using **Destination** LAN1.

The screenshot shows the 'IP Route' configuration window. The title bar is blue with the IP address '10.64.0.0'. The window has a tab labeled 'IP Route'. The configuration fields are as follows:

IP Address	10 . 64 . 0 . 0
IP Mask	255 . 255 . 0 . 0
Gateway IP Address	10 . 80 . 150 . 1
Destination	LAN1
Metric	0
<input type="checkbox"/> Proxy ARP	

The IP Office LAN2 port is physically connected to the service provider and has a default gateway of 1.1.1.1. Right-click **IP Route** from the Navigation pane, and select **New** to add another route. The following screen shows the Details pane with the relevant route using **Destination** LAN2

The screenshot shows the 'IP Route' configuration window. The title bar is blue with the IP address '0.0.0.0'. The window has a tab labeled 'IP Route'. The configuration fields are as follows:

IP Address	0 . 0 . 0 . 0
IP Mask	0 . 0 . 0 . 0
Gateway IP Address	1 . 1 . 1 . 1
Destination	LAN2
Metric	0
<input type="checkbox"/> Proxy ARP	

To facilitate use of Avaya IP Office Softphone, https was enabled in the sample configuration. To check whether https is enabled, navigate to **File → Advanced → Security Settings**. A screen such as the following is presented. Log in with the appropriate security credentials.

The screenshot shows the 'Security Service User Login' dialog box. The title bar is blue with the text 'Security Service User Login'. The dialog box contains the following fields and buttons:

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

After logging in, select **Services** from the Navigation pane and **HTTP** from the Group pane. In the Details pane, verify the **Service Security Level** is configured as intended, as shown below.

Security Settings	Services (6)	Service : HTTP																								
<ul style="list-style-type: none"> <li>Security <ul style="list-style-type: none"> <li>General</li> <li>System (1)</li> <li>Services (6)</li> <li>Rights Groups (15)</li> <li>Service Users (8)</li> </ul> </li> </ul>	<table border="1"> <thead> <tr> <th>Name</th> <th>Security Level</th> </tr> </thead> <tbody> <tr> <td>Configuration</td> <td>Unsecure Only</td> </tr> <tr> <td>Security Administrati...</td> <td>Unsecure Only</td> </tr> <tr> <td>System Status Interf...</td> <td>Unsecure Only</td> </tr> <tr> <td>Enhanced TSPI</td> <td>Unsecure Only</td> </tr> <tr> <td>HTTP</td> <td>Unsecure + Secure</td> </tr> <tr> <td>Web Services</td> <td>Secure, Medium</td> </tr> </tbody> </table>	Name	Security Level	Configuration	Unsecure Only	Security Administrati...	Unsecure Only	System Status Interf...	Unsecure Only	Enhanced TSPI	Unsecure Only	HTTP	Unsecure + Secure	Web Services	Secure, Medium	<table border="1"> <thead> <tr> <th colspan="2">Service Details</th> </tr> </thead> <tbody> <tr> <td>Name</td> <td>HTTP</td> </tr> <tr> <td>Host System</td> <td>Verizon</td> </tr> <tr> <td>Service Port</td> <td>80, 443</td> </tr> <tr> <td>Service Security Level</td> <td>Unsecure + Secure</td> </tr> </tbody> </table>	Service Details		Name	HTTP	Host System	Verizon	Service Port	80, 443	Service Security Level	Unsecure + Secure
Name	Security Level																									
Configuration	Unsecure Only																									
Security Administrati...	Unsecure Only																									
System Status Interf...	Unsecure Only																									
Enhanced TSPI	Unsecure Only																									
HTTP	Unsecure + Secure																									
Web Services	Secure, Medium																									
Service Details																										
Name	HTTP																									
Host System	Verizon																									
Service Port	80, 443																									
Service Security Level	Unsecure + Secure																									

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

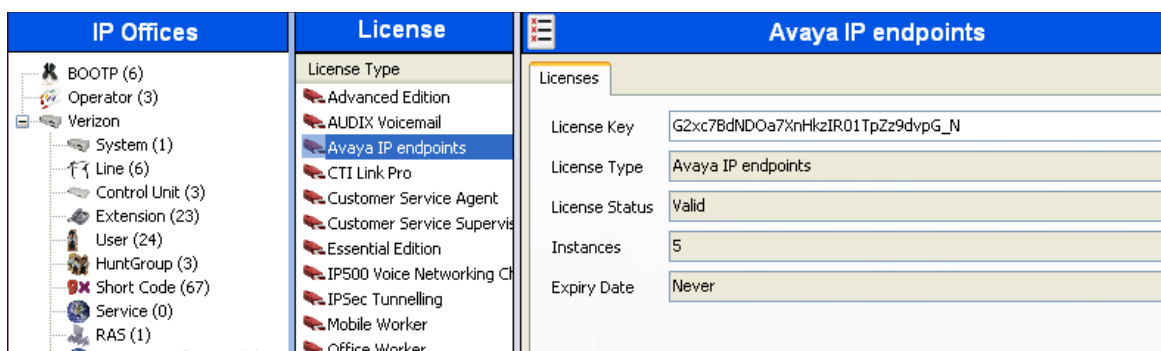
## 1.5. Licensing

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

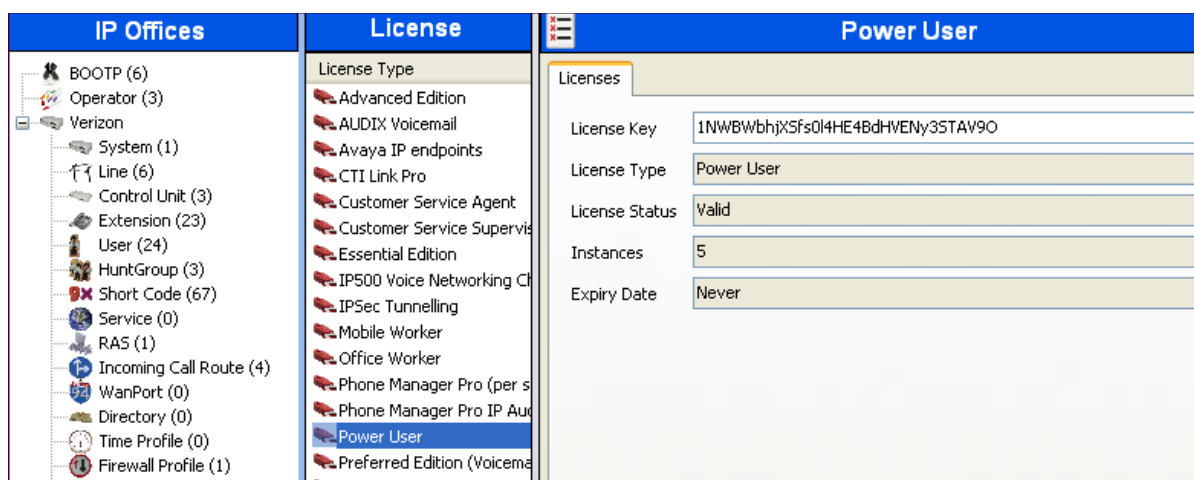
To verify that there is a SIP Trunk Channels License with sufficient capacity, click **License** in the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm a valid license with sufficient “Instances” (trunk channels) in the Details pane.

IP Offices	License	SIP Trunk Channels																																	
<ul style="list-style-type: none"> <li>BOOTP (6)</li> <li>Operator (3)</li> <li>Verizon</li> <li>System (1)</li> <li>Line (6)</li> <li>Control Unit (3)</li> <li>Extension (23)</li> <li>User (24)</li> <li>HuntGroup (3)</li> <li>Short Code (67)</li> <li>Service (0)</li> <li>RAS (1)</li> <li>Incoming Call Route (4)</li> <li>WanPort (0)</li> <li>Directory (0)</li> <li>Time Profile (0)</li> <li>Firewall Profile (1)</li> <li>IP Route (5)</li> <li>Account Code (0)</li> <li>License (22)</li> <li>Tunnel (0)</li> <li>User Rights (8)</li> </ul>	<table border="1"> <thead> <tr> <th>License Type</th> </tr> </thead> <tbody> <tr><td>Advanced Edition</td></tr> <tr><td>AUDIX Voicemail</td></tr> <tr><td>Avaya IP endpoints</td></tr> <tr><td>CTI Link Pro</td></tr> <tr><td>Customer Service Agent</td></tr> <tr><td>Customer Service Supervis</td></tr> <tr><td>Essential Edition</td></tr> <tr><td>IP500 Voice Networking Cl</td></tr> <tr><td>IPSec Tunneling</td></tr> <tr><td>Mobile Worker</td></tr> <tr><td>Office Worker</td></tr> <tr><td>Phone Manager Pro (per s</td></tr> <tr><td>Phone Manager Pro IP Au</td></tr> <tr><td>Power User</td></tr> <tr><td>Preferred Edition (Voicema</td></tr> <tr><td>Receptionist</td></tr> <tr><td>SIP Trunk Channels</td></tr> <tr><td>Software Upgrade 255</td></tr> <tr><td>Teleworker</td></tr> <tr><td>VMPro Networked Messagi</td></tr> </tbody> </table>	License Type	Advanced Edition	AUDIX Voicemail	Avaya IP endpoints	CTI Link Pro	Customer Service Agent	Customer Service Supervis	Essential Edition	IP500 Voice Networking Cl	IPSec Tunneling	Mobile Worker	Office Worker	Phone Manager Pro (per s	Phone Manager Pro IP Au	Power User	Preferred Edition (Voicema	Receptionist	SIP Trunk Channels	Software Upgrade 255	Teleworker	VMPro Networked Messagi	<table border="1"> <thead> <tr> <th colspan="2">Licenses</th> </tr> </thead> <tbody> <tr> <td>License Key</td> <td>t@HYRX6RAvHOIp8FoCkpxU3K3_Lww4rX</td> </tr> <tr> <td>License Type</td> <td>SIP Trunk Channels</td> </tr> <tr> <td>License Status</td> <td>Valid</td> </tr> <tr> <td>Instances</td> <td>5</td> </tr> <tr> <td>Expiry Date</td> <td>Never</td> </tr> </tbody> </table>	Licenses		License Key	t@HYRX6RAvHOIp8FoCkpxU3K3_Lww4rX	License Type	SIP Trunk Channels	License Status	Valid	Instances	5	Expiry Date	Never
License Type																																			
Advanced Edition																																			
AUDIX Voicemail																																			
Avaya IP endpoints																																			
CTI Link Pro																																			
Customer Service Agent																																			
Customer Service Supervis																																			
Essential Edition																																			
IP500 Voice Networking Cl																																			
IPSec Tunneling																																			
Mobile Worker																																			
Office Worker																																			
Phone Manager Pro (per s																																			
Phone Manager Pro IP Au																																			
Power User																																			
Preferred Edition (Voicema																																			
Receptionist																																			
SIP Trunk Channels																																			
Software Upgrade 255																																			
Teleworker																																			
VMPro Networked Messagi																																			
Licenses																																			
License Key	t@HYRX6RAvHOIp8FoCkpxU3K3_Lww4rX																																		
License Type	SIP Trunk Channels																																		
License Status	Valid																																		
Instances	5																																		
Expiry Date	Never																																		

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.



A similar process can be used to check the license status for other desired features. For example, the following screen shows the availability of a valid license for **Power User** features. In the sample configuration, the user with extension 234 will be configured as a “Power User” and will be capable of using the Avaya IP Office Softphone.

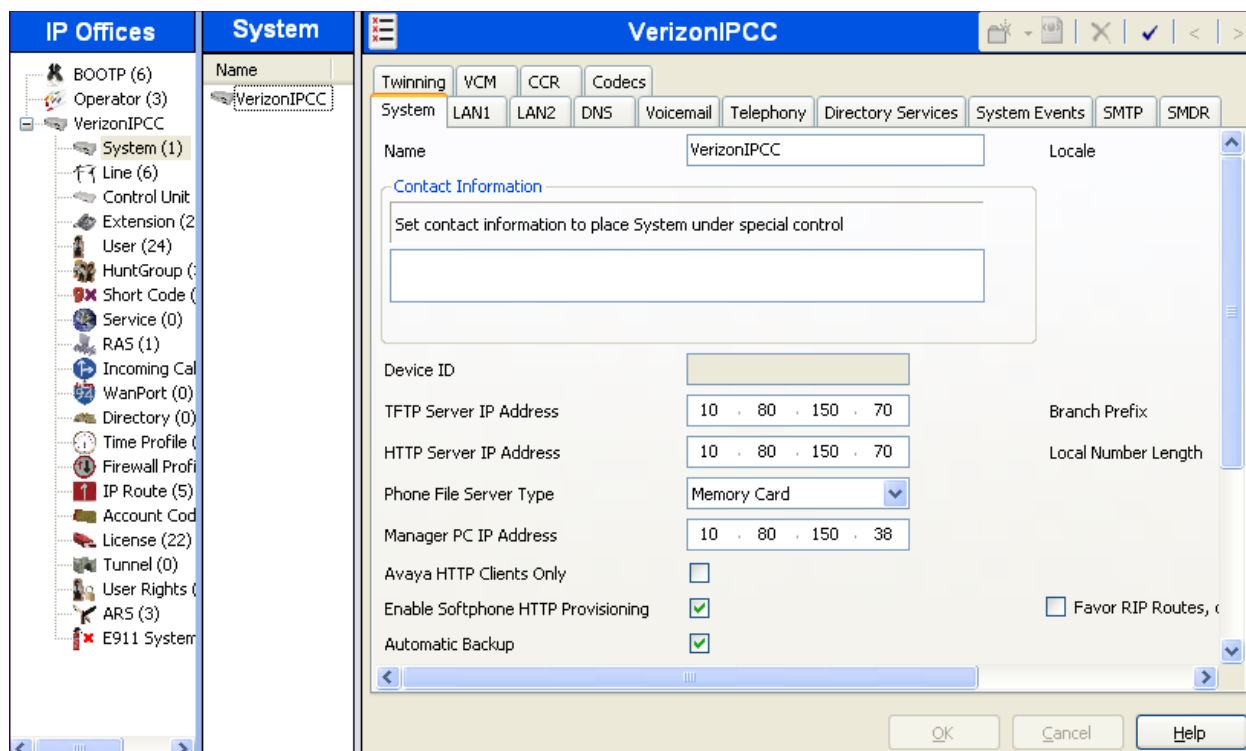


## 1.6. System Settings

This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings. The subsection order corresponds to a left to right navigation of the tabs in the Details pane for System settings.

### 1.6.1. 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, “VerizonIPCC” is used as the name. The **Enable SoftPhone HTTP Provisioning** box is checked to facilitate Avaya IP Office Softphone usage.



### 1.6.2. LAN 1 Settings

The IP500/IP500 V2 control units have 2 RJ45 Ethernet ports, physically marked as LAN and WAN. Within the system configuration, the physical LAN port is LAN1, the physical WAN port is LAN2.

In the sample configuration, LAN1 was used to connect the 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. As shown in **Figure 1**, the IP Address of the IP Office is 10.80.150.70. Other parameters on this screen may be set according to customer requirements. In the example screen, the **DHCP Mode** was set to “Server” to allow IP Office to facilitate provisioning for the IP Telephones in the sample configuration.

The screenshot shows the Avaya configuration interface with the following settings:

- System Tabs:** SMDR, Twinning, VCM, CCR, Codecs, System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP.
- LAN Settings Tabs:** LAN Settings, VoIP, Network Topology, DHCP Pools, SIP Registrar.
- IP Address:** 10 . 80 . 150 . 70
- IP Mask:** 255 . 255 . 255 . 0
- Primary Trans. IP Address:** 0 . 0 . 0 . 0
- RIP Mode:** None
- Enable NAT:** ☐
- Number Of DHCP IP Addresses:** 200
- DHCP Mode:**
  - ☒ Server
  - ☐ Client
  - ☐ Dialin
  - ☐ Disabled
- Advanced:**

Select the **VoIP** tab as shown in the following screen. 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 and 9600-Series Telephones used in the sample configuration. The **SIP Registrar Enable** box is checked to allow Avaya 1140E, Avaya Flare Experience, and Avaya IP Office Softphone usage. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Verizon Business

**RTP Port Number:** 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 (i.e., 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, that being the range defined by the Internet Assigned Numbers Authority (IANA) for dynamic usage. **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.

System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events SMTP SMDR Twinning VCM CCR Codecs

LAN Settings VoIP Network Topology DHCP Pools SIP Registrar

☒ H.323 Gatekeeper Enable  
☒ SIP Trunks Enable  
☒ SIP Registrar Enable

☐ H.323 Auto-create Extn  
☐ H.323 Auto-create User  
☐ H.323 Remote Extn Enable  
☒ Enable RTCP Monitoring On Port 5005

RTP Port Number Range  
 Port Range (Minimum) 49152  
 Port Range (Maximum) 53246

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

DHCP Settings  
 Primary Site Specific Option Number (SSON) 176  
 Secondary Site Specific Option Number (SSON) 242  
 VLAN Not Present  
 1100 Voice VLAN Site Specific Option Number (SSON) 232  
 1100 Voice VLAN IDs

RTP Keepalives  
 Scope Disabled Periodic timeout 0

Select the **Network Topology** tab as shown in the following screen. In the sample configuration, the default settings were used.

Twining VCM CCR Codecs

System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events SMTP SMDR

LAN Settings VoIP Network Topology DHCP Pools SIP Registrar

Network Topology Discovery

STUN Server IP Address 0 . 0 . 0 . 0 STUN Port 3478

Firewall/NAT Type Unknown

Binding Refresh Time (seconds) 120

Public IP Address 0 . 0 . 0 . 0

Public Port UDP 0

Run STUN Cancel

☐ Run STUN on startup



If using IP Office as a DHCP server and DHCP Server mode has been selected from the **LAN1** → **Lan Settings** Tab, click the **DHCP Pools** tab. Although beyond the intended scope of these Application Notes, the following screen is shown as a simple example.

Start Address	Subnet Mask	Default Router	Pool Size
10.80.150.72	255.255.255.0	10.80.150.1	15

Optionally, select the **SIP Registrar** tab. The following screen shows the settings used in the sample configuration. The **Domain Name** has been set to the customer premises equipment domain “avayalab.com”. If the **Domain Name** is left at the default blank setting, SIP registrations may use the IP Office LAN 1 IP Address. All other parameters shown are default values.

### 1.6.3. LAN 2 Settings

In the sample configuration, LAN2 was used to connect the IP Office to the Verizon network. To view or configure the **IP Address** of LAN2, select the **LAN2** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP Address of the IP Office, known to Verizon, is 1.1.1.2. **DHCP Mode** is set to **Disabled** since DHCP is unnecessary towards Verizon. Other parameters on this screen may be set according to customer requirements.

The screenshot displays a configuration window with a top navigation bar containing tabs: Twinning, VCM, CCR, Codecs, System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, and SMDR. Below this, a sub-navigation bar includes LAN Settings, VoIP, and Network Topology. The main configuration area is titled 'LAN Settings' and contains the following fields:

- IP Address: 1 . 1 . 1 . 2
- IP Mask: 255 . 255 . 255 . 0
- Primary Trans. IP Address: 0 . 0 . 0 . 0
- Firewall Profile: <None> (dropdown menu)
- RIP Mode: None (dropdown menu)
- ☒ Enable NAT
- Number Of DHCP IP Addresses: 1 (spinner)
- DHCP Mode: Server, Client, Dialin, Disabled (radio buttons, with 'Disabled' selected)
- Advanced (button)

Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** and **SIP Registrar Enable** boxes are unchecked since IP telephones will not be registering on this link. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Verizon Business.

If desired, the **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media paths from Verizon Business to IP Office. The defaults are used here. See Section 5.3.2 for more information on these RTP settings.

If desired, 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. In the sample configuration shown below, IP Office will mark SIP signaling with a value associated with “Assured Forwarding” using DSCP decimal 34 (**SIG DSCP** parameter). IP Office will mark the RTP media with a value associated with “Expedited Forwarding” using DSCP decimal 46 (**DSCP** parameter). This screen enables flexibility in IP Office DiffServ markings (RFC 2474) to allow alignment with network routing policies, which are outside the scope of these Application Notes. Other parameters on this screen may be set according to customer requirements.

System	LAN1	LAN2	DNS	Voicemail	Telephony	Directory Services	System Events	SMTP	SMDR	Twinning	VCM	CCR	Codecs
LAN Settings			VoIP		Network Topology								

☐ H.323 Gatekeeper Enable  
☒ SIP Trunks Enable  
☐ SIP Registrar Enable

☐ H.323 Auto-create Extn  
☐ H.323 Auto-create User  
☐ H.323 Remote Extn Enable  
☐ Enable RTP Monitoring On Port 5005

**RTP Port Number Range**  
 Port Range (Minimum) 49152  
 Port Range (Maximum) 53246

**DiffServ Settings**  

B8	DSCP(Hex)	FC	DSCP Mask (Hex)	88	SIG DSCP (Hex)
46	DSCP	63	DSCP Mask	34	SIG DSCP

**DHCP Settings**  
 Primary Site Specific Option Number (SSON) 176  
 Secondary Site Specific Option Number (SSON) 242  
 VLAN Not Present  
 1100 Voice VLAN Site Specific Option Number (SSON) 232  
 1100 Voice VLAN IDs

**RTP Keepalives**  
 Scope Disabled Periodic timeout 3

Select the **Network Topology** tab as shown in the following screen. In the sample configuration, the default settings were used and the **Use Network Topology Info** in the **SIP Line** was set to “None” in Section 5.4.2. The **Binding Refresh Time (seconds)** can still be used to lower the SIP OPTIONS timing from the default of 300 seconds. During the testing, the Binding Refresh Time was varied (e.g., 60 seconds, 120 seconds to test SIP OPTIONS timing).

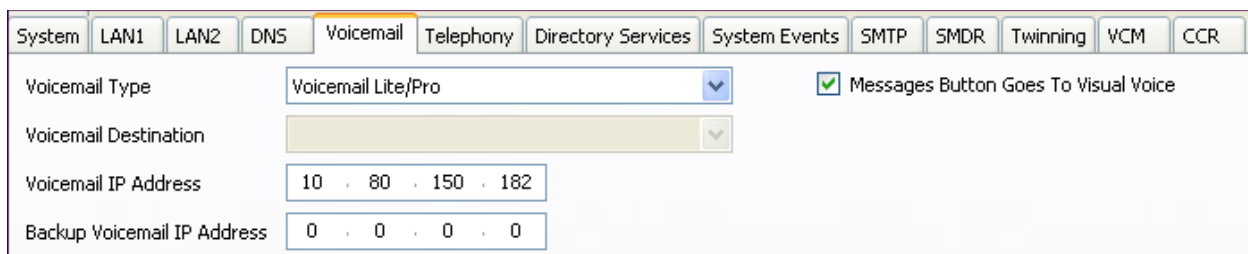
System	LAN1	LAN2	DNS	Voicemail	Telephony	Directory Services	System Events	SMTP	SMDR	Twinning	VCM	CCR	Codecs
LAN Settings			VoIP		Network Topology								

**Network Topology Discovery**  
 STUN Server IP Address 0 . 0 . 0 . 0 STUN Port 3478  
 Firewall/NAT Type Unknown  
 Binding Refresh Time (seconds) 120  
 Public IP Address 0 . 0 . 0 . 0  
 Public Port UDP 0  
 Run STUN Cancel  
☐ Run STUN on startup

Since **SIP Registrar Enable** was unchecked on the VOIP tab, the SIP Registrar Tab is not present on LAN2.

#### 1.6.4. Voicemail

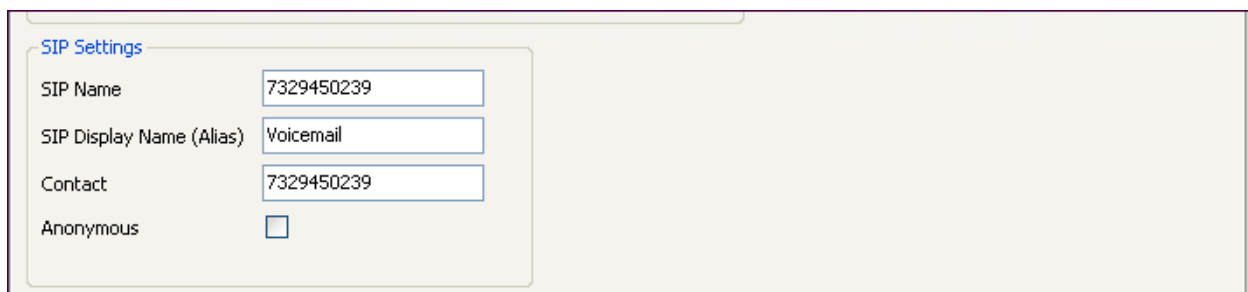
To view or change voicemail settings, select the **Voicemail** tab as shown in the following screen. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. The **Voicemail Type** in the sample configuration is “Voicemail Lite/Pro”. Other Voicemail types may be used. The **Voicemail IP Address** in the sample configuration is 10.80.150.182, the IP Address of the PC running the Voicemail Pro software, as shown in **Figure 1**.



The screenshot shows a configuration window with multiple tabs at the top: System, LAN1, LAN2, DNS, Voicemail (selected), Telephony, Directory Services, System Events, SMTP, SMDR, Twinning, VCM, and CCR. The Voicemail tab contains the following settings:

- Voicemail Type:** A dropdown menu set to "Voicemail Lite/Pro".
- Voicemail Destination:** A dropdown menu.
- Voicemail IP Address:** A text field containing "10 . 80 . 150 . 182".
- Backup Voicemail IP Address:** A text field containing "0 . 0 . 0 . 0".
- Messages Button Goes To Visual Voice:** A checkbox that is checked.

As described in [VZBIPT-IPO81], the “Callback” application of Avaya Voicemail Pro was used to allow Voicemail Pro to call out via the SIP Line to Verizon Business IP Trunk service when a message is left in a voice mailbox.



The screenshot shows a "SIP Settings" configuration window with the following fields:

- SIP Name:** A text field containing "7329450239".
- SIP Display Name (Alias):** A text field containing "Voicemail".
- Contact:** A text field containing "7329450239".
- Anonymous:** An unchecked checkbox.

#### 1.6.5. 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 sample configuration and are not intended to be prescriptive. In the sample configuration, the **Inhibit Off-Switch Forward/Transfer** box is unchecked so that call forwarding and call transfer to PSTN destinations via the Verizon Business IP Trunk service can be tested. That is, a call can arrive to IP Office via the Verizon Business IPCC service, and be forwarded or transferred back to the PSTN with the outbound leg of the call using the Verizon Business IP Trunk service. The **Companding Law** parameters are set to “ULAW” as is typical in North American locales. Other parameters on this screen may be set according to customer requirements.

The **Default Name Priority** is a new field in IP Office Release 8 and 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. With the option to “Favor Directory”, 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 will be illustrated further in the context of the SIP Line to Verizon. A user’s personal directory example is shown in Section 5.5.1.

The screenshot shows the 'Telephony' configuration window with the 'Telephony' tab selected. The 'Tones & Music' sub-tab is also visible. The window is divided into several sections:

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

To view or change settings associated with tones or music, select the **Telephony** tab and **Tones & Music** sub-tab as shown in the following screen. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. In the sample configuration, music on hold was provided via a WAV file from IP Office. For conferences, entry tone and exit tones are provided.

The screenshot shows the 'Telephony' configuration window with the 'Tones & Music' tab selected. The 'Telephony' tab is highlighted in the top navigation bar. The 'Tones & Music' sub-tab is also selected. The settings are as follows:

- Conferencing Tone: Entry & Exit Tones
- Disconnect Tone: Default
- Tone Plan: Tone Plan 1
- CLI Type: (empty dropdown)
- Local Dial Tone: ☒
- Local Busy Tone: ☐
- Beep on listen: ☒
- GSM Silence Suppression: ☐
- Busy Tone Detection: ☐
  - Mode: System Frequency
- Hold Music:
  - System Source: WAV File

### 1.6.6. System Twinning Configuration

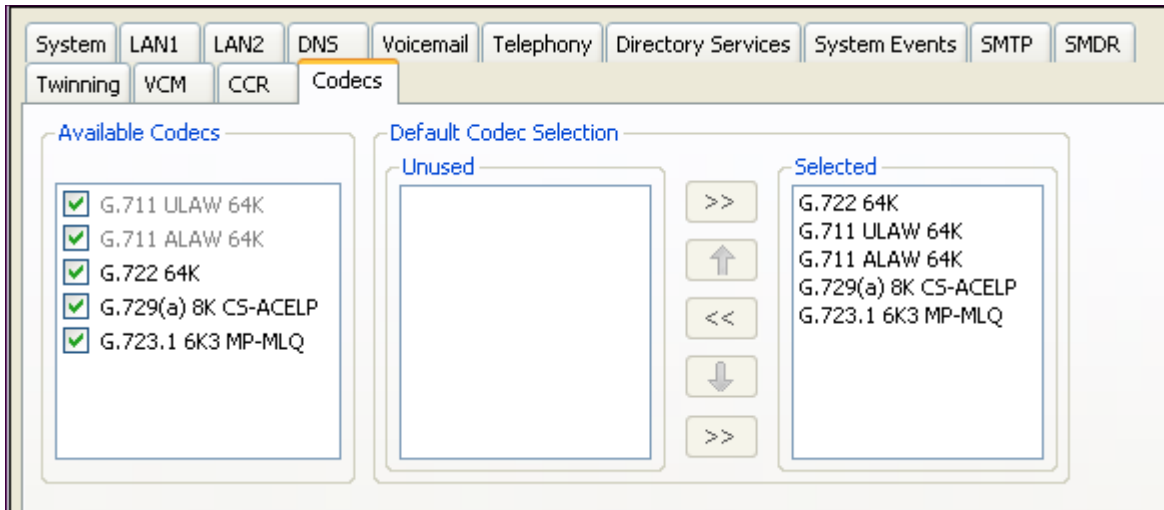
To view or change Twinning settings, select the **Twining** tab as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank. With this configuration, and related configuration of “Diversion header” on the SIP Line to Verizon Business IP Trunk service (Section 5.4.1 of reference [VZBIPT-IPO81]), the true identity of the caller can be presented to the twinning destination (e.g., a user’s mobile phone) when a call is twinned out via the Verizon Business IP Trunk service. That is, a call can arrive via a Verizon Business IPCC service toll-free number, and be twinned out to a mobile telephone using the Verizon Business IP Trunk service, with the twinned mobile phone seeing the identity of the caller that dialed the Verizon toll-free number.

The screenshot shows the 'Twining' configuration window. The 'Twining' tab is highlighted in the top navigation bar. The settings are as follows:

- ☐ Send original calling party information for Mobile Twinning
- Calling party information for Mobile Twinning: (empty text box)

### 1.6.7. System Codecs Configuration

The **System → Codecs** tab was introduced in IP Office Release 8. On the left, observe the list of **Available Codecs**. In the example screen below, which is not intended to be prescriptive, the box next to each codec is checked, making all the codecs available in other screens where codec configuration may be performed (such as the SIP Line in Section 5.4). The **Default Codec Selection** area enables the codec preference order to be configured on a system-wide basis, using the up, down, left, and right arrows. By default, all IP (SIP and H.323) lines and extensions will assume the system default codec selection, unless configured otherwise for the specific line or extension.



## 1.7. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 8.1. The Appendix in Section 11 contains an example SIP Trunk template file that was generated from the SIP Line configured in this section.

To add a new SIP Line, right click on **Line** in the Navigation pane, and select **New → SIP Line**. A new Line Number will be assigned automatically. To edit an existing SIP Line, click **Line** in the Navigation pane, and the SIP Line to be configured in the Group pane.

### 1.7.1. SIP Line - SIP Line Tab

The **SIP Line** tab in the Details pane is shown below for Line Number 18, used for the Verizon Business IP Contact Center service. The **ITSP Domain Name** is configured to the IP Office LAN2 address (1.1.1.2). By default, the **In Service** and **Check OOS** boxes are checked.

The **Call Routing Method** can retain the default “Request URI” setting, or may be changed to “To Header”, to match Incoming Call Routes based on the contents of the “To Header”. In the sample configuration, the default “Request URI” setting was used. As can be observed in the sample INVITE header contents in **Figure 1** and Section 7.2, the Request-URI and the To header do not necessarily contain the same number. In the tested configuration, the Request-URI contained the toll-free number, and the “To” header contained 1 followed by the toll-free number.

In the sample configuration, the IP Office **Country Code** was set to 1. The “From” and “PAI” headers received from Verizon for calls from U.S. PSTN numbers contain “+1” before the calling PSTN number. By configuring the IP Office **Country Code** to 1, the caller ID display presented to IP Office users will be the PSTN number without any codes or prefixes. For example, a call from 3035387006 would display 3035387006. If the **Country Code** does not match the value following the “+” from Verizon, the IP Office user display would show the contents of the **International Prefix** field, followed by the value following the “+”, followed by the PSTN number. For example, if the Country Code parameter were left blank, the IP Office

user would see a display such as “0013035387006”. Aside from display implications, if the **Country Code** is not configured, other patterns may also fail to match as expected, such as a match on the **Incoming CLI** field of the Incoming Call Route. See Section 5.7.3 for configuration of incoming call routing based on the calling number.

The area of the screen entitled **REFER Support** was introduced in IP Office Release 6.1. In the following screen, the default automatic determination of REFER support is shown. Alternatively, the default can be overridden with “Never” to explicitly disable use of REFER, or “Always” to explicitly enable use of REFER. The **Association Method** parameter was introduced in IP Office Release 7.0, and the screen below shows the default value, which is sufficient in the sample configuration. The various alternatives for the **Association Method** may be useful when multiple SIP Trunks with different SIP domains resolve to a single IP Address. The default option associates incoming requests with SIP Lines by comparing the source IP Address and port of the incoming message against the configured far-end of the SIP Line.

The **Name Priority** parameter was introduced in IP Office Release 8.0. The **Name Priority** parameter can retain the default “System Default” setting, or can be configured to “Favor Trunk” or “Favor Directory” as shown in the sample screen below. “System Default” will use the setting displayed on the System → Telephony → Telephony Tab. The “Favor Directory” setting enables IP Office to match the caller’s telephone number against available system or personal directories, and display the name obtained from a match in the directory, if any, rather than name information received in the SIP signaling from Verizon. Click **OK** (not shown).

**SIP Line - Line 18**

SIP Line | Transport | SIP URI | VoIP | T38 Fax | SIP Credentials

Line Number: 18

ITSP Domain Name: 1.1.1.2

Prefix:

National Prefix:

Country Code: 1

International Prefix: 00

Send Caller ID: None

Association Method: By Source IP address

In Service: ☒

Use Tel URI: ☐

Check OOS: ☒

Call Routing Method: Request URI

Originator number for forwarded and twinning calls:

Name Priority: System Default

Caller ID from From header: ☐

Send From In Clear: ☐

User-Agent and Server Headers:

☒ REFER Support

Incoming: Auto

Outgoing: Auto

UPDATE Supported: Auto



### 1.7.2. SIP Line - Transport Tab

Select the **Transport** tab. This tab was introduced in Release 6.1. Some information configured in this tab had been under the **SIP Line** tab in Release 6.0.

The **ITSP Proxy Address** is set to the IP Address provided by Verizon Business. As shown in **Figure 1**, this IP Address is 172.30.205.55. In the **Network Configuration** area, UDP is selected as the **Layer 4 Protocol**, and the **Send Port** is set to the port number provided by Verizon Business. As shown in **Figure 1**, this port is 5072 in the sample configuration. The **Use Network Topology Info** parameter is set to “None”. The **Use Network Topology Info** parameter is set to “None”.

The screenshot shows the 'Transport' tab of the SIP Line configuration. The 'ITSP Proxy Address' is set to '172.30.205.55'. The 'Network Configuration' section includes a 'Layer 4 Protocol' dropdown set to 'UDP', a 'Send Port' spinner set to '5072', a 'Use Network Topology Info' dropdown set to 'None', and a 'Listen Port' spinner set to '5060'. Below this, 'Explicit DNS Server(s)' are shown as two sets of IP address input fields, both containing '0 . 0 . 0 . 0'. The 'Calls Route via Registrar' checkbox is checked. A 'Separate Registrar' text input field is at the bottom.

### 1.7.3. SIP Line - SIP URI Tab

Select the **SIP URI** tab. To add a new SIP URI, click the **Add...** button. In the bottom of the screen, a New Channel area will be opened. To edit an existing entry, click an entry in the list at the top, and click the **Edit...** button. In the bottom of the screen, the Edit Channel area will be opened.

In the sample configuration, each of the Verizon-provided toll free numbers are entered as a SIP URI, with the specific number entered in the **Local URI**, **Contact**, and **Display Name** fields. The **PAI** parameter was introduced in IP Office Release 6.1, and the value “None” is shown selected from the drop-down menu. The **Registration** parameter is set to the default “0: <None>” since Verizon Business IP Contact Center service does not require registration. The **Incoming Group** parameter, set here to 18, will be referenced when configuring Incoming Call Routes to map inbound SIP trunk calls to IP Office destinations in Section 5.7. The **Outgoing Group** parameter, also set here to 18, is not relevant in that this SIP Line will not be chosen for outbound calls, since the Verizon Business IPCC service will only be used for inbound toll-free calls. Click **OK**.

Channel	Groups	Via	Local URI	Contact	Display Name	PAI
1	18 18	<... 866851...	8668...	8668510107	8668510107	N...
2	18 18	<... 866850...	8668...	8668502380	8668502380	N...
3	18 18	<... 866851...	8668...	8668512649	8668512649	N...
4	18 18	<... 866850...	8668...	8668506850	8668506850	N...
5	18 18	<... 866852...	8668...	8668523221	8668523221	N...

Add...  
Remove  
Edit...

Edit Channel

Via: <None>  
Local URI: 8668510107  
Contact: 8668510107  
Display Name: 8668510107  
PAI: None  
Registration: 0: <None>  
Incoming Group: 18  
Outgoing Group: 18  
Max Calls per Channel: 10

OK  
Cancel

IP Office allows the number of simultaneous calls to a specific SIP URI to be managed using the **Max Calls per Channel** field. In the following screen, note that the **Max Calls per Channel** field has been changed from the default 10 to 2. With this configuration, two simultaneous calls to the number 866-850-6850 will be allowed. Once two calls are active, and a third call is attempted to 866-850-6850, IP Office will return a SIP 4xx response. Calls to other toll-free numbers using this same SIP Line are unaffected by the Max Calls per Channel for a different URI. Therefore, this approach could be used to control the maximum number of calls to each of the specific toll-free numbers, preventing a surge of calls to a given toll-free number from monopolizing the available call handling capacity of the access line or IP Office resources. An alternative means to restrict the number of simultaneous calls to a toll-free number that terminates on a hunt group would be to limit the queue size of the destination hunt group. If a non-priority call arrives to IP Office to a hunt group with a fixed size queue, and the queue is full, and there is no voice mail for the hunt group, IP Office returns a 486 Busy Here. See Section 5.5.4 for hunt group configuration.

Via	<None>
Local URI	8668506850
Contact	8668506850
Display Name	8668506850
PAI	None
Registration	0: <None>
Incoming Group	18
Outgoing Group	18
Max Calls per Channel	2

#### 1.7.4. SIP Line - VoIP Tab

Select the **VoIP** tab. In the sample configuration, the **Codec Selection** was configured using the “Custom” option, allowing an explicit ordered list of codecs to be specified, different from the system default (see Section 5.3.6). The arrow buttons can be used such that **G.729(a) 8K CS-ACELP** and **G.711 ULAW 64K** codecs are listed in the **Selected** column. This configures IP Office to support either G.729a or G.711MU for this SIP Line. 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. The **Use Offerer’s Preferred Codec** parameter can be left at the default unchecked value, or may be checked. In the sample configuration, Verizon preferred the G.729A codec in SDP, while also allowing the G.711MU codec. The IP Office configuration shown below matches these Verizon preferences. In the course of testing, the IP Office configuration was varied such that G.711MU was the preferred or only codec listed, and G.711MU calls were also successfully verified. The **PRACK/100rel Supported** parameter was introduced in IP Office Release 8, and should be left at the default unchecked value. Since the Verizon Business IP Contact Center service does not support fax, the **Fax Transport Support** parameter is set to “None”, and the **T38 Fax** tab need not be visited. Since the Verizon Business IPCC service does not require registration, the **SIP Credentials** tab need not be visited. Click **OK** (not shown).

SIP Line	Transport	SIP URI	VoIP	T38 Fax	SIP Credentials
<div> <div> <div>Codec Selection</div> <div>Custom</div> <div> <div>Unused</div> <div> G.711 ALAW 64K  G.722 64K  G.723.1 6K3 MP-MLQ </div> <div> &gt;&gt;  ↑  &lt;&lt;  ↓  &gt;&gt; </div> <div> <div>Selected</div> <div> G.729(a) 8K CS-ACELP  G.711 ULAW 64K </div> </div> </div> </div> <div> <input type="checkbox"/> VoIP Silence Suppression  <input checked="" type="checkbox"/> Re-invite Supported  <input type="checkbox"/> Use Offerer's Preferred Codec  <input type="checkbox"/> Codec Lockdown  <input type="checkbox"/> PRACK/100rel Supported </div> </div>					
<div>Fax Transport Support</div> <div>None</div>					
<div>Call Initiation Timeout (s)</div> <div>4</div>					
<div>DTMF Support</div> <div>RFC2833</div>					

## 1.8. Users, Extensions, and Hunt Groups

In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in the Navigation pane, and select **New**. To edit an existing User, select **User** in the Navigation pane, and select the appropriate user to be configured in the Group pane.

### 1.8.1. Digital User 232

The following screen shows the **User** tab for User 232. As shown in **Figure 1**, this user corresponds to the Avaya Digital 9508.

User		Avaya9508: 232	
Name	Extension	Button Programming   Menu Programming   Mobility   Phone Manager Options   Hunt Group Membership   Announcements	
RemoteMa...		Personal Directory	
NoUser		User   Voicemail   DND   ShortCodes   Source Numbers   Telephony   Forwarding   Dial In   Voice Recording	
Extn202	202	Name: Avaya9508 Password: **** Confirm Password: **** Full Name: Extension: 232 Email Address: Locale: Priority: 5 System Phone Rights: None Profile: Power User <input type="checkbox"/> Receptionist <input checked="" type="checkbox"/> Enable Softphone <input checked="" type="checkbox"/> Enable one-X Portal Services <input checked="" type="checkbox"/> Enable one-X TeleCommuter <input checked="" type="checkbox"/> Enable Remote Worker <input type="checkbox"/> Enable Flare Flare Mode: Simultaneous <input type="checkbox"/> Send Mobility Email <input type="checkbox"/> Ex Directory Device Type: Avaya 9508	
Extn203	203		
Extn204	204		
Extn205	205		
Extn206	206		
Extn207	207		
Extn208	208		
Extn210	210		
Extn211	211		
Extn212	212		
Extn213	213		
Extn214	214		
Extn216	216		
T7316E	231		
Avaya9508	232		
Avaya1616	233		
Softphone	234		
Avaya1140E	235		
Avaya9630	236		
Avaya9611	237		
Avaya9621	238		
Analog	241		

The following screen shows the **SIP** tab for User 232. In the sample configuration, the **SIP Name** and **Contact** parameters are configured with a Verizon Business IP Trunk DID number for the user, 7329450232. As shown in [VZBIPT-IPO81], 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 from Verizon IP Trunk service, without having to enter this number as an explicit SIP URI for the SIP Line. 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.

Avaya9508: 232	
User	Avaya9508: 232
Voicemail	
DND	
ShortCodes	
Source Numbers	
Telephony	
Forwarding	
Dial In	
Voice Recording	
Button Programming	
Menu Programming	
Mobility	
Phone Manager	
Hunt Group Membership	
Announcements	
SIP	
Personal Directory	
SIP Name	7329450232
SIP Display Name (Alias)	Avaya9508
Contact	7329450232
<input type="checkbox"/> Anonymous	

From **Figure 1**, note that user 232 will use the Mobile Twinning feature. The following screen shows the **Mobility** tab for User 232. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case 913035387024. Other options can be set according to customer requirements. In the sample configuration, the **Mobile Call Control** and **Mobile Callback** boxes were checked, and both mobile call control feature and mobile callback were tested using a Verizon-provided Toll Free number. In the case of mobile callback, a Verizon provided toll-free number was used to call in to IP Office and hang up. The mobile callback outbound leg used the Verizon Business IP Trunk service provisioned in [VZBIPT-IPO81].

The screenshot displays the Avaya IP Office configuration interface for user 232. The top navigation bar includes tabs for User, Voicemail, DND, ShortCodes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, Button Programming, and Menu Programming. The Mobility tab is selected, showing options for Internal Twinning, Mobility Features, and Mobile Twinning. The Mobile Twinning section is expanded, showing fields for Twinned Mobile Number (917326870755), Twinning Time Profile (<None>), Mobile Dial Delay (secs) (4), and Mobile Answer Guard (secs) (0). Checkboxes for Mobile Call Control and Mobile Callback are also visible.

As described in Section 5.3.4, names can be entered in directories to allow IP Office to match the caller ID for incoming calls and display the names from the directory. The following screen shows the **Personal Directory** tab for user 232. With the configuration shown below and on the SIP Line in Section 5.4.1 (where “Favor Directory” is selected), if user 232 receives an inbound Verizon IP Toll Free call from the telephone number 13035387006, the phone will display the name “Avaya Lab 1” (along with the number).

User	Voicemail	DND	ShortCodes	Source Numbers	Telephony	Forwarding	Dial In	Voice Recording	Button Programming	Menu Programming
Mobility	Phone Manager Options	Hunt Group Membership	Announcements	SIP	Personal Directory					

Index	Name	Number	
00	Avaya Lab 1	3035387006	
01	Avaya Lab 2	3035387024	

Add...  
Remove  
Edit...

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.

Extension				Digital Extension: 25 232	
Id	Extension	Module	Port		
1	231	BD1	1		
2	202	BD1	2		
3	203	BD1	3		
4	204	BD1	4		
5	205	BD1	5		
6	206	BD1	6		
7	207	BD1	7		
8	208	BD1	8		
25	232	BD2	1		
26	210	BD2	2		
27	211	BD2	3		
28	212	BD2	4		
29	213	BD2	5		
30	214	BD2	6		
31	241	BP2	7		
32	216	BP2	8		

Extn	
Extension Id	25
Base Extension	232
Caller Display Type	On
Reset Volume After Calls	<input type="checkbox"/>
Device Type	Avaya 9508
Module	BD2
Port	1
Disable Speakerphone	<input type="checkbox"/>

### 1.8.2. Avaya Flare Experience User 234 with IP Office Softphone Privileges

The following screen shows the **User** tab for User 234. This user corresponds to a user that will be granted “Power User”, Flare features and Avaya IP Office Softphone features. The **Profile** parameter is set to “Power User”. The **Enable Softphone** and **Enable Flare** boxes are checked, along with the **Flare Mode** set to “Standalone”.

**Softphone: 234**

Hunt Group Membership | Announcements | SIP | Personal Directory | **User** | Voicemail | DND | ShortCodes | Source Numbers | Telephony | Forwarding | Dial In | Voice Recording | Button Programming | Menu Programming

Name: Softphone  
Password: \*\*\*\*  
Confirm Password: \*\*\*\*  
Full Name:   
Extension: 234  
Email Address:   
Locale:   
Priority: 5  
System Phone Rights: None  
Profile: Power User

☐ Receptionist  
☒ Enable Softphone  
☒ Enable one-X Portal Services  
☒ Enable one-X TeleCommuter  
☒ Enable Remote Worker  
☒ Enable Flare  
Flare Mode: Standalone  
☐ Send Mobility Email  
☐ Ex Directory

Device Type: Unknown SIP device

User Rights  
User Rights view: User data

Like the user with extension 232, the **SIP** tab for the user with extension 234 is configured with a **SIP Name** and **Contact** specifying the user’s Verizon Business DID number using the Verizon Business IP Trunk service, as detailed in [VZBIPT-IPO81].



Softphone: 234	
User	Voicemail
DND	ShortCodes
Source Numbers	Telephony
Forwarding	Dial In
Voice Recording	Button Programming
Menu Programming	Mobility
Phone Manager	
Hunt Group Membership	Announcements
SIP	Personal Directory
SIP Name	7329450234
SIP Display Name (Alias)	Softphone
Contact	7329450234
<input type="checkbox"/> Anonymous	

The following screen shows the Voicemail tab for the user with extension 234. The **Voicemail On** box is checked, and a voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes, incoming calls from the Verizon Business IP Contact Center service to this user were redirected to Voicemail Pro after no answer. Voicemail messages were recorded and retrieved successfully. Voice mail navigation and retrieval were performed locally and from PSTN telephones using the Verizon Business IPCC toll-free numbers, to test DTMF using RFC 2833, and to test assignment of a Verizon-provided toll free number to the “Voicemail Collect” feature (e.g., via the \*17 short code shown in Section 5.6).

Softphone: 234	
Hunt Group Membership	Announcements
SIP	Personal Directory
User	Voicemail
DND	ShortCodes
Source Numbers	Telephony
Forwarding	Dial In
Voice Recording	Button Programming
Voicemail Code	****
Confirm Voicemail Code	****
Voicemail Email	
<input checked="" type="checkbox"/> Voicemail On <input type="checkbox"/> Voicemail Help <input type="checkbox"/> Voicemail Ringback <input type="checkbox"/> Voicemail Email Reading <input type="checkbox"/> UMS Web Services	
Voicemail Email <input checked="" type="radio"/> Off <input type="radio"/> Copy <input type="radio"/> Forward <input type="radio"/> Alert	
DTMF Breakout Reception / Breakout (DTMF *0/0)   System Default () Breakout (DTMF 2)   System Default () Breakout (DTMF 3)   System Default ()	

Select the **Telephony** tab. Then select the **Supervisor Settings** tab as shown below. The **Login Code** will be used by the Flare Experience and IP Office Softphone user as the login password.

The screenshot shows the 'Softphone: 234' configuration interface. The 'Telephony' tab is selected, and within it, the 'Supervisor Settings' sub-tab is active. The interface includes a navigation pane on the left with tabs like 'Hunt Group Membership', 'Announcements', 'SIP', 'Personal Directory', 'User', 'Voicemail', 'DND', 'ShortCodes', 'Source Numbers', 'Telephony', 'Forwarding', 'Dial In', 'Voice Recording', 'Button Programming', and 'Menu Programming'. The main content area has sub-tabs for 'Call Settings', 'Supervisor Settings', 'Multi-line Options', and 'Call Log'. Under 'Supervisor Settings', there are fields for 'Login Code' (set to '\*\*\*\*'), 'Login Idle Period (secs)', 'Monitor Group' (set to '<None>'), 'Coverage Group' (set to '<None>'), and 'Status on No-Answer' (set to 'Logged On (No change)'). There are also checkboxes for 'Force Login', 'Force Account Code', 'Outgoing Call Bar', 'Inhibit Off-Switch Forward/Transfer', 'Can Intrude', 'Cannot be Intruded' (checked), 'Can Trace Calls', 'CCR Agent', 'Automatic After Call Work', and 'Deny Auto Intercom Calls'. A section for 'Reset Longest Idle Time' has radio buttons for 'All Calls' (selected) and 'External Incoming'. At the bottom, 'After Call Work Time (secs)' is set to 'System Default (10)'.

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

The screenshot shows the 'Softphone: 234' configuration interface, now with the 'Call Settings' sub-tab selected. The 'Call Waiting On' checkbox is checked. Other settings include '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), and 'Call Cost Mark-Up' (100). Other checkboxes include 'Answer Call Waiting On Hold' (checked), 'Busy On Held', and 'Offhook Station' (checked).

The following screen shows the Extension information for this user, simply to illustrate the **VoIP** tab available for a SIP Telephone. To view, select **Extension** from the Navigation pane, and the appropriate extension from the Group pane. Select **VoIP** in the Details pane. The new **Codec Selection** parameter may retain the default setting “System Default” to follow the system configuration shown in Section 5.3.6. Alternatively, “Custom” may be selected to allow the codecs to be configured for this extension, using the arrow keys to select and order the codecs.

**SIP Extension: 8001 234**

Extn

VoIP

T38 Fax

IP Address 0 . 0 . 0 . 0

Codec Selection System Default ▼

Unused

>>

↑

<<

↓

>>

Selected

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

- ☐ VoIP Silence Suppression
- ☐ Local Hold Music
- ☒ Allow Direct Media Path
- ☒ Re-invite Supported
- ☐ Use Offerer's Preferred Codec
- ☐ Codec Lockdown
- ☐ Reserve Avaya IP endpoint license
- ☐ Reserve 3rd party IP endpoint license

Fax Transport Support None ▼

TDM->IP Gain Default ▼

IP->TDM Gain Default ▼

DTMF Support RFC2833 ▼

### 1.8.3. Hunt Groups

During the verification of these Application Notes, users could 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 a hunt group with Extension 401 and Name “Sales”. This hunt group was configured to contain various telephones from **Figure 1**. The **Ring Mode** was set to “LongestWaiting” (i.e., “longest waiting”, most idle user receives next call). Click the **Edit** button to change the **User List**.

Longest Waiting Group Sales: 401		
Hunt Group		
Name	Sales	<input type="checkbox"/> CCR Agent Group
Extension	401	
Ring Mode	LongestWaiting	No Answer Time (secs) System Default (15)
Hold Music Source	No Change	
Agent's Status on No-Answer Applies To	None	
User List		
Extension	Name	
<input checked="" type="checkbox"/>	235	Avaya1140E
<input checked="" type="checkbox"/>	232	Avaya9508
<input checked="" type="checkbox"/>	237	Avaya9611
<input checked="" type="checkbox"/>	234	Softphone
<input checked="" type="checkbox"/>	231	T7316E
<div>Edit... Remove</div>		

The following screen shows the **Queuing** tab for hunt group 401. In the sample configuration, the hunt group was configured to allow queuing so that incoming Verizon toll-free calls could be queued when all the members of the hunt group were busy on calls. In the testing associated with these Application Notes, the **Queue Length** was varied using both “No Limit” and specifically sized queues. For example, if the **Queue Length** is configured to 2, and if two calls are already in queue, a third call to the Verizon toll-free number corresponding to this hunt group will get busy tone because IP Office will send a 486 Busy Here (i.e., if there is no Voicemail for the hunt group). As another example, if the **Queue Length** has a fixed limit of 2, and if two calls are already in queue, a third call to the Verizon toll-free number destined for this hunt group from a priority caller (see Section 5.7.3) will be queued ahead of non-priority callers, temporarily expanding the queue.

The screenshot shows a configuration window titled "Longest Waiting Group Sales: 401". It has several tabs: "Hunt Group", "Queuing", "Overflow", "Fallback", "Voicemail", "Voice Recording", "Announcements", and "SIP". The "Queuing" tab is selected. Inside the "Queuing" tab, there is a "Queuing On" section with a checked checkbox. Below it, "Queue Length" is set to "No Limit" and "Normalize Queue Length" is checked. "Queue Type" is set to "Assign Call On Agent Answer". There is also a "Calls In Queue Alarm" section with "Calls In Queue Threshold" set to 1 and "Analog Extension to Notify" set to "<None>".

The following screen shows the **Announcements** tab for hunt group 220. In the sample configuration, when a call arrives when all members of the hunt group are busy on calls, the caller will first hear ring back tone. If a member of the hunt group does not become available after 10 seconds, the call will be answered by IP Office (i.e., 200 OK will be sent to Verizon), and the toll-free caller will hear a first announcement. Note that the **Flag call as answered** box is relevant for reporting applications, but does not change the fact that IP Office will answer the call when the first announcement is played. If the call is still not answered after the first announcement completes, the caller will hear music, a repeating second announcement, music, and so on until the call is answered by a member of the hunt group, or answered by voicemail for the hunt group (if configured). If a member of the hunt group becomes available while the caller is listening to ring back, music, or an announcement, the call is de-queued and delivered to the available member.

IP Office supports priority for queuing. For example, if low priority calls are waiting in queue, a higher priority call entering queue can be moved to the front of the queue and serviced before lower priority callers. For an inbound SIP trunk call, the priority can be specified on the Incoming Call Route as shown in Section 5.7.3.

**Longest Waiting Group Sales: 401**

Hunt Group | Queuing | Overflow | Fallback | Voicemail | Voice Recording | **Announcements** | SIP

☒ Announcements On

Wait before 1st announcement (seconds) 10 ☐ Synchronize Calls

Flag call as answered ☐

Play 1st announcement

Post announcement tone Music on hold

2nd Announcement ☒

Wait before 2nd announcement (seconds) 20

Play 2nd announcement

Repeat last announcement ☒

Wait before repeat (seconds) 20

## 1.9. Short Codes

In this section, various examples of IP Office short codes will be illustrated. To add a short code, right click on **Short Code** in the Navigation pane, and select **New**. To edit an existing short code, click **Short Code** in the Navigation pane, and the short code to be configured in the Group pane.

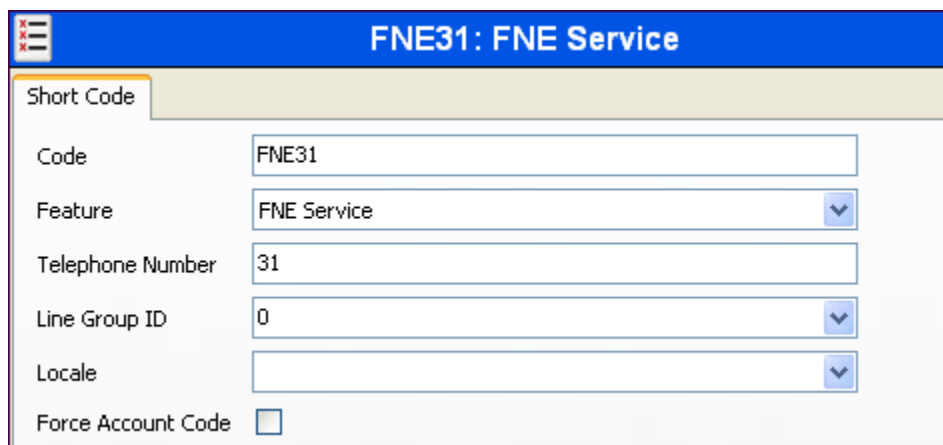
In the screen shown below, the short code “7N;” is repeated from references [VZBIPT-IPO81]. The **Code** parameter is set to “7N;”. The **Feature** parameter is set to “Dial”. The **Telephone Number** parameter is set to N“@Domain Name or IP Address of Verizon Business IP Trunk Service”. The **Telephone Number** 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 1, matching the number of the **Outgoing Group** configured on the **SIP URI** tab of SIP Line 17 to Verizon Business documented in [VZBIPT-IPO81]. Although Verizon Business IPCC service, the focus of these Application Notes, is used for inbound toll-free numbers, inbound toll-free calls can be twinned, forwarded, or transferred back to the PSTN via the Verizon Business IP Trunk SIP Line. In addition, inbound IPCC toll-free calls used to access the IP Office mobile call back feature can have the call back occur using the Verizon Business IP Trunk SIP Line. For more information on outbound calls, short codes, and ARS, see reference [VZBIPT-IPO81].

7N;; Dial	
Short Code	
Code	7N;;
Feature	Dial
Telephone Number	N"@pcelban0001.avayaalincroft.globalipcom.com"
Line Group ID	1
Locale	United States (US English)
Force Account Code	<input type="checkbox"/>

The following screen illustrates a short code that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code** “\*17” is defined for **Feature** “Voicemail Collect”. This short code will be used as one means to allow a Verizon toll-free number to be programmed to route directly to voice messaging, via inclusion of this short code as the destination of an Incoming Call Route. See Section 5.7 for configuration of Incoming Call Routes.

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

The following screen illustrates another short code. In this case, the **Code** “FNE31” is defined for **Feature** “FNE Service” to **Telephone Number** “31” (Mobile Call Control). This short code will be used as means to allow a Verizon DID to be programmed to route directly to this feature, via inclusion of this short code as the destination of an Incoming Call Route. See Section 1.10. This feature is used to provide dial tone to twinned mobile devices (e.g., cell phone) directly from IP Office; once dial tone is received the user can perform dialing actions including making calls and activating Short Codes.



FNE31: FNE Service	
Short Code	
Code	FNE31
Feature	FNE Service
Telephone Number	31
Line Group ID	0
Locale	
Force Account Code	<input type="checkbox"/>

## 1.10. Incoming Call Routes

In this section, IP Office Incoming Call Routes are illustrated. Each Incoming Call Route will map a Verizon Business toll-free number to a destination user, group, or function on IP Office. In some cases, the destination will be chosen based on the combination of the toll-free number and the caller id of the caller. Example mappings are summarized in **Table 1** in Section 3. 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.



### 1.10.1. Incoming Call Route to a Specific Telephone Extension

In the screen shown below, the incoming call route for **Incoming Number** “8668502380” is illustrated. The **Line Group Id** is 18, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to the Verizon Business IPCC service, in Section 5.4.

The screenshot shows the 'Standard' tab of a configuration window for the incoming call route '18 8668502380'. The window has three tabs: 'Standard', 'Voice Recording', and 'Destinations'. The 'Standard' tab is active. It contains several fields with dropdown menus:

Bearer Capability	Any Voice
Line Group ID	18
Incoming Number	8668502380
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

Select the **Destinations** tab. From the **Destination** drop-down, select an extension to receive the call when a PSTN user dials 8668502380. As shown in **Table 1**, 8668502380 is the number associated with IP Office user extension 232. (The **Destination** was changed in the course of testing to associate different destinations with the toll-free numbers.)

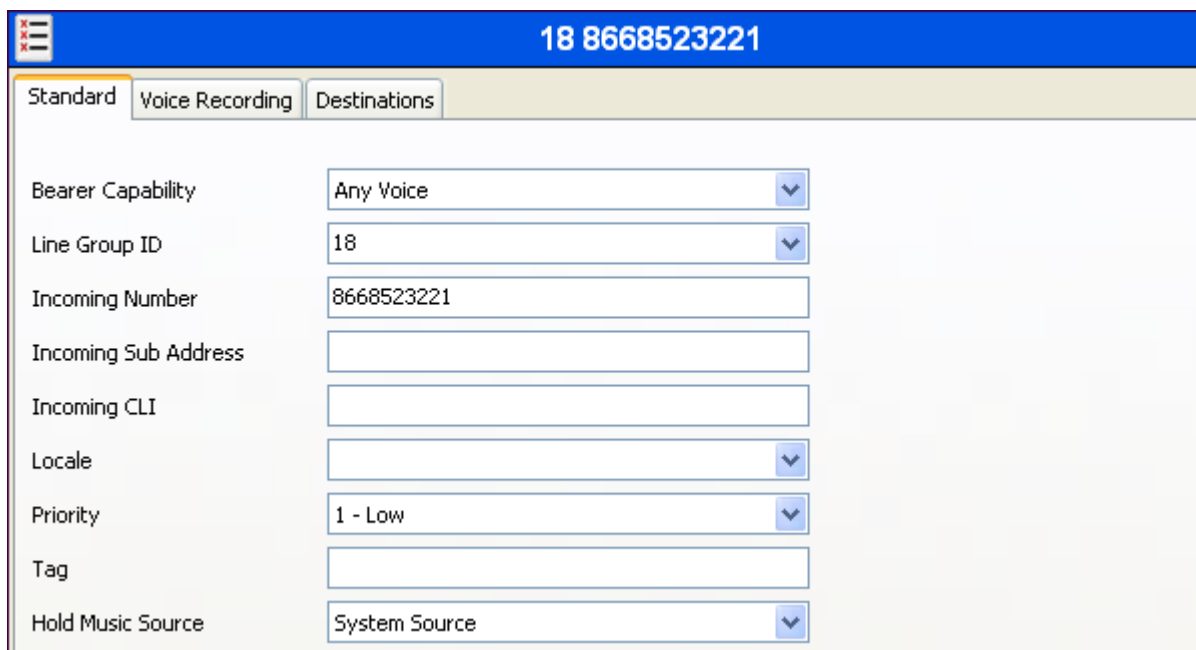
The screenshot shows the 'Destinations' tab of the same configuration window. It contains a table with three columns: 'TimeProfile', 'Destination', and 'Fallback Extension'. The 'Default Value' row shows '232 Avaya9508' selected in the 'Destination' dropdown.

TimeProfile	Destination	Fallback Extension
Default Value	232 Avaya9508	

Incoming Call Routes for other direct mappings of toll-free numbers to IP Office users are not presented here, but are configured in the same fashion.

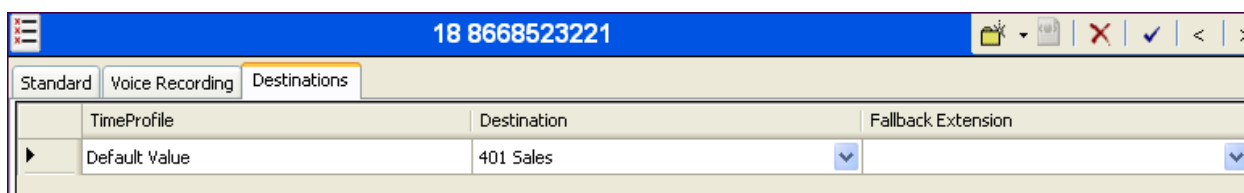
### 1.10.2. Incoming Call Routes to a Hunt Group by Dialed Toll-Free Number

In the screen shown below, an incoming call route for **Incoming Number** “8668523221” is illustrated. The **Line Group Id** is 18, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 5.4.



18 8668523221	
Standard Voice Recording Destinations	
Bearer Capability	Any Voice
Line Group ID	18
Incoming Number	8668523221
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

Select the **Destinations** tab. From the **Destination** drop-down, select the destination to receive the call when an arbitrary PSTN user dials 8668523221. As shown in **Table 1**, 8668510107 is the toll-free number associated with IP Office hunt group extension 401, the “Sales” hunt group.



18 8668523221		
Standard Voice Recording Destinations		
TimeProfile	Destination	Fallback Extension
Default Value	401 Sales	

### 1.10.3. Incoming Call Routes Based on Calling Party Number

This section presents simple examples showing that IP Office can use the calling party number to distinguish call priority or call destination, for calls to the same toll-free number. While the matching shown here is based on the full calling number, partial matching is also possible (e.g., to match a calling area code for a targeted geographic treatment).

In the screen shown below, the incoming call route for **Incoming Number** “8668523221” and **Incoming CLI** “3035387022” is illustrated. The **Line Group Id** is 18, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 5.4.

Note that the **Incoming Number** is the same as the toll-free number configured in the previous section. This route will be used for calls to the toll-free number specifically from a caller with caller ID “3035387022”. In this case, to allow this caller to be treated with priority when calling in for “Sales”, the **Priority** field is set to “3 – High”.

18 8668523221

Standard

Voice Recording

Destinations

Bearer Capability

Any Voice

Line Group ID

18

Incoming Number

8668523221

Incoming Sub Address

Incoming CLI

3035387022

Locale

Priority

3 - High

Tag

Hold Music Source

System Source

Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when PSTN user 13035387022 dials 8668523221. In this case, the **Destination** is also the hunt group “401 Sales”, but since high priority has been configured via the **Standard** tab, incoming calls from this caller will move to the front of the queue, and be serviced before calls waiting in queue from other non-priority callers.

18 8668523221

Standard

Voice Recording

Destinations

	TimeProfile	Destination	Fallback Extension
▶	Default Value	401 Sales	

**1.10.4. Incoming Call Routes to Various IP Office Features**

In the sample configuration, the incoming call route for **Incoming Number** “8668506850” was varied to test different destination features, such as Voice Mail, Mobile Call Control, and Conference Bridge, as shown in **Table 1** in **Section 3**. The screen showing the **Standard** tab for this toll-free number is shown below.

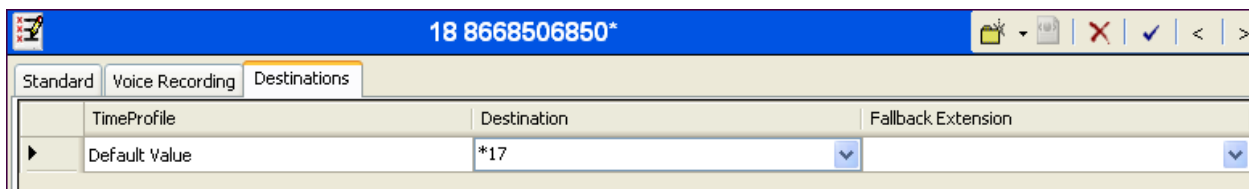


18 8668506850	
Standard	Voice Recording Destinations
Bearer Capability	Any Voice
Line Group ID	18
Incoming Number	8668506850
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

When configuring an Incoming Call Route, the **Destination** field can be manually configured with a number such as a short code, or certain keywords available from the drop-down list. At different times during testing, the **Destinations** tab for 8668506850 was configured to contain the following destinations:

- “\*17” (short code “Voicemail Collect”, as shown in Section 5.6). With this destination, an incoming call to 8668506850 will be delivered directly to voice mail, allowing the caller to log-in to voice mail and access messages.
- “FNE31” (short code for accessing the Mobile Call Control feature directly, as shown in Section 5.6) With this destination, an incoming call to 8668506850 from configured mobile callers will be provided dial tone to make calls from the mobile phone as if the user were using their IP Office extension.
- “VM:MeetMeConf” With this destination, an incoming call to 8668506850 will be delivered directly to the Voicemail Pro Module “MeetMeConf” created for use as a conference bridge.

An example screen showing the short code configured for Voicemail Collect is shown below.

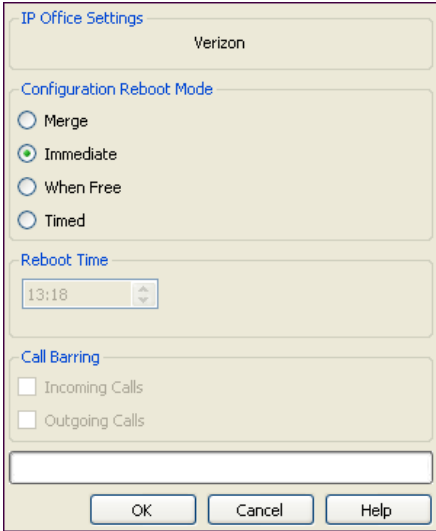


18 8668506850*		
Standard	Voice Recording	Destinations
TimeProfile	Destination	Fallback Extension
▶ Default Value	*17	

## 1.11. Saving Configuration Changes to IP Office

Navigate to **File → Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

The following will appear, with either **Merge** or **Immediate** selected, based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a service disruption. Click **OK** if desired.



The image shows a dialog box titled "IP Office Settings". It contains several sections: "IP Office Settings" with a dropdown menu showing "Verizon"; "Configuration Reboot Mode" with four radio button options: "Merge", "Immediate" (which is selected), "When Free", and "Timed"; "Reboot Time" with a time selection field showing "13:18"; and "Call Barring" with two checkboxes: "Incoming Calls" and "Outgoing Calls", both of which are unchecked. At the bottom of the dialog are three buttons: "OK", "Cancel", and "Help".

## Verizon Business Configuration

Information regarding Verizon Business IP Contact Center service offer can be found by contacting a Verizon Business sales representative, or by visiting <http://www.verizonbusiness.com/Products/communications/contact-center/>

The configuration described in these Application Notes was located in the Avaya Solutions and Interoperability Lab. The Verizon Business IP Contact Center service was accessed via a Verizon Private IP (PIP) T1 connection as described in Section 1. Verizon Business provided the necessary service provisioning, which included the domain *adevc.avaya.globalipcom.com* for the Avaya IP Office location.

For service provisioning, Verizon will require the customer IP address of the Avaya IP Office. Verizon provided the following information for the compliance testing: the IP address and port used by the Verizon SBC, and the toll-free numbers shown in **Figure 1** and **Table 1**. This information was used to complete the IP Office configuration shown in Section 5.

## Verification

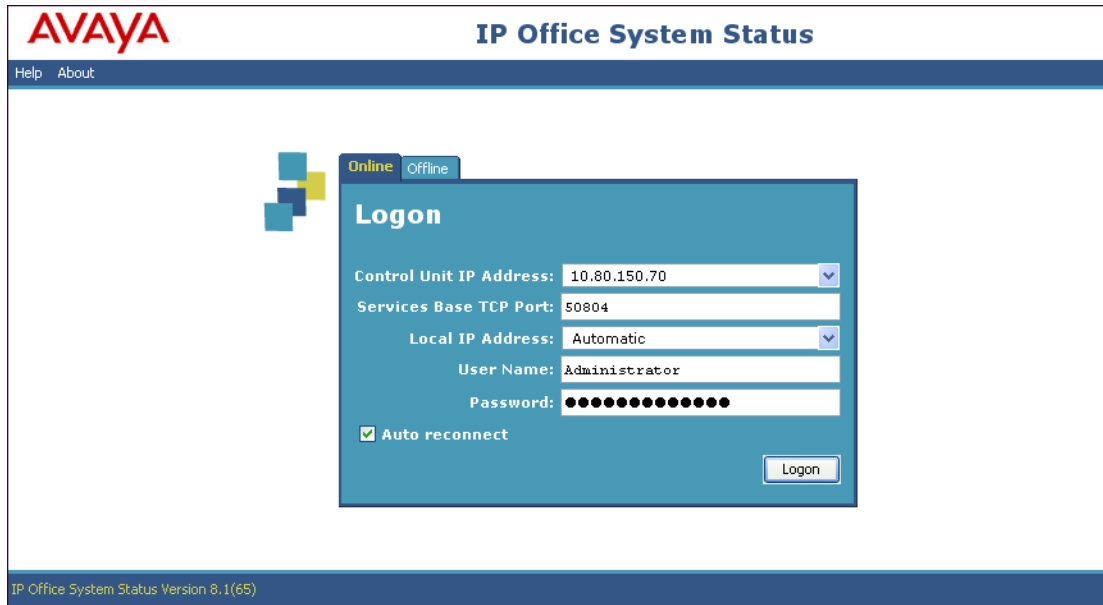
This section illustrates means to verify the configuration besides simply making the types of calls detailed in Section 2.1.

### 1.11.1. System Status

The System Status application is used to monitor and troubleshoot IP Office. Use the System Status application to verify the state of the SIP trunk. System Status can be accessed from **Start → Programs → IP Office → System Status**. Or by opening an Internet browser and type the URL: `http://ipaddress` where *ipaddress* is the IP address of the Avaya IP Office LAN1 interface. See reference [4] for more information. Click on **System Status** to launch the application.



The following screen shows an example **Logon** screen. Enter the IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.



**AVAYA** IP Office System Status

Help About

Online Offline

### Logon

Control Unit IP Address: 10.80.150.70

Services Base TCP Port: 50804

Local IP Address: Automatic

User Name: Administrator

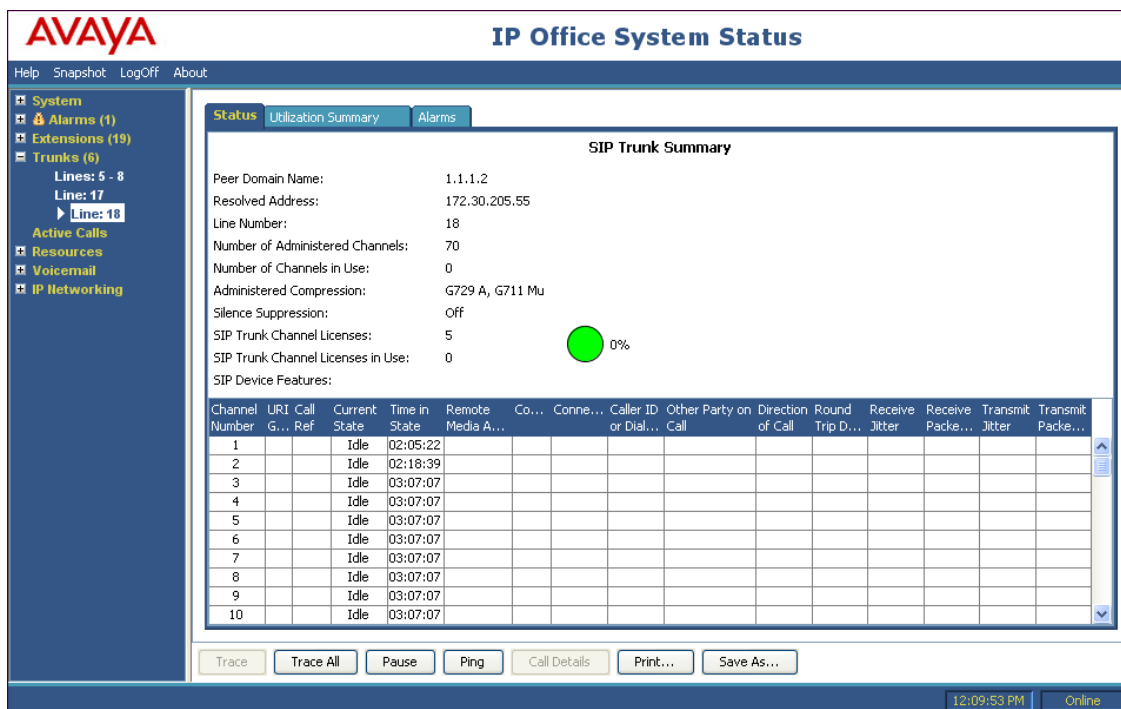
Password: ●●●●●●●●●●

☒ Auto reconnect

Logon

IP Office System Status Version 8.1(65)

Select the SIP line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current State** is *Idle* for each channel.



**AVAYA** IP Office System Status

Help Snapshot LogOff About

- System
  - Alarms (1)
  - Extensions (19)
  - Trunks (6)
    - Lines: 5 - 8
    - Line: 17
    - Line: 18
  - Active Calls
  - Resources
  - Voicemail
  - IP Networking

**Status** Utilization Summary Alarms

### SIP Trunk Summary

Peer Domain Name: 1.1.1.2

Resolved Address: 172.30.205.55

Line Number: 18

Number of Administered Channels: 70

Number of Channels in Use: 0

Administered Compression: G729 A, G711 Mu

Silence Suppression: Off

SIP Trunk Channel Licenses: 5

SIP Trunk Channel Licenses in Use: 0

SIP Device Features:

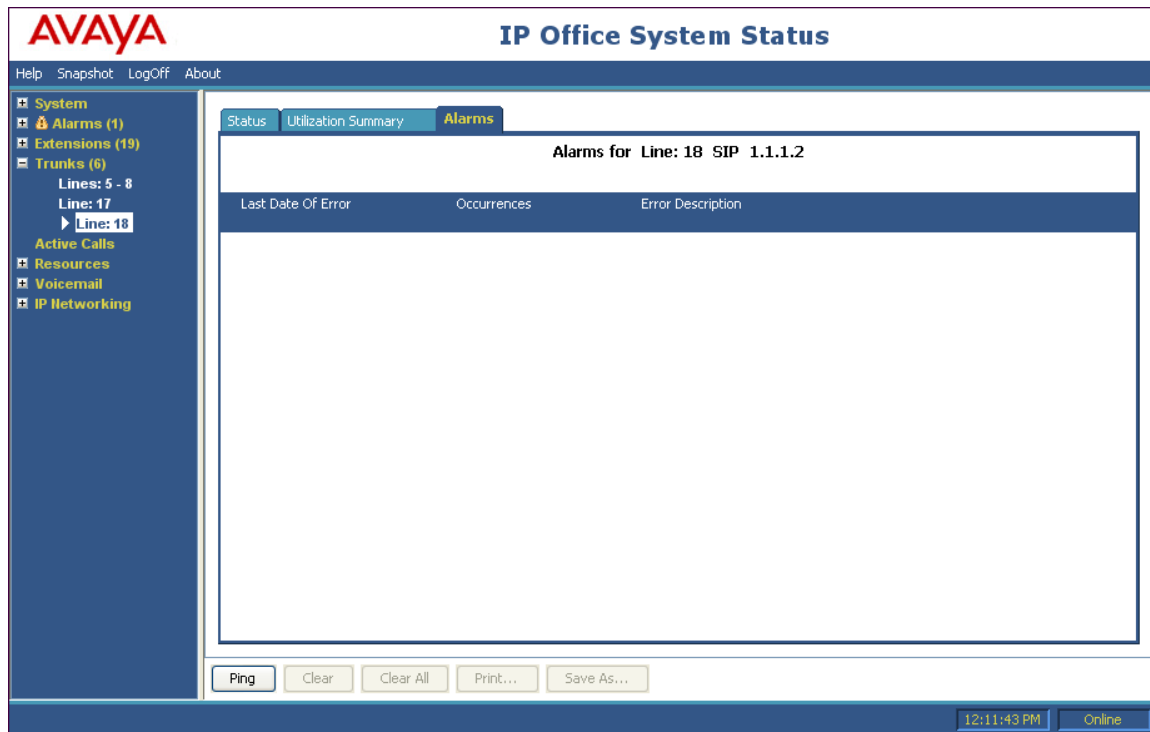
Channel Number	URI G...	Call Ref	Current State	Time in State	Remote Media A...	Co...	Conne...	Caller ID or Dial...	Other Party on Call	Direction of Call	Round Trip D...	Receive Jitter	Receive Packe...	Transmit Jitter	Transmit Packe...
1			Idle	02:05:22											
2			Idle	02:18:39											
3			Idle	03:07:07											
4			Idle	03:07:07											
5			Idle	03:07:07											
6			Idle	03:07:07											
7			Idle	03:07:07											
8			Idle	03:07:07											
9			Idle	03:07:07											
10			Idle	03:07:07											

Trace Trace All Pause Ping Call Details Print... Save As...

12:09:53 PM Online

Select the **Alarms** tab and verify that no alarms are active on the SIP line.





### 1.11.2. System Monitor

The System Monitor application can also be used to monitor and troubleshoot IP Office. System Monitor can be accessed from **Start → Programs → IP Office → Monitor**. The application allows the monitored information to be customized. To customize, select the button that is third from the right in the screen below, or select **Filters → 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

T1

ATM

ISDN

VComp

Call

Key/Lamp

VPN

DTE

Directory

VPN

EConf

Media

WAN

Frame Relay

PPP

SCN

R2

SSI

GDD

Jade

H.323

Interface

SIP

System

Events

☐ Sip

Low

☐ STUN

☐ SIP Dect

Packets

☐ SIP Reg/Opt Rx

☐ SIP Reg/Opt Tx

☐ SIP Call Rx

☐ SIP Call Tx

☐ SIP Misc Rx

☐ SIP Misc Tx

☐ Cm Notify Rx

☐ Cm Notify Tx

☒ Sip Rx

☐ hex

☒ Sip Tx

☐ hex

IP Filter (nnn.nnn.nnn.nnn)

Default All

Clear All

Tab Clear All

Tab Set All

OK

Cancel

Save File

Load File

Load Partial File

Select File

The screenshot displays a network packet capture tool interface. At the top, there is a menu bar with 'File', 'Edit', 'View', 'Filters', 'Status', and 'Help'. Below the menu is a toolbar with various icons for file operations, navigation, and analysis. The main display area shows a list of captured packets on the left and a detailed view of the selected packet on the right.

The packet list on the left shows two packets:

- 2013-03-12T12:12:52 11421362mS SIP Rx: UDP 172.30.205.55:5072 -> 1.1.1.2:5060
- 2013-03-12T12:12:52 11421367mS SIP Tx: UDP 1.1.1.2:5060 -> 172.30.205.55:5072

The detailed view of the first packet (2013-03-12T12:12:52 11421362mS SIP Rx) shows the following structure:

- 2013-03-12T12:12:52 11421362mS SIP Rx: UDP 172.30.205.55:5072 -> 1.1.1.2:5060
- INVITE sip:8668502380@adevc.avaya.globalipcom.com:5060 SIP/2.0
- Via: SIP/2.0/UDP 172.30.205.55:5072;branch=z9hG4bKs5arhql04o30ls4vm5m0.1
- Call-ID: 5407146481614113783@10.10.40.29
- From: <sip:+13035387006@199.173.94.24:5060;user=phone>;tag=942921091.5.kakaicnbnkbbombakbjldkda
- To: sip:18668502380@1.1.1.2
- CSeq: 1 INVITE
- Contact: <sip:+13035387006@172.30.205.55:5072;transport=udp>
- Allow: INVITE, ACK, BYE, OPTIONS, CANCEL, SUBSCRIBE, REFER
- P-Asserted-Identity: "THORNTON ,CO" <sip:+13035387006@199.173.94.24;user=phone>
- Accept: application/sdp
- Content-Type: application/sdp
- Content-Length: 204
- Max-Forwards: 69
- v=0
- o=- 1363111973294 0 IN IP4 172.30.205.164
- s=-
- c=IN IP4 172.30.205.164
- t=0 0
- m=audio 10406 RTP/AVP 18 0 8 101
- a=rtpmap:101 telephone-event/8000
- a=fmtp:101 0-15
- a=ptime:20
- a=fmtp:18 annexb=no

The detailed view of the second packet (2013-03-12T12:12:52 11421367mS SIP Tx) shows the following structure:

- 2013-03-12T12:12:52 11421367mS SIP Tx: UDP 1.1.1.2:5060 -> 172.30.205.55:5072
- SIP/2.0 100 Trying
- Via: SIP/2.0/UDP 172.30.205.55:5072;branch=z9hG4bKs5arhql04o30ls4vm5m0.1
- From: <sip:+13035387006@199.173.94.24:5060;user=phone>;tag=942921091.5.kakaicnbnkbbombakbjldkda
- To: <sip:18668502380@1.1.1.2>;tag=96930972ce016d87
- Call-ID: 5407146481614113783@10.10.40.29
- CSeq: 1 INVITE
- Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
- Supported: timer

IP Office is a highly modular IP telephone system designed to meet the needs of home offices, standalone businesses, and networked offices for small and medium enterprises.

**IP Office Release 8.1 has not been independently certified by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.**

## Additional References

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

- [1] *IP Office 8.1 Installation Manual*, Document Number 15-601042, August 2012
- [2] *IP Office Manager Manual 10.0*, Document Number 15-601011, August 2012
- [3] *IP Office Release 8.1 Implementing Voicemail Pro*, Document Number 15-601064, June 2012
- [4] *IP Office System Status Application*, Document Number 15-601758, November 2011
- [5] *Avaya IP Office Knowledgebase*, <http://marketingtools.avaya.com/knowledgebase>

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 8.0 with Verizon IPCC Service Suite.

[VZB-IPCCIPOR8FT] Application Notes for Configuring SIP Trunking using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 8, Issue 1.0

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 8.1 with Verizon IP Trunk Service Suite.

[VZBIPT-IPO81] Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service and Avaya IP Office Release 8.1– Issue 1.1

[RFC-3261] RFC 3261 *SIP: Session Initiation Protocol* <http://www.ietf.org/rfc/rfc3261.txt>

[RFC-2833] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals* <http://www.ietf.org/rfc/rfc2833.txt>

Information in the following Verizon documents was also used for these Application Notes. Contact a Verizon Business Account Representative for additional information.

- [VZ-Test-Plan] Test Suite for CPE IP Trunking Interoperability v1.3
- [VZ-Spec] Verizon Business IPCC Trunk Interface Network Interface Specification, Document Version 2.2.1.9

## Appendix A: 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.

Note that not all of the configuration information, particularly items relevant to a specific installation environment, is included in the SIP Line Template. Therefore, it is critical that the SIP Line configuration be verified/updated after a template has been imported and additional configuration be supplemented using Section 5.4 in these Application Notes as a reference.

The SIP Line Template created from the configuration as documented in these Application Notes is as follows:

```
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20130312</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>VerizonIPCC_IP081</DescriptiveName>
  <ITSPDomainName>1.1.1.2</ITSPDomainName>
  <SendCallerID>CallerIDNone</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>SystemDefault</LineNamePriority>
  <UpdateSupport>UpdateAuto</UpdateSupport>
  <UserAgentServerHeader />
  <CallerIDfromFromheader>false</CallerIDfromFromheader>
  <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
  <ITSPProxy>172.30.205.55</ITSPProxy>
  <LayerFourProtocol>SipUDP</LayerFourProtocol>
  <SendPort>5072</SendPort>
  <ListenPort>5060</ListenPort>
  <DNSServerOne>0.0.0.0</DNSServerOne>
  <DNSServerTwo>0.0.0.0</DNSServerTwo>
  <CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
  <SeparateRegistrar />
  <CompressionMode>AUTOSELECT</CompressionMode>
  <UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs>
  <AdvCodecPref>G.729(a) 8K CS-ACELP,G.711 ULAW 64K</AdvCodecPref>
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
```

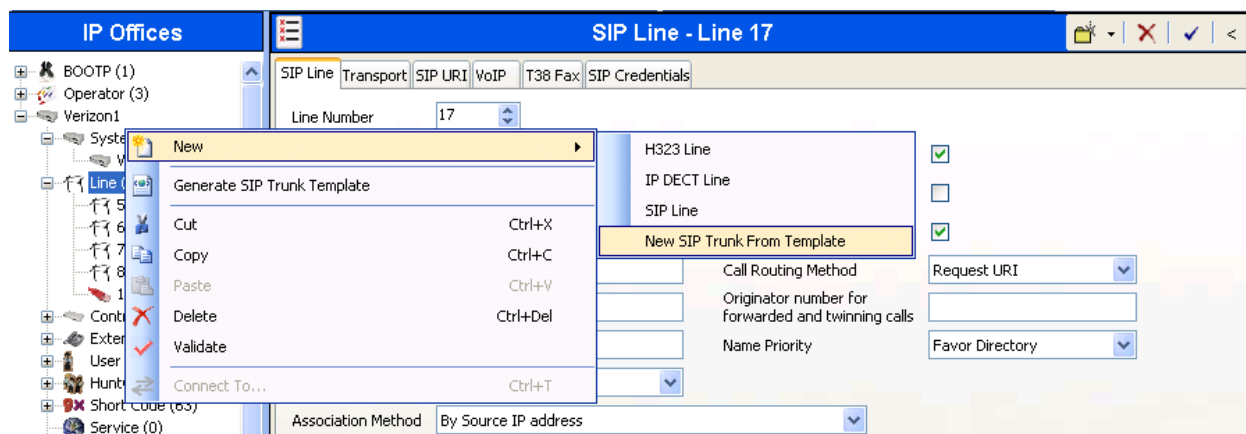
```

<VoipSilenceSupression>>false</VoipSilenceSupression>
<ReinviteSupported>>true</ReinviteSupported>
<FaxTransportSupport>FOIP_NONE</FaxTransportSupport>
<UseOffererPrefferedCodec>>false</UseOffererPrefferedCodec>
<CodecLockdown>>false</CodecLockdown>
<Rel100Supported>>false</Rel100Supported>
<T38FaxVersion>3</T38FaxVersion>
<Transport>UDPTL</Transport>
<LowSpeed>0</LowSpeed>
<HighSpeed>0</HighSpeed>
<TCFMethod>Trans_TCF</TCFMethod>
<MaxBitRate>FaxRate_14400</MaxBitRate>
<EflagStartTimer>2600</EflagStartTimer>
<EflagStopTimer>2300</EflagStopTimer>
<UseDefaultValues>>true</UseDefaultValues>
<ScanLineFixup>>true</ScanLineFixup>
<TFOPENhancement>>true</TFOPENhancement>
<DisableT30ECM>>false</DisableT30ECM>
<DisableEflagsForFirstDIS>>false</DisableEflagsForFirstDIS>
<DisableT30MRCompression>>false</DisableT30MRCompression>
<NSFOVERRIDE>>false</NSFOVERRIDE>
</Template>

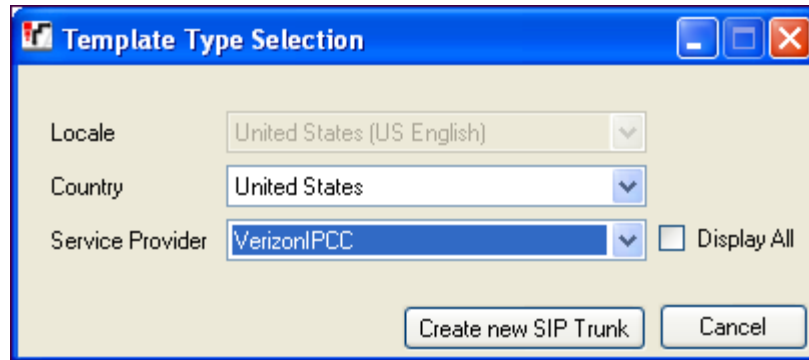
```

To import the above template into a new installation:

1. On the PC where IP Office Manager was installed, copy and paste the above template into a text document named **US\_VerizonIPCC\_SIPTrunk.xml**. Move the .xml file to the IP Office Manager template directory (C:\Program Files\Avaya\IP Office\Manager\Templates). It may be necessary to create this directory.
2. Import the template into an IP Office installation 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**:



1. Verify that “United States” is automatically populated for **Country** and “VerizonIPCC” is automatically populated for **Service Provider** in the resulting Template Type Selection screen as shown below. Click **Create new SIP Trunk** to finish the importing process.



Template Type Selection

Locale: United States (US English)

Country: United States

Service Provider: VerizonIPCC ☐ Display All

Create new SIP Trunk Cancel

---

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