



## Avaya Solution & Interoperability Test Lab

---

# Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service and Avaya IP Office Release 6.1, Using REFER and DNS SRV – Issue 1.0

### Abstract

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Trunk SIP Trunk Service Offer and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500 Release 6.1 Preferred Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, digital, and analog endpoints.

These Application Notes complement previously published Application Notes by illustrating the configuration and verification of new capabilities of IP Office Release 6.1, including transfer using SIP REFER to Verizon, and use of DNS SRV to determine the Verizon Business SIP signaling information from a Verizon DNS Server. **Although these new IP Office Release 6.1 capabilities have not been independently certified by Verizon labs, these Application Notes can be used to facilitate customer engagements via the Verizon field trial process for either or both new features.**

The Verizon Business IP Trunk service offer referenced within these Application Notes is designed for business customers. The service enables local and long distance PSTN calling via standards-based SIP trunks directly, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

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

# Table of Contents

Table of Contents.....	2
1. Introduction.....	3
1.1. Interoperability Compliance Testing .....	4
1.2. Support.....	4
1.2.1. Avaya .....	4
1.2.2. Verizon.....	4
1.3. Known Limitations .....	4
2. Reference Configuration.....	6
3. Equipment and Software Validated .....	8
4. Avaya IP Office Configuration .....	8
4.1. Physical, Network, and Security Configuration .....	9
4.2. Licensing.....	11
4.3. System Settings.....	13
4.3.1. System Tab.....	13
4.3.2. LAN Settings .....	14
4.3.3. Voicemail.....	16
4.3.4. System Telephony Configuration .....	17
4.3.5. System Twinning Configuration.....	17
4.4. SIP Line .....	18
4.4.1. SIP Line - SIP Line Tab.....	18
4.4.2. SIP Line - Transport Tab .....	20
4.4.3. SIP Line - SIP URI Tab .....	21
4.4.4. SIP Line - VoIP Tab .....	22
4.5. Users, Extensions, and Hunt Groups .....	22
4.5.1. User 201 .....	23
4.5.2. User 30026.....	26
4.5.3. Hunt Groups.....	29
4.6. Short Codes.....	32
4.7. Incoming Call Routes .....	34
4.8. ARS and Alternate Routing .....	37
4.9. Privacy / Anonymous Calls .....	39
4.10. SIP Options Frequency .....	40
4.11. Saving Configuration Changes to IP Office .....	41
5. Verizon Business Configuration.....	43
6. General Test Approach and Results.....	43
7. Verifications.....	44
7.1. REFER Testing .....	44
7.1.1. Wireshark Trace Illustration for REFER-Transfer .....	45
7.2. DNS SRV Testing.....	48
7.2.1. Wireshark Trace Illustration for DNS SRV.....	48
7.3. Verification Summary.....	50
8. Conclusion .....	53
9. References.....	53

# 1. Introduction

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Trunk SIP Trunk Service Offer and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500 Release 6.1 Preferred Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, digital, and analog endpoints.

These Application Notes are based on the previously published Application Notes in reference [JRR-IPOR6] which covered IP Office Release 6. These Application Notes complement previously published Application Notes by illustrating the configuration and verification of new capabilities of IP Office Release 6.1, including transfer using SIP REFER to Verizon, and use of DNS SRV to determine the Verizon Business SIP signaling information from a Verizon DNS Server. Although these new IP Office Release 6.1 capabilities have not been independently certified by Verizon labs, these Application Notes can be used to facilitate customer engagements via the Verizon field trial process for either or both new features. REFER-based transfer can allow the trunks to the IP Office enterprise location to be released when a call is transferred, resulting in a connection that no longer uses IP Office resources. The use of DNS SRV obviates the need to statically configure the Verizon SIP Signaling IP Address and port information in IP Office, thus allowing dynamic, automatic updates to SIP signaling to occur without manual intervention by the IP Office administrator, if Verizon network conditions change.

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

Verizon Business IP Trunk service offer 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 IP Trunk service terminated via a PIP network connection, the solution validated in this document applies equally to IP Trunk services delivered via IDA service terminations.

For more information on the Verizon Business IP Trunking service, including access alternatives, visit <http://www.verizonbusiness.com/us/products/voip/trunking/>

## 1.1. Interoperability Compliance Testing

The Avaya IP Office location was connected to the Verizon Business IP Trunk Service, as depicted in **Figure 1**. Avaya IP Office was configured to use the commercially available SIP Trunking solution provided by the Verizon Business IP Trunk SIP Trunk Service. This allowed Avaya IP Office users to make calls to the PSTN and receive calls from the PSTN via the Verizon Business IP Trunk SIP Trunk Service.

This document supplements reference [JRR-IPOR6], which included detailed illustrations of the verification of representative calls using the IP Office System Status application, the IP Office System Monitor application, and Wireshark.

The focus of the verification testing associated with this version of the Application Notes was the use of SIP REFER messaging for call transfer scenarios, and the use of DNS SRV rather than static provisioning of the Verizon SIP signaling information. Note that the REFER testing and the DNS SRV testing are not dependent on one another. That is, IP Office can use REFER for call transfer without using DNS SRV, and IP office can use DNS SRV without using REFER for call transfer. These Application Notes are intended to facilitate Verizon field trial testing of either or both capabilities.

## 1.2. Support

### 1.2.1. Avaya

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

### 1.2.2. Verizon

For technical support on Verizon Business IP Trunk service offer, visit the online support site at <http://www.verizonbusiness.com/us/customer/>.

## 1.3. Known Limitations

Interoperability testing of the sample configuration was completed with successful results, with the successful verifications detailed in Section 7. The following observations were noted:

1. For **Compression Mode**, either **G.729a 8K CS-ACELP** or **G.711 ULAW 64K** can be selected for voice calls. Note that T.38 fax is not supported by the Verizon IP Trunk SIP Trunk Service, and T.38 fax is the only fax method supported using SIP Lines on IP Office. Although the Verizon Business IP Trunk Service does not support T.38 fax, and T.38 fax is the only fax method supported by IP Office for SIP Lines, calls were nevertheless made to and from an analog fax machine connected to IP Office. Fax calls may succeed using G.711 but cannot be guaranteed. Although not supported, if fax is to be attempted, **G.711 ULAW 64K** can be selected as the only allowed codec on the SIP Line. Alternatively, if G.729a is desired for voice calls and G.711 for fax calls, the **Re-Invite Supported** option for the SIP Line can be checked to allow re-negotiation to G.711 for a call involving a fax machine that begins at G.729a.

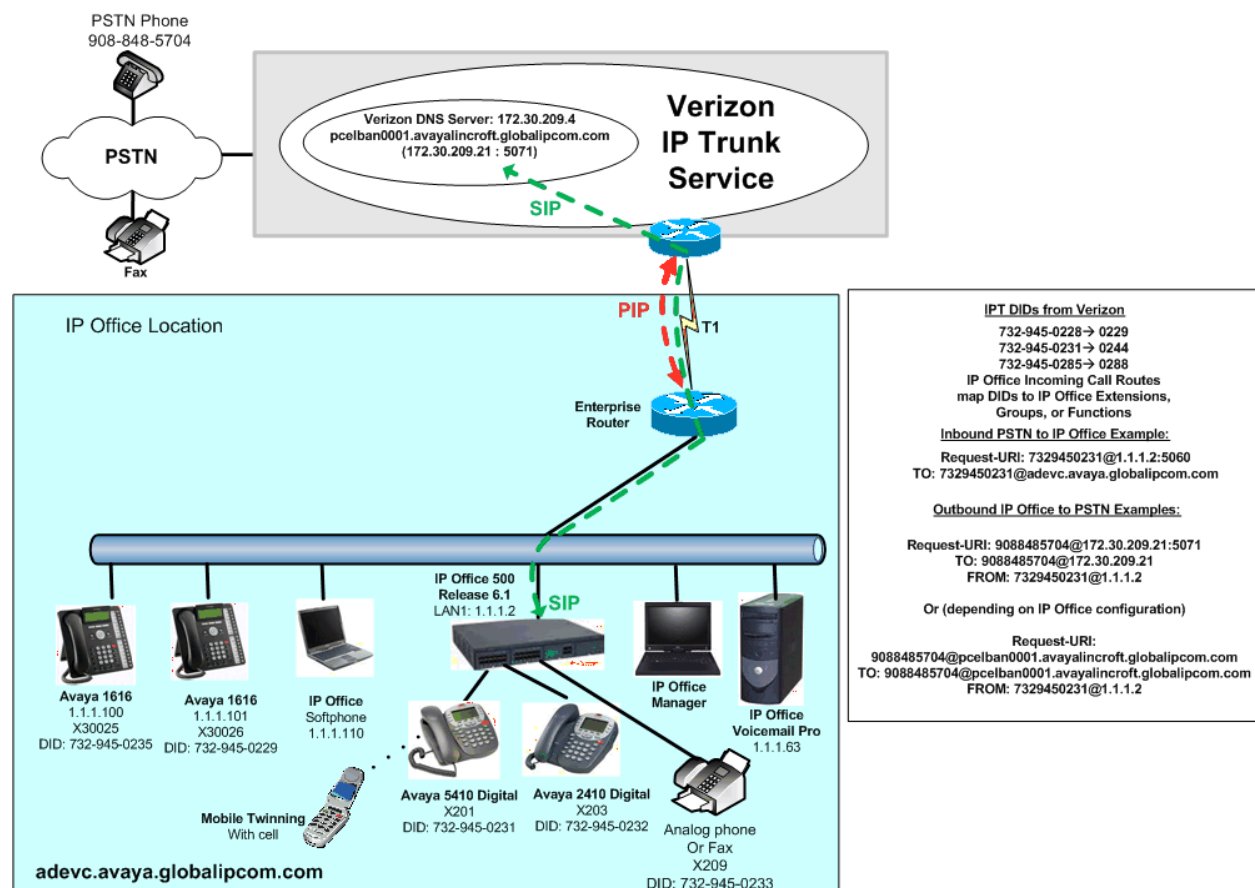
2. When a call is put on hold by an IP Office user, there is no indication sent via SIP messaging to Verizon. This is transparent to the users on the call.
3. When using the IP Office Softphone, inbound PSTN calls from the Verizon Business IP Trunk service to the IP Office Softphone may negotiate to the G.711MU codec, even if the SIP Line configuration lists G.729a first on the **VoIP** tab (as shown in Section 4.4.4). Specifically, if the IP Office Softphone user has logged in with the “IP Office: Default” profile, and the **Automatic Codec Preference** parameter on the **System → Telephony** tab is set to “G.711 ULAW 64K”, an inbound call from the Verizon Business IP Trunk service to the IP Office Softphone will use G.711MU. The IP Office Softphone user can log in with the “IP Office: Low Bandwidth” profile to ensure use of G.729a for both inbound and outbound calls via the SIP Line to Verizon Business.

## 2. Reference Configuration

**Figure 1** illustrates an example Avaya IP Office solution connected to the Verizon Business IP Trunk SIP Trunk service. The Avaya equipment is located on a private IP subnet. An enterprise edge router provides access to the Verizon Business IP Trunk service network via a Verizon Business T1 circuit. This circuit is provisioned for the Verizon Business Private IP (PIP) service. The Verizon configuration is identical to the configuration used in reference [JRR-IPOR6].

In the sample configuration, IP Office receives traffic from the Verizon Business IP Trunk service on port 5060. IP Office uses DNS SRV, using UDP for transport, to determine the IP Address and port to be used to send SIP signaling to Verizon. In the sample configuration, the DNS process will result in SIP signaling being sent to IP Address 172.30.209.21 and port 5071, but these values are not statically configured in IP Office. As shown in **Table 1**, the Verizon Business IP Trunk service provided Direct Inward Dial (DID) numbers. These DID numbers were mapped to IP Office destinations via Incoming Call Routes in the IP Office configuration.

Verizon Business used FQDN *pcelban0001.avayalincroft.globalipcom.com*. The Avaya IP Office environment was assigned FQDN *adevc.avaya.globalipcom.com* by Verizon Business.



**Figure 1: Avaya IP Office with Verizon IP Trunk SIP Trunk Service**

**Table 1** shows the mapping of Verizon-provided DID numbers to IP Office users, groups, or functions. The associated IP Office configuration is shown in Section 4. Selected verifications are illustrated in Section 7.

<b>Verizon Provided DID</b>	<b>Avaya IP Office Destination</b>	<b>Notes</b>
732-945-0228	Auto-Attendant on Voicemail Pro	See Section 4.6
732-945-0229	x30026	Avaya 1616 Telephone, or Avaya IP Office Softphone logged in as x30026
732-945-0231	x201	Digital Telephone with Mobile Twinning Active
732-945-0232	x203	Digital Telephone
732-945-0233	x209	Analog telephone or Fax machine, see Section 1.3
732-945-0234	Voicemail Collect on Voicemail Pro	See Section 4.6, also used for Voicemail Pro Callback
732-945-0235	x30025	Avaya 1616 IP Telephone
732-945-0236	“200 Main” Hunt Group	Collective Ring Hunt Group
732-945-0237	30200 Hunt Group	Sequential Ring Hunt Group

**Table 1: Verizon DID to IP Office Mappings**

### 3. Equipment and Software Validated

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

Equipment	Software
Avaya IP Office 500	Release 6.1 (6.1.5) (Preferred Edition)
Avaya IP Office Manager	Release 8.1 (8.1.5) (Preferred Edition)
Avaya IP Office Voicemail Pro	Release 6.1.16
Avaya IP Office Voicemail Pro Client	Version 6.1 (16)
Avaya 1600-Series Telephones (H.323)	Release 1.3
Avaya 2400-Series and 5400-Series Digital Telephones	REL: 6.00 (downloaded from IP Office)
Avaya IP Office Softphone	Release 3.1.2.17 59616
Brother Intellifax 1360 (analog fax)	N/A

Table 2: Equipment and Software Tested

### 4. Avaya IP Office Configuration

The Avaya IP Office configuration shown in this section is effectively the same as the configuration shown in reference [JRR-IPOR61]. The only substantive configuration screen differences are shown in Section 4.4 for the SIP Line.

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

**WELCOME to IP Office Administration**

**What would you like to do ?**

[Create an Offline Configuration](#)

[Open Configuration from System](#)

[Read a Configuration from File](#)

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



## 4.1. Physical, Network, and Security Configuration

This section describes attributes of the sample configuration, but is not meant to be prescriptive. Consult reference [IPO-INSTALL] 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 blank (i.e., no module is inserted). The next slots from left to right contain a VCM64, a Digital station module, and a “Phone8” analog module. The VCM64 is a Voice Compression Module supporting VoIP codecs. The Digital module allows connection of Avaya 5400-Series and Avaya 2400-Series Digital telephones. Referring to **Figure 1**, the Avaya 5410 telephone with extension 201 is connected to port 1 of the Digital module, and the Avaya 2410 telephone with extension 203 is connected to port 3 of the Digital module. The “Phone8” module allows connection of analog devices such as simple analog telephones or fax machines. In the testing of the sample configuration, an analog telephone or a fax machine is connected to port 1 of the “Phone8” module.

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** is selected in the Group pane, revealing additional information about the IP 500 in the Details pane.

The screenshot displays the IP Office configuration interface. It is divided into three main panes: IP Offices, Control Unit, and IP 500.

- IP Offices:** A tree view on the left showing various components. 'Control Unit (5)' is highlighted.
- Control Unit:** A table listing modules installed in the Control Unit.
- IP 500:** A details pane on the right showing configuration parameters for the selected IP 500 unit.

Dev No.	Dev Type	Version
1	IP 500	6.1 (5)
2	CARRIER/PRID T1	5.0 (8)
3	VCM64	6.1 (5)
4	DIGSTA8/ATM4	6.1 (5)
5	PHONE8/ATM4	6.1 (5)

Unit	
Device Number	1
Unit Type	IP 500
Version	6.1 (5)
Serial Number	00e007026f2d
Unit IP Address	1.1.1.2
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 1.1.1.1. To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New**. To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the Details pane with the relevant default route using **Destination** LAN1.

IP Route

IP Address: 0 . 0 . 0 . 0

IP Mask: 0 . 0 . 0 . 0

Gateway IP Address: 1 . 1 . 1 . 1

Destination: LAN1

Metric: 0

☐ 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.

**Security Service User Login**

IP Office : 00E007026F2D - IP 500

Service User Name: security

Service User Password: ••••••••••

OK Cancel Help

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

**System : 00E007026F2D**

System Details | Unsecured Interfaces

---

**Base Configuration**

Services Base TCP Port

Maximum Service Users

Maximum Rights Groups

---

**System Discovery**

TCP Discovery Active ☒ UDP Discovery Active ☒


---

**Security**

Session ID Cache (Hours)

Allow HTTPS ☒

**Server Certificate**

Offer Certificate ☒ 

Private Key

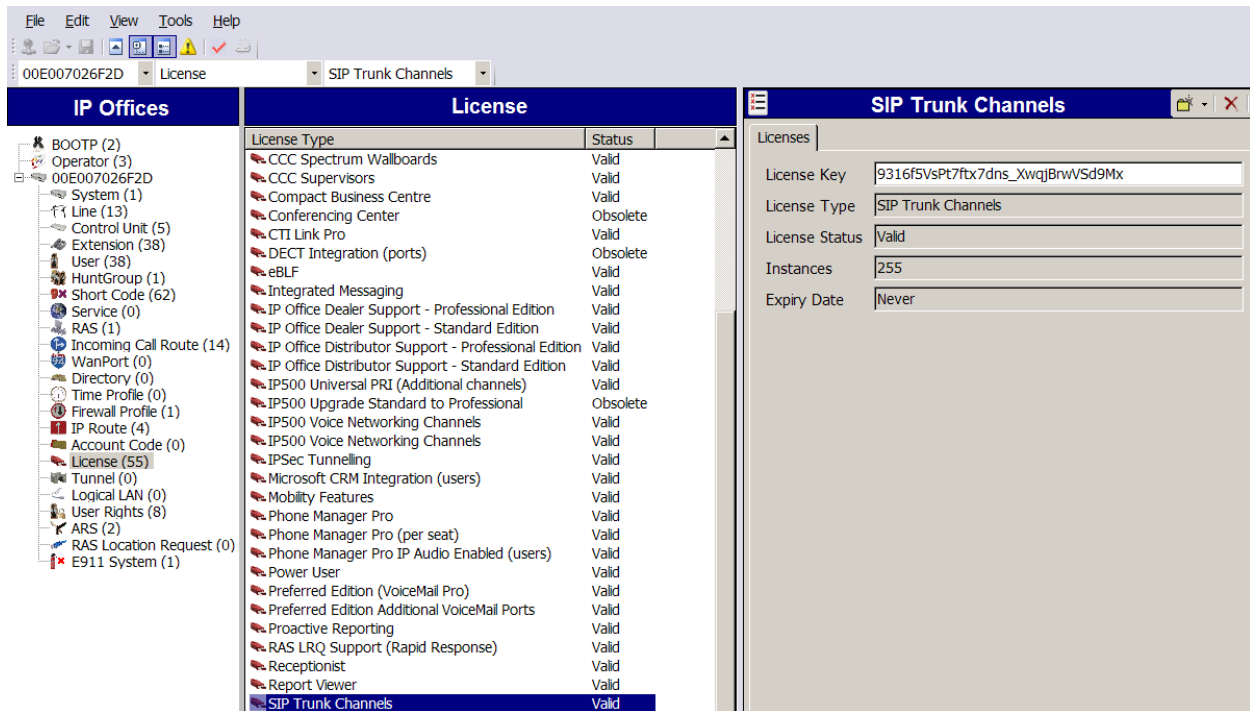
Issued to : IP Office 00e007026f2d

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

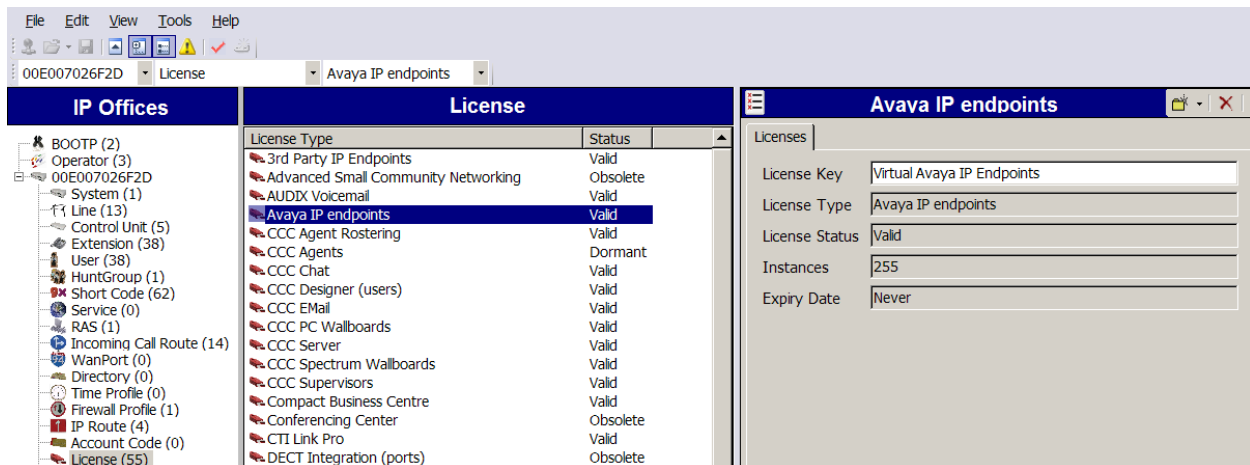
## 4.2. Licensing

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

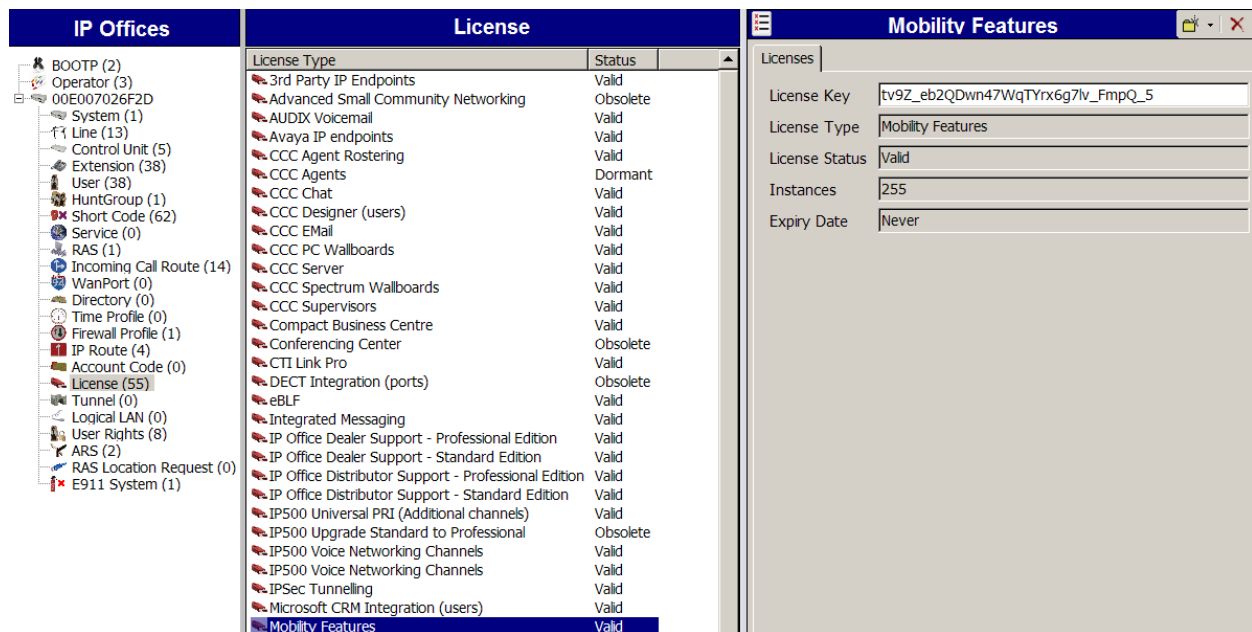
To verify that there is a SIP Trunk Channels License with sufficient capacity, click **License** in the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm a valid license with sufficient “Instances” (trunk channels) in the Details pane.



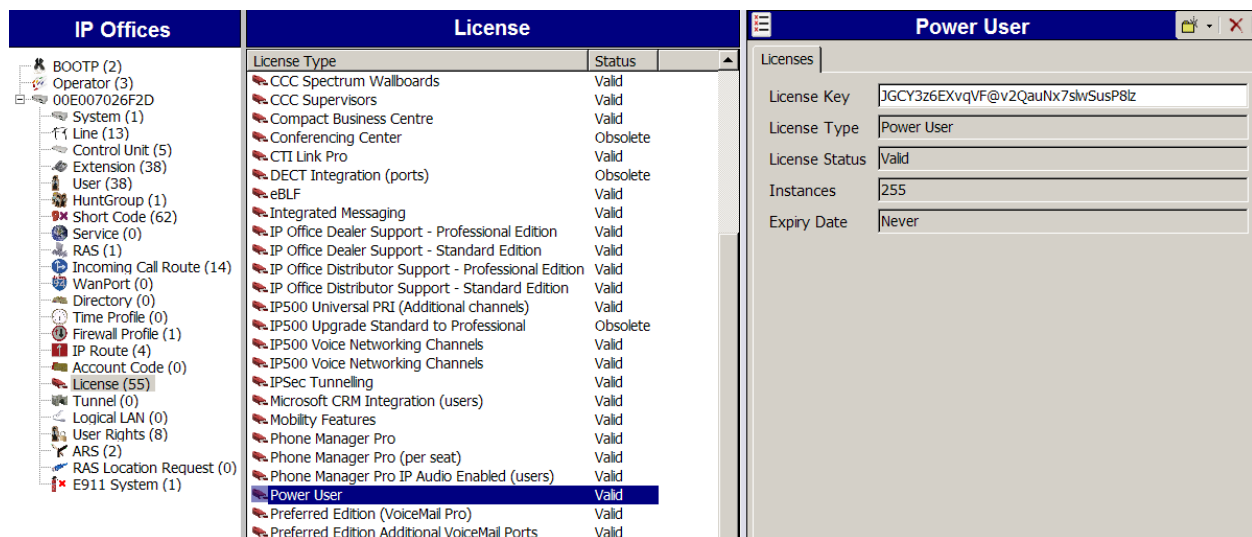
If Avaya IP Telephones will be used, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints** in the Group pane. Confirm a valid license with sufficient “Instances” in the Details pane.



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 Mobility features. In the sample configuration, various mobility features including Mobile Twinning are used.



The following screen shows the availability of a valid license for **Power User** features. In the sample configuration, the user with extension 30026 will be configured as a “Power User” and will be capable of using the Avaya IP Office Softphone.



## 4.3. 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.

### 4.3.1. System Tab

With the proper system name selected in the Group pane, select the **System** tab in the Details pane. The following screen shows a portion of the **System** tab. The **Name** field can be used for

a descriptive name of the system. In this case, the MAC address is used as the name. The **Avaya HTTP Clients Only** and **Enable SoftPhone HTTP Provisioning** boxes are checked to facilitate Avaya IP Office Softphone usage.

### 4.3.2. LAN Settings

In the sample configuration, LAN1 was used to connect the IP Office to the enterprise network. Other LAN choices (e.g., LAN2) may also be used. 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, known to Verizon Business, is 1.1.1.2. Other parameters on this screen may be set according to customer requirements.

Select the **VoIP** tab as shown in the following screen. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Verizon Business. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such

as the Avaya 1600-Series Telephones used in the sample configuration. The **SIP Registrar Enable** box is checked to allow Avaya IP Office Softphone usage.

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. That is, for SIP Trunk calls to and from Verizon Business, the SIP protocol exchanges will result in Verizon Business sending RTP media to IP Office using a UDP port in the configurable range shown below.

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 28 (**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.

The screenshot displays the Avaya IP Office configuration interface. On the left, a tree view shows the system hierarchy under 'IP Offices', including BOOTP (1), Operator (3), System (1), Line (13), Control Unit (5), Extension (38), User (38), HuntGroup (4), Short Code (65), Service (0), RAS (1), Incoming Call Route (23), WanPort (0), Directory (0), Time Profile (0), Firewall Profile (1), IP Route (4), Account Code (0), License (55), Tunnel (0), User Rights (8), ARS (2), RAS Location Request (0), and E911 System (1). The main panel is titled 'System' and shows the configuration for system '00E007026F2D'. The 'SIP Registrar' tab is selected, showing the following settings:

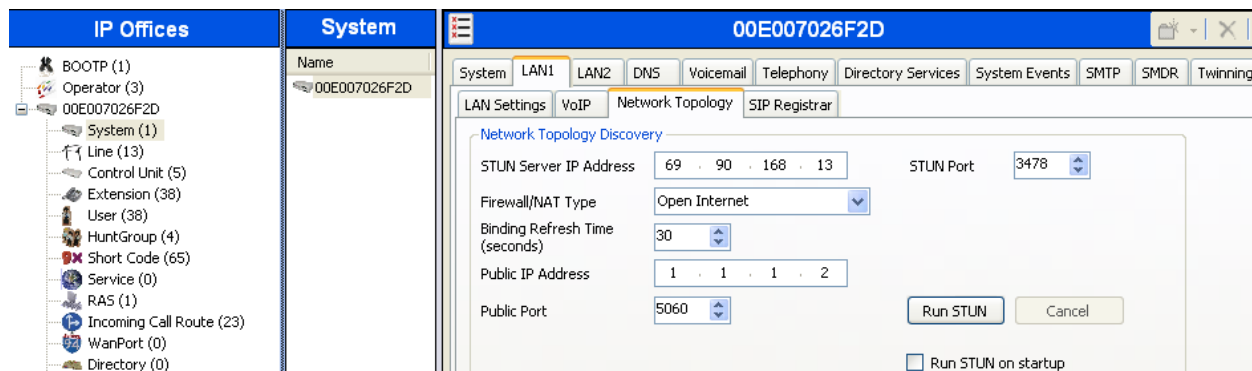
- ☒ H323 Gatekeeper Enable
- ☒ SIP Trunks Enable
- ☒ SIP Registrar Enable
- ☐ H323 Auto-create Extn
- ☐ H323 Auto-create User
- ☒ Enable RTCP Monitoring On Port 5005
- RTP Port Number Range**
  - Port Range (Minimum): 49152
  - Port Range (Maximum): 53246
- DiffServ Settings**

Parameter	Value
B8 DSCP(Hex)	FC
DSCP Mask (Hex)	70
SIG DSCP (Hex)	28
46 DSCP	63
DSCP Mask	28
SIG DSCP	28
- DHCP Settings**

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

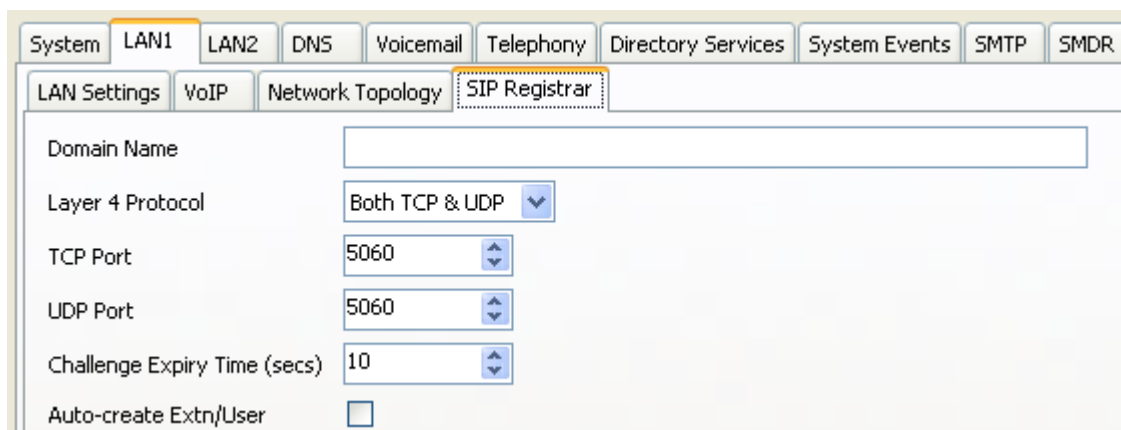
Select the **Network Topology** tab as shown in the following screen. For **Public IP Address**, enter the Avaya IP Office LAN1 IP address. Set the **Public Port** to 5060. In the sample configuration, the **Firewall/NAT Type** is set to “Open Internet”. With this configuration, STUN will not be used. During the testing, the **Binding Refresh Time** was varied (e.g., 30 seconds, 90 seconds to test SIP OPTIONS timing). Click the **OK** button.





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

Optionally, select the **SIP Registrar** tab. The following screen shows the settings used in the sample configuration.



### 4.3.3. 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 1.1.1.63, the IP Address of the PC running the Voicemail Pro software, as shown in **Figure 1**. In the sample configuration, the “Callback” application of Avaya Voicemail Pro was used to allow Voicemail Pro to call out via the SIP Line to Verizon Business when a message is left in a voice mailbox. The **SIP Settings** shown in the screen below enable IP Office to populate the SIP headers for an outbound “callback” call from Voicemail Pro, similar to the way the fields with these same names apply to calls made from telephone users (e.g., see Section 4.5). Other parameters on this screen may be set according to customer requirements.



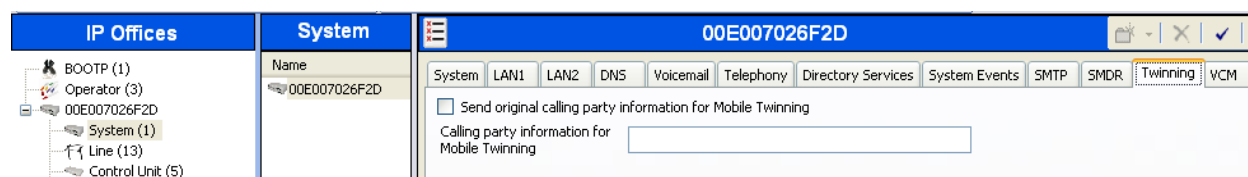
#### 4.3.4. System Telephony Configuration

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. 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.

#### 4.3.5. 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 (Section 4.4), the true identity of a PSTN caller can be presented to the twinning destination (e.g., a user’s mobile phone) when a call is twinned out via the Verizon Business IP Trunk service.



## 4.4. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 6.1. Since IP Office Release 6.1 introduced new SIP Line parameters and re-oriented existing parameters, this section has the most substantive changes in these Application Notes, compared to the configuration documented in references [JRR-IPOR6] and [JRR-IPOR61].

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.

### 4.4.1. SIP Line - SIP Line Tab

The **SIP Line** tab in the Details pane is shown below for Line Number 7, used for the Verizon Business IP Trunk service. The **ITSP Domain Name** is configured to the IP Office LAN1 address (1.1.1.2) so that IP Office uses 1.1.1.2 as the host portion of SIP headers such as the From header and Diversion header. By default, the **In Service** and **Check OOS** boxes are checked. In the sample configuration, IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office will use the **Binding Refresh Time** for LAN1, as shown in Section 4.3.2. See Section 4.10 for additional information related to configuring the periodicity of SIP OPTIONS.

The **Send Caller ID** parameter is set to “Diversion Header”. With this setting and the related configuration in Section 4.3.5, IP Office will include the Diversion Header for calls that are directed via Mobile Twinning out the SIP Line to Verizon. The Diversion Header will contain the number associated with the Twinning user, allowing Verizon to admit the call, and the From Header will be populated with the true calling party identity, allowing the twinning destination (e.g., mobile phone) to see the true caller id. 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”. Click **OK** (not shown).

The area of the screen entitled **REFER Support** is new for IP Office Release 6.1. In the following screen, “Always” has been selected from the drop-down menu for the **Incoming** and **Outgoing** parameters, to enable use of SIP REFER in the sample configuration.

**SIP Line - Line 7**

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

Line Number: 7

ITSP Domain Name: 1.1.1.2

Prefix:

National Prefix: 0

Country Code:

International Prefix: 00

Send Caller ID: Diversion Header

In Service: ☒

Use Tel URI: ☐

Check OOS: ☒

Call Routing Method: Request URI

Originator number for forwarded and twinning calls:

☒ REFER Support

Incoming: Always

Outgoing: Always

#### 4.4.2. SIP Line - Transport Tab

Select the **Transport** tab. This tab is new 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 Verizon domain provided by Verizon Business. As shown in **Figure 1**, this domain is “pcelban0001.avayalincroft.globalipcom.com”. Optionally, the domain configured in the ITSP Proxy Address field can be suffixed with a number. For example, “pcelban0001.avayalincroft.globalipcom.com(4)” may be entered. In the **Network Configuration** area, UDP is selected as the **Layer 4 Protocol**. Since DNS SRV will be used, the **Send Port** can retain the default value 5060. The port to which IP Office sends SIP messages will be determined via the DNS procedures. The **Use Network Topology Info** parameter is set to “LAN 1”. This associates the SIP Line with the parameters in the **System → LAN1 → Network Topology** tab. The **Explicit DNS Server(s)** is configured with the DNS Server IP address provided by Verizon Business, which is 172.30.209.4 in the sample configuration.

The screenshot shows the 'SIP Line - Line 7\*' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' field contains 'pcelban0001.avayalincroft.globalipcom.com'. The 'Network Configuration' section includes 'Layer 4 Protocol' set to 'UDP', 'Send Port' set to '5060', 'Use Network Topology Info' set to 'LAN 1', and 'Listen Port' set to '5060'. The 'Explicit DNS Server(s)' field shows the IP address '172.30.209.4'. The 'Calls Route via Registrar' checkbox is checked. The 'Separate Registrar' field is empty.

SIP Line - Line 7*	
SIP Line   <b>Transport</b>   SIP URI   VoIP   T38 Fax   SIP Credentials	
ITSP Proxy Address: pcelban0001.avayalincroft.globalipcom.com	
<b>Network Configuration</b>	
Layer 4 Protocol: UDP	Send Port: 5060
Use Network Topology Info: LAN 1	Listen Port: 5060
Explicit DNS Server(s): 172 . 30 . 209 . 4    0 . 0 . 0 . 0	
Calls Route via Registrar: <input checked="" type="checkbox"/>	
Separate Registrar:	

### 4.4.3. 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 example screen below, a previously configured entry is edited. “Use Internal Data” is selected for the **Local URI**, **Contact**, and **Display Name**. Information configured on the SIP Tab for individual users will be used to populate the SIP headers. The **PAI** parameter is new for IP Office Release 6.1, and the value “None” is shown selected from the drop-down menu. With PAI set to “none”, IP Office Release 6.1 will behave like IP Office Release 6.0 with respect to the SIP P-Asserted-Identity header (e.g., IP Office will not include a PAI header for an outbound call unless privacy is asserted). The **Registration** parameter is set to the default “0: <None>” since Verizon Business IP Trunk service does not require registration. The **Incoming Group** parameter, set here to 7, will be referenced when configuring Incoming Call Routes to map inbound SIP trunk calls to IP Office destinations in Section 4.7. The **Outgoing Group** parameter, set here to 8, will be used for routing outbound calls to Verizon via the Short Codes (Section 4.6) or ARS configuration (Section 4.8). The **Max Calls per Channel** parameter, configured here to 10, sets the maximum number of simultaneous calls that can use the URI before IP Office returns busy to any further calls. Click **OK**.

**SIP Line - Line 7\***

Tabs: SIP Line | Transport | **SIP URI** | VoIP | T38 Fax | SIP Credentials

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	7 8	1.1.1.2				None	0: <Non...	10
2	7 8	1.1.1.2	7329450234	7329450234		None	0: <Non...	10
3	7 8	1.1.1.2	7329450228			None	0: <Non...	10
4	7 8	1.1.1.2	7329450238	7329450238	7329450238	None	0: <Non...	10
5	7 8	1.1.1.2	7329450239	7329450239	7329450239	None	0: <Non...	10
6	7 8	1.1.1.2	7329450235	7329450235	7329450235	None	0: <Non...	10

Buttons: Add..., Remove, Edit...

**Edit Channel**

Via: 1.1.1.2

Local URI: Use Internal Data

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: None

Registration: 0: <None>

Incoming Group: 7

Outgoing Group: 8

Max Calls per Channel: 10

Buttons: OK, Cancel

In the sample configuration, the single SIP URI shown above was sufficient to allow incoming calls for Verizon DID numbers destined for specific IP Office users or IP Office hunt groups. The calls are accepted by IP Office since the incoming number will match the SIP Name configured for the user or hunt group that is the destination for the call. For service numbers, such as a DID number routed directly to voicemail, or a DID number routed to an auto-attendant service on Voicemail Pro, the DID numbers that IP Office should admit can be entered into the **Local URI** and **Contact** fields instead of “Use Internal Data”.

#### 4.4.4. SIP Line - VoIP Tab

Select the **VoIP** tab. In the sample configuration, the **Compression Mode** was configured using the **Advanced** button, allowing an explicit ordered list of codecs to be specified. Place a check mark next to the **G.729(a) 8K CS-ACELP** and **G.711 ULAW 64K** codecs to cause IP Office to include both G.729a and G.711MU in the Session Description Protocol (SDP) offer, in that order. 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 setting. In the sample configuration, Verizon also 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. Since the Verizon Business IP Trunk service does not support T.38 fax, the **Fax Transport Support** parameter is not checked, and the **T38 Fax** tab need not be visited. Since the Verizon Business IP Trunk service does not require registration, the **SIP Credentials** tab need not be visited. Click **OK** (not shown).

SIP Line - Line 7	
<div> <div>SIP Line</div> <div>Transport</div> <div>SIP URI</div> <div>VoIP</div> <div>T38 Fax</div> <div>SIP Credentials</div> </div>	
Compression Mode <div>Advanced</div>	<div> <input checked="" type="checkbox"/> G.729(a) 8K CS-ACELP  <input checked="" type="checkbox"/> G.711 ULAW 64K  <input type="checkbox"/> G.711 ALAW 64K  <input type="checkbox"/> G.723.1 6K3 MP-MLQ         </div>
Call Initiation Timeout (s) <div>6</div>	<input type="checkbox"/> VoIP Silence Suppression <input type="checkbox"/> Fax Transport Support <input checked="" type="checkbox"/> Re-invite Supported <input type="checkbox"/> Use Offerer's Preferred Codec
DTMF Support <div>RFC2833</div>	

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

In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests of brevity, not all users and extensions 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.

### 4.5.1. User 201

The following screen shows the **User** tab for User 201. As shown in **Figure 1**, this user corresponds to the digital telephone 5410.

IP Offices		User		Extn201: 201	
BOOTP (2)		Name	Extens	Menu Programming	Mobility
Operator (3)		Extn201	201	User	Voicemail
00E007026F2D		Extn202	202	DND	ShortCodes
System (1)		Extn203	203	Source Numbers	Telephony
Line (13)		Extn204	204	Forwarding	Dial
Control Unit (5)		Extn205	205		
Extension (38)		Extn206	206	Name	Extn201
User (38)		Extn207	207	Password	
HuntGroup (1)		Extn208	208	Confirm Password	
Short Code (62)		Extn210	210	Full Name	Joey Dig5410
Service (0)		Extn211	211	Extension	201
RAS (1)		Extn212	212	Locale	
Incoming Call Route (14)		Extn213	213	Priority	5
WanPort (0)		Extn214	214	System Phone Rights	None
Directory (0)		Extn215	215	Profile	Basic User
Time Profile (0)		Extn216	216	<input type="checkbox"/> Receptionist	
Firewall Profile (1)		Extn30000	30000	<input type="checkbox"/> Enable SoftPhone	
IP Route (4)		Extn30025	30025	<input type="checkbox"/> Enable one-X Portal Services	
Account Code (0)		Extn30026	30026	<input type="checkbox"/> Enable one-X TeleCommuter	
License (55)		Extn30027	30027	<input type="checkbox"/> Ex Directory	
Tunnel (0)		Extn30028	30028		
Logical LAN (0)		Extn30029	30029		
User Rights (8)		Extn30030	30030		
ARS (2)		Extn50000	50000		
RAS Location Request (0)		Extn51007	51007		
E911 System (1)		Extn51010	51010		
		Extn51020	51020		
		Extn51021	51021		
		Extn51022	51022		

Extn201: 201	
Device	Avaya 5410
Type	

The following screen shows the **SIP** tab for User 201. The **SIP Name** and **Contact** parameters are configured with the DID number of the user, 7329450231. These parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk calls, and allow matching of the SIP URI for incoming calls, without having to enter this number as an explicit SIP URI for the SIP Line. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network. See Section 4.6 for a method of using a short code (rather than static user provisioning) to place an anonymous call.

Extn201: 201*	
SIP Name	7329450231
SIP Display Name (Alias)	Joey-Dig5410
Contact	7329450231
<input type="checkbox"/> Anonymous	

From **Figure 1**, note that user 201 will use the Mobile Twinning feature. The following screen shows the **Mobility** tab for User 201. 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 917326870755. Other options can be set according to customer requirements.

**Ext201: 201**

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

☐ Internal Twinning

Twinned Handset: <None>

Maximum Number of Calls: 1

☐ Twin Bridge Appearances

☐ Twin Coverage Appearances

☐ Twin Line Appearances

☒ Mobility Features

☒ Mobile Twinning

Twinned Mobile Number (including dial access code): 917326870755

Twinning Time Profile: <None>

Mobile Dial Delay (secs): 2

Mobile Answer Guard (secs): 0

☐ Hunt group calls eligible for mobile twinning

☐ Forwarded calls eligible for mobile twinning

☐ Twin When Logged Out


☐ one-X Mobile Client

☒ Mobile Call Control

☒ Mobile Callback



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. As stated in Section 4.1, the Avaya 5410 telephone user with extension 201 is connected to port 1 of the digital module.

IP Offices		Extension				Digital Extension: 49 201																																																																	
<ul style="list-style-type: none"> <li>BOOTP (2)</li> <li>Operator (3)</li> <li>00E007026F2D</li> <li>System (1)</li> <li>Line (13)</li> <li>Control Unit (5)</li> <li>Extension (38)</li> <li>User (38)</li> <li>HuntGroup (1)</li> <li>Short Code (62)</li> <li>Service (0)</li> <li>RAS (1)</li> <li>Incoming Call Route (14)</li> <li>WanPort (0)</li> <li>Directory (0)</li> <li>Time Profile (0)</li> <li>Firewall Profile (1)</li> <li>IP Route (4)</li> <li>Account Code (0)</li> </ul>	<table border="1"> <thead> <tr> <th>Id</th> <th>Extension</th> <th>Module</th> <th>Port</th> </tr> </thead> <tbody> <tr><td>8020</td><td></td><td>0</td><td>0</td></tr> <tr><td>8021</td><td></td><td>0</td><td>0</td></tr> <tr><td>49</td><td>201</td><td>BD3</td><td>1</td></tr> <tr><td>50</td><td>202</td><td>BD3</td><td>2</td></tr> <tr><td>51</td><td>203</td><td>BD3</td><td>3</td></tr> <tr><td>52</td><td>204</td><td>BD3</td><td>4</td></tr> <tr><td>53</td><td>205</td><td>BD3</td><td>5</td></tr> <tr><td>54</td><td>206</td><td>BD3</td><td>6</td></tr> <tr><td>55</td><td>207</td><td>BD3</td><td>7</td></tr> <tr><td>56</td><td>208</td><td>BD3</td><td>8</td></tr> <tr><td>73</td><td>209</td><td>BP4</td><td>1</td></tr> <tr><td>74</td><td>210</td><td>BP4</td><td>2</td></tr> <tr><td>75</td><td>211</td><td>BP4</td><td>3</td></tr> <tr><td>76</td><td>212</td><td>BP4</td><td>4</td></tr> <tr><td>77</td><td>213</td><td>BP4</td><td>5</td></tr> <tr><td>78</td><td>214</td><td>BP4</td><td>6</td></tr> </tbody> </table>	Id	Extension	Module	Port	8020		0	0	8021		0	0	49	201	BD3	1	50	202	BD3	2	51	203	BD3	3	52	204	BD3	4	53	205	BD3	5	54	206	BD3	6	55	207	BD3	7	56	208	BD3	8	73	209	BP4	1	74	210	BP4	2	75	211	BP4	3	76	212	BP4	4	77	213	BP4	5	78	214	BP4	6	<div>Extn</div> <div>Extension Id <input type="text" value="49"/></div> <div>Base Extension <input type="text" value="201"/></div> <div>Caller Display Type <input type="text" value="Off"/></div> <div>Reset Volume After Calls <input type="checkbox"/></div> <div>Device type  <input type="text" value="Avaya 5410"/></div> <div>Module <input type="text" value="BD3"/></div> <div>Port <input type="text" value="1"/></div> <div>Disable Speakerphone <input type="checkbox"/></div>	
Id	Extension	Module	Port																																																																				
8020		0	0																																																																				
8021		0	0																																																																				
49	201	BD3	1																																																																				
50	202	BD3	2																																																																				
51	203	BD3	3																																																																				
52	204	BD3	4																																																																				
53	205	BD3	5																																																																				
54	206	BD3	6																																																																				
55	207	BD3	7																																																																				
56	208	BD3	8																																																																				
73	209	BP4	1																																																																				
74	210	BP4	2																																																																				
75	211	BP4	3																																																																				
76	212	BP4	4																																																																				
77	213	BP4	5																																																																				
78	214	BP4	6																																																																				

### 4.5.2. User 30026

The following screen shows the **User** tab for User 30026. This user corresponds to an Avaya 1616 IP Telephone that will be granted “Power User” and Avaya IP Office Softphone features. The **Profile** parameter is set to “Power User”. The **Enable Softphone** box is checked, along with other advanced capabilities.

Ext30026: 30026

Menu Programming | Mobility | Phone Manager Options | Hunt Group Membership | Announcements | SIP | Personal Directory |

User | Voicemail | DND | ShortCodes | Source Numbers | Telephony | Forwarding | Dial In | Voice Recording | Button Programming |

Name: Ext30026

Password: \*\*\*\*\*

Confirm Password: \*\*\*\*\*

Full Name: Monica IP-1616

Extension: 30026

Locale: [dropdown]

Priority: 5

System Phone Rights: None

Profile: Power User

☐ Receptionist

☒ Enable SoftPhone

☒ Enable one-X Portal Services

☒ Enable one-X TeleCommuter

☐ Ex Directory

Device Type: Avaya 1616

Like the user with extension 201, the **SIP** tab for the user with extension 30026 is configured with a **SIP Name** and **Contact** specifying the user’s Verizon Business DID number.

Ext30026: 30026

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 |

SIP Name: 7329450229

SIP Display Name (Alias): Monica-IP-1616

Contact: 7329450229

☐ Anonymous

The following screen shows the **Voicemail** tab for the user with extension 30026. 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 Trunk 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, to test DTMF using RFC 2833, and to test assignment of a Verizon DID number to the “Voicemail Collect” feature (e.g., via the \*17 short code shown in Section 4.6).

The screenshot shows the 'Ext30026: 30026' configuration window with the 'Voicemail' tab selected. The window has a title bar with a menu icon, the title 'Ext30026: 30026', and standard window controls. Below the title bar is a navigation bar with tabs: Menu Programming, Mobility, Phone Manager Options, Hunt Group Membership, Announcements, SIP, Personal Directory, User, Voicemail (selected), DND, ShortCodes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, and Button Programming. The main content area is divided into two columns. The left column contains text boxes for 'Voicemail Code' (filled with '\*\*\*\*\*'), 'Confirm Voicemail Code' (filled with '\*\*\*\*\*'), and 'Voicemail Email'. The right column contains a list of checkboxes: 'Voicemail On' (checked), 'Voicemail Help' (unchecked), 'Voicemail Ringback' (unchecked), 'Voicemail Email Reading' (unchecked), and 'UMS Web Services' (checked). Below these is a 'Voicemail Email' section with radio buttons for 'Off' (selected), 'Copy', 'Forward', and 'Alert'. At the bottom is a 'DTMF Breakout' section with three text boxes: 'Reception / Breakout (DTMF 0)' (filled with 'System Default ()'), 'Breakout (DTMF 2)' (filled with 'System Default ()'), and 'Breakout (DTMF 3)' (filled with 'System Default ()').

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

The screenshot shows the 'Ext30026: 30026\*' configuration window with the 'Supervisor Settings' tab selected. The window has a title bar with a menu icon, the title 'Ext30026: 30026\*', and standard window controls. Below the title bar is a navigation bar with tabs: SIP, Personal Directory, Button Programming, Menu Programming, Mobility, Phone Manager Options, Hunt Group Membership, Announcements, User, Voicemail, DND, ShortCodes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, Call Settings, Supervisor Settings (selected), Multi-line Options, and Call Log. The main content area is divided into two columns. The left column contains text boxes for 'Login Code' (filled with '\*\*\*\*'), 'Login Idle Period (secs)', 'Monitor Group' (filled with '<None>'), 'Coverage Group' (filled with '<None>'), and 'Status on No-Answer' (filled with 'Logged On (No change)'). Below these is a 'Reset Longest Idle Time' section with radio buttons for 'All Calls' (selected) and 'External Incoming'. The right column contains a list of checkboxes: 'Force Login' (unchecked), 'Force Account Code' (unchecked), 'Outgoing Call Bar' (unchecked), 'Inhibit Off-Switch Forward/Transfer' (unchecked), 'Can Intrude' (unchecked), 'Cannot be Intruded' (checked), 'Can Trace Calls' (unchecked), and 'CCR Agent' (unchecked).

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

The screenshot shows the 'Ext30026: 30026' configuration window with the 'Call Settings' tab selected. The window has a menu bar with options: Button Programming, Menu Programming, Mobility, Phone Manager Options, Hunt Group Membership, Announcements, SIP, Personal Directory, User, Voicemail, DND, ShortCodes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording. The 'Call Settings' sub-tab is active, showing options for Outside Call Sequence, Inside Call Sequence, Ringback Sequence, No Answer Time (secs), Wrap-up Time (secs), Transfer Return Time (secs), and Call Cost Mark-Up. On the right, there are checkboxes for 'Call Waiting On' (checked), 'Answer Call Waiting On Hold (Analogue)' (checked), 'Busy On Held' (unchecked), and 'Offhook Station' (checked).

The following screen shows the **Source Numbers** tab for the user with extension 30026. Although the Voicemail Pro configuration is beyond the scope of these Application Notes, the “Callback” feature has been enabled on Voicemail Pro for this user, and the Source Number “P917326870755” has been previously added. With this configuration, when a message is left in this user’s Voicemail Pro mailbox, a “callback” call will be initiated to “917326870755”. The callback call will be sent to Verizon via SIP Line 7, and the From and Contact headers in the SIP INVITE will be populated with the information configured in the **System → Voicemail** tab shown in Section 4.3.3. It is possible (and more typical) for the end user to configure callback numbers via the Voicemail Pro Telephony User Interface, rather than the administrator configuring a callback number via the “P” Source Number in IP Office, as shown below.

To add a new Source Number, Press the **Add...** button to the right of the list of any previously configured Source Numbers. To edit an existing Source Number, select the Source Number from the list, and click **Edit...** When finished, click **OK**.

The screenshot shows the 'Ext30026: 30026\*' configuration window with the 'Source Numbers' tab selected. The window has a menu bar with options: Menu Programming, Mobility, Phone Manager Options, Hunt Group Membership, Announcements, SIP, Personal Directory, User, Voicemail, DND, ShortCodes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, Button Programming. The 'Source Numbers' sub-tab is active, showing a list of source numbers: V30026 and P917326870755. To the right of the list are buttons for 'Add...', 'Remove', and 'Edit...'.

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

The screenshot displays the configuration interface for an H323 extension. The title bar reads "H323 Extension: 8014 30026". Below the title bar, there are two tabs: "Extn" and "VoIP", with "VoIP" being the active tab. The main configuration area is divided into two columns. The left column contains fields for "IP Address" (0 . 0 . 0 . 0), "MAC Address" (00 00 00 00 00 00), "Compression Mode" (Automatic Select), "TDM->IP Gain" (Default), "IP->TDM Gain" (Default), and "Supplementary Services" (None). The right column contains a list of checkboxes: "VoIP Silence Suppression", "Enable Faststart for non-Avaya IP phones", "Out Of Band DTMF" (checked), "Local Tones", "Allow Direct Media Path" (checked), "Reserve Avaya IP endpoint license", and "Reserve 3rd party IP endpoint license".

### 4.5.3. 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 hunt group 200. This hunt group was configured to contain the two digital telephones x201 and x203 in **Figure 1**. These telephones will both ring when the hunt group number is called, due to the **Ring Mode** setting "Collective". Click the **Edit** button to change the **User List**.

**Collective Group Main: 200**

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

Name: Main  
 Extension: 200  
 Ring Mode: Collective  
 Overflow Mode: Group  
 Hold Music Source: No Change  
 Agent's Status on No-Answer Applies To: None

☐ CCR Agent Group

No Answer Time (secs): System Default (15)  
 Overflow Time (secs): Off  
 Voicemail Answer Time (secs): 45

**User List**

Extension	Name
<input checked="" type="checkbox"/>	201 Extn201
<input type="checkbox"/>	202 Extn202
<input checked="" type="checkbox"/>	203 Extn203
<input type="checkbox"/>	204 Extn204
<input type="checkbox"/>	205 Extn205
<input type="checkbox"/>	206 Extn206
<input type="checkbox"/>	207 Extn207
<input type="checkbox"/>	208 Extn208
<input type="checkbox"/>	209 Justa Fax

**Overflow Group List**

Group Name
------------

Edit... Remove... Add... Remove...

The following screen shows the **SIP** tab for hunt group 200. The **SIP Name** and **Contact** are configured with Verizon DID 7329450236. Later, in Section 4.7, an Incoming Call Route will map 7329450236 to this hunt group.

**Collective Group Main: 200**

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

SIP Name: 7329450236  
 SIP Display Name (Alias): Main  
 Contact: 7329450236

☐ Anonymous

The following screen shows the **Hunt Group** tab for another hunt group 30200. This hunt group was configured to contain the two IP telephones x30025 and x30026 in **Figure 1**. These telephones will ring sequentially when the hunt group number is called, due to the **Ring Mode** setting "Sequential". That is, extension 30025 will ring first. If unanswered, extension 30026 will ring, and so on (for larger groups).

**Sequential Group IP Stations Seq: 30200**

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

Name: IP Stations Seq ☐ CCR Agent Group

Extension: 30200

Ring Mode: Sequential

Overflow Mode: Group

Hold Music Source: No Change

Agent's Status on No-Answer Applies To: None

No Answer Time (secs): System Default (15)

Overflow Time (secs): Off

Voicemail Answer Time (secs): 45

**User List**

Extension	Name
<input checked="" type="checkbox"/> 30025	Extn30025
<input checked="" type="checkbox"/> 30026	Extn30026

**Overflow Group List**

Group Name
------------

Edit... Remove Add... Remove

The following screen shows the **SIP** tab for hunt group 30200. The **SIP Name** and **Contact** are configured with Verizon DID 7329450237. Later, in Section 4.7, an incoming call route will map this same Verizon DID number to this hunt group.

**Sequential Group IP Stations Seq: 30200**

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

SIP Name: 7329450237

SIP Display Name (Alias): IP Stations Seq

Contact: 7329450237

☐ Anonymous

## 4.6. Short Codes

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

In the screen shown below, the short code “9N;” is illustrated. The **Code** parameter is set to “9N;”. 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” with the text string beginning with @ in quotes. Below, the Verizon provided domain shown in **Figure 1** is configured. 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 8, matching the number of the **Outgoing Group** configured on the **SIP URI** tab of SIP Line 7 to Verizon Business (Section 4.4).

This simple short code will allow an IP Office user to dial the digit 9 followed by any telephone number, symbolized by the letter N, to reach the SIP Line to Verizon business. N can be any number such as a 10-digit number, a 1+10 digit number, a toll free number, directory assistance (e.g., 411), etc. This short code approach has the virtue of simplicity, but does not provide for alternate routing or an awareness of end of user dialing. When a users dial 9 plus the number, IP Office must wait for an end of dialing timeout before sending the SIP INVITE to Verizon Business. Click the **OK** button (not shown).

The screenshot displays the IP Office configuration interface. On the left, the 'IP Offices' navigation pane shows a tree structure with 'Short Code (62)' selected. The main area is divided into two panes. The left pane, titled 'Short Code', lists various short codes with their corresponding telephone numbers, such as '9X\*19', '9X\*20\*N# N', and '9X\*37\*N# N'. The right pane, titled '9N:: Dial', shows the configuration for a specific short code. The 'Code' field is set to '9N;', the 'Feature' is set to 'Dial', the 'Telephone Number' is set to 'N"@pcelban0001.avayaipcom.', the 'Line Group Id' is set to '8', and the 'Locale' is set to 'United States (US English)'. The 'Force Account Code' checkbox is unchecked.

Optionally, add or edit a short code that can be used to access the SIP Line anonymously. In the screen shown below, the short code “8N;” is illustrated. This short code is similar to the “9N;” short code except that the Verizon IP Address rather than the domain is entered in the **Telephone Number** field. This is done for variety; either method can be used. The Telephone Number field begins with the letter **W**, which means “withhold the outgoing calling line identification”. In the case of the SIP Line to Verizon documented in these Application Notes, when a user dials 8 plus the number, IP Office will include the user’s telephone number in the P-Asserted-Identity (PAI) header along with “Privacy: Id”. Verizon will allow the call due to the presence of a valid DID in the PAI header, but will prevent presentation of the caller id to the called PSTN destination.



IP Offices	Short Code	8N:: Dial																													
<ul style="list-style-type: none"> <li>BOOTP (2)</li> <li>Operator (3)</li> <li>00E007026F2D</li> <li>System (1)</li> <li>Line (13)</li> <li>Control Unit (5)</li> <li>Extension (38)</li> <li>User (38)</li> <li>HuntGroup (1)</li> <li>Short Code (62)</li> <li>Service (0)</li> <li>RAS (1)</li> </ul>	<table border="1"> <thead> <tr> <th>Code</th> <th>Telephone Nun</th> </tr> </thead> <tbody> <tr><td>*19</td><td></td></tr> <tr><td>*20*N#</td><td>N</td></tr> <tr><td>*21*N#</td><td>N</td></tr> <tr><td>*29</td><td></td></tr> <tr><td>*30</td><td></td></tr> <tr><td>*31</td><td></td></tr> <tr><td>*32*N#</td><td>N</td></tr> <tr><td>*33*N#</td><td>N</td></tr> <tr><td>*34N;</td><td>N</td></tr> <tr><td>*35*N#</td><td>N</td></tr> </tbody> </table>	Code	Telephone Nun	*19		*20*N#	N	*21*N#	N	*29		*30		*31		*32*N#	N	*33*N#	N	*34N;	N	*35*N#	N	<table border="1"> <thead> <tr> <th>Short Code</th> </tr> </thead> <tbody> <tr> <td>Code: 8N;</td> </tr> <tr> <td>Feature: Dial</td> </tr> <tr> <td>Telephone Number: WN"@172.30.209.21"</td> </tr> <tr> <td>Line Group Id: 8</td> </tr> <tr> <td>Locale:</td> </tr> <tr> <td>Force Account Code: <input type="checkbox"/></td> </tr> </tbody> </table>	Short Code	Code: 8N;	Feature: Dial	Telephone Number: WN"@172.30.209.21"	Line Group Id: 8	Locale:	Force Account Code: <input type="checkbox"/>
Code	Telephone Nun																														
*19																															
*20*N#	N																														
*21*N#	N																														
*29																															
*30																															
*31																															
*32*N#	N																														
*33*N#	N																														
*34N;	N																														
*35*N#	N																														
Short Code																															
Code: 8N;																															
Feature: Dial																															
Telephone Number: WN"@172.30.209.21"																															
Line Group Id: 8																															
Locale:																															
Force Account Code: <input type="checkbox"/>																															

The simple “9N;” and “8N;” short codes illustrated previously do not provide a means of alternate routing if the primary Verizon SIP line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. In the following example screen, the short code 7N is illustrated for access to ARS. When the IP Office user dials 7 plus any number N, rather than being directed to a specific **Line Group Id**, the call is directed to **Line Group ID** “50: Main”, configurable via ARS. See Section 4.8 for example ARS route configuration for “50: Main” as well as a backup route.

IP Offices	Short Code	7N: Dial																													
<ul style="list-style-type: none"> <li>BOOTP (2)</li> <li>Operator (3)</li> <li>00E007026F2D</li> <li>System (1)</li> <li>Line (13)</li> <li>Control Unit (5)</li> <li>Extension (38)</li> <li>User (38)</li> <li>HuntGroup (1)</li> <li>Short Code (62)</li> <li>Service (0)</li> <li>RAS (1)</li> </ul>	<table border="1"> <thead> <tr> <th>Code</th> <th>Telephone Nun</th> </tr> </thead> <tbody> <tr><td>*19</td><td></td></tr> <tr><td>*20*N#</td><td>N</td></tr> <tr><td>*21*N#</td><td>N</td></tr> <tr><td>*29</td><td></td></tr> <tr><td>*30</td><td></td></tr> <tr><td>*31</td><td></td></tr> <tr><td>*32*N#</td><td>N</td></tr> <tr><td>*33*N#</td><td>N</td></tr> <tr><td>*34N;</td><td>N</td></tr> <tr><td>*35*N#</td><td>N</td></tr> </tbody> </table>	Code	Telephone Nun	*19		*20*N#	N	*21*N#	N	*29		*30		*31		*32*N#	N	*33*N#	N	*34N;	N	*35*N#	N	<table border="1"> <thead> <tr> <th>Short Code</th> </tr> </thead> <tbody> <tr> <td>Code: 7N</td> </tr> <tr> <td>Feature: Dial</td> </tr> <tr> <td>Telephone Number: N</td> </tr> <tr> <td>Line Group Id: 50: Main</td> </tr> <tr> <td>Locale:</td> </tr> <tr> <td>Force Account Code: <input type="checkbox"/></td> </tr> </tbody> </table>	Short Code	Code: 7N	Feature: Dial	Telephone Number: N	Line Group Id: 50: Main	Locale:	Force Account Code: <input type="checkbox"/>
Code	Telephone Nun																														
*19																															
*20*N#	N																														
*21*N#	N																														
*29																															
*30																															
*31																															
*32*N#	N																														
*33*N#	N																														
*34N;	N																														
*35*N#	N																														
Short Code																															
Code: 7N																															
Feature: Dial																															
Telephone Number: N																															
Line Group Id: 50: Main																															
Locale:																															
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 DID 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 4.7.

IP Offices	Short Code	*17: Voicemail Collect																																							
<ul style="list-style-type: none"> <li>BOOTP (2)</li> <li>Operator (3)</li> <li>00E007026F2D</li> <li>System (1)</li> <li>Line (13)</li> <li>Control Unit (5)</li> <li>Extension (38)</li> <li>User (38)</li> <li>HuntGroup (1)</li> <li>Short Code (62)</li> <li>Service (0)</li> <li>RAS (1)</li> <li>Incoming Call Route (14)</li> <li>WanPort (0)</li> <li>Directory (0)</li> <li>Time Profile (0)</li> <li>Firewall Profile (1)</li> <li>IP Route (4)</li> </ul>	<table border="1"> <thead> <tr> <th>Code</th> <th>Telephone Nun</th> </tr> </thead> <tbody> <tr><td>*03</td><td></td></tr> <tr><td>*04</td><td></td></tr> <tr><td>*05</td><td></td></tr> <tr><td>*06</td><td></td></tr> <tr><td>*07*N#</td><td>N</td></tr> <tr><td>*08</td><td></td></tr> <tr><td>*09</td><td></td></tr> <tr><td>*10*N#</td><td>N</td></tr> <tr><td>*11*N#</td><td>N</td></tr> <tr><td>*12*N#</td><td>N</td></tr> <tr><td>*13*N#</td><td>N</td></tr> <tr><td>*14*N#</td><td>N</td></tr> <tr><td>*15</td><td></td></tr> <tr><td>*16</td><td></td></tr> <tr><td>*17</td><td>?U</td></tr> </tbody> </table>	Code	Telephone Nun	*03		*04		*05		*06		*07*N#	N	*08		*09		*10*N#	N	*11*N#	N	*12*N#	N	*13*N#	N	*14*N#	N	*15		*16		*17	?U	<table border="1"> <thead> <tr> <th>Short Code</th> </tr> </thead> <tbody> <tr> <td>Code: *17</td> </tr> <tr> <td>Feature: Voicemail Collect</td> </tr> <tr> <td>Telephone Number: ?U</td> </tr> <tr> <td>Line Group Id: 0</td> </tr> <tr> <td>Locale:</td> </tr> <tr> <td>Force Account Code: <input type="checkbox"/></td> </tr> </tbody> </table>	Short Code	Code: *17	Feature: Voicemail Collect	Telephone Number: ?U	Line Group Id: 0	Locale:	Force Account Code: <input type="checkbox"/>
Code	Telephone Nun																																								
*03																																									
*04																																									
*05																																									
*06																																									
*07*N#	N																																								
*08																																									
*09																																									
*10*N#	N																																								
*11*N#	N																																								
*12*N#	N																																								
*13*N#	N																																								
*14*N#	N																																								
*15																																									
*16																																									
*17	?U																																								
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 that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code** “\*99” is defined for **Feature** “Voicemail Collect” and Telephone Number “Attendant”. This short code will be used as a means to allow a Verizon DID to be programmed to route directly to a Voicemail Pro “Attendant” (auto-attendant)

application, via inclusion of this short code as the destination of an Incoming Call Route. See Section 4.7.

**\*99: Voicemail Collect**

Short Code

Code: \*99

Feature: Voicemail Collect

Telephone Number: "Attendant"

Line Group Id: 0

Locale:

Force Account Code: ☐

## 4.7. Incoming Call Routes

In this section, IP Office Incoming Call Routes are illustrated. Each Incoming Call Route will map a specific Verizon Business DID number to a destination user, group, or function on IP Office. To add an incoming call route, right click on **Incoming Call Route** in the Navigation pane, and select **New**. To edit an existing incoming call route, select **Incoming Call Route** in the Navigation pane, and the appropriate incoming call route to be configured in the Group pane.

In the screen shown below, the incoming call route for **Incoming Number** “7329450231” is illustrated. The **Line Group Id** is 7, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to Verizon Business in Section 4.4.

**IP Offices**

- BOOTP (2)
- Operator (3)
- 00E007026F2D
- System (1)
- Line (13)
- Control Unit (5)
- Extension (38)
- User (38)
- HuntGroup (1)
- Short Code (62)
- Service (0)
- RAS (1)
- Incoming Call Route (14)**
- WanPort (0)
- Directory (0)
- Time Profile (0)
- Firewall Profile (1)
- IP Route (4)

**Incoming Call Route**

Line Group Id	Incoming Number
0	
0	
7	8668506850
7	8668523221
7	8668512649
7	8668510107
7	7329450235
7	7329450233
7	7329450229
7	7329450228
7	7329450232
<b>7</b>	<b>7329450231</b>
17	9727289390
21	8668502380

**7 7329450231**

Standard | Voice Recording | Destinations

Bearer Capability: Any Voice

Line Group Id: 7

Incoming Number: 7329450231

Incoming Sub Address:

Incoming CLI:

Locale: United States (US English)

Priority: 1 - Low

Tag:

Hold Music Source: System Source

Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when a PSTN user dials 7329450231. As shown in **Table 1**, 7329450231 is the DID number associated with IP Office user extension 201.

7 7329450231			
Standard		Voice Recording	Destinations
	TimeProfile	Destination	Fallback Extension
▶	Default Value	201 Extn201	201 Extn201

In the screen shown below, the incoming call route for **Incoming Number** “7329450229” is illustrated. The **Line Group Id** is 7, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 4.4.

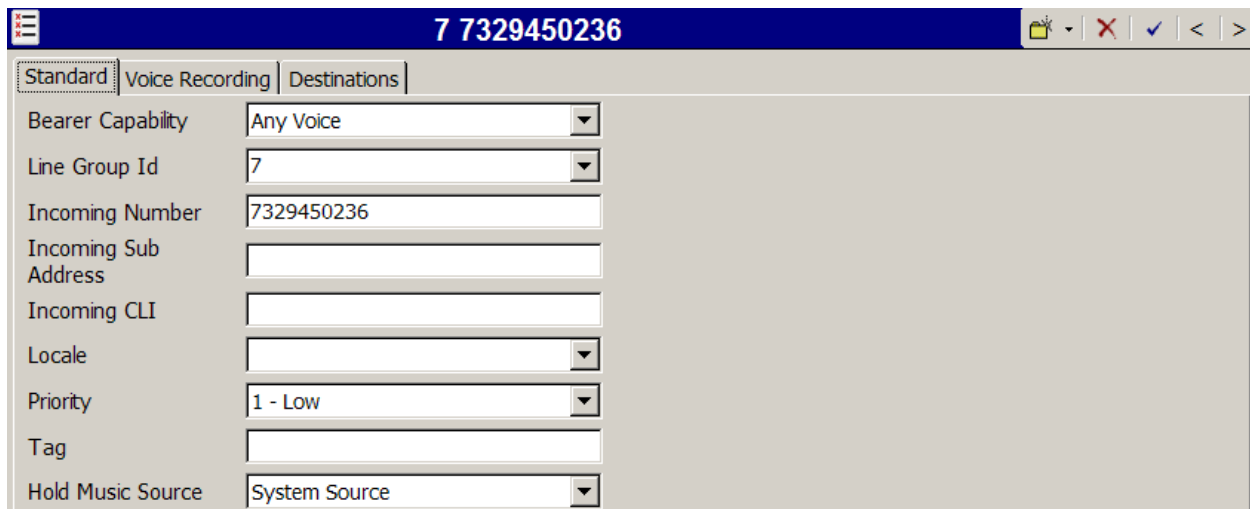
7 7329450229	
Standard   Voice Recording   Destinations	
Bearer Capability	Any Voice
Line Group Id	7
Incoming Number	7329450229
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when a PSTN user dials 7329450229. As shown in **Table 1**, 7329450229 is the DID number associated with IP Office user extension 30026.

7 7329450229			
Standard		Voice Recording	Destinations
	TimeProfile	Destination	Fallback Extension
▶	Default Value	30026 Extn30026	30026 Extn30026

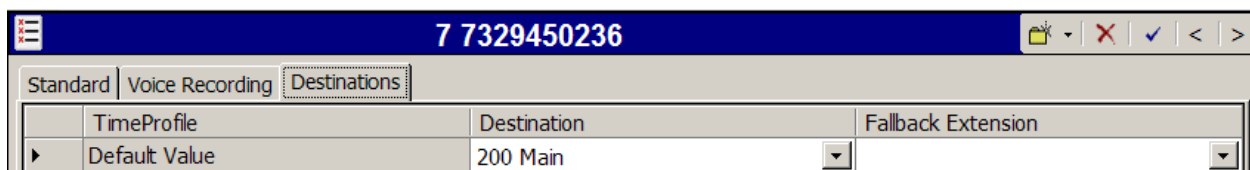
Incoming Call Routes for other direct mappings of DID numbers to IP Office users listed in **Table 1** are omitted here, but can be configured in the same fashion.

In the screen shown below, the incoming call route for **Incoming Number** “7329450236” is illustrated. The **Line Group Id** is 7, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 4.4.



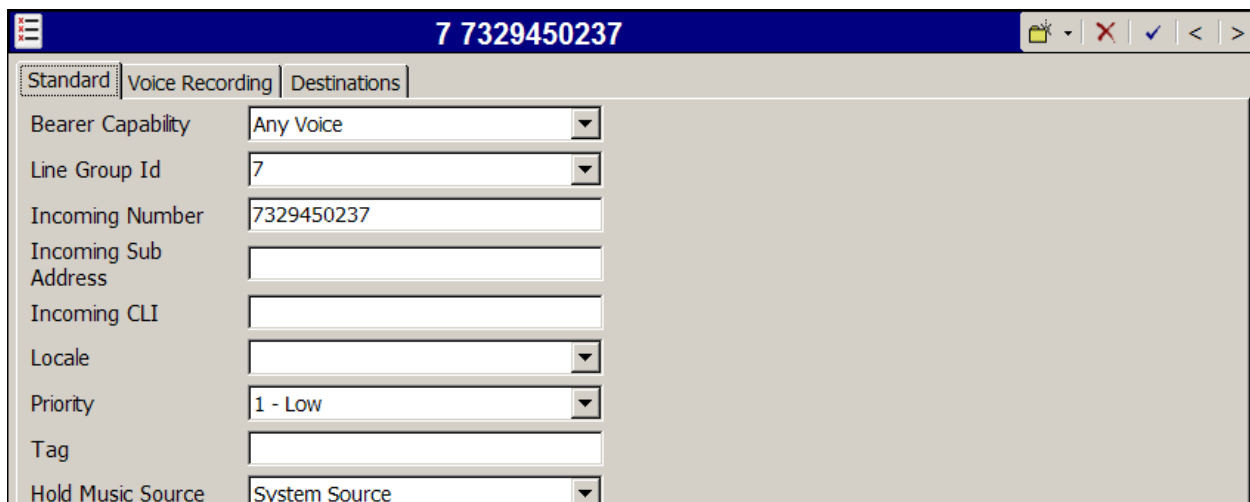
7 7329450236	
Standard	Voice Recording Destinations
Bearer Capability	Any Voice
Line Group Id	7
Incoming Number	7329450236
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when a PSTN user dials 7329450236. In this case, the destination is the hunt group “200 Main” whose configuration is shown in Section 4.5.3.



TimeProfile	Destination	Fallback Extension
Default Value	200 Main	

In the screen shown below, the incoming call route for **Incoming Number** “7329450237” is illustrated. The **Line Group Id** is 7, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 4.4.



7 7329450237	
Standard	Voice Recording Destinations
Bearer Capability	Any Voice
Line Group Id	7
Incoming Number	7329450237
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 extension to receive the call when a PSTN user dials 7329450237. In this case, the destination is the hunt group “30200 IP Stations Seq” whose configuration is shown in Section 4.5.3.

7 7329450237			
Standard   Voice Recording   Destinations			
	TimeProfile	Destination	Fallback Extension
▶	Default Value	30200 IP Stations Seq	

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. For example, the following **Destinations** tab for an incoming call route contains the **Destination** “\*17” entered manually. The dial string “\*17” is the short code for “Voicemail Collect”, as shown in Section 4.6. An incoming call to 732-945-0234 will be delivered directly to voice mail, allowing the caller to log-in to voicemail and access messages.

7 7329450234			
Standard   Voice Recording   Destinations			
	TimeProfile	Destination	Fallback Extension
▶	Default Value	*17	

Similar, the following **Destinations** tab for an incoming call route contains the **Destination** “\*99” entered manually. The dial string “\*99” is the short code for accessing the “Attendant” application on Voicemail Pro, as shown in Section 4.6. An incoming call to 732-945-0228 will be delivered directly to the Voicemail Pro “Attendant” application, which will allow the caller to be prompted with announcements, navigate via DTMF, and transfer to IP Office users. The configuration of the “Attendant” application on Voicemail Pro is outside the intended scope of these Application Notes.

7 7329450228			
Standard   Voice Recording   Destinations			
	TimeProfile	Destination	Fallback Extension
▶	Default Value	*99	*99

## 4.8. ARS and Alternate Routing

While detailed coverage of ARS is beyond the scope of these Application Notes, this section includes basic ARS screen illustrations and considerations. ARS is illustrated here mainly to illustrate alternate routing should the SIP Line be out of service or temporarily not responding.

Optionally, Automatic Route Selection (ARS) can be used rather than the simple “9N;” short code approach documented in Section 4.6. With ARS, secondary dial tone can be provided after the access code, time-based routing criteria can be introduced, and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. Although not shown in this section, ARS also facilitates more specific dialed telephone number matching, enabling immediate routing and alternate treatment for different types of numbers following the access code. For example, if all 1+10 digit calls following an

access code should use the SIP Line preferentially, but other local or service numbers following the access code should prefer a different outgoing line group, ARS can be used to distinguish the call behaviors.

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

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

The screenshot displays the IP Office configuration interface for the ARS (Automatic Route Selection) route named "Main". The left pane shows the navigation tree with "ARS" selected. The main pane shows the configuration for the "Main" route.

**ARS Configuration Parameters:**

- ARS Route Id: 50
- Route Name: Main
- Dial Delay Time: System Default
- In Service: ☒ (checked)
- Time Profile: <None>
- Secondary Dial tone: SystemTone
- Check User Call Barring: ☒ (checked)
- Out of Service Route: 51: backup
- Out of Hours Route: <None>

**Short Code Table:**

Code	Telephone Number	Feature	Line Gr
11	911	Dial Emergency	0
911	911	Dial Emergency	0
0N;	0N	Dial 3K1	0
1N;	1N"@63.79.179.178"	Dial 3K1	18
XN;	N	Dial 3K1	0

**Alternate Routing Parameters:**

- Alternate Route Priority Level: 3
- Alternate Route Wait Time: 30
- Additional Route: 51: backup

Assuming the primary route is in-service, the number passed from the short code used to access ARS (e.g., 7N in Section 4.6) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 7-1-908-848-5704, the call would be directed to Line Group 18, another SIP Line that exists in the configuration that is not described in these Application Notes. If Line Group 18 cannot be used, the call can automatically route to the route name configured in the **Additional Route** parameter in the lower right of the screen. Since alternate routing can be considered a privilege not available to all callers, IP Office can control access to the alternate route by comparing the calling user’s priority to the value in the **Alternate Route Priority Level** field.

The following screen shows an example ARS configuration for the route named “backup”, ARS Route ID 51. Continuing the example, if the user dialed 7-1-908-848-5704, and the call could

not be routed via the primary route “50: Main” described above, the call will be delivered to this “backup” route. Per the configuration shown below, the call will be delivered to Line Group 8, using the SIP Line to Verizon Business IP Trunk service described in these Application Notes. The configuration of the **Code**, **Telephone Number**, **Feature**, and **Line Group ID** for an ARS route is similar to the configuration already shown for short codes in Section 4.6. In this case, the originally dialed number (sans the short code 7) is delivered in the Request URI along with the Verizon FQDN (i.e., the contents of the INVITE sent to Verizon are the same as the 9-1-908-848-5704 short code 9 approach from Section 4.6). Additional codes (e.g., 411, 0+10, etc.) can be added to the ARS route by pressing the **Add...** button to the right of the list of previously configured codes.

The screenshot displays the Avaya IP Office configuration interface. On the left, the 'IP Offices' tree shows the hierarchy: BOOTP (2), Operator (3), 00E007026F2D, System (1), Line (13), Control Unit (5), Extension (38), User (38), HuntGroup (1), Short Code (62), Service (0), RAS (1), Incoming Call Route, WanPort (0), Directory (0), Time Profile (0), Firewall Profile (1), IP Route (4), Account Code (0), License (55), Tunnel (0), Logical LAN (0), User Rights (8), ARS (2), RAS Location Reque, and E911 System (1). The 'ARS' tab is selected, showing a list of routes: 'Main' and 'backup'. The 'backup' route is selected, and its configuration is shown on the right.

The 'backup' route configuration includes the following fields:

- ARS Route Id: 51
- Route Name: backup
- Dial Delay Time: System Default
- Secondary Dial tone: SystemTone
- Check User Call Barring: ☐
- In Service: ☒ (Out of Service Route: <None>)
- Time Profile: <None> (Out of Hours Route: <None>)

Below these fields is a table of configured codes:

Code	Telephone Number	Feature	Line Gr
11	911	Dial Emergency	0
911	911	Dial Emergency	0
1N;	1N"@pcelban0001.avayalincroft.glo...	Dial	8

Buttons for 'Add...', 'Remove', and 'Edit...' are located to the right of the table. Below the table, the 'Alternate Route Priority Level' is set to 3, and the 'Alternate Route Wait Time' is set to 30. The 'Additional Route' is set to <None>.

In the testing associated with the configuration, calls were successfully delivered to SIP Line 8 via both the primary ARS route “50: Main” (via changes to “50: Main”) as well as the backup ARS route shown above. If a primary route experiences a network outage such that no response is received to an outbound INVITE, IP Office successfully routes the call via the backup route. The user receives an audible tone when the re-routing occurs and may briefly see “Waiting for Line” on the display. The testing verified that the INVITE was sent to the primary route, and the call re-routed upon timeout. The call was made right after a failure of the primary route was induced, so IP Office had not yet marked the SIP Line out of service as a result of no response to SIP OPTIONS. Testing also verified that calls can be delivered to Verizon via the alternate route when the primary route was manually marked out-of-service, or known to be out-of-service due to prior failure of SIP OPTIONS.

## 4.9. Privacy / Anonymous Calls

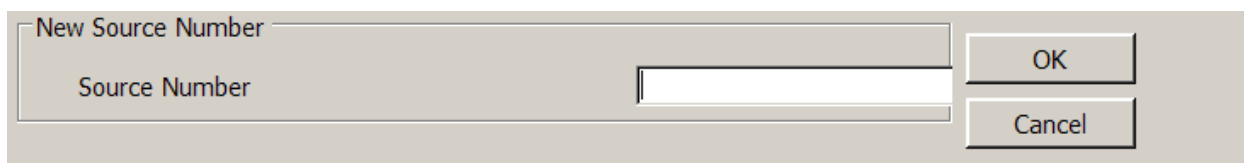
As described in Section 4.6, an IP Office user whose calling line identification is not typically withheld from the network can request privacy in the sample configuration by dialing the short code 8 to access the SIP Line. The Avaya 1600-Series IP Telephones can also request privacy



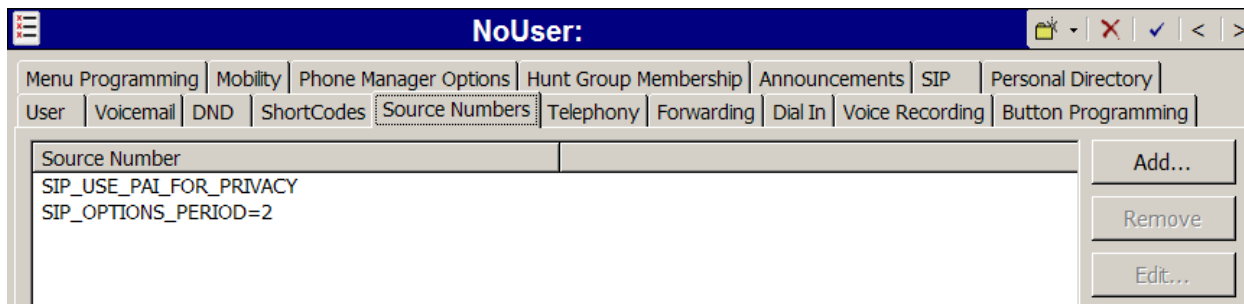
for a specific call, without dialing a unique short code, using **Features → Call Settings → Withhold Number**. Specific users may be configured to always withhold calling line identification by checking the **Anonymous** field in the **SIP** tab for the user (Section 4.5).

To configure IP Office to include the caller's DID number in the P-Asserted-Identity SIP header, required by Verizon Business to admit an otherwise anonymous caller to the network, the following procedure may be used.

From the Navigation pane, select **User**. From the Group pane, scroll down past the configured users and select the user named **NoUser**. From the NoUser Details pane, select the tab **Source Numbers**. Press the **Add...** button to the right of the list of any previously configured Source Numbers. In the **Source Number** field shown below, type **SIP\_USE\_PAI\_FOR\_PRIVACY**. Click **OK**.



The source number **SIP\_USE\_PAI\_FOR\_PRIVACY** should now appear in the list of Source Numbers as shown below.



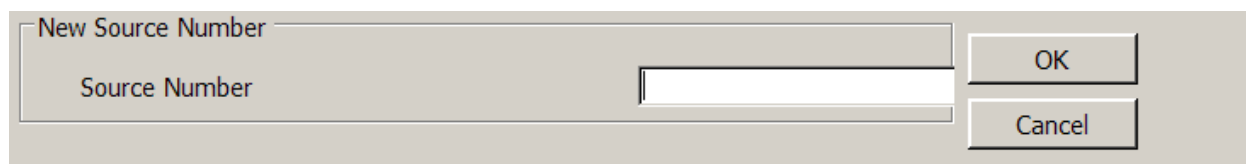
## 4.10. SIP Options Frequency

In Section 4.4, the SIP Line to Verizon Business is shown with the **Check OOS** box checked. In the sample configuration, IP Office periodically checks the health of the SIP Line by sending a SIP OPTIONS message. If there is no response, IP Office can mark the trunk out of service. Although ARS as shown in Section 4.8 can include alternate routes to complete calls even if the far-end is not responding, IP Office must wait for the outbound INVITE to timeout before route advance. Once the SIP OPTIONS maintenance recognizes that the SIP Line is out-of-service, new calls will no longer be delayed before route advance. Also, once the problem with the SIP Line is resolved, the SIP OPTIONS maintenance will automatically bring the link back to the in-service state.

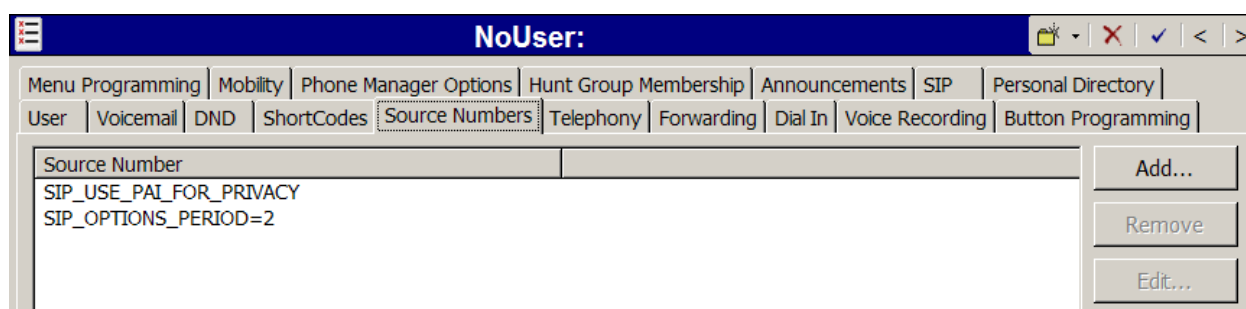


If a customer wishes to control how often SIP OPTIONS messages are sent by IP Office, a NoUser Source Number can be configured as follows. This configuration complements the configuration presented in Section 4.3.2 and Section 4.4.

From the Navigation pane, select **User**. From the Group pane, scroll down past the configured users and select the user named **NoUser**. From the NoUser Details pane, select the tab **Source Numbers**. Press the **Add...** button to the right of the list of any previously configured Source Numbers. In the **Source Number** field shown below, type **SIP\_OPTIONS\_PERIOD=X**. X is a value (in minutes) representing a longer time than the interval configured (in seconds) in the **Binding Refresh Interval**. In the sample configuration, the value used for X was 2 minutes. Click **OK**.



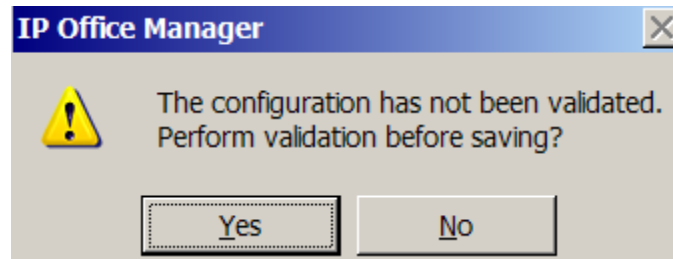
The source number **SIP\_OPTIONS\_PERIOD=2** should now appear in the list of Source Numbers as shown below.



With this configuration, Binding Refresh Intervals of 30 seconds and 90 seconds were tested successfully. That is, IP Office sourced SIP OPTIONS every 30 or 90 seconds, depending on the value configured in the Binding Refresh Interval, since the Binding Refresh Interval was less than the value configured via the SIP\_OPTIONS\_PERIOD source number.

## 4.11. Saving Configuration Changes to IP Office

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



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

A screenshot of a "Send Configuration" dialog box. The title bar is blue with standard window controls. The dialog is divided into several sections: "IP Office Settings" with a text field containing "00E007026F2D"; "Configuration Reboot Mode" with four radio button options: "Merge" (selected), "Immediate", "When Free", and "Timed"; "Reboot Time" with a time picker set to "10:09"; and "Call Barring" with two unchecked checkboxes: "Incoming Calls" and "Outgoing Calls". At the bottom, there are three buttons: "OK", "Cancel", and "Help".

## 5. Verizon Business Configuration

Information regarding Verizon Business IP Trunk service offer can be found by contacting a Verizon Business sales representative, or by visiting <http://www.verizonbusiness.com/us/products/voip/trunking/>.

The reference configuration described in these Application Notes was located in the Avaya Solutions and Interoperability Lab. The Verizon Business IP trunk service was accessed via a Verizon Private IP (PIP) T1 connection. Verizon Business provided the necessary service provisioning.

The following Fully Qualified Domain Names (FQDNs) were provided by Verizon for the reference configuration.

CPE (Avaya)	Verizon Network
<i>adevc.avaya.globalipcom.com</i>	<i>pcelban0001.avayalincroft.globalipcom.com</i>

For service provisioning, Verizon will require the customer IP address used to reach the Avaya IP Office server. Verizon provided the following information for the compliance testing: the IP address and port used by the Verizon SIP SBC, and the Direct Inward Dialed (DID) numbers shown in **Figure 1** and **Table 1**. This information was used to complete the Avaya IP Office configuration shown in Section 4.

## 6. General Test Approach and Results

The Avaya IP Office location was connected to the Verizon Business IP Trunk Service, as depicted in **Figure 1**. Avaya IP Office was configured to use the commercially available SIP Trunking solution provided by the Verizon Business IP Trunk SIP Trunk Service. This allowed Avaya IP Office users to make calls to the PSTN and receive calls from the PSTN via the Verizon Business IP Trunk SIP Trunk Service.

The focus of the verification testing associated with this version of the IP Office with Verizon IP Trunk Application Notes was the use of SIP REFER messaging for call transfer scenarios, and the use of DNS SRV to Verizon, rather than static provisioning of the Verizon SIP signaling information. Note that the REFER testing and the DNS SRV testing are not dependent on one another. That is, IP Office can use REFER for call transfer without using DNS SRV, and IP office can use DNS SRV without using REFER for call transfer.

Testing was successful. Any limitations related to the overall configuration are noted in Section 1.3.

## 7. Verifications

This section summarizes and provides detailed illustrations of the verification of REFER and DNS SRV with Verizon IP Trunk Service.

### 7.1. REFER Testing

The following scenarios will result in IP Office sending REFER to the Verizon network. Each scenario was tested successfully.

- PSTN user makes call to Verizon IP Trunk DID and IP Office user answers. IP Office user performs an attended transfer of the inbound call to a PSTN destination using the Verizon IP Trunk service. In this context, an attended transfer implies that the destination of the outbound call answers before the IP Office user completes the transfer. In terms of SIP signaling, this means that IP Office sends the REFER after the 200 OK is received from Verizon. This scenario is illustrated with Wireshark in Section 7.1.1.
- IP Office user makes an outbound call to the PSTN via the Verizon IP Trunk service. The IP Office user then performs an attended transfer of the call to another PSTN destination using the Verizon IP Trunk service. In this context, an attended transfer implies that the destination of the outbound call answers before the IP Office user completes the transfer. In terms of SIP signaling, this means that IP Office sends the REFER after the 200 OK is received from Verizon.
- PSTN user makes call to Verizon IP Trunk DID and IP Office user answers. The IP Office user performs an unattended transfer of the inbound call to a PSTN destination using the Verizon IP Trunk service. In this context, an unattended transfer implies that the destination of the outbound call does not answer before the IP Office user completes the transfer. In terms of SIP signaling, this means that IP Office sends the REFER after a message such as 183 Session Progress is received from Verizon but before a 200 OK is received from Verizon.
- IP Office user makes an outbound call to the PSTN via the Verizon IP Trunk service. The IP Office user performs an unattended transfer of the call to another PSTN destination using the Verizon IP Trunk service. In this context, an unattended transfer implies that the destination of the outbound call does not answer before the IP Office user completes the transfer. In terms of SIP signaling, this means that IP Office sends the REFER after a message such as 183 Session Progress is received from Verizon but before a 200 OK is received from Verizon.

### 7.1.1. Wireshark Trace Illustration for REFER-Transfer

This section illustrates the SIP signaling for an inbound Verizon IP Trunk call that is transferred back to the PSTN by an IP Office user. IP Office will use SIP REFER.

The following screen shows the portion of the trace until the point where the IP Office user answers the inbound call. Frame 5 is selected and expanded to show the contents of the INVITE message from Verizon. In frame 8, IP Office answers the call with a 200 OK.

Filter: sip && ip.addr == 172.30.209.21					
Expression... Clear Apply					
No. .	Time	Source	Destination	Protocol	Info
5	4.378172	172.30.209.21	1.1.1.2	SIP/SDP	Request: INVITE sip:7329450231@1.1.1.2:5060, with session description
6	4.383762	1.1.1.2	172.30.209.21	SIP	Status: 100 Trying
7	4.399999	1.1.1.2	172.30.209.21	SIP	Status: 180 Ringing
8	4.918998	1.1.1.2	172.30.209.21	SIP/SDP	Status: 200 ok, with session description
15	5.187682	172.30.209.21	1.1.1.2	SIP	Request: ACK sip:7329450231@1.1.1.2:5060;transport=udp
Session Initiation Protocol					
Request-Line: INVITE sip:7329450231@1.1.1.2:5060 SIP/2.0					
Message Header					
Via: SIP/2.0/UDP 172.30.209.21:5071;branch=z9hG4bK45ftej103gtgbugfj1u0.1					
From: "AVAYA ALPHA"<sip:9088485704@65.211.120.226;user=phone>;tag=1449953991-1291127879280-					
To: "Lincroft Lab LINCROFT LAB"<sip:7329450231@adevc.avaya.globalipcom.com>					
Call-ID: BW093759280301110-935807537@65.211.120.226					
CSeq: 245103417 INVITE					
Contact: <sip:9088485704@172.30.209.21:5071;transport=udp>					
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY					
Accept: multipart/mixed,application/media_control+xml,application/sdp					
Supported:					
Max-Forwards: 69					
Content-Type: application/sdp					
Content-Length: 208					
Message Body					
Session Description Protocol					
Session Description Protocol Version (v): 0					
Owner/Creator, Session Id (o): Broadworks 42605053 1 IN IP4 172.30.209.132					
Session Name (s): -					
Connection Information (c): IN IP4 172.30.209.132					
Time Description, active time (t): 0 0					
Media Description, name and address (m): audio 10250 RTP/AVP 18 0 8 101					
Media Attribute (a): rtpmap:101 telephone-event/8000					
Media Attribute (a): fmp:101 0-15					
Media Attribute (a):ptime:20					
Media Attribute (a):fmp:18 annexb=no					

After answering the inbound call, the IP Office user presses the transfer button, and dials the number 9-1-732-687-0755 which is routed by IP Office to the Verizon IP Trunk service. In this “attended transfer” example, the IP Office user performing the transfer waits for the outbound call to be answered before pressing the transfer button a second time to complete the transfer.

Scrolling down in the same trace in the screen below, frame 1805 is selected to show the contents of an outbound INVITE sent by IP Office to the Verizon IP Trunk service. In frame 3327, Verizon sends the 200 OK when the called party answers the call.

Filter: sip && ip.addr == 172.30.209.21 Expression... Clear Apply					
No. .	Time	Source	Destination	Protocol	Info
1805	22.779040	1.1.1.2	172.30.209.21	SIP/SDP	Request: INVITE sip:17326870755@pcelban0001.avaya!ncroft.globalipcom.com,
1818	22.895856	172.30.209.21	1.1.1.2	SIP	Status: 100 Trying
1994	24.533600	172.30.209.21	1.1.1.2	SIP/SDP	Status: 183 Session Progress, with session description
3327	31.132818	172.30.209.21	1.1.1.2	SIP/SDP	Status: 200 OK, with session description
3329	31.137673	1.1.1.2	172.30.209.21	SIP	Request: ACK sip:17326870755@172.30.209.21:5071;transport=udp
Session Initiation Protocol					
Request-Line: INVITE sip:17326870755@pcelban0001.avaya!ncroft.globalipcom.com SIP/2.0					
Message Header					
Via: SIP/2.0/UDP 1.1.1.2:5060;rport;branch=z9hG4bKe5ba745192bc01d409f9c1d733318760					
From: "Joey-Dig5410" <sip:7329450231@1.1.1.2>;tag=38487ff86ffadfeb					
To: <sip:17326870755@pcelban0001.avaya!ncroft.globalipcom.com>					
Call-ID: 9e647a848a929dbc5506d241e5282a03@1.1.1.2					
CSeq: 1136372479 INVITE					
Contact: "Joey-Dig5410" <sip:7329450231@1.1.1.2:5060;transport=udp>					
Max-Forwards: 70					
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE					
Content-Type: application/sdp					
Supported: timer					
Content-Length: 240					
Message Body					
Session Description Protocol					
Session Description Protocol Version (v): 0					
Owner/Creator, Session Id (o): UserA 607296314 4055416929 IN IP4 1.1.1.2					
Session Name (s): Session SDP					
Connection Information (c): IN IP4 1.1.1.2					
Time Description, active time (t): 0 0					
Media Description, name and address (m): audio 49154 RTP/AVP 18 0 101					
Media Attribute (a): rtpmap:18 G729/8000					
Media Attribute (a): fmtp:18 annexb=no					
Media Attribute (a): rtpmap:0 PCMU/8000					
Media Attribute (a): rtpmap:101 telephone-event/8000					
Media Attribute (a): fmtp:101 0-15					

Filter: sip && ip.addr == 172.30.209.21		Expression... Clear Apply			
No. .	Time	Source	Destination	Protocol	Info
6262	45.558569	1.1.1.2	172.30.209.21	SIP	Request: REFER sip:17326870755@172.30.209.21:5071;transport=udp
6283	45.677297	172.30.209.21	1.1.1.2	SIP	Status: 202 Accepted
6285	45.683267	172.30.209.21	1.1.1.2	SIP	Request: BYE sip:7329450231@1.1.1.2:5060;transport=udp
6287	45.686118	1.1.1.2	172.30.209.21	SIP	Status: 200 OK
6288	45.688720	172.30.209.21	1.1.1.2	SIP	Request: BYE sip:7329450231@1.1.1.2:5060;transport=udp
<div> <div>Internet Protocol, Src: 1.1.1.2 (1.1.1.2), Dst: 172.30.209.21 (172.30.209.21)</div> <div>User Datagram Protocol, Src Port: sip (5060), Dst Port: powerschool (5071)</div> <div>Session Initiation Protocol <div> <div>Request-Line: REFER sip:17326870755@172.30.209.21:5071;transport=udp SIP/2.0</div> <div>Message Header <div> <div>Via: SIP/2.0/UDP 1.1.1.2:5060;rport;branch=z9hG4bK6411c9e935671b5153f844e459d9c95d</div> <div>From: "Joey-Dig5410" &lt;sip:7329450231@1.1.1.2&gt;;tag=38487ff86ffadfeb</div> <div>To: &lt;sip:17326870755@pcelban0001.avayalincroft.globalipcom.com&gt;;tag=261679990-1291127899438</div> <div>Call-ID: 9e647a84a92dbdc5506d241e5282a03@1.1.1.2</div> <div>CSeq: 1136372480 REFER</div> <div>Contact: "Joey-Dig5410" &lt;sip:7329450231@1.1.1.2:5060;transport=udp&gt;</div> <div>Max-Forwards: 70</div> <div>Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE</div> <div>Supported: timer</div> <div>Content-Length: 0</div> <div>Refer-To: &lt;sip:9088485704@172.30.209.21?replaces=Bw093759280301110-935807537%4065.211.120.226%3Bto-tag%3D1449953991-1291127879280-%3Bfrom-tag</div> </div> </div> </div> </div> </div>					

No.	Time	Source	Destination	Protocol	Info
6262	45.558569	1.1.1.2	172.30.209.21	SIP	Request: REFER sip:17326870755@172.30.209.21:5071;transport=udp
6283	45.677297	172.30.209.21	1.1.1.2	SIP	Status: 202 Accepted
6285	45.683267	172.30.209.21	1.1.1.2	SIP	Request: BYE sip:7329450231@1.1.1.2:5060;transport=udp
6287	45.686118	1.1.1.2	172.30.209.21	SIP	Status: 200 ok
6288	45.688720	172.30.209.21	1.1.1.2	SIP	Request: BYE sip:7329450231@1.1.1.2:5060;transport=udp
⊞ Frame 6285 (472 bytes on wire, 472 bytes captured)					
⊞ Ethernet II, Src: Netscree_91:64:e5 (00:10:db:91:64:e5), Dst: AvayaEcs_02:6f:2d (00:e0:07:02:6f:2d)					
⊞ Internet Protocol, Src: 172.30.209.21 (172.30.209.21), Dst: 1.1.1.2 (1.1.1.2)					
⊞ User Datagram Protocol, Src Port: powerschool (5071), Dst Port: sip (5060)					
⊞ Session Initiation Protocol					
⊞ Request-Line: BYE sip:7329450231@1.1.1.2:5060;transport=udp SIP/2.0					
⊞ Message Header					
⊞ Via: SIP/2.0/UDP 172.30.209.21:5071;branch=z9hg4bk7pggti2018eh6tsfelq1sde07v1l1.1					
⊞ From: <sip:17326870755@nelban0001.avayalincroft.globalipcom.com>;tag=261679990-1291127899438					
⊞ To: "Joey-Dig5410" <sip:7329450231@adevc.avaya.globalipcom.com>;tag=38487ff86ffadfeb					
Call-ID: 9e647a848a929dbc5506d241e5282a03@1.1.1.2					
⊞ Cseq: 245112682 BYE					

No.	Time	Source	Destination	Protocol	Info
6262	45.558569	1.1.1.2	172.30.209.21	SIP	Request: REFER sip:17326870755@172.30.209.21:5071;transport=udp
6283	45.677297	172.30.209.21	1.1.1.2	SIP	Status: 202 Accepted
6285	45.683267	172.30.209.21	1.1.1.2	SIP	Request: BYE sip:7329450231@1.1.1.2:5060;transport=udp
6287	45.686118	1.1.1.2	172.30.209.21	SIP	Status: 200 ok
6288	45.688720	172.30.209.21	1.1.1.2	SIP	Request: BYE sip:7329450231@1.1.1.2:5060;transport=udp

Session Initiation Protocol

Request-Line: BYE sip:7329450231@1.1.1.2:5060;transport=udp SIP/2.0

Message Header

Via: SIP/2.0/UDP 172.30.209.21:5071;branch=z9hG4bKhd08qg300o9hvtkup7r1cdeo6nft0.1

From: "AVAYA ALPHA" <sip:9088485704@65.211.120.226;user=phone>;tag=1449953991-1291127879280-

To: "Lincroft Lab LINCROFT LAB" <sip:7329450231@adevc.avaya.globalipcom.com>;tag=7c4aa5141b088fbb

Call-ID: BW093759280301110-935807537@65.211.120.226

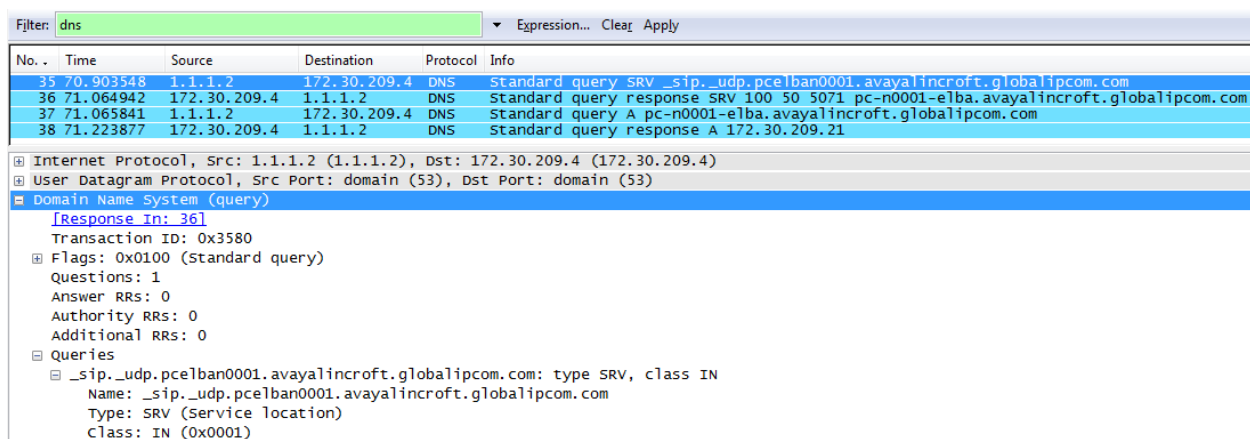
CSeq: 245103418 BYE

## 7.2. DNS SRV Testing

The new IP Office Release 6.1 capability to determine the Verizon SIP signaling address and port using DNS procedures was tested using the production Verizon PIP circuit. Rather than statically configure IP Office with the Verizon IP Address and SIP signaling port, as was the case shown in reference [JRR-IPOR6], IP Office determined the Verizon IP Address and signaling port dynamically using DNS. On the production circuit used for testing, Verizon responded with one “answer”.

### 7.2.1. Wireshark Trace Illustration for DNS SRV

This section illustrates the DNS signaling used when the SIP Line in IP Office is configured to use DNS SRV. Please reference Section 4.4.2 of these Application Notes for the relevant configuration. In the filtered Wireshark trace shown below, IP Office (1.1.1.2) sends a DNS SRV query to the Verizon DNS server (172.30.209.4) configured in IP Office for the SIP Line. Frame 35 is highlighted and expanded. Note that the query contains “\_sip.\_udp.pcelban0001.avayalincroft.globalipcom.com” because the IP Office SIP Line has been configured with “pcelban0001.avayalincroft.globalipcom.com” as the Verizon domain, using UDP for transport.



The image shows a Wireshark packet capture trace filtered for 'dns'. The packet list shows four frames: Frame 35 (70.903548) is a DNS Standard query SRV from 1.1.1.2 to 172.30.209.4; Frame 36 (71.064942) is a DNS Standard query response SRV from 172.30.209.4 to 1.1.1.2; Frame 37 (71.065841) is a DNS Standard query A from 1.1.1.2 to 172.30.209.4; and Frame 38 (71.223877) is a DNS Standard query response A from 172.30.209.4 to 1.1.1.2. The details pane for Frame 35 is expanded, showing the Internet Protocol, User Datagram Protocol, and Domain Name System (query) sections. The DNS query is for the domain \_sip.\_udp.pcelban0001.avayalincroft.globalipcom.com, type SRV, class IN.

No.	Time	Source	Destination	Protocol	Info
35	70.903548	1.1.1.2	172.30.209.4	DNS	Standard query SRV _sip._udp.pcelban0001.avayalincroft.globalipcom.com
36	71.064942	172.30.209.4	1.1.1.2	DNS	Standard query response SRV 100 50 5071 pc-n0001-elba.avayalincroft.globalipcom.com
37	71.065841	1.1.1.2	172.30.209.4	DNS	Standard query A pc-n0001-elba.avayalincroft.globalipcom.com
38	71.223877	172.30.209.4	1.1.1.2	DNS	Standard query response A 172.30.209.21

Internet Protocol, Src: 1.1.1.2 (1.1.1.2), Dst: 172.30.209.4 (172.30.209.4)

User Datagram Protocol, Src Port: domain (53), Dst Port: domain (53)

Domain Name System (query)

[Response in: 36]

Transaction ID: 0x3580

Flags: 0x0100 (Standard query)

Questions: 1

Answer RRs: 0

Authority RRs: 0

Additional RRs: 0

Queries

- \_sip.\_udp.pcelban0001.avayalincroft.globalipcom.com: type SRV, class IN
- Name: \_sip.\_udp.pcelban0001.avayalincroft.globalipcom.com
- Type: SRV (Service location)
- Class: IN (0x0001)



The Verizon DNS response in frame 36 is highlighted and expanded in the following screen. Note that the “Answer” contains Target “pc-n0001-elba.avayalincroft.globalipcom.com” and port 5071.

No. .	Time	Source	Destination	Protocol	Info
35	70.903548	1.1.1.2	172.30.209.4	DNS	Standard query SRV _sip._udp.pcelban0001.avayalincroft.globalipcom.com
36	71.064942	172.30.209.4	1.1.1.2	DNS	Standard query response SRV 100 50 5071 pc-n0001-elba.avayalincroft.globalipcom.com
37	71.065841	1.1.1.2	172.30.209.4	DNS	Standard query A pc-n0001-elba.avayalincroft.globalipcom.com
38	71.223877	172.30.209.4	1.1.1.2	DNS	Standard query response A 172.30.209.21

User Datagram Protocol, Src Port: domain (53), Dst Port: domain (53)

Domain Name System (response)

[Request In: 35]

[Time: 0.161394000 seconds]

Transaction ID: 0x3580

Flags: 0x8180 (Standard query response, No error)

Questions: 1

Answer RRs: 1

Authority RRs: 0

Additional RRs: 0

Queries

\_sip.\_udp.pcelban0001.avayalincroft.globalipcom.com: type SRV, class IN

Name: \_sip.\_udp.pcelban0001.avayalincroft.globalipcom.com

Type: SRV (Service location)

Class: IN (0x0001)

Answers

\_sip.\_udp.pcelban0001.avayalincroft.globalipcom.com: type SRV, class IN, priority 100, weight 50, port 5071, target pc-n0001-elba.i

Name: \_sip.\_udp.pcelban0001.avayalincroft.globalipcom.com

Type: SRV (Service location)

Class: IN (0x0001)

Time to live: 6 hours

Data length: 22

Priority: 100

weight: 50

Port: 5071

Target: pc-n0001-elba.avayalincroft.globalipcom.com

Frame 37 is expanded below to illustrate the IP Office DNS A-query to determine the IP Address associated with the name “pc-n0001-elba.avayalincroft.globalipcom.com” (i.e., the “Target” returned by Verizon as shown in the prior screen).

No. .	Time	Source	Destination	Protocol	Info
35	70.903548	1.1.1.2	172.30.209.4	DNS	Standard query SRV _sip._udp.pcelban0001.avayalincroft.globalipcom.com
36	71.064942	172.30.209.4	1.1.1.2	DNS	Standard query response SRV 100 50 5071 pc-n0001-elba.avayalincroft.globalipcom.com
37	71.065841	1.1.1.2	172.30.209.4	DNS	Standard query A pc-n0001-elba.avayalincroft.globalipcom.com
38	71.223877	172.30.209.4	1.1.1.2	DNS	Standard query response A 172.30.209.21

User Datagram Protocol, Src Port: domain (53), Dst Port: domain (53)

Domain Name System (query)

[Response In: 38]

Transaction ID: 0x3581

Flags: 0x0100 (Standard query)

Questions: 1

Answer RRs: 0

Authority RRs: 0

Additional RRs: 0

Queries

pc-n0001-elba.avayalincroft.globalipcom.com: type A, class IN

Name: pc-n0001-elba.avayalincroft.globalipcom.com

Type: A (Host address)

Class: IN (0x0001)



application, and Wireshark. The verification testing from reference [JRR-IPOR6] included the following successful SIP trunk interoperability compliance testing:

- Incoming calls from the PSTN were routed to the DID numbers assigned by Verizon Business to the Avaya IP Office location. These incoming PSTN calls arrived via the SIP Line and were answered by Avaya H.323 telephones, Avaya digital telephones, analog telephones, Avaya IP Office Softphone, and Avaya IP Office Voicemail Pro voicemail and auto-attendant applications. The display of caller ID on display-equipped Avaya IP Office telephones was verified.
- Incoming calls answered by members of collective and sequential Hunt Groups were verified.
- Outgoing calls from the Avaya IP Office location to the PSTN were routed via the SIP Line to Verizon Business. These outgoing PSTN calls were originated from Avaya H.323 telephones, Avaya digital telephones, analog endpoints, and Avaya IP Office Softphone. The display of caller ID on display-equipped PSTN telephones was verified.
- Proper disconnect when the caller abandoned a call before answer for both inbound and outbound calls.
- Proper disconnect when the IP Office party or the PSTN party terminated an active call.
- Proper busy tone heard when an IP Office user called a busy PSTN user, or a PSTN user called a busy IP Office user (i.e., if no redirection was configured for user busy conditions)
- Various outbound PSTN call types were tested including long distance, international, toll-free, operator assisted, and directory assistance calls.
- Requests for privacy (i.e., caller anonymity) for IP Office outbound calls to the PSTN were verified. That is, when privacy is requested by IP Office, outbound PSTN calls were successfully completed while withholding the caller ID from the displays of display-equipped PSTN telephones.
- Privacy requests for inbound calls from the PSTN to IP Office users were verified. That is, when privacy is requested by a PSTN caller, the inbound PSTN call was successfully completed to an IP Office user while presenting an “anonymous” display to the IP Office user.
- SIP OPTIONS monitoring of the health of the SIP trunk was verified. Both Verizon Business and IP Office were able to monitor health using SIP OPTIONS. The Avaya IP Office configurable control of SIP OPTIONS timing was exercised successfully.
- IP Office outbound calls were placed with simple short codes as well as using ARS. Using ARS, the ability of IP Office to route-advance to an alternate route was exercised when the primary SIP line was not responding. The Line Group associated with the Verizon Business SIP Line was the primary line group chosen for a call, or an alternate line group selected upon failure of a primary line.
- Incoming and outgoing calls using the G.729(a) and G.711 ULAW codecs.
- DTMF transmission (RFC 2833) with successful voice mail navigation using G.729a and G.711MU for incoming and outgoing calls. Successful navigation of a simple auto-attendant application configured on IP Office Voicemail Pro.
- The “callback” feature of Avaya Voicemail Pro was tested successfully. When a message was left for a voice mail subscriber with “callback” configured, an outbound call

was placed to the subscriber's configured mobile telephone via the SIP Line to Verizon Business. Upon answer, Voicemail Pro announced the call and prompted the user to enter the “#” key to accept the call. The user had the opportunity to navigate the voicemail TUI via DTMF (e.g., to listen to the voice message that stimulated the callback).

- Inbound and outbound long holding time call stability.
- Telephony features such as call waiting, hold, transfer, and conference.
- Inbound calls from Verizon IP Trunk Service that were call forwarded back to PSTN destinations via Verizon IP Trunk Service.
- Mobile twinning to a mobile phone, presenting true calling party information to the mobile phone. Outbound mobile call control was also verified successfully (e.g., using DTMF on a twinned call to place new calls and create a conference via a mobile phone).
- Proper DiffServ markings for IP Office SIP signaling and RTP media.

## 8. Conclusion

IP Office is a highly modular IP telephone system designed to meet the needs of home offices, standalone businesses, and networked branch and head offices for small and medium enterprises. These Application Notes demonstrated how IP Office can be successfully combined with a Verizon Business IP Trunk SIP trunk service connection to create an end-to-end SIP Telephony business solution. By following the example configurations provided in this document, customers using Avaya IP Office can connect to the PSTN via a Verizon Business IP Trunk SIP Trunk service connection, thus eliminating the costs of analog or digital trunk connections previously required to access the PSTN. Utilizing this solution, IP Office customers can leverage the operational efficiencies and cost savings associated with SIP trunking while gaining the advanced technical features provided through the marriage of best of breed technologies from Avaya and Verizon.

As noted in the Introduction, the focus of this version of the IP Office with Verizon IP Trunk Application Notes is the use of two newly configurable IP Office Release 6.1 SIP trunk features: REFER-based transfer and DNS-SRV with Verizon IP Trunk Service. Although these new features have not been independently certified by Verizon labs, these Application Notes may be used to facilitate field trial testing of either or both of these new capabilities of IP Office Release 6.1 with Verizon IP Trunk service.

## 9. References

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

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

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

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

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

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

Additional IP Office documentation can be found at:

<http://marketingtools.avaya.com/knowledgebase/>

The following Application Notes formed the base configuration for this document. The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 6.0.

[JRR-IPOR6] Application Notes for Configuring SIP Trunk using Verizon Business IP Trunk SIP Trunk Service Offer and Avaya IP Office Release 6, Issue 1.0

<https://support.avaya.com/css/P8/documents/100082703>

The following Application Notes provide the configuration that will enable IP Office Release 6.1 to perform like IP Office Release 6.0. That is, in reference [JRR-IPOR61], the newly configurable SIP features of IP Office Release 6.1, such as SIP REFER and DNS SRV, are disabled. The Application Notes referenced below were written to support Verizon certification of IP Office Release 6.1 based on formal compliance testing of IP Office Release 6.0.

[JRR-IPOR61] Application Notes for Configuring SIP Trunk using Verizon Business IP Trunk SIP Trunk Service Offer and Avaya IP Office Release 6.1, Issue 1.0

[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>

---

**©2010 Avaya Inc. All Rights Reserved.**

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

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