



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

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# **Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service and Avaya IP Office Release 10 – 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 Server Edition Primary Server Release 10, an IP500 V2 Expansion System Release 10.0, Voicemail Pro, Avaya one-X® Portal for IP Office, Avaya Communicator for Windows, Avaya Communicator for Web, and Avaya SIP, H.323, digital, and analog endpoints.

These Application Notes complement previously published Application Notes by illustrating the configuration screens and Avaya testing of IP Office Release 10.

The Verizon Business IP Trunk SIP 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.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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

# 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 (Verizon Business IP Trunk) and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office Server Edition Primary Server, an IP500 V2 Expansion System, Voicemail Pro, Avaya one-X® Portal for IP Office, Avaya Communicator for Windows, Avaya Communicator for Web, Avaya SIP, H.323, digital, and analog endpoints. The single Server Edition Primary server provided IP Office Server Edition, Voicemail Pro, and Avaya one-X® Portal for IP Office.

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

For more information on the Verizon Business IP Trunk service, including access alternatives, visit <http://www.verizonenterprise.com/products/business-communications/voice-over-ip/>.

## 2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site 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 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 service.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

### 2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability, the following features and functionality were covered during the interoperability compliance test:

- 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 SIP endpoints, Avaya H.323 endpoints, Avaya digital endpoints, analog endpoints, analog fax machines, Avaya Communicator for Windows, Avaya Communicator for Web and Avaya Voicemail Pro. The display of caller ID on display-equipped Avaya IP Office endpoints was verified.

- Incoming calls answered by members of 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 SIP endpoints, Avaya H.323 endpoints, Avaya digital endpoints, analog endpoints, Avaya Communicator for Windows, Avaya Communicator for Web and Avaya Voicemail Pro. The display of caller ID on display-equipped PSTN telephones was verified.
- Inbound / Outbound fax using G.711 and T.38 were 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 SIP trunk health using SIP OPTIONS.
- 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 IP Trunk was the primary line group chosen for a call, or an alternate line group was selected upon failure of a primary line.
- Incoming and outgoing calls using the G.729A and G.711MU 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 Avaya Voicemail Pro.
- Inbound and outbound long holding time call stability.
- Telephony features such as call waiting, hold, transfer, and conference.
- Attended call transfer using the SIP REFER method.
- Unattended or “blind” call transfer using the SIP REFER method.
- Inbound calls from Verizon Business IP Trunk service that were call forwarded back to PSTN destinations, presenting true calling party information to the PSTN telephone, via Verizon Business 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).

- DiffServ markings in accordance with network requirements for IP Office SIP signaling and RTP media.
- Mobility Features such as Mobile Callback and Mobile Call Control.

## 2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results. The following observations were noted.

- **SIP endpoint transfers:** When Refer based call transfers are performed, Verizon did not send NOTIFY SIP messages to Avaya IP Office to signal transfer completion. Some Avaya SIP endpoints (e.g., Avaya 1140E, and Avaya Communicator for Windows) require receipt of a NOTIFY when Refer based call transfers are performed. The IP Office SIP Line option, **Emulate NOTIFY for REFER** will send the necessary NOTIFY messages to these endpoints (see **Section 5.4.7**).
- **DiffServ markings:** For IP Office Server Edition, the IP header in SIP signaling packets sent from the IP Office server do not contain the DSCP values configured in IP Office Manager for Quality of Service policies (See **Section 5.2.2**). The IP headers in RTP media packets have the correct values. Also, this only affects Server Edition systems; the IP headers in SIP signaling packets from IP 500V2 systems have the correct values. This anomaly is under investigated by IP Office product development (IPOFFICE- 112012).

## 2.3. Support

### 2.3.1. Avaya

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

### 2.3.2. Verizon

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

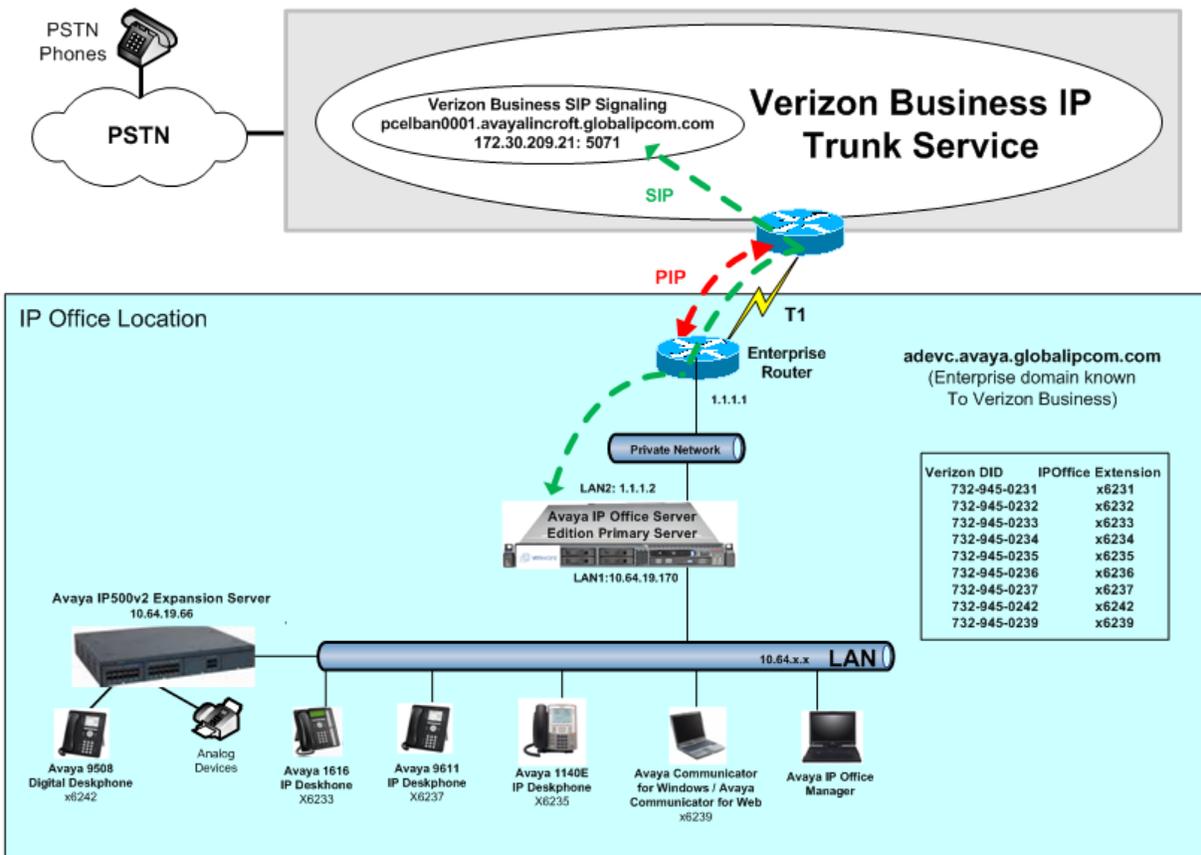
### 3. Reference Configuration

**Figure 1** illustrates a sample Avaya IP Office solution connected to the Verizon Business IP 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 PIP service.

In the sample configuration, IP Office receives traffic from the Verizon Business IP Trunk service on port 5060 and sends traffic to port 5071, using UDP for network transport, as required by the Verizon Business IP Trunk service. As shown in **Figure 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 the Fully Qualified Domain Name (FQDN) *pcelban0001.avayalincroft.globalipcom.com*.

The Avaya CPE environment was assigned FQDN *adevc.avaya.globalipcom.com* by Verizon Business.



**Figure 1: Avaya Interoperability Test Lab Configuration**

## 4. Equipment and Software Validated

Table 1 shows the equipment and software used in the sample configuration.

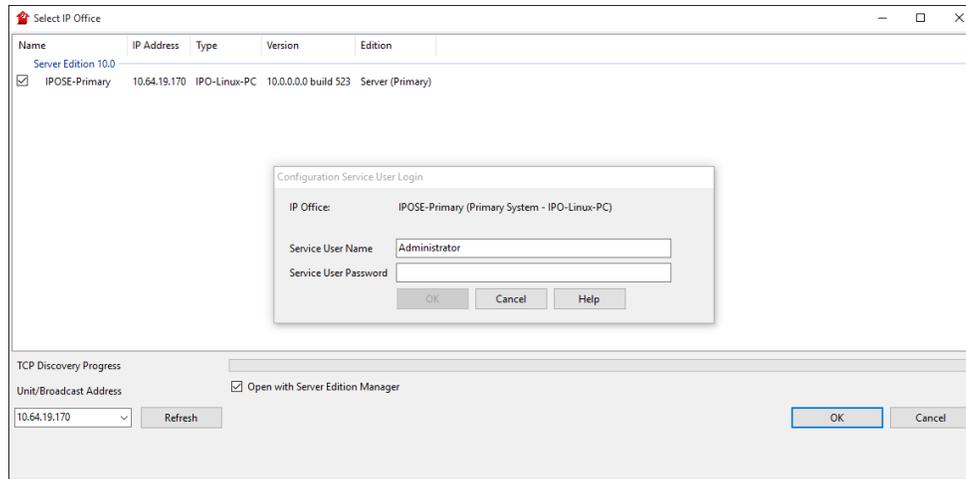
Avaya IP Telephony Solution Components	
Equipment/Software	Release/Version
Avaya IP Office Server Edition (Primary Server) <ul style="list-style-type: none"> <li>▪ IP Office</li> <li>▪ Voicemail Pro</li> <li>▪ Avaya WebRTC Gateway</li> <li>▪ Avaya one-X® Portal for IP Office</li> </ul>	Release 10.0.0.0 build 550 Release 10.0.0.0 build 469 Release 10.0.0.0 build 140 Release 10.0.0.0 build 980
Avaya IP Office IP500 V2 (Expansion System) <ul style="list-style-type: none"> <li>▪ Avaya IP Office TCM 8</li> <li>▪ Avaya IP Office COMBO6210/ATM4</li> </ul>	Release 10.0.0.0 Build 550 Release 10.0.0.0 Build 550
Avaya IP Office Manager	Release 10.0.0.0 Build 550
Avaya 9611SW IP Deskphone (H.323)	Release 6.6302
Avaya 1616 IP Deskphone (H.323)	Release 1.3.10
Avaya 1140E IP Deskphone (SIP)	Release 04.04.23
Avaya 9508 Digital Deskphone	Release 0.59
Avaya Communicator for Windows	Release 2.1.3.237
Avaya Communicator for Web	Release 1.0.16.2010
Analog Fax device	Ventafax 6.3

**Table 1: Equipment and Software Tested**

Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2, and also when deployed with all configurations of IP Office Server Edition. Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

## 5. Avaya IP Office Primary Configuration

This section illustrates relevant aspects of the Primary server used in the verification of these Application Notes. The Primary server is configured via the IP Office Manager program. For more information on IP Office Manager, consult reference [2]. From the IP Office Manager PC, select **Start** → **Programs** → **IP Office** → **Manager** to launch the Manager application. Navigate to **File** → **Open Configuration** (not shown), select the proper Avaya IP Office system from the pop-up window, and log in using the appropriate credentials.



On Server Edition systems, the Solution View screen will appear, similar to the one shown below. If the left navigation pane does not immediately appear, click on the **Configuration** link as highlighted below. In the sample configuration, IP users registered to the Primary server and failover to the Secondary server. Digital and Analog users are configured on the Expansion System. A SIP trunk to Verizon Business is configured on the Primary server. Clicking the “plus” sign next to the Primary server system name, e.g., **IPOSE-Primary**, on the left navigation pane will expand the menu on this server.



## 5.1. Licensing

In the sample configuration, **IPOSE-Primary** was used as the system name of the Primary Server and **IP500 Expansion** was used as the system name of the Expansion System. All navigation described in the following sections (e.g., **License**) appears as submenus underneath the system name in the Navigation Pane.

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. Confirm a valid **SIP Trunk Channels** license with sufficient **Instances** (trunk channels). If Avaya IP Deskphones will be used as is the case in these Application Notes, verify the **Avaya IP endpoints** license.

License Type	Status
License	Remote Server
License Mode	WebLM Normal
Licensed Version	10.0

Feature	Instances	Status	Expiration Date	Source
Additional Voicemail Pro Ports	152	Valid	Never	WebLM
VMPro TTS Professional	1	Valid	Never	WebLM
Power User	6	Valid	Never	WebLM
Avaya IP endpoints	9	Valid	Never	WebLM
SIP Trunk Channels	50	Valid	Never	WebLM
CTI Link Pro	1	Valid	Never	WebLM
Server Edition R10	1	Valid	Never	WebLM
Web Collaboration	5	Valid	Never	WebLM
UMS Web Services	1	Valid	Never	WebLM
Basic User	5	Valid	Never	WebLM

License Type	Status
License	Remote Server

Remote Server Configuration

License Source: WebLM

Domain Name (URL): 10.64.19.170

Path: WebLM/LicenseServer

Port Number: 52233

WebLM client ID: 000C29140005-sillipose

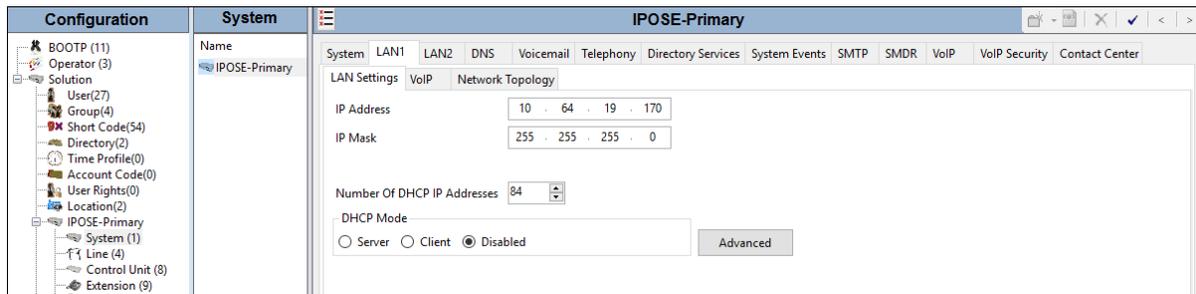
Reserved Licenses	Instances	Server Edition	Instances
SIP Trunk Sessions	50	Server Edition	1
SM Trunk Sessions	0	Avaya IP Endpoints	9
Voicemail Pro Ports	152	3rd Party IP Endpoints	0
VMPro Recordings Administrators	0	Receptionist	0
VMPro TTS Professional	1	Basic User	5
CTI Link Pro	1	Office Worker	0
UMS Web Services	1	Power User	6
Mac Softphones	0	Avaya Softphone	0
Avaya Contact Center Select	0	Web Collaboration	5
Third Party Recorder	0		

## 5.2. System Settings

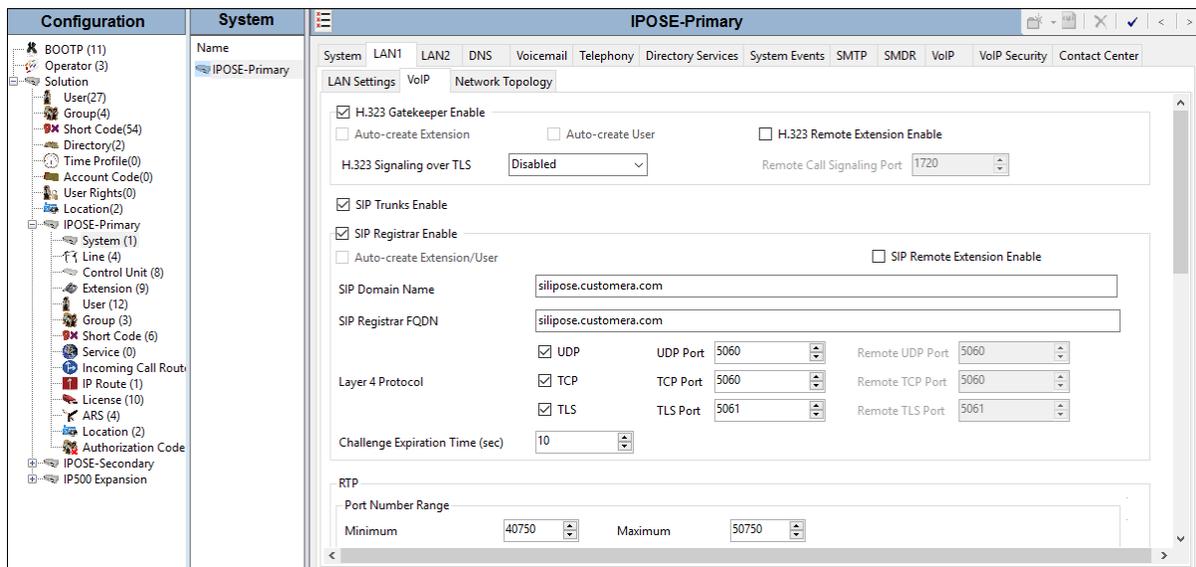
This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings. The subsection order corresponds to a left to right navigation of the tabs in the Details pane for System settings. For all of the following configuration sections, the **OK** button (not shown) must be selected in order for any changes to be saved

### 5.2.1. LAN 1 Settings

In the sample configuration, LAN1 is used to connect the Primary server to the enterprise network. To view or configure the **IP Address** of LAN1, select the **LAN1** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP address of the Primary server is “**10.64.19.170**”. Other parameters on this screen may be set according to customer requirements.

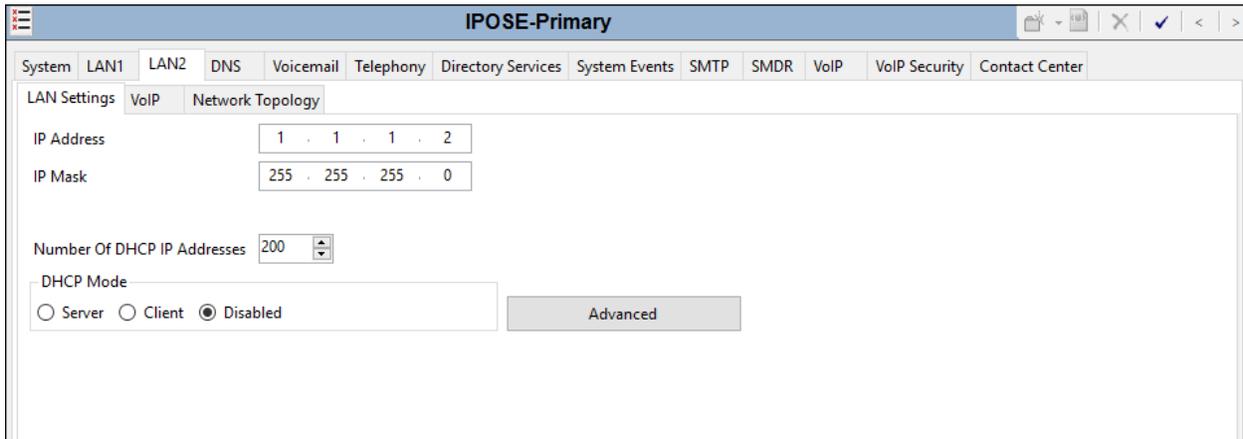


Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** parameter is checked to allow the use of Avaya IP Deskphones using the H.323 protocol, such as the Avaya 1616 and 9611 used in the sample configuration. The **SIP Registrar Enable** parameter is checked to allow Avaya 1140E and Avaya Communicator usage. The **SIP Domain Name** and **SIP Registrar FQDN** may be set according to customer requirements.



## 5.2.2. LAN 2 Settings

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



The screenshot displays the configuration interface for IPOSE-Primary. The main title is "IPOSE-Primary". Below the title is a navigation bar with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VoIP, VoIP Security, and Contact Center. The "LAN2" tab is selected. Underneath, there are sub-tabs for LAN Settings, VoIP, and Network Topology. The "LAN Settings" sub-tab is active. The configuration fields are as follows:

IP Address	1 . 1 . 1 . 2
IP Mask	255 . 255 . 255 . 0
Number Of DHCP IP Addresses	200
DHCP Mode	<input type="radio"/> Server <input type="radio"/> Client <input checked="" type="radio"/> Disabled

An "Advanced" button is located to the right of the DHCP Mode options.

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

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

The screenshot displays the configuration page for IPOSE-Primary, specifically the VoIP tab. The interface includes a navigation menu at the top with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VoIP, VoIP Security, and Contact Center. The VoIP tab is selected, and the sub-tab is Network Topology. The configuration is divided into several sections:

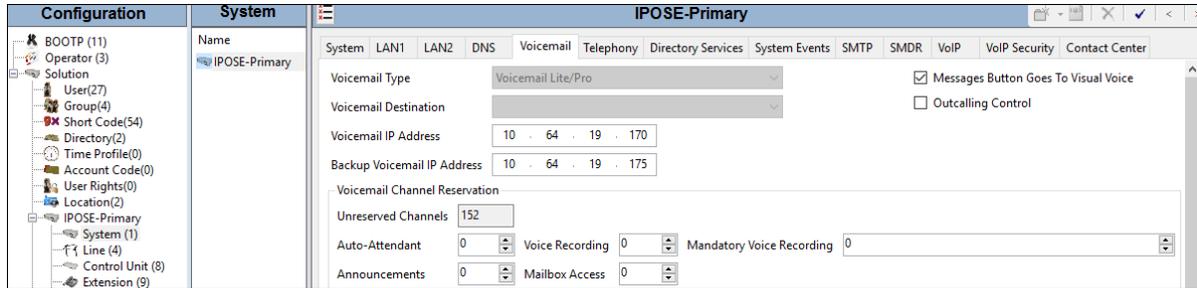
- H.323 Settings:** Includes checkboxes for H.323 Gatekeeper Enable, Auto-create Extension, Auto-create User, and H.323 Remote Extension Enable. A dropdown menu for H.323 Signaling over TLS is set to Disabled, and the Remote Call Signaling Port is 1720.
- SIP Settings:** Includes a checked checkbox for SIP Trunks Enable, and unchecked checkboxes for SIP Registrar Enable and SIP Remote Extension Enable. There is an Auto-create Extension/User checkbox. The SIP Domain Name and SIP Registrar FQDN fields are empty. Under Layer 4 Protocol, UDP and TCP are checked, with ports 5060 for both local and remote. TLS is unchecked, with port 5061 for both local and remote. The Challenge Expiration Time (sec) is set to 10.
- RTP Settings:** Includes two Port Number Range sections. The first section has a Minimum of 16384 and a Maximum of 32766. The second section (NAT) has a Minimum of 40750 and a Maximum of 50750. The Enable RTCP Monitoring on Port 5005 checkbox is checked. The RTCP collector IP address for phones is 0.0.0.0. The Keepalives section has a Scope of RTP-RTCP, a Periodic timeout of 30, and Initial keepalives set to Enabled.

Scrolling down, 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). See **Section 2.2** for limitations with IP Office Server Edition. 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.

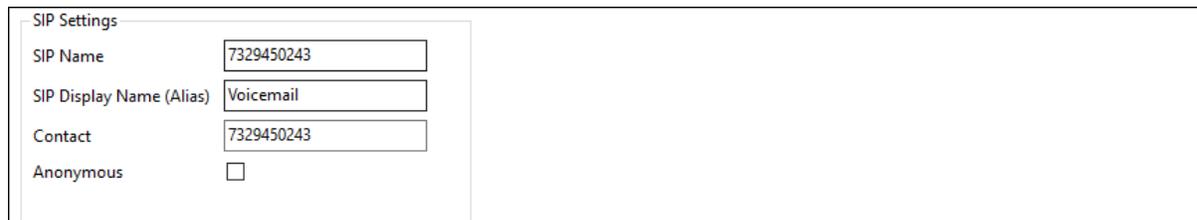
Select the **Network Topology** tab as shown in the following screen. The **Firewall/NAT Type** is set to “**Open Internet**” in the sample configuration. Note that the **Firewall/NAT Type** parameter may need to be set differently, depending on the type of firewall or Network Address Translation device used at the customer premise. The **Binding Refresh Time (sec)** can be used to lower the SIP OPTIONS timing from the default of 300 seconds. During the testing, the Binding Refresh Time was varied (e.g., 90 seconds, 120 seconds) to test SIP OPTIONS timing. The **Public IP Address** is set to the IP address known to Verizon. In the sample configuration, this is “**1.1.1.2**”. The **UDP Public Port** is set to “**5060**”.

### 5.2.3. Voicemail Settings

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**”. The **Voicemail IP Address** in the sample configuration is “**10.64.19.170**”, the IP address of the Primary server running the Voicemail Pro software. The **Backup Voicemail IP Address** is “**10.64.19.175**”, the IP address of the Secondary server.

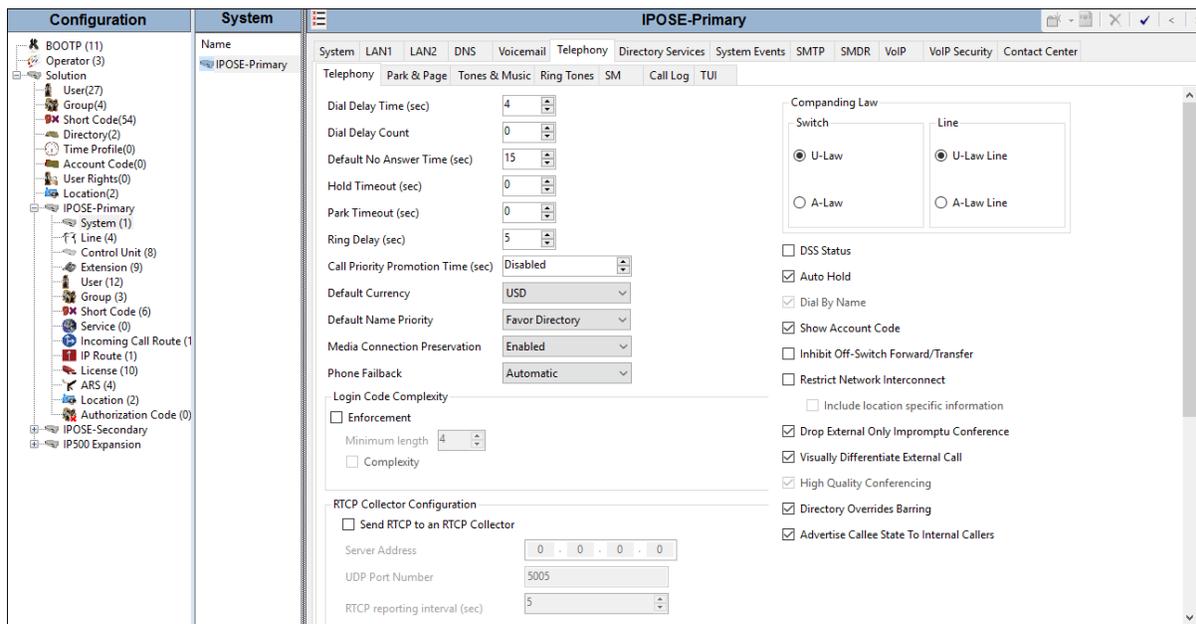


In the sample configuration, the “Callback” application of Avaya Voicemail Pro was used to allow Voicemail Pro to call out via the SIP Line to Verizon Business when a message is left in a voice mailbox. The **SIP Settings** shown in the screen below enable the Primary server to populate the SIP headers 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 5.5**).



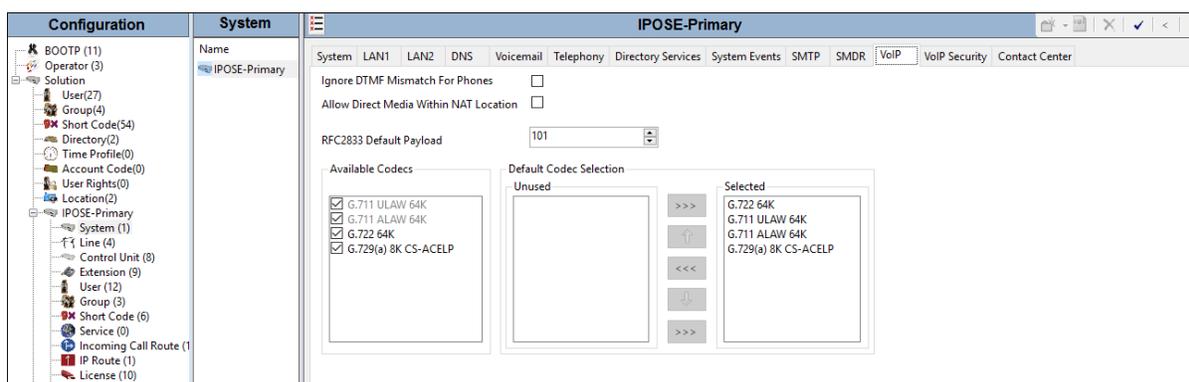
## 5.2.4. System Telephony Configuration

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** parameter is unchecked so that call forwarding and call transfer to PSTN destinations via the Verizon Business IP Trunk service can be tested. That is, a call can arrive to IP Office via the Verizon Business IP Trunk, and be forwarded or transferred back to the PSTN with the outbound leg of the call using the Verizon Business IP Trunk service. The **Companing Law** parameters are set to “U-Law” as is typical in North American locales. Other parameters on this screen may be set according to customer requirements.



## 5.2.5. System Codecs Configuration

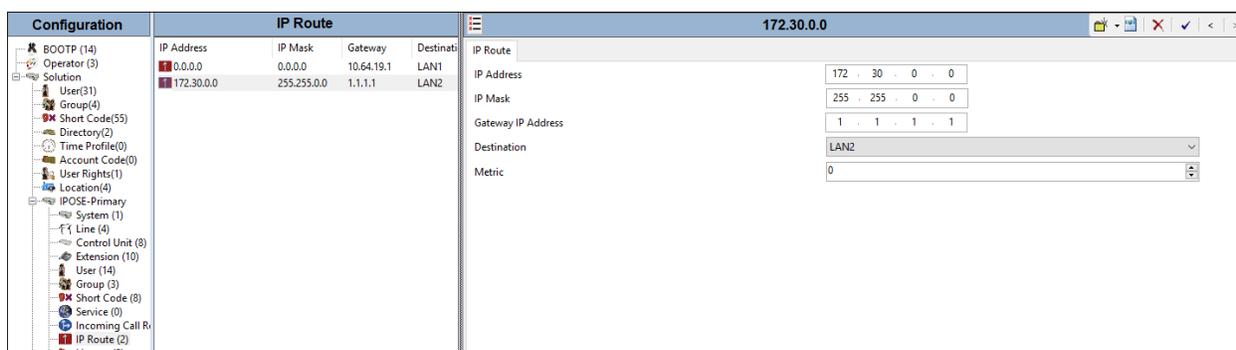
To view or change system codec settings, select the **VoIP** tab. On the left, observe the list of **Available Codecs**. In the example screen below, which is not intended to be prescriptive, the parameter next to each codec is checked, making all the codecs available in other screens where codec configuration may be performed (such as the SIP Line in **Section 5.4.6**). The **Default Codec Selection** area enables the codec preference order to be configured on a system-wide basis, using the up, down, left, and right arrows. By default, all IP (SIP and H.323) lines and extensions will assume the system default codec selection, unless configured otherwise for the specific line or extension. The **RFC2833 Default Payload** parameter is set to “**101**”, the value preferred by Verizon Business.



## 5.3. IP Route

In the sample configuration, the IP Office LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is “**10.64.19.1**”.

The IP Office LAN2 port is physically connected to the Verizon PIP network and has a default gateway of “**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 route using **Destination** “**LAN2**”.



## 5.4. SIP Line

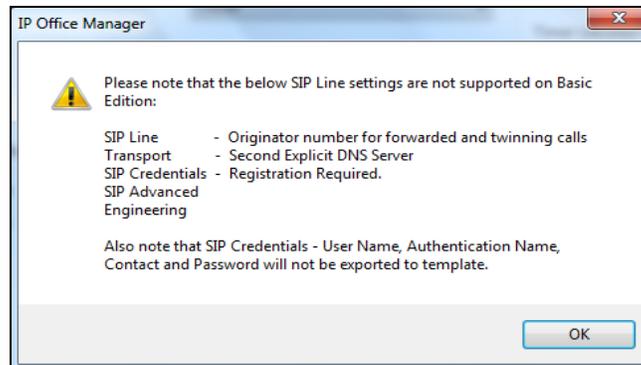
This section shows the configuration screens for the SIP Line in IP Office Release 10.0. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.4.2** to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses
- SIP credentials (if applicable)
- SIP URI entries
- Setting of the **Use Network Topology Info** field on the Transport tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.3 – 5.4.7**.

In addition, the following SIP Line settings are not supported on Basic Edition:

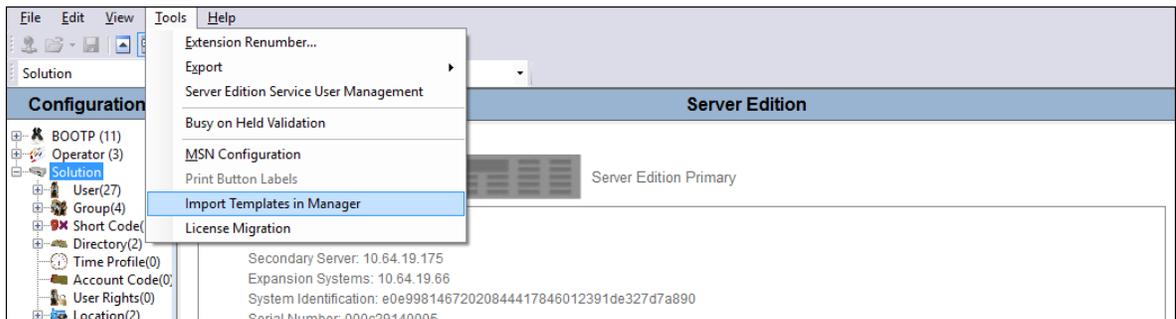


Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.3 – 5.4.7**.

### 5.4.1. Importing a SIP Line Template

**Note** – DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500v2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer’s environment.

1. Copy a previously created template file to a location (e.g., *\temp*) on the same computer where IP Office Manager is installed.
2. Import the template into IP Office Manager. From IP Office Manager, select **Tools** → **Import Templates in Manager**.

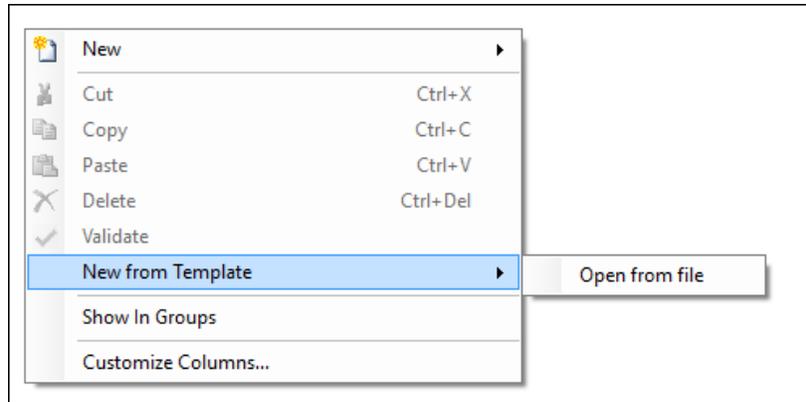


3. A folder browser will open (not shown). Select the directory used in **step 1** to store the template(s) (e.g., *\temp*). In the sample configuration, template file **New-Template.xml** was imported. The template files are automatically copied into the IP Office default template location, **C:\Program Files\Avaya\IP Office\Manager\Templates**.
4. After the import is complete, a final import status pop-up window will open stating success or failure.

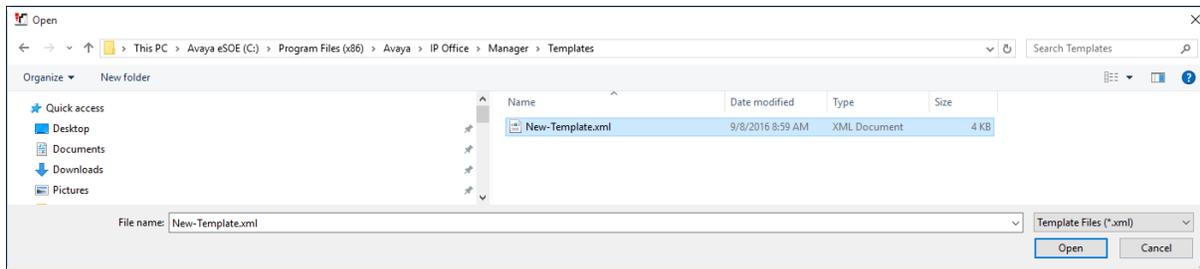


### 5.4.2. Creating a SIP Trunk from an XML Template

1. To create the SIP Trunk from a template, right-click on **Line** in the Navigation pane, and hover over **New from Template**, and select **Open from file**.



Navigate to **C:\Program Files\Avaya\IP Office\Manager\Templates**. Select **\*.xml** as the file type, find the template, and click **Open**.



The newly created SIP Line will appear in the Navigation pane (e.g., SIP Line 2).

Line Number	Line Type	Line SubType
1	IP Office Line	WebSocket Server SCN
3	IP Office Line	WebSocket Server SCN
2	SIP Line	

Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.3 –5.4.7**.

### 5.4.3. SIP Line – SIP Line Tab

The **SIP Line** tab in the Details pane is shown below for **Line Number 21**, used for the Verizon Business IP Trunk service. The **ITSP Domain Name** is configured to the IP address supplied by Verizon (“**172.30.209.21**”). The **Local Domain** is set to the IP address of the Avaya IP Office LAN2 SIP trunking interface (e.g., “**1.1.1.2**”). 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 LAN2, as shown in **Section 5.2.2**.

Under **Session Timers**, the **Refresh Method** is set to “**Re-invite**” and the **Timer (seconds)** is set to “**1800**”. With this configuration, IP Office will send re-INVITES every 15 minutes (half of the set value) to keep the active session alive.

Under **Redirect and Transfer**, the default automatic determination of **Incoming Supervised REFER** and **Outgoing Supervised REFER** is “**Auto**”. Alternatively, the default can be overridden with “**Never**” to explicitly disable use of supervised REFER, or “**Always**” to explicitly enable use of supervised REFER, as shown below. The **Send 302 Moved Temporarily** setting is unchecked, as Verizon does not support receiving a 302 Moved Temporarily message. Optionally, the **Outgoing Blind REFER** parameter can be checked to enable use of REFER for blind transfers.

The screenshot shows the 'SIP Line' configuration page with the 'SIP Advanced' tab selected. The 'Redirect and Transfer' section is expanded, showing the following settings:

- Incoming Supervised REFER: Always
- Outgoing Supervised REFER: Always
- Send 302 Moved Temporarily:
- Outgoing Blind REFER:

#### 5.4.4. SIP Line - Transport Tab

Select the **Transport** tab. The **ITSP Proxy Address** is set to the IP address provided by Verizon as shown in **Figure 1**. In the **Network Configuration** area, “**UDP**” is selected as the **Layer 4 Protocol**. The **Send Port** can retain the default value “**5060**”. The **Use Network Topology Info** parameter is set to “**LAN 2**”.

The screenshot shows the 'SIP Line' configuration page with the 'Transport' tab selected. The 'Network Configuration' section is expanded, showing the following settings:

- ITSP Proxy Address: 172.30.209.21
- Layer 4 Protocol: UDP
- Send Port: 5060
- Use Network Topology Info: LAN 2
- Listen Port: 5060
- Explicit DNS Server(s): 0 . 0 . 0 . 0
- Calls Route via Registrar:
- Separate Registrar:

### 5.4.5. 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 URI** area will be opened.

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header	Credential	Max Calls	
1	21 21	<Internal>	<Internal>	<Internal>	None	PAI		Diversion	None	0: <Non...	10	Add...
2	21 0	7329450240	7329450240	7329450240	None	PAI		None	None	0: <Non...	10	Remove
3	21 0	7329450288	7329450288	7329450288	None	PAI		None	None	0: <Non...	10	Edit...

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 URI** tab for individual users will be used to populate the SIP headers. The **Identity** parameter is set to “**None**”, and the **Header** parameter is set to “**P Asserted ID**”.

Edit URI

Local URI:

Contact:

Display Name:

Identity:

Header:

Under **Forwarding and Twinning**, the **Originator Number** is left blank, and the **Send Caller ID** parameter is set to “**Diversion Header**”. With this setting 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. 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. IP Office will also include the Diversion header for calls that are call forwarded out the SIP Line to Verizon.

Forwarding And Twinning

Originator Number:

Send Caller ID:

The **Diversion Header** field is new in IP Office 10.0, and can be configured with a value to allow IP Office to include a Diversion Header in the original outbound SIP INVITE message. This can be used to send a known billing number to accompany a branch office number, toll-free number, or extension number in the From Header to present as the caller ID of an outbound call. An example of this would be the use of Verizon’s Unscreened ANI service offer, as shown in

**Section 11.** For this URI entry, the **Diversion Header** field is set to “None”. 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 “21”, will be referenced when configuring Incoming Call Routes to map inbound SIP trunk calls to IP Office destinations in **Section 5.8**. The **Outgoing Group** parameter, set here to “21”, will be used for routing outbound calls to Verizon via the Short Codes (**Section 5.7**). The **Max Sessions** 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.

Diversion Header	None
Registration	0: <None>
Incoming Group	21
Outgoing Group	21
Max Sessions	10

In the sample configuration, the single SIP URI shown below 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.

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header	Credential	Max Calls
1	21 21	<Internal>	<Internal>	<Internal>	None	PAI		Diversion	None	0: <Non...	10
2	21 0	7329450240	7329450240	7329450240	None	PAI		None	None	0: <Non...	10
3	21 0	7329450288	7329450288	7329450288	None	PAI		None	None	0: <Non...	10

Edit URI	
Local URI	Use Internal Data
Contact	Use Internal Data
Display Name	Use Internal Data
Identity	
Identity	None
Header	P Asserted ID
Forwarding And Twinning	
Originator Number	
Send Caller ID	Diversion Header
Diversion Header	None
Registration	0: <None>
Incoming Group	21
Outgoing Group	21
Max Sessions	10

URIs “2” and “3” display service numbers, such as a DID number routed directly to voicemail or DID used for Mobile Call Control. DID numbers that IP Office should admit can be entered into the **Local URI** and **Contact** fields instead of “Use Internal Data”. The numbers 732-945-0240 and 732-945-0288 will be assigned as service numbers in the Incoming Call Routes in **Section 5.8**. These URI entries will not be used for outbound dialing; therefore an unused number is specified for the **Outgoing Group**.

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header	Credential	Max Calls
1	21 21	<Internal>	<Internal>	<Internal>	None	PAI		Diversion	None	0: <Non...	10
2	21 0	7329450240	7329450240	7329450240	None	PAI		None	None	0: <Non...	10
3	21 0	7329450288	7329450288	7329450288	None	PAI		None	None	0: <Non...	10

**Edit URI**

Local URI:

Contact:

Display Name:

Identity:

Header:

**Forwarding And Twinning**

Originator Number:

Send Caller ID:

Diversion Header:

Registration:

Incoming Group:

Outgoing Group:

SIP Line Transport SIP URI VoIP SIP Credentials SIP Advanced Engineering

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header	Credential	Max Calls
1	21 21	<Internal>	<Internal>	<Internal>	None	PAI		Diversion	None	0: <Non...	10
2	21 0	7329450240	7329450240	7329450240	None	PAI		None	None	0: <Non...	10
3	21 0	7329450288	7329450288	7329450288	None	PAI		None	None	0: <Non...	10

**Edit URI**

Local URI:

Contact:

Display Name:

Identity:

Header:

**Forwarding And Twinning**

Originator Number:

Send Caller ID:

Diversion Header:

Registration:

Incoming Group:

Outgoing Group:

OK

Cancel

### 5.4.6. SIP Line - VoIP Tab

Select the **VoIP** tab. The **Codec Selection** drop-down parameter “**System Default**” (default) will match the codecs set in the system wide Default Selection list (**System → Codecs**). In the sample configuration, “**Custom**” is selected and codecs preferred by Verizon are included (i.e., G.729(a) 8K CS-ACELP and G.711 ULAW 64K). This will cause IP Office to include G.729a and G.711MU in the Session Description Protocol (SDP) offer, in that order. Set the **Fax Transport Support** drop-down to “**T38 Fallback**”. This enables T.38 to be used if supported and will fall back to G.711 if not. The **DTMF Support** parameter can remain set to the default value “**RFC2833/RFC4733**”. The **Media Security** parameter can retain its default value of “**Disabled**” as Verizon does not support media encryption. 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 **PRACK/100rel Supported** parameter can be checked to enable support for the PRACK (Provisional Reliable Acknowledgement) message on SIP trunks.

For PSTN originations, Verizon preferred the G.729a codec in the SDP, while also allowing the G.711MU codec. During testing, the IP Office configuration was varied such that G.711MU was the preferred or only codec listed, and G.711MU calls were also successfully verified.

The screenshot shows the configuration interface for a SIP Line, specifically the VoIP tab. The interface includes several sections:

- Codec Selection:** A dropdown menu is set to "Custom". Below it are two lists: "Unused" (G.711 ALAW 64K, G.722 64K) and "Selected" (G.729(a) 8K CS-ACELP, G.711 ULAW 64K). Navigation buttons (right arrow, up arrow, left arrow, down arrow, right arrow) are positioned between the lists.
- Fax Transport Support:** A dropdown menu set to "T38 Fallback".
- DTMF Support:** A dropdown menu set to "RFC2833/RFC4733".
- Media Security:** A dropdown menu set to "Disabled".
- Checkboxes:** A group of checkboxes on the right side: "Local Hold Music" (unchecked), "Re-invite Supported" (checked), "Codec Lockdown" (unchecked), "Allow Direct Media Path" (unchecked), "Force direct media with phones" (unchecked), and "PRACK/100rel Supported" (checked).

### 5.4.7. SIP Line – SIP Advanced Tab

Select the **SIP Advanced** tab. In the **Identity** area, the **Use PAI for Privacy** parameter is checked to include the caller's DID number in the P-Asserted-Identity (PAI) SIP header for a privacy requested call. This PAI SIP header is required by Verizon Business to admit an otherwise anonymous caller to the network. The **Caller ID from From header** parameter is checked to have IP Office use the Caller ID information in the From SIP header rather than the PAI or Contact SIP header for inbound calls. This will allow the Caller Name presented in the From SIP header by Verizon Business to also be included in the Caller ID.

In the **Media** area, the **Indicate HOLD** parameter is checked to have IP Office send an INVITE with media attribute "sendonly", indicating the call was placed on hold. This is the preferred behavior for Verizon Business to indicate placing a call on hold.

In the **Call Control** area, the **Emulate NOTIFY for REFER** parameter is checked. This is required for SIP endpoints that perform refer based transfers across the SIP line. See **Section 2.2** for more details. The **No REFER if using Diversion** parameter is checked to prevent IP Office from using the SIP REFER method on call forwarded scenarios that use a Diversion SIP header. Verizon does not support this type of refer, and would respond with a "603 Decline" SIP message.

The screenshot shows the SIP Line configuration interface, specifically the SIP Advanced tab. The interface is divided into several sections:

- Addressing:** Association Method is set to "By Source IP address", Call Routing Method is "Request URI", and Suppress DNS SRV Lookups is unchecked.
- Identity:** A list of checkboxes includes "Use 'phone-context'", "Add user=phone", "Use + for International", "Use PAI for Privacy" (checked), "Use Domain for PAI", "Swap From and PAI/Diversion", "Caller ID from From header" (checked), "Send From In Clear", "Cache Auth Credentials" (checked), "User-Agent and Server Headers" (empty text box), and "Send Location Info" (set to "Emergency Calls").
- Media:** A list of checkboxes includes "Allow Empty INVITE", "Send Empty re-INVITE", "Allow To Tag Change", "P-Early-Media Support" (set to "None"), "Send SilenceSup=Off", "Force Early Direct Media", "Media Connection Preservation" (set to "Disabled"), and "Indicate HOLD" (checked).
- Call Control:** A list of settings includes "Call Initiation Timeout (s)" (4), "Call Queuing Timeout (mins)" (5), "Service Busy Response" (486 - Busy Here), "on No User Responding Send" (408-Request Timeout), "Action on CAC Location Limit" (Allow Voicemail), "Suppress Q.850 Reason Header" (unchecked), "Emulate NOTIFY for REFER" (checked), and "No REFER if using Diversion" (checked).

**Note** – 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 \*67 to access the SIP Line, as described in **Section 5.7**. Certain Avaya endpoints can also request privacy, without dialing a unique short code, by accessing the telephone user interface screen **Features** → **Call Settings** → **Withhold Number**. Consult reference [7] for more information. The **Withhold Number** parameter may be set to “On” (i.e., for privacy). Specific users may be configured to always withhold calling line identification by checking the **Anonymous** field in the **SIP** tab for the user (**Section 5.6.1**).

## 5.5. IP Office Line

IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. Below is the IP Office Line to the Expansion System.

Line Number	Line Type	Line SubType
1	IP Office Line	WebSocket Server SCN
3	IP Office Line	WebSocket Server SCN
9	SIP Line	

**IP Office Line - Line 1**

Line Short Codes VoIP Settings

Line Number: 1 Telephone Number: [ ]

Transport Type: WebSocket Server Prefix: [ ]

Networking Level: SCN Outgoing Group ID: 99001

Security: Medium Number of Channels: 250

Outgoing Channels: 250

Gateway

Address: 10 . 64 . 19 . 66

Location: 2: Denver

Password: [ ]

Confirm Password: [ ]

SCN Resiliency Options

Supports Resiliency

Backs up my IP phones

Backs up my hunt groups

Backs up my IP DECT phones

Description: [ ]

In the sample configuration, a fax machine is connected to one of the analog ports on the Expansion System. To accommodate T.38 fax, set the **Fax Transport Support** drop-down to “**T38 Fallback**” on the **VoIP Settings** tab.

**IP Office Line - Line 1**

Line Short Codes VoIP Settings

Out Of Band DTMF

Allow Direct Media Path

Codec Selection: System Default

Unused: [ ]

Selected: G.722 64K, G.711 ULAW 64K, G.711 ALAW 64K, G.729(a) 8K CS-ACELP

Fax Transport Support: T38 Fallback

Call Initiation Timeout (s): 4

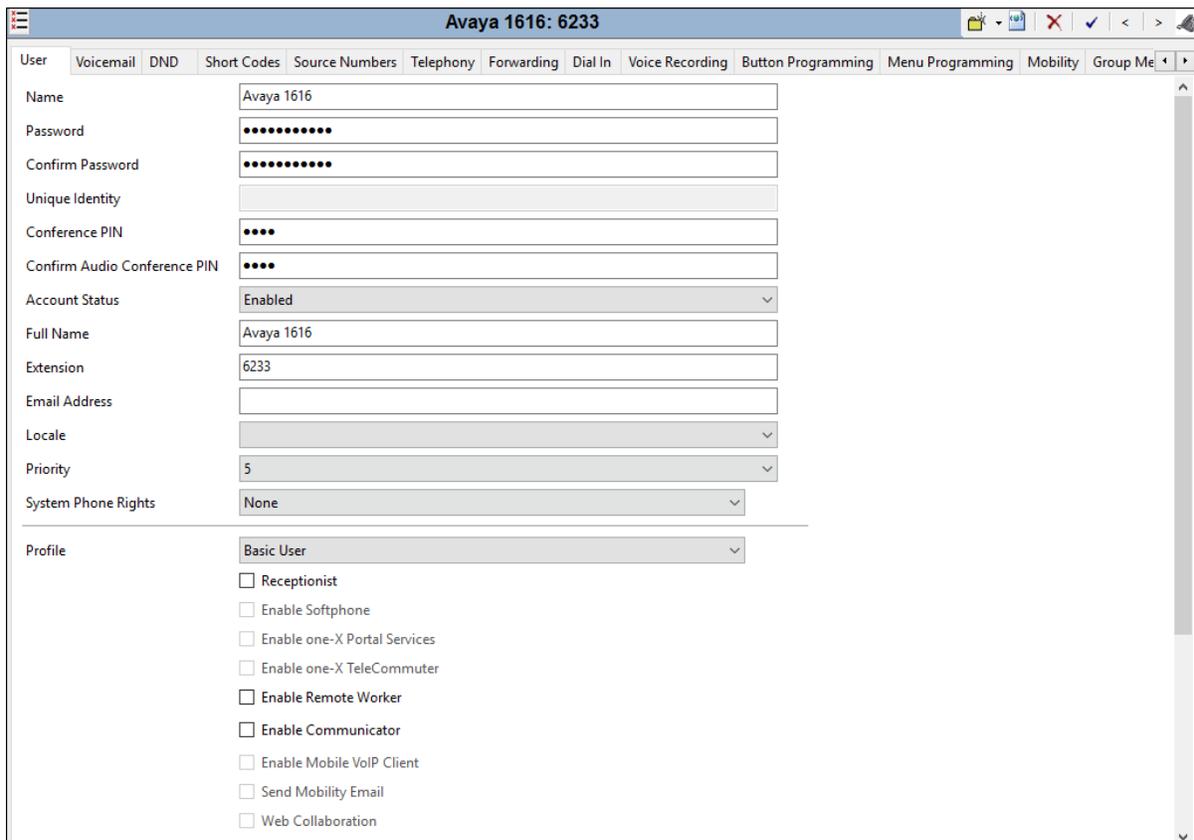
Media Security: Same as System (Disabled)

## 5.6. Users, Extensions, and Hunt Groups

In this section, examples of an IP Office User, Extension, and Hunt Group 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.

### 5.6.1. H.323 User 6233

The following screen shows the **User** tab for user 6233. As shown in **Figure 1**, this user corresponds to the Avaya 1616 H.323 endpoint.



The screenshot displays the configuration interface for user 6233. The title bar reads "Avaya 1616: 6233". The interface includes a navigation pane with tabs for "User", "Voicemail", "DND", "Short Codes", "Source Numbers", "Telephony", "Forwarding", "Dial In", "Voice Recording", "Button Programming", "Menu Programming", "Mobility", and "Group Me". The "User" tab is active, showing the following configuration fields:

- Name: Avaya 1616
- Password: [Redacted]
- Confirm Password: [Redacted]
- Unique Identity: [Empty]
- Conference PIN: [Redacted]
- Confirm Audio Conference PIN: [Redacted]
- Account Status: Enabled
- Full Name: Avaya 1616
- Extension: 6233
- Email Address: [Empty]
- Locale: [Empty]
- Priority: 5
- System Phone Rights: None
- Profile: Basic User

Below the Profile dropdown, there are several checkboxes for additional features:

- Receptionist
- Enable Softphone
- Enable one-X Portal Services
- Enable one-X TeleCommuter
- Enable Remote Worker
- Enable Communicator
- Enable Mobile VoIP Client
- Send Mobility Email
- Web Collaboration

The following screen shows the **SIP** tab for user 6233. The **SIP Name** and **Contact** parameters are configured with the DID number of the user, “**7329450233**”. 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** parameter may be checked to withhold the user’s information from the network. See **Section 5.7** for a method of using a short code (rather than static user provisioning) to place an anonymous call.

The screenshot shows the configuration page for user 6233, specifically the SIP tab. The fields are as follows:

SIP Name	7329450233
SIP Display Name (Alias)	Avaya 1616
Contact	7329450233

The **Anonymous** checkbox is currently unchecked.

The following screen shows the **Mobility** tab for user 6233. 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, including the dial access code for ARS, in this case “**913035382177**”. Other options can be set according to customer requirements.

The screenshot shows the configuration page for user 6233, specifically the Mobility tab. The configuration is as follows:

- 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): 913035382177
    - Twinning Time Profile: <None>
    - Mobile Dial Delay (sec): 0
    - Mobile Answer Guard (sec): 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.

## 5.6.2. 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 **Group** from the Navigation pane, and select **New**. To view or edit an existing hunt group, select **Group** from the Navigation pane, and the appropriate hunt group from the Group pane.

The following screen shows the **Group** tab for hunt group 401. The telephone extensions in the **User List** are rung based on the extension that has been unused for the longest period, due to the **Ring Mode** setting “**Longest Waiting**” (i.e., most idle user to receive the next call). Click the **Edit** button to change the **User List**.

Extension	Name	System
<input checked="" type="checkbox"/>	6242	Avaya 9508 IP500 Expansion
<input checked="" type="checkbox"/>	6237	Avaya 9611 IPOSE-Primary

The following screen shows the **SIP** tab for hunt group 401. The **SIP Name** and **Contact** are configured with Verizon DID “7329450245”. Later, in **Section 5.8**, an Incoming Call Route will map 7329450245 to this hunt group based on the information entered on this tab.

Group	Queuing	Overflow	Fallback	Voicemail	Voice Recording	Announcements	SIP
SIP Name	7329450245						
SIP Display Name (Alias)	Call Center						
Contact	7329450245						
<input type="checkbox"/> Anonymous							

## 5.7. 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** (not shown) 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”. 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 “50: Main”, configurable via ARS. See **Section 5.9** for example ARS route configuration for “50: Main” as well as an alternate route.

Short Code	
Code	9N
Feature	Dial
Telephone Number	N
Line Group ID	50: Main
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

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 “\*67N” is illustrated. This short code is similar to the “9N” short code except that the **Telephone Number** field begins with the letter “W”, which means “withhold the outgoing calling line identification”. In the case of the SIP Line connecting to Verizon documented in these Application Notes, when a user dials \*67 plus any number “N”, IP Office will include the user’s telephone number in the P-Asserted-Identity (PAI) header (see **Section 5.4.7**) 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.

*67N: Dial	
Code	*67N
Feature	Dial
Telephone Number	WN
Line Group ID	50: Main
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

The following screen illustrates a solution level short code, common to all servers, that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code “FNE31”** is defined for **Feature “FNE Service”** to **Telephone Number “31”** (Mobile Call Control). This short code will be used as means to allow a Verizon DID to be programmed to route directly to this feature, via inclusion of this short code as the destination of an Incoming Call Route. See **Section 5.8**. This feature is used to provide dial tone to twinned mobile devices (e.g., cell phone) directly from IP Office; once dial tone is received the user can perform dialing actions including making calls and activating Short Codes.

IP Office	Code	Teleph
<Shared>	FNE31	31
<Shared>	FNE33	33
<Shared>	57N	"Modr
<Shared>	56N	"Modr
<Shared>	*99	"edit_r
<Shared>	*71'N#	N
<Shared>	*70'N#	N
<Shared>	*69	"VzIPC
<Shared>	*57'N#	N
<Shared>	*559	
<Shared>	*556	"menu
<Shared>	555	"M...

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

## 5.8. Incoming Call Routes

In this section, IP Office Incoming Call Routes are illustrated. 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, a simple incoming call route is illustrated. The **Line Group Id** is “21”, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to Verizon Business in **Section 5.4.5**. The **Incoming Number** field is left blank to match all details of the number in the To header.

Line Group ID	Incoming Number	Destination
21	7329450240	FNE31
21	7329450288	VM:TestMeetMe

The following **Destinations** tab for the incoming call route contains the **Destination “.”** entered manually. This will match the **Incoming Number** field as the destination and route the call based on the information in the SIP tab for the user or hunt group as illustrated in **Section 5.6**. For example, a call to 732-945-0233 will be routed to user 6233, because this user has 7329450233 configured for the **SIP Name** and **Contact** parameters.

TimeProfile	Destination	Fallback Extension
Default Value	.	

In the following screen, the incoming call route for **Incoming Number “7329450240”** is illustrated. The **Line Group Id** is “21”, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to Verizon Business in **Section 5.4.5**.

Line Group ID	Incoming Number	Destination
21	7329450240	FNE31
21	7329450288	VM:TestMeetMe

The following **Destinations** tab for the incoming call route contains the **Destination “FNE31”** entered manually. The name “FNE31” is the short code for accessing the “Mobile Call Control” application configured in **Section 5.7**, and 732-945-0240 was configured in **Section 5.4.5** on the SIP URI tab as an incoming number. An incoming call to 732-945-0240 will be delivered directly to internal dial tone from the IP Office, allowing the caller to perform dialing actions including making calls and activating Short Codes. The incoming caller ID must match the Twinned Mobile Number entered in the User Mobility tab in **Section 5.6.1**; otherwise the IP Office responds with a “486 Busy Here” and the caller will hear a busy tone.

Configuration	Incoming Call Route	21 7329450240		
<ul style="list-style-type: none"> <li>BOOTP (14)</li> <li>Operator (3)</li> <li>Solution</li> <li>User(31)</li> <li>Group(4)</li> <li>Short Code(55)</li> <li>Directory(2)</li> <li>Time Profile(0)</li> </ul>	Line Group ID	Incoming Number	Destination	Standard
	21	7329450240	FNE31	Voice Recording
	21	7329450288	VM:TestMeetMe	Destinations
				TimeProfile
				Destination
				Fallback Extension
				Default Value
				FNE31

Similarly, the following **Destinations** tab for an incoming call route contains the **Destination “VM:TestMeetMe”** entered manually. An incoming call to 732-945-0288 will be delivered directly to the Voicemail Pro Module “TestMeetMe”.

Configuration	Incoming Call Route	21 7329450288		
<ul style="list-style-type: none"> <li>BOOTP (14)</li> <li>Operator (3)</li> <li>Solution</li> <li>User(31)</li> <li>Group(4)</li> <li>Short Code(55)</li> <li>Directory(2)</li> </ul>	Line Group ID	Incoming Number	Destination	Standard
	21	7329450240	FNE31	Voice Recording
	21	7329450288	VM:TestMeetMe	Destinations
				TimeProfile
				Destination
				Fallback Extension
				Default Value
				VM:TestMeetMe

## 5.9. ARS

Alternate Route Selection (ARS) is used to route outbound traffic to the SIP line. 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. In the Details pane that appears, a collection of matching patterns (similar to short codes) can be entered to route calls as shown below.

The following screen shows a sample ARS configuration for the route named “**Main**”. The sequence of **Xs** used in the **Code** column specify the exact number of digits to be expected following the access. The first entry below shows that for calls to area codes in the North American Numbering Plan, the user dials 9 (stripped by the short code configuration), followed by 10 digits. The list of codes defined below is simply an example and not intended to be prescriptive. Other dialing codes may be appropriate for different customer networks. The **Line Group ID** is set to “**21**” matching the number of the **Outgoing Group** configured on the **SIP URI** tab of SIP Line 21 to Verizon Business (**Section 5.4.5**).

The screenshot shows the ARS configuration window for a route named "Main". The configuration includes the following fields and options:

- ARS Route ID: 50
- Route Name: Main
- Dial Delay Time: System Default (4)
- Description: (empty)
- In Service:  (checked)
- Time Profile: <None>
- Secondary Dial tone:  (checked), SystemTone
- Check User Call Barring:  (checked)
- Out of Service Route: <None>
- Out of Hours Route: <None>

Code	Telephone Number	Feature	Line Group ID
xxxxxxxx	.	Dial	21
x11	.	Dial	21
945xxxx	.	Dial	21
5551212	.	Dial	21
1411	.	Dial	21
911	.	Dial Emergency	21
1xxxxxxxx	.	Dial	21

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

Alternate Route Priority Level: 1

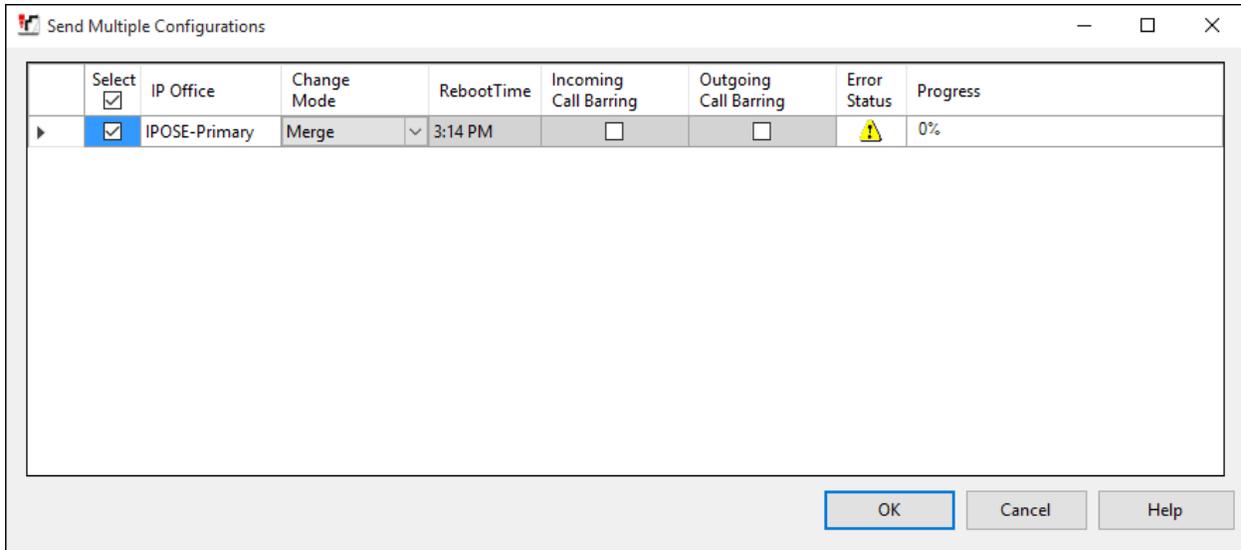
Alternate Route Wait Time: 5

Alternate Route: <None>

## 5.10. Save Configuration

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

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



## 6. Avaya IP Office Expansion Configuration

The section illustrates relevant aspects of the Expansion System used in the verification of these Application Notes. The Expansion System is configured by logging in to the Primary server. Navigate to **File** → **Open Configuration** (not shown), select the proper Primary server system from the pop-up window, and log in using the appropriate credentials. Clicking the “plus” sign next to **IP500 Expansion** on the left navigation pane will expand the menu on this server.

The screenshot displays the Avaya IP Office configuration interface, divided into two main panes: **Configuration** on the left and **System Inventory** on the right. The **Configuration** pane shows a tree view of system components, with **IP500 Expansion** selected and expanded. The **System Inventory** pane displays the configuration for the **Server Edition Expansion System**, including hardware and system settings.

**Configuration Pane (Left):**

- BOOTP (11)
- Operator (3)
- Solution
  - User (27)
    - Group (4)
    - Short Code (54)
    - Directory (2)
    - Time Profile (0)
    - Account Code (0)
    - User Rights (0)
    - Location (2)
    - IPOSE-Primary
    - IPOSE-Secondary
    - IP500 Expansion**
      - System (1)
      - Line (6)
      - Control Unit (3)
      - Extension (16)
      - User (17)
        - Group (1)
        - Short Code (14)
      - Service (0)
      - RAS (1)
      - Incoming Call Route (1)
      - WAN Port (0)
      - Firewall Profile (1)
      - IP Route (1)
      - License (4)
      - Tunnel (0)
      - ARS (3)
      - Location (2)
      - Authorization Code (0)

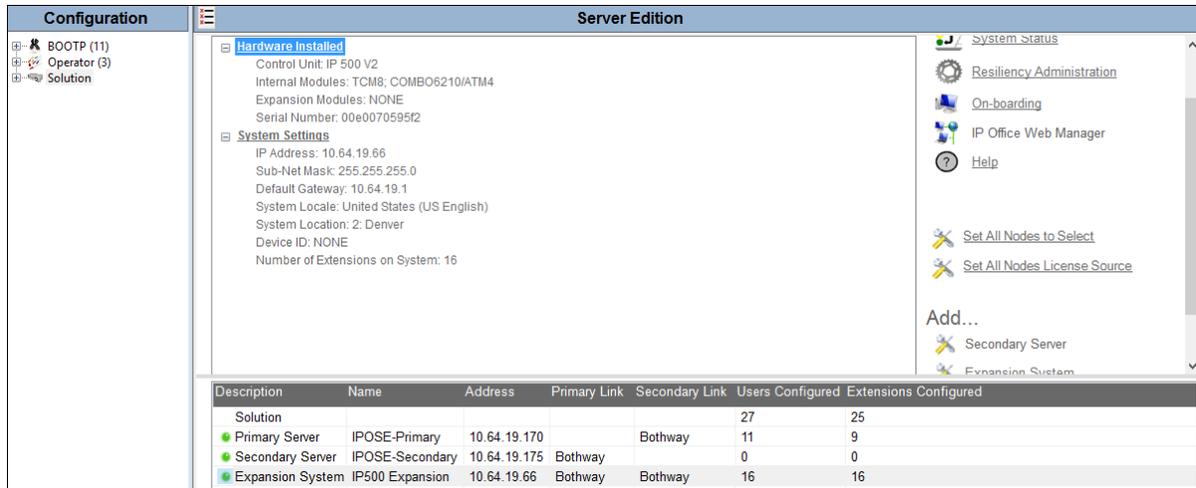
**System Inventory Pane (Right):**

Server Edition Expansion System

- Hardware Installed**
  - Control Unit: IP 500 V2
  - Internal Modules: TCM8; COMBO6210/ATM4
  - Expansion Modules: NONE
  - Serial Number: 00e0070595f2
- System Settings**
  - IP Address: 10.64.19.66
  - Sub-Net Mask: 255.255.255.0
  - Default Gateway: 10.64.19.1
  - System Locale: United States (US English)
  - System Location: 2: Denver
  - Device ID: NONE
  - Number of Extensions on System: 16
- Features Configured**
  - Licenses Installed: Power User(2); SIP Trunk Channels(25); Server Edition R10(1); Basic User(14)
  - Connected Extensions: 201; 6242
  - Users NOT Configured for Voicemail: Fax
  - Users assigned as Ex-Directory: NONE
  - Users assigned for Twinning: NONE
  - Users barred from making Outgoing Calls: NONE
  - Music on Hold: WAV File

### 6.1. Physical Hardware

In the sample configuration, looking at the Expansion System IP500 V2 from left to right, the first module is a TCM 8 Digital Station Module. This module supports BCM / Norstar T-Series and M-Series telephones. The second module is a COMBO6210/ATM4 module. This module is used to add a combination of ports to an IP500 V2 control unit and is not supported by IP500 control units. The module supports 10 voice compression channels. Codec support is G.711, G729A and G.723 with 64ms echo cancellation. G.722 is supported by IP Office Release 8.0 and higher. The “Combo” card will support 6 Digital Station ports for digital stations in slots 1-6 (except 3800, 4100, 4400, 7400, M and T-Series), 2 Analog Extension ports in slots 7-8, and 4 Analog Trunk ports in slots 9-12.

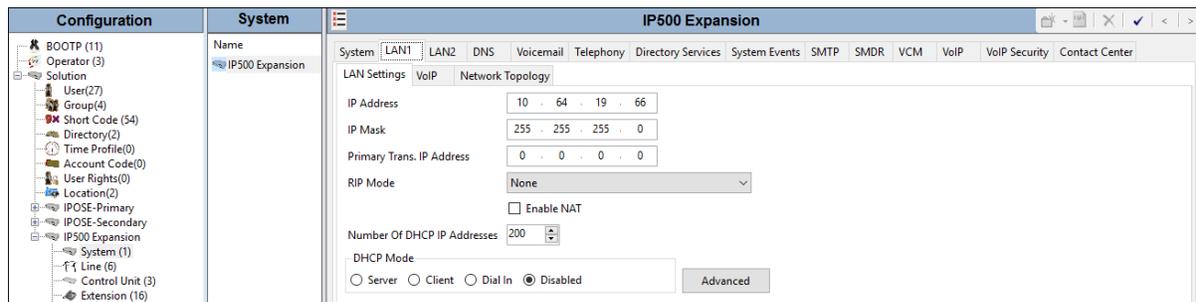


## 6.2. System Settings

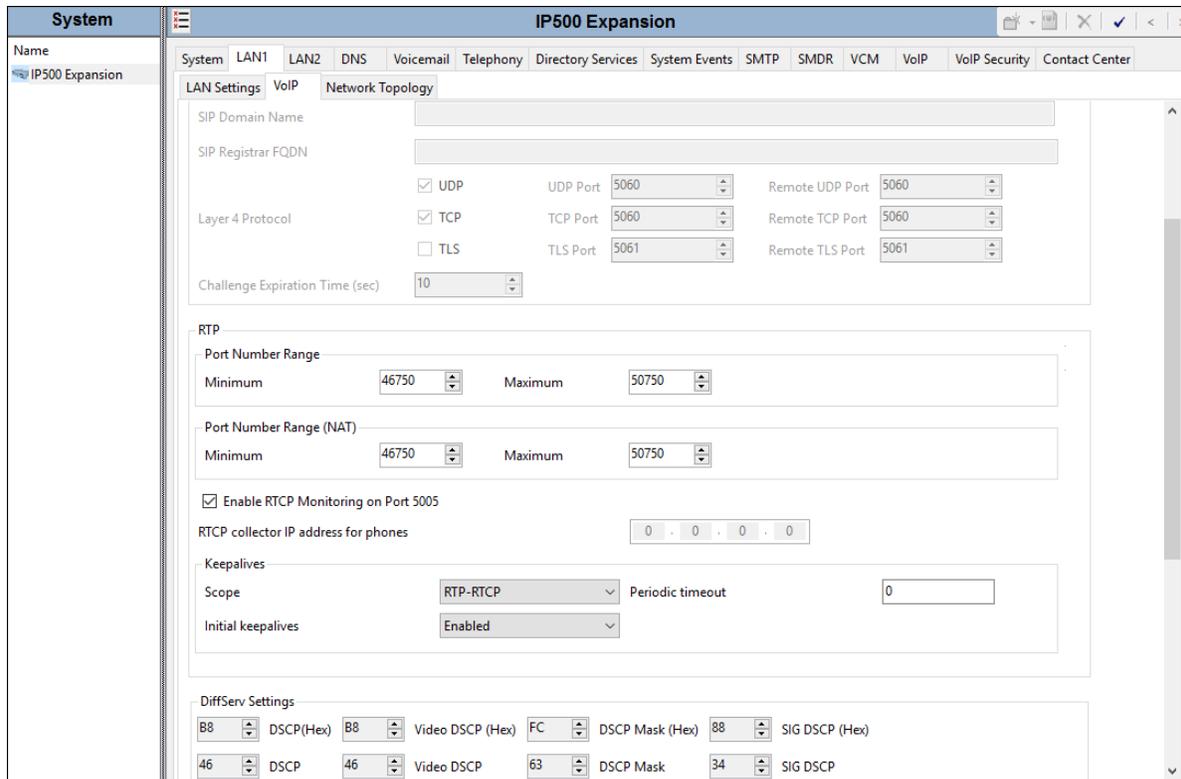
This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings. The subsection order corresponds to a left to right navigation of the tabs in the Details pane for System settings. For all of the following configuration sections, the **OK** button (not shown) must be selected in order for any changes to be saved.

### 6.2.1. LAN Settings

In the sample configuration, LAN1 is used to connect the Expansion System to the enterprise network. To view or configure the **IP Address** of LAN1, select the **LAN1** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP address of the Expansion System is “**10.64.19.66**”. Other parameters on this screen may be set according to customer requirements.

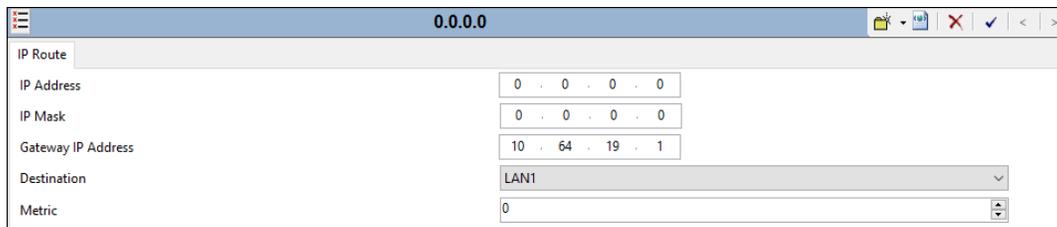


Select the **VoIP** tab as shown in the following screen. If desired, the **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media paths from Verizon to IP Office. The defaults are used here.



### 6.3. IP Route

Configuration is similar to the Primary server, as shown in **Section 5.3**. In the sample configuration, the Expansion System LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is “**10.64.19.1**”.

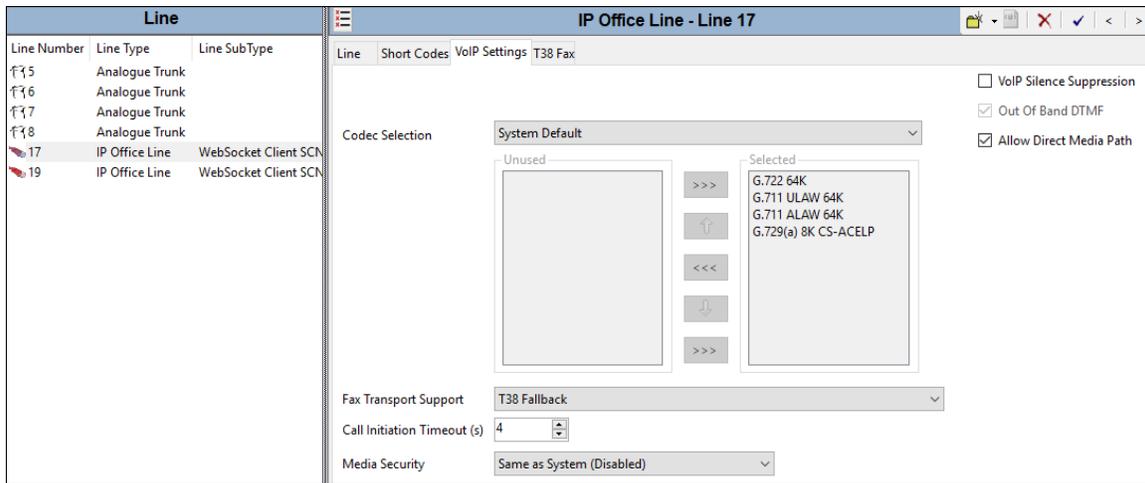


### 6.4. IP Office Line

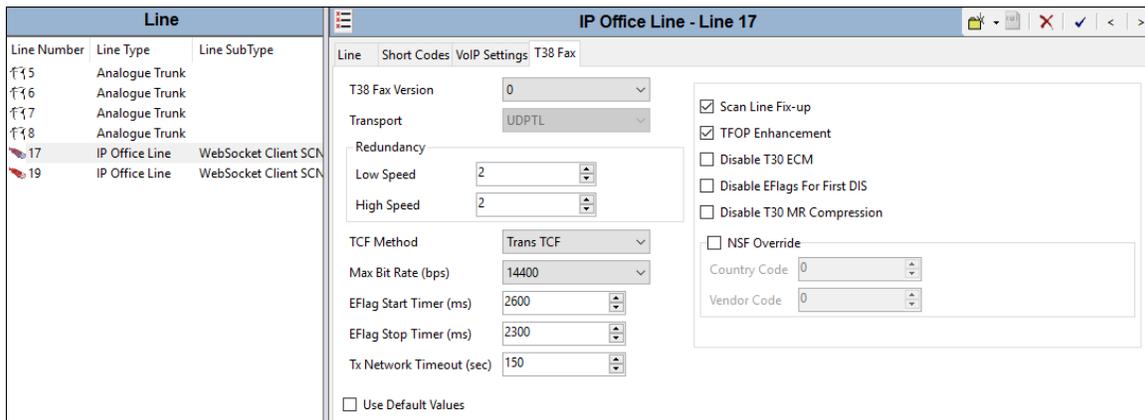
The IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. Below is the IP Office Line to the Primary server.

Line			IP Office Line - Line 17	
Line Number	Line Type	Line SubType	Line Short Codes VoIP Settings T38 Fax	
15	Analogue Trunk		Line Number	17
16	Analogue Trunk		Transport Type	WebSocket Client
17	Analogue Trunk		Networking Level	SCN
17	IP Office Line	WebSocket Client SCN	Security	Medium
19	IP Office Line	WebSocket Client SCN	Telephone Number	
			Prefix	
			Outgoing Group ID	99999
			Number of Channels	250
			Outgoing Channels	250
			Gateway	
			Address	10 . 64 . 19 . 170
			Location	2: Denver
			Port	443
			Password	.....
			Confirm Password	.....
			SCN Resiliency Options	
				<input checked="" type="checkbox"/> Supports Resiliency
				<input checked="" type="checkbox"/> Backs up my IP phones
				<input type="checkbox"/> Backs up my hunt groups
				<input type="checkbox"/> Backs up my IP DECT phones
			Description	

In the sample configuration, a fax machine is connected to one of the analog ports on the Expansion System. To accommodate T.38 fax, set the **Fax Transport Support** drop-down to **“T38 Fallback”** on the **VoIP Settings** tab.



Select the **T38 Fax** tab. The **T38 Fax Version** is set to **“0”**. In the **Redundancy** area, the **Low Speed** and **High Speed** parameters are set to **“2”**. All other values are left at default.



## 6.5. Short Codes

Similar to the configuration of the Primary server in **Section 5.7**, create a Short Code to access ARS. In the sample configuration, the **Line Group ID** is set to an ARS route illustrated in the next section.

Configuration	Short Code	Telep
BOOTP (11)		
Operator (3)		
Solution		
User (27)		
Group(4)		
Short Code (54)		
Directory(2)		
Time Profile(0)		
Account Code(0)		
User Rights(0)		
Location (4)		
IPOSE-Primary		
IPOSE-Secondary		
IP500 Expansion		
System (1)		
Line (6)		
Control Unit (3)		
Extension (16)		
User (17)		

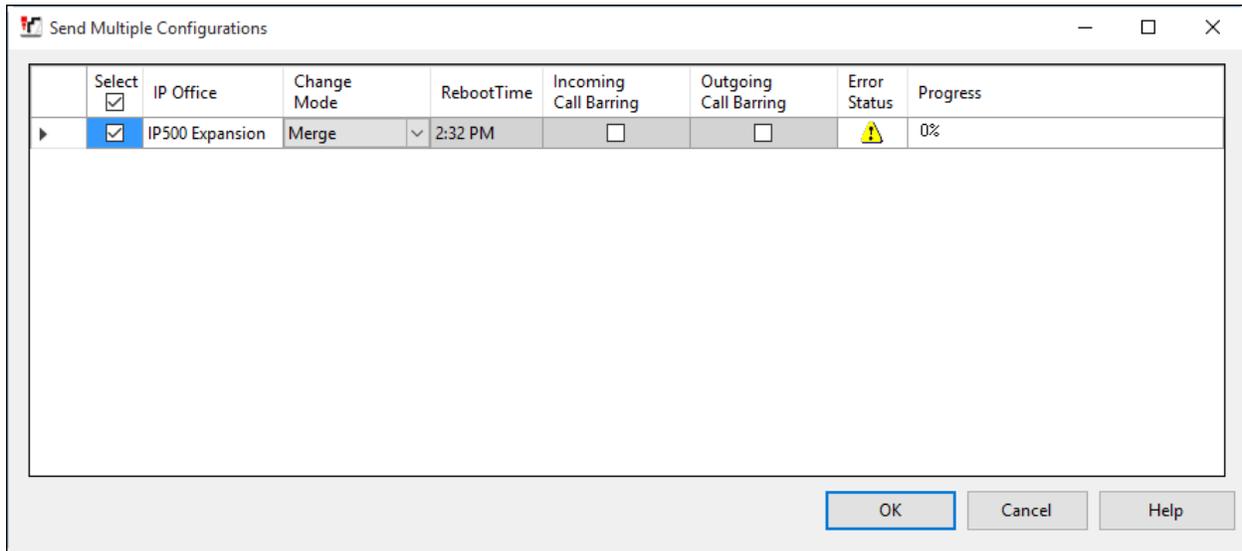
## 6.6. ARS

The following screen shows a sample ARS configuration for the route named “**To-Primary**” on the Expansion System. The **Line Group ID** is set to “**99999**” matching the number of the **Outgoing Group** configured on the IP Office Line 17 to the Primary server (**Section 6.4**). The **Alternate Route** is set to “**To-Secondary**” to route calls to the Secondary server, if present, in the event the Primary server is unavailable.

Code	Telephone Number	Feature	Line Group ID
xN	9N	Dial	99999
911	9911	Dial Emergency	99999

## 6.7. Save Configuration

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



## 7. 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 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. Verizon provided the following information for the compliance testing: the IP address and port used by the Verizon SIP SBC, and the DID numbers shown in **Figure 1**. This information was used to complete the Avaya IP Office configuration.

## 8. Verifications

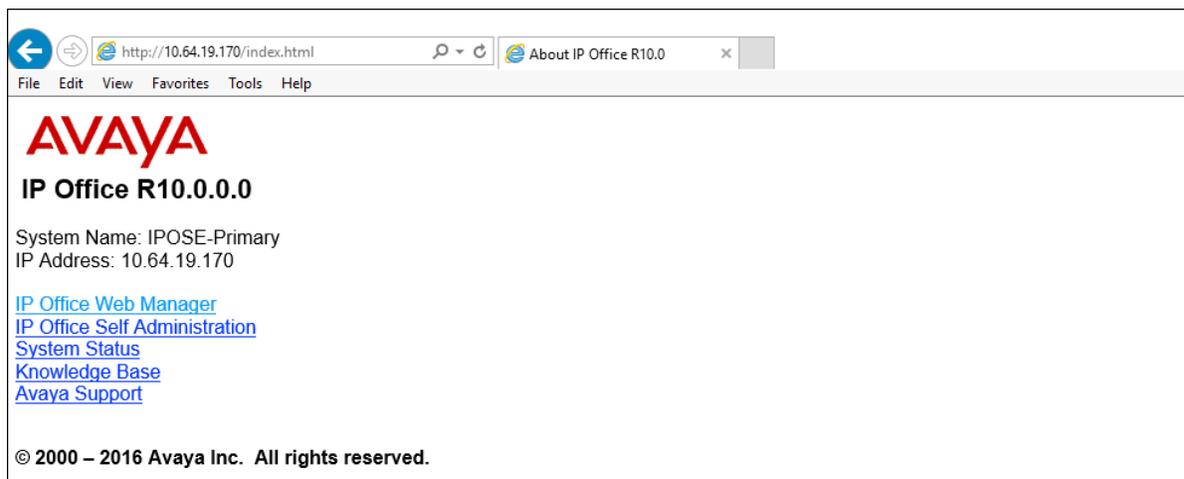
This section provides sample verifications of the Avaya configuration with Verizon Business IP Trunk service.

### 8.1. IP Office

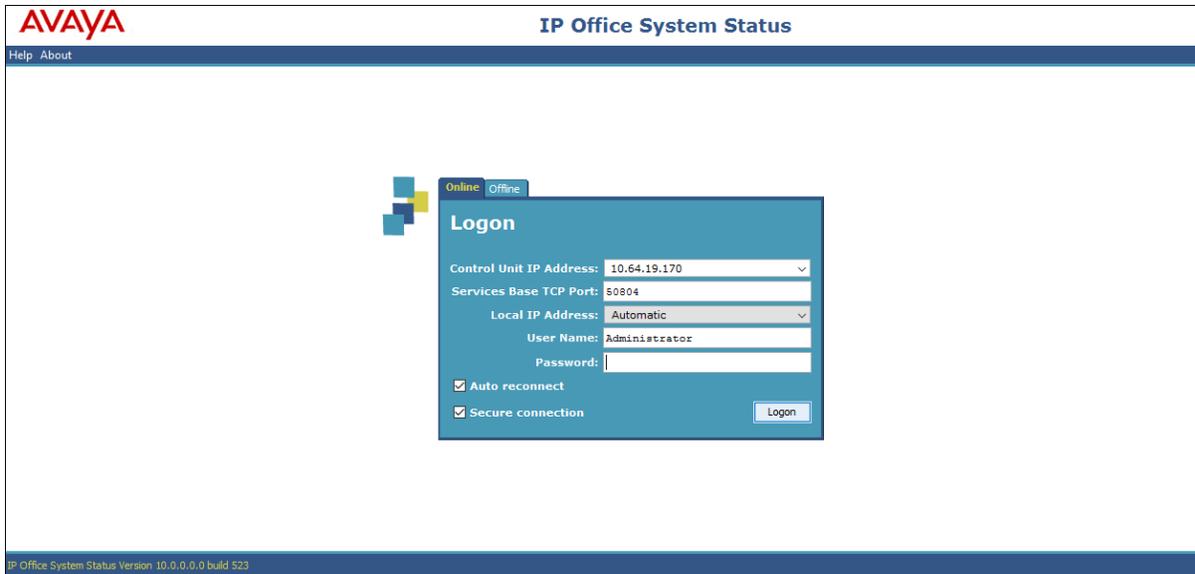
This section provides verification steps that may be performed with the IP Office.

#### 8.1.1. System Status

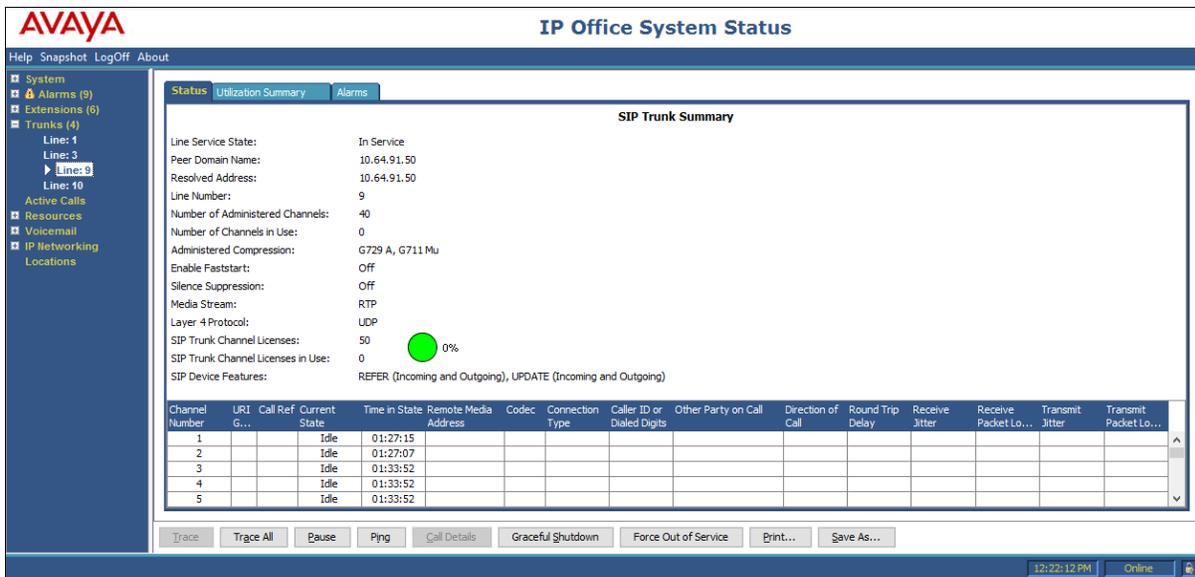
The System Status application is used to monitor and troubleshoot IP Office. Use the System Status application to verify the state of the SIP trunk. System Status can be accessed from **Start → Programs → IP Office → System Status**. Or by opening an Internet browser and type the URL: `http://ipaddress`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



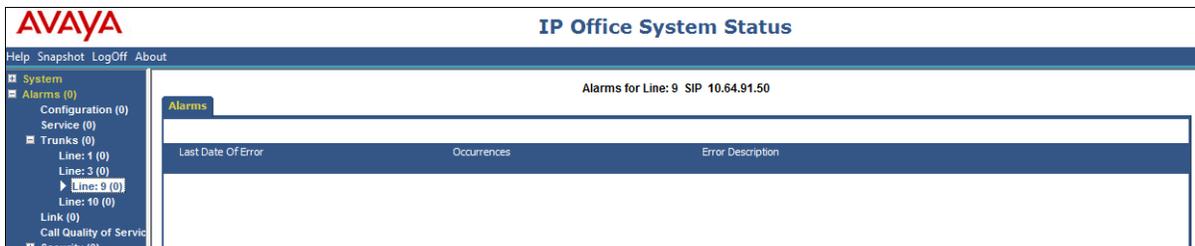
The following screen shows a sample **Logon** screen. Enter the Primary server **IP** address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.



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



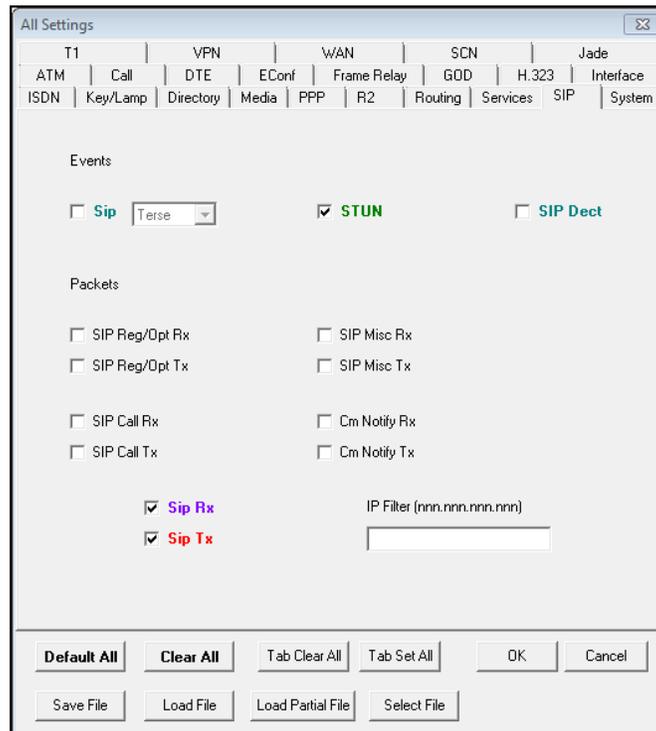
Select the **Alarms** tab and verify that no alarms are active on the SIP line.



### 8.1.2. Monitor

The Monitor application can also be used to monitor and troubleshoot IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor**. The application allows the monitored information to be customized. To customize, select **Filters → Trace Options** (not shown).

The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked. All SIP messages will appear in the trace with the color blue. To customize the color, right-click on **SIP Rx** or **SIP Tx** and select the desired color.



As an example, the following shows a portion of the monitoring window for an outbound call from extension 6242, whose DID is 732-945-0242, calling out to the PSTN via the Verizon Business IP Trunk service. The telephone user dialed 9-1-303-538-2177.

```
Avaya IP Office SysMonitor - [STOPPED] Monitoring 10.64.19.170 (IPOSE-Primary (Server Edition(P))); Log Settings - C:\Users\...\sysmonitorsettings.ini
File Edit View Filters Status Help
14:41:43 157338mS SIP Tx: UDP 1.1.1.2:5060 -> 172.30.209.21:5071
INVITE sip:13035382177@172.30.209.21 SIP/2.0
Via: SIP/2.0/UDP 1.1.1.2:5060;rport:branch=z9hG4bKf4cd939ff72cf43a01eb727ba7a348c
From: "Avaya 9508" <sip:7329450242@1.1.1.2>;tag=0b6d8efd60cee010
To: <sip:13035382177@172.30.209.21>
Call-ID: c40512798a6a961b9a9f99e191fd70ef
CSeq: 296176814 INVITE
Contact: "Avaya 9508" <sip:7329450242@1.1.1.2:5060;transport=udp>
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,REFER,NOTIFY
Supported: timer,100rel
Min-SE: 1800
Session-Expires: 1800;refresher=uac
User-Agent: IP Office 10.0.0.0 build 550
Content-Type: application/sdp
Content-Length: 239

v=0
o=UserA 203193726 980591525 IN IP4 1.1.1.2
s=Session SDP
c=IN IP4 1.1.1.2
t=0 0
m=audio 16388 RTP/AVP 18 0 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
14:41:43 157338mS Sip: 0a4013aa000003f4 21.1012.0 3 SIFTrunk Endpoint(f6e45678) UpdateSIPCallState SIPDialog::INITIAL(0) -> SIPDialog::INVITE_SENT(1)
14:41:43 157338mS CMCallEvt: 0000000000000000 0.1010.0 -1 TargetingEP: StateChange: END-X CMCSOffering->CMCSDelete
14:41:43 157338mS CMCallEvt: 0000000000000000 0.1010.0 -1 BaseEP: DELETE CMEndpoint f6e527d0 TOTAL NOW=3 CALL_LIST=1
14:41:43 157338mS CD: CALL: 1.3.1 BState=Idle Cut=3 Music=0.0 Aend="Line 1" (251.1) Bend="" [Line 21] (0.0) CalledNum=913035382177 () CallingNum=6242 (Avaya 9508) Int
14:41:43 157338mS CD2: CALL:S 1.3.1.0.1010.0,8,0,1,0,1,0,Line 1,Line 21,251.1,0.0,100.100,913035382177,100.101,6242,,,100,913035382177,100,,0,16,0,1,0,0,,,,,0,6,0,
14:41:43 157338mS H323Evt: v=0 stackNum=1 State: new=ICProceeding, old=Present id=3
14:41:43 157403mS SIP Rx: UDP 172.30.209.21:5071 -> 1.1.1.2:5060
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 1.1.1.2:5060;received=1.1.1.2;branch=z9hG4bKf4cd939ff72cf43a01eb727ba7a348c;rport=5060
From: "Avaya 9508" <sip:7329450242@1.1.1.2>;tag=0b6d8efd60cee010
To: <sip:13035382177@172.30.209.21>
Call-ID: c40512798a6a961b9a9f99e191fd70ef
CSeq: 296176814 INVITE
14:41:43 157403mS Sip: Find End Point2 0a4013aa000003f4 21.1012.0 3 SIFTrunk Endpoint (f6e437c8) Sip CallId c40512798a6a961b9a9f99e191fd70ef
14:41:43 157403mS Sip: 0a4013aa000003f4 21.1012.0 3 SIFTrunk Endpoint(f6e45678) Ignoring Q.850 Reason dialog f6e45678, method INVITE, CodeNum 100 in state SIPDialog::
14:41:43 157403mS Sip: 0a4013aa000003f4 21.1012.0 3 SIFTrunk Endpoint(f6e45678) Process SIP response dialog f6e45678, method INVITE, CodeNum 100 in state SIPDialog::I
14:41:43 157404mS CMLineRx: v=0
```

## 9. 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 Release 10 can be successfully combined with a Verizon Business IP Trunk service connection to create an end-to-end SIP Telephony business solution. By following the sample configurations provided in this document, customers using Avaya IP Office can connect to the PSTN via a Verizon Business IP 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.

## 10. Additional References

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

- [1] *IP Office™ Platform 10.0, Deploying Avaya IP Office™ Platform Servers as Virtual Machines*, Document Number 15-601011 Issue 03j, July 2016
- [2] *Administering Avaya IP Office™ Platform with Manager*, February 2016
- [3] *IP Office™ Platform 10.0, Installing and Maintaining the Avaya IP Office™ Platform Application Server*, Document Number 15-601011 Issue 10d, July 2016
- [4] *IP Office™ Platform 10.0, Deploying Avaya IP Office™ Platform IP500 V2*, Document Number 15-601042, July 2016
- [5] *IP Office™ Platform 10.0, Using Avaya IP Office™ System Status*, Document Number 15-601758, July 2016
- [6] *Administering Avaya Communicator on IP Office*, February 2016
- [7] *IP Office™ Platform 10.0, 1608/1616 Phone User Guide*, Document Number 15-601040 Issue 10b, June 2016
- [8] RFC 3261 *SIP: Session Initiation Protocol* <http://www.ietf.org/rfc/rfc3261.txt>

Additional IP Office documentation can be found at:

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

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 10.0 with Avaya Session Border Controller for Enterprise Release 7.1 with Verizon Business IP Trunking.

[VZIPT-IPO10SBC] Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service and Avaya IP Office Release 10.0 with Avaya Session Border Controller for Enterprise Release 7.1 – Issue 1.0

## 11. Appendix A - Unscreened ANI Testing and Configuration

Unscreened ANI is a Verizon offered service (available with VoIP IP Integrated Access and VoIP IP Trunking) and is a feature being offered with IP Office Release 10.0. This service was tested successfully in this test configuration and can be implemented by following the steps here.

This optional feature allows customers to send an “unscreened” ANI to Verizon’s network which is then displayed to the called party as Caller ID. An “unscreened” ANI can be any telephone number that the customer passes through Verizon’s network for Caller ID display purposes only. If this feature is enabled on the Verizon Business IP Trunk services, Verizon will designate one of the assigned telephone numbers as a “Screened Telephone Number” for each unique location. Verizon will use this Screened Telephone Number to determine call origination for billing, call routing, and E911. The customer is responsible for configuring its IP-PBX, PBX or other devices to accommodate and properly process the Screened Telephone Number.

Select the SIP Line, SIP URI tab as shown in **Section 5.4.5**.

The Screened Telephone Number (STN) provided by Verizon for this test is 732-945-0821. Typically, customers would have one or more STN; one for every location. A central Primary server could be used to pass multiple STNs to Verizon based on the Outgoing Group selected. The STN would then be entered in the **Diversion Header** field as shown below.

The screenshot displays the 'SIP Line' configuration window with the 'SIP URI' tab selected. A table lists four SIP URIs, with the fourth one (URI 4) highlighted in blue. Below the table, the 'Edit URI' dialog box is open, showing the configuration for the selected URI. The 'Diversion Header' field is set to '7329450821'.

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header	Credential	Max Calls
1	17 17	<Internal>	<Internal>	<Internal>	None	PAI		Diversion	None	0: <Non...	10
2	17 117	7329450240	7329450240	7329450240	None	PAI		Diversion	None	0: <Non...	10
3	17 117	7329450238	7329450238	7329450238	None	PAI		Diversion	None	0: <Non...	10
4	17 417	<Internal>	<Internal>	<Internal>	None	PAI		None	7329450821	0: <Non...	10

**Edit URI**

Local URI: Use Internal Data

Contact: Use Internal Data

Display Name: Use Internal Data

Identity: None

Header: P Asserted ID

Forwarding And Twinning

Originator Number: [Empty]

Send Caller ID: None

Diversion Header: 7329450821

Registration: 0: <None>

Incoming Group: 17

Outgoing Group: 417

Max Sessions: 10

The following screen shows a sample ARS configuration for the route named “**Unscreened ANI**”. The **Line Group ID** is set to “**417**” matching the number of the **Outgoing Group** configured for Unscreened ANI.

ARS

ARS Route ID: 52

Route Name: Unscreened ANI

Dial Delay Time: System Default (4)

Description:

In Service:  Out of Service Route: <None>

Time Profile: <None> Out of Hours Route: <None>

Code	Telephone Number	Feature	Line Group ID
xxxxxxxx	.	Dial	417
1xxxxxxxx	.	Dial	417
911	911	Dial Emergency	417

Alternate Route Priority Level: 1

Alternate Route Wait Time: 5

Alternate Route: <None>

A Short Code is created to route outbound calls using the Unscreened ANI ARS. A Short Code can be created for an entire system, a particular user, a User Rights profile to be applied to a group of users, or line.

A sample system Short Code to route calls out the ARS pattern designated for Unscreened ANI is shown below.

Short Code

Code: 8N

Feature: Dial

Telephone Number: N

Line Group ID: 52: Unscreened ANI

Locale:

Force Account Code:

Force Authorization Code:

To limit the use of this feature to a particular user, a user Short Code can be created to have a user dial 9 to route calls out the Unscreened ANI ARS as shown below. A User Rights profile is created in a similar manner (not shown).

Code	Telephone Number	Feature	Line Group ID
*DCP	4000000,0,1,1,0	Dial	0
*FWD0	913035551234=913035551234	Forward Number	0
9N	N	Dial	52

Edit Short Code	
Code	9N
Feature	Dial
Telephone Number	N
Line Group ID	52: Unscreened ANI
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

An entire Small Community Network location can be specified to use the Unscreened ANI feature by creating a Line Short Code as shown below. In this example, any outbound call destined to the PSTN from this location will use the Unscreened ANI ARS.

The screenshot displays a software interface for configuring a line. On the left, a table lists several lines:

Line Number	Line Type	Line Sub Type
1	IP Office Line	WebSocket Sen
3	IP Office Line	WebSocket Sen
21	SIP Line	
22	SIP Line	

The main window is titled "IP Office Line - Line 1" and contains a "Short Codes" tab. It features a table with the following data:

Code	Telephone Number	Feature	Line Group ID
9N	N	Dial	52

Below the table is an "Edit Short Code" form with the following fields:

- Code: 9N
- Feature: Dial
- Telephone Number: N
- Line Group ID: 52: Unscreened ANI
- Locale: (empty)
- Force Account Code:

Buttons for "Add...", "Remove", "Edit...", "OK", and "Cancel" are visible on the right side of the configuration window.

## Verification

In the following Monitor trace, it is observed that the From header contains the DID number, 732-945-0242 and a Diversion header has been added with the screened ANI (732-945-0821).

From: "Avaya 9508" <sip:7329450242@1.1.1.2>;tag=237e8d3f3fb1977f

Diversion: <sip:7329450821@1.1.1.2:5060>;reason=direct;screen=no;privacy=off;counter=1

```
Avaya IP Office SysMonitor - [STOPPED] Monitoring 10.64.19.170; Log Settings - C:\Users\...\sysmonitorsettings.ini
File Edit View Filters Status Help
13:40:10 340323069mS Sip: 0a4013aa00000484 21.1156.0 27 SIPTrunk Endpoint(f6f5b8d8) SetLocalRTPAddress to 1.1.1.2:16432
13:40:10 340323069mS Sip: 0a4013aa00000484 21.1156.0 27 SIPTrunk Endpoint(f6f5b8d8) SdpClone
13:40:10 340323070mS Sip: 0a4013aa00000484 21.1156.0 27 SIPTrunk Endpoint(f6f5b8d8) UpdateSDPState SIPDialog::IDLE(0) -> SIPDialog::OFFER_SENT(1)
13:40:10 340323070mS Sip: 0a4013aa00000484 21.1156.0 27 SIPTrunk Endpoint(f6f5b8d8) INVITE SENT TO 172.30.209.21:5071 (reg required 0 registered 0)
13:40:10 340323070mS SIP Tx: UDP 1.1.1.2:5060 -> 172.30.209.21:5071
INVITE sip:130353821778172.30.209.21 SIP/2.0
Via: SIP/2.0/UDP 1.1.1.2:5060;rport=branch=z9hG4bK0592c0e2f8e47f565b9921f5bbd0a03a
From: "Avaya 9508" <sip:7329450242@1.1.1.2>;tag=237e8d3f3fb1977f
To: <sip:130353821778172.30.209.21>
Call-ID: fb748970f286e337acfa5baacd78483d
CSeq: 2044453694 INVITE
Contact: "Avaya 9508" <sip:7329450242@1.1.1.2:5060;transport=udp>
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,REFER,NOTIFY
Supported: timer,100rel
Min-SE: 1800
Session-Expires: 1800;refresher=uac
User-Agent: IP Office 10.0.0.0 build 550
Diversion: <sip:7329450821@1.1.1.2:5060>;reason=direct;screen=no;privacy=off;counter=1
Content-Type: application/sdp
Content-Length: 239

v=0
o=UserA 492554925 470785411 IN IP4 1.1.1.2
s=Session SDP
c=IN IP4 1.1.1.2
t=0 0
m=audio 16432 RTP/AVP 18 0 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
13:40:10 340323070mS Sip: 0a4013aa00000484 21.1156.0 27 SIPTrunk Endpoint(f6f5b8d8) UpdateSIPCallState SIPDialog::INITIAL(0) -> SIPDialog::INVITE_SENT(1)
13:40:10 340323070mS CMCalleEvt: 0000000000000000 0.1154.0 -1 TargetingEP: StateChange: END-X CMCSOffering->CMCSDelete
13:40:10 340323070mS CMCalleEvt: 0000000000000000 0.1154.0 -1 BaseEP: DELETE CMEndpoint f6f63090 TOTAL NOW=3 CALL_LIST=1
13:40:10 340323070mS CD: CALL: 1.3.1 SState=Idle Cut=3 Music=0.0 Aend="Line 1" (251.1) Bend="" [Line 21] (0.0) CalledNum=913035382177 ( ) CallingNum=6242 (Avaya 9508) Int
13:40:10 340323070mS CD2: CALL:S 1.3.1.0.1154.0,8,0,1,0,1,0,Line 1,,Line 21,251.1,0.0,100,913035382177,100.101,6242,,100,913035382177,100,,0,16,0,1,0.0,,,,,0,5,0,
13:40:10 340323137mS SIP Rx: UDP 172.30.209.21:5071 -> 1.1.1.2:5060
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 1.1.1.2:5060;received=1.1.1.2;branch=z9hG4bK0592c0e2f8e47f565b9921f5bbd0a03a;rport=5060
From: "Avaya 9508" <sip:7329450242@1.1.1.2>;tag=237e8d3f3fb1977f
To: <sip:130353821778172.30.209.21>
Call-ID: fb748970f286e337acfa5baacd78483d
CSeq: 2044453694 INVITE
```

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