



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

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# **Application Notes for Configuring SIP Trunking Using Verizon Business IP Contact Center VoIP Inbound with Avaya IP Office Release 11.1 – Issue 1.0**

## **Abstract**

These Application Notes describe a reference configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Contact Center VoIP Inbound SIP Trunk Service and an Avaya IP Office solution. In the reference configuration, the Avaya IP Office solution consists of Avaya IP Office Server Edition Release 11.1, Voicemail Pro 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 11.1.

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

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 Verizon Business IP Contact Center service.

# 1. Introduction

These Application Notes describe a reference configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Contact Center VoIP Inbound Service (Verizon Business IPCC) and an Avaya IP Office solution. In the reference configuration, the Avaya IP Office solution consists of an Avaya IP Office Server Edition Primary Server, an IP500 V2 Expansion System, Voicemail Pro 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 11.1.

Customers using Avaya IP Office with the Verizon Business IPCC service are able to receive inbound toll-free calls from the PSTN via the SIP protocol. The converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI. This service provides toll free inbound calling via standards-based SIP trunks as well as re-routing of inbound toll-free calls to alternate destinations based upon SIP messages (i.e., REFER) generated by the Avaya IP Office.

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

For more information on the Verizon Business IPCC service, visit <https://enterprise.verizon.com/products/customer-experience-services/transport-and-intelligent-routing/ip-contact-center/>

## 2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the Verizon Business IPCC service, as depicted in **Figure 1**. The Avaya IP Office were configured to use the commercially available SIP Trunking solution provided by the Verizon Business IPCC service. This allowed Avaya IP Office to receive inbound toll-free calls from the PSTN via the SIP protocol.

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and Verizon Business IPCC Services did not include use of any specific encryption features as requested by Verizon.

Encryption (TLS/SRTP) was used internal to the enterprise between Avaya products wherever possible.

## 2.1. Interoperability Compliance Testing

The testing included executing the test cases detailed in **Section 100**, reference [VZ-Test-Plan], which contains the Verizon Business IPCC Interoperability Lab Test Plan. To summarize, the testing included the following successful SIP trunk interoperability compliance testing:

- SIP OPTIONS monitoring of the health of the SIP trunk was verified. Verizon Business and IP Office can both monitor health using SIP OPTIONS.
- Proper recovery from induced failure conditions such as IP Office reboots, and IP network outages between Verizon and IP Office, of short and long durations.
- Incoming calls from the PSTN were routed to the toll-free numbers assigned by Verizon Business to the Avaya IP Office location. These incoming calls arrived via the SIP Line configured in **Section 5.4** and were answered by Avaya H.323 telephones, Avaya SIP telephones, Avaya digital telephones, analog telephones, and Avaya IP Office Voicemail Pro.
- Proper disconnect when either party hangs up an active call.
- Proper disconnect when the PSTN caller abandons (i.e., hangs up) a toll-free call before the IP Office party has answered.
- Proper SIP 486 response and busy tone heard by the caller when a PSTN user calls a toll-free number directed to a busy IP Office user, an IP Office user with Do-not-disturb active, or an IP Office user that is logged out (i.e., assuming no redirection is configured for these conditions). Similarly, busy tone is heard when a PSTN user calls a toll-free number directed to a hunt group whose queue is “full” (i.e. if no redirection is configured for hunt group busy conditions, see **Section 5.6.2**).
- Proper termination of an inbound IP Toll Free call left in a ringing state for a relatively long duration.
- The display of caller ID on display-equipped Avaya IP Office telephones was verified. The IP Office capability to use the caller ID received from Verizon to look up and display a name from a configurable directory was also exercised successfully.
- Privacy requests for inbound toll-free calls from the PSTN were verified. That is, when privacy is requested by a PSTN caller (e.g., dialing \*67 from a mobile phone), the inbound toll-free call can be successfully completed to an IP Office telephone user while presenting a “WITHHELD” or anonymous display to an IP Office user (i.e., rather than the caller’s telephone number).
- Inbound toll-free long holding time call stability.
- Inbound fax calls using T.38 and G.711.

- Telephony features such as hold and resume, transfer of toll-free calls to other IP Office users, and conference of toll-free calls.
- Incoming voice calls using the G.729(a) and G.711 ULAW codecs, and proper protocol procedures related to media.
- DTMF transmission using RFC 2833. Successful IP Office Voicemail Pro menu navigation for incoming toll-free calls.
- Incoming toll-free calls directed to the Hunt Groups configured in **Section 5.6.2** were verified. Incoming calls could be queued, queued with priority, and be answered by members of the hunt group as members become available.
- Incoming toll-free calls that were redirected by a SIP REFER message, generated by Avaya IP Office/Voicemail Pro, back to the Verizon Business IPCC service for redirection to an alternate destination.
- Outgoing calls from the Avaya IP Office location to the PSTN were routed via a SIP Line to the Verizon Business IP Trunk service described in reference [VzIPT-IPO11\_1]. As detailed in reference [VzIPT-IPO11\_1], these outgoing PSTN calls can be originated from Avaya H.323 telephones, Avaya SIP telephones, Avaya digital telephones, and analog endpoints. The display of caller ID on display-equipped PSTN telephones was verified. In the context of inbound toll-free calls using Verizon Business IPCC, inbound toll-free calls arriving via the SIP Line configured in **Section 5.4** could be forwarded or twinned out the Verizon Business IP Trunk service SIP Line. Inbound toll-free calls from the Verizon Business IPCC SIP Line could also trigger mobile callback calls that use the Verizon Business IP Trunk service SIP Line.
- Call Forwarding of Verizon toll-free calls to PSTN destinations via the Verizon Business IP Trunk service documented in reference [VzIPT-IPO11\_1], presenting true calling party information to the mobile phone.
- Mobile twinning of Verizon toll-free calls to a mobile phone via the Verizon Business IP Trunk service documented in reference [VzIPT-IPO11\_1], presenting true calling party information to the mobile phone.
- Proper DiffServ markings for Avaya IP Office SIP signaling and RTP media.

## 2.2. Test Results

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

- During the compliance test, some inbound toll-free calls are transferred back to the Verizon Business IPCC service to be routed to alternate destinations. Since the Verizon Business IPCC in the reference configuration is an inbound only service, the call transfer is achieved by IP Office sending a SIP REFER message, with a Refer-To without a Replaces header, back to Verizon. This functionality is limited to Voicemail Pro modules and certain SIP endpoints using blind transfer. Other endpoints like J100 series IP telephones, H.323, digital, or analog endpoints do not support this type of transfer.
- On inbound toll-free calls that are transferred back out to the Verizon Business IPCC service using SIP REFER, Verizon sends a NOTIFY message after the “referred” call has been released from the IP Office. Since IP Office cannot associate this NOTIFY with any active call, it sends a "481 Call/Transaction Does Not Exist" to Verizon in response. No user perceived problem occurs because of this anomaly.
- When transferring an inbound Verizon Business IPCC toll-free call back out the Verizon Business IPCC service using SIP REFER to a third party that is busy, Verizon disconnects the call and no error message or busy tone is heard by the originating caller.
- The Verizon Business IPCC Services suite does not support the SIP 302 Redirect method.

## 2.3. Support

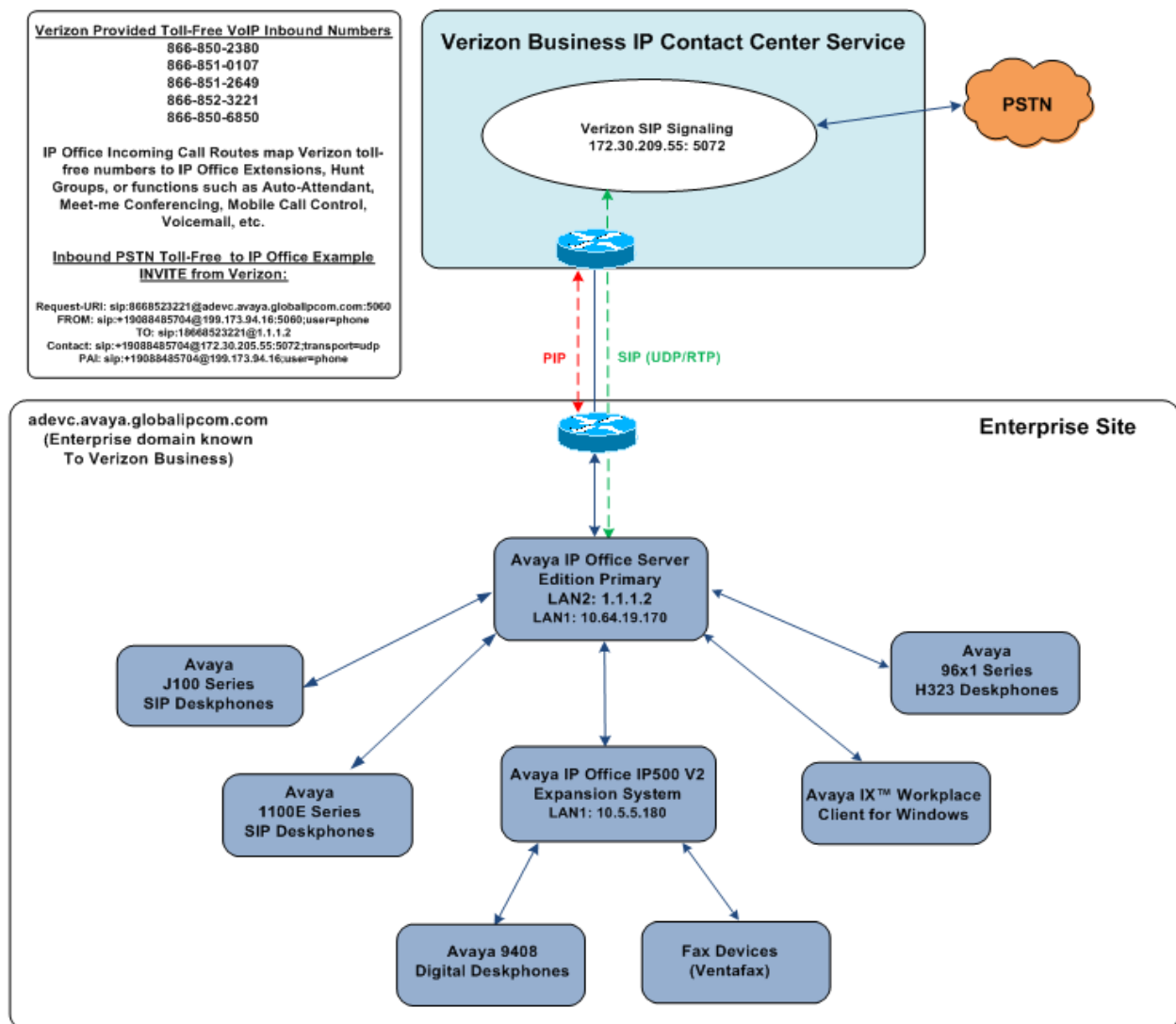
For technical support on Verizon Business IPCC Services offer, visit online support at <http://www.verizonenterprise.com/support/>

Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

### 3. Reference Configuration

**Figure 1** illustrates an example Avaya IP Office solution connected to the Verizon Business IP Contact Center SIP Trunk service. The Avaya equipment is located on a private IP subnet. An enterprise edge router provides access to the Verizon Business network via a Verizon Business T1 circuit. This circuit is provisioned for the Verizon Business Private IP (PIP) service.

In the reference configuration, the Avaya IP Office receives traffic from the Verizon Business IPCC service on port 5060 and sends traffic to port 5072, using UDP for network transport, as required by the Verizon Business IPCC service. Verizon provided five toll-free numbers associated with the IP Contact Center service. These toll-free numbers were mapped to IP Office destinations via Incoming Call Routes as shown in **Section 5.8**. The Avaya CPE environment domain known to Verizon was *adevc.avaya.globalipcom.com*.



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

The Verizon Business IP Trunk service used for outgoing calls, as described in **Section 2.1**, required different SIP line configuration parameters than what were needed for the Verizon Business IPCC service. A new SIP line was created in IP Office to support the Verizon Business IPCC service. This SIP line is separate from the SIP line previously created for Verizon Business IP Trunk service as described in reference [VzIPT-IPO11\_1]. Having separate SIP lines for each Verizon service will allow for unique parameters to be set on IP Office to accommodate the differences between the two services.

## 4. Equipment and Software Validated

**Table 1** shows the equipment and software used in the reference configuration.

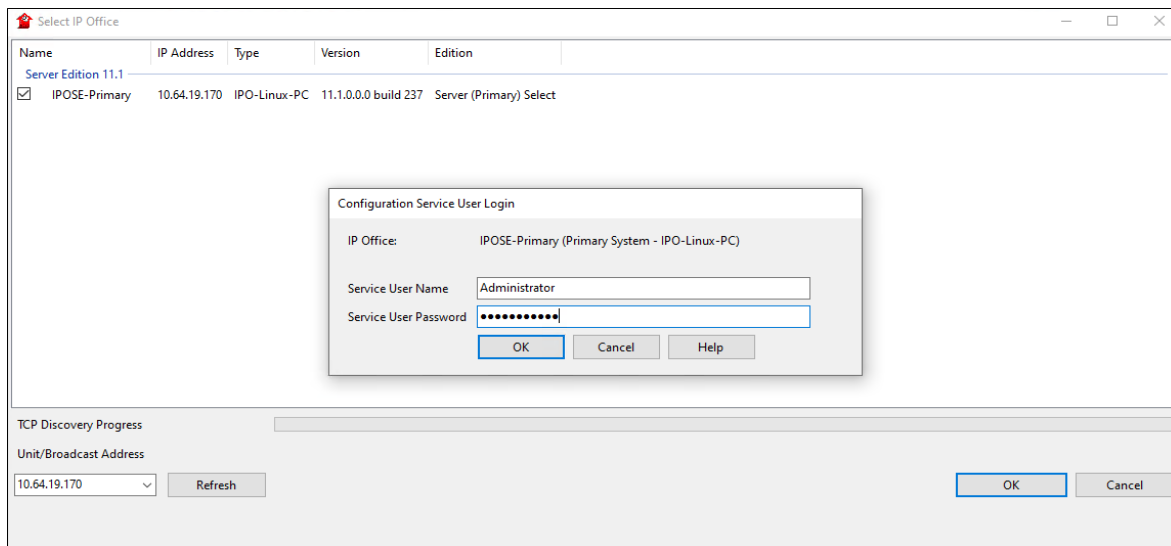
Avaya IP Telephony Solution Components	
Equipment	Software
Avaya IP Office Server Edition	Release 11.1.0.0.0 Build 237
- Avaya IP Office Voicemail Pro	Release 11.1.0.0.0 Build 234
Avaya IP Office 500 V2 Expansion System	Release 11.1.0.0.0 Build 237
Avaya IP Office Manager	Release 11.1.0.0.0 Build 237
Avaya 96x1 Series IP Deskphone (H.323)	Release 6.8304
Avaya 1140E IP Deskphone (SIP)	Release 04.04.23.00
Avaya J169 IP Deskphone (SIP)	Release 4.0.4.0.10
Avaya IX Workplace Client for Windows	Release 3.8.4.10.2
Avaya 9508 Digital Deskphone	Release 0.60

**Table 1: Equipment and Software Tested**

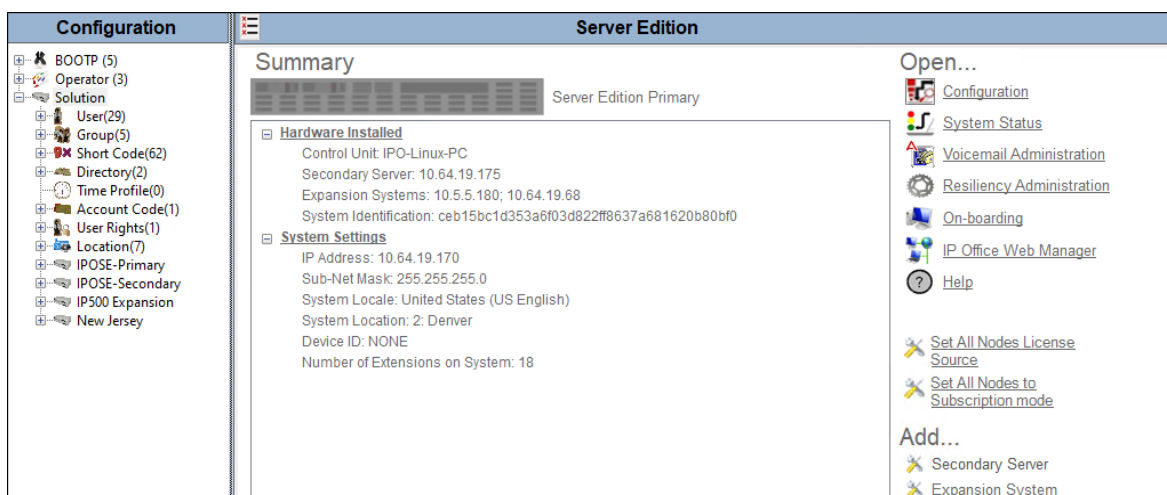
Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2, and also when deployed with all configurations of IP Office Server Edition. Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

## 5. Avaya IP Office Primary Server Configuration

IP Office is configured via the IP Office Manager program. For more information on IP Office Manager, consult reference **Error! Reference source not found.** From the IP Office Manager PC, select **Start → All Apps → 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. All the Avaya IP Office configurable components are shown in the left pane, known as the Navigation Pane. Clicking the “plus” sign next to the Primary server system name, e.g., **IPOSE-Primary**, on the navigation pane will expand the menu on this server. In the sample configuration, IP users registered to the Primary server. Digital and Analog users are configured on the Expansion System. The SIP trunk to Verizon Business is configured on the Primary server.



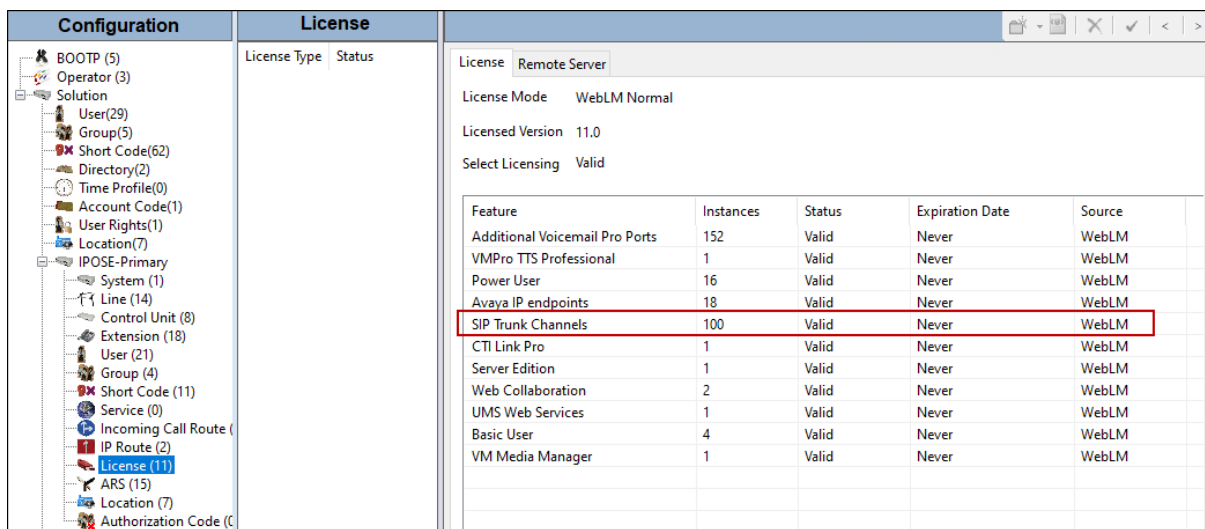


## 5.1. Licensing

In the reference 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.

Navigate to **License** in the Navigation Pane. In the Details Pane verify that the **License Status** for **SIP Trunk Channels** is **Valid** and that the number of **Instances** is sufficient to support the number of channels provisioned for the SIP trunk.



The screenshot displays the Avaya IP Office configuration interface. The left pane shows the navigation tree with 'License (11)' selected under 'IPOSE-Primary'. The right pane shows the 'License' details for 'Remote Server'.

License Mode: WebLM Normal  
Licensed Version: 11.0  
Select Licensing: Valid

Feature	Instances	Status	Expiration Date	Source
Additional Voicemail Pro Ports	152	Valid	Never	WebLM
VMPro TTS Professional	1	Valid	Never	WebLM
Power User	16	Valid	Never	WebLM
Avaya IP endpoints	18	Valid	Never	WebLM
SIP Trunk Channels	100	Valid	Never	WebLM
CTI Link Pro	1	Valid	Never	WebLM
Server Edition	1	Valid	Never	WebLM
Web Collaboration	2	Valid	Never	WebLM
UMS Web Services	1	Valid	Never	WebLM
Basic User	4	Valid	Never	WebLM
VM Media Manager	1	Valid	Never	WebLM

## 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. LAN1 Settings

In the reference 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.

The screenshot displays the configuration interface for the IPOSE-Primary system. On the left, a 'Configuration' tree shows the hierarchy: BOOTP (3), Operator (3), Solution, User(49), Group(7), Short Code(60), Directory(2), Time Profile(0), Account Code(1), User Rights(1), Location(7), IPOSE-Primary, System (1), Line (14), and Control Unit (9). The 'System' tab is selected, showing the 'Name' as 'IPOSE-Primary'. The main panel, titled 'IPOSE-Primary', contains several tabs: System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, and SMTP. The 'LAN1' tab is active, and within it, the 'LAN Settings' sub-tab is selected. The 'Network Topology' sub-tab is also visible. The 'IP Address' field is set to '10 . 64 . 19 . 170' and the 'IP Mask' field is set to '255 . 255 . 255 . 0'. The 'Number Of DHCP IP Addresses' is set to '84'. The 'DHCP Mode' is set to 'Disabled' (radio button selected). An 'Advanced' button is located at the bottom right of the LAN Settings section.

Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** parameter is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 96x1 deskphones used in the reference configuration. The **H.323 Signaling over TLS** should be set based on customer needs. In the reference configuration it was set to **Preferred**. The **SIP Trunks Enable** parameter must be checked to enable the configuration of SIP trunks to Verizon Business. The **SIP Registrar Enable** parameter is checked to allow Avaya J169, Avaya 1140E, and Avaya IX Workplace usage.

The **SIP Domain Name** and **SIP Registrar FQDN** may be set according to customer requirements. Set the **Layer 4 Protocol** section based on customer needs. In the reference configuration **TCP/5055** and **TLS/5056** were configured.

System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events SMTP SMDR VoIP Contact Center Avaya Cloud Services

LAN Settings VoIP Network Topology

☒ H.323 Gatekeeper Enable  
☐ Auto-create Extension ☐ Auto-create User ☐ H.323 Remote Extension Enable  
H.323 Signaling over TLS Preferred Remote Call Signaling Port 1720

☒ SIP Trunks Enable  
☒ SIP Registrar Enable  
☐ Auto-create Extension/User ☒ SIP Remote Extension Enable Allowed SIP User Agents Block blacklist only  
SIP Domain Name sipipose.customer.com  
SIP Registrar FQDN sipipose.customer.com

Layer 4 Protocol  
☐ UDP UDP Port 5060 Remote UDP Port 5060  
☒ TCP TCP Port 5055 Remote TCP Port 5055  
☒ TLS TLS Port 5056 Remote TLS Port 5056

Challenge Expiration Time (sec) 10

RTP  
Port Number Range  
Minimum 40750 Maximum 50750  
Port Number Range (NAT)  
Minimum 40750 Maximum 50750

### 5.2.2. LAN2 Settings

In the sample configuration, LAN2 is used to connect the IP Office to the Verizon Business 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.

The screenshot shows the IPOSE-Primary configuration window. The top navigation bar includes tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VoIP, Contact Center, and Avaya Cloud. The LAN2 tab is selected. Below it, the LAN Settings sub-tab is active. The IP Address field is set to 1.1.1.2 and the IP Mask is 255.255.255.0. The Number Of DHCP IP Addresses is set to 200. The DHCP Mode is set to Disabled, with radio buttons for Server, Client, and Disabled. An Advanced button is visible at the bottom right.

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 screenshot shows the IPOSE-Primary configuration window with the LAN2 tab and VoIP sub-tab selected. The H.323 Gatekeeper Enable checkbox is unchecked. The SIP Trunks Enable checkbox is checked. The SIP Registrar Enable checkbox is unchecked. The SIP Domain Name and SIP Registrar FQDN fields are empty. The Layer 4 Protocol section shows UDP, TCP, and TLS protocols with their respective ports. The RTP Port Number Range is set to Minimum 16384 and Maximum 32766. The Port Number Range (NAT) is set to Minimum 40750 and Maximum 50750.

Scrolling down the page, on the **Keepalives** section, set the **Scope** to **RTP-RTCP**. Set the **Periodic timeout** to **30** and the **Initial keepalives** parameter to **Enabled**. These settings will cause the Primary server to send RTP and RTCP keepalive packets starting at the time of initial connection and every 30 seconds thereafter if no other RTP or RTCP traffic is present. This facilitates the flow of media in cases where each end of the connection is waiting to see media from the other, as well as helping to keep ports open for the duration of the call.

In the **DiffServ Settings** section, the Primary server can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services (QoS) policies for both signaling and media. The **DSCP** field is the value used for media, while the **SIG DSCP** is the value used for signaling. These settings should be set according to the customer's QoS policies in place. In the sample configuration shown below, IP Office will mark the RTP media with a value associated with "Expedited Forwarding" using DSCP decimal 46 (**DSCP** parameter). IP Office will mark SIP signaling with a value associated with "Assured Forwarding" using DSCP decimal 28 (**SIG DSCP** parameter).

The screenshot displays the 'IPOSE-Primary' configuration window. The 'LAN2' tab is selected, and the 'VoIP' sub-tab is active. Under 'LAN Settings', the 'Enable RTCP Monitoring on Port 5005' checkbox is checked, and the 'RTCP collector IP address for phones' is set to 0.0.0.0. In the 'Keepalives' section, the 'Scope' is set to 'RTP-RTCP', the 'Periodic timeout' is 30, and 'Initial keepalives' is 'Enabled'. The 'DiffServ Settings' section shows 'DSCP (Hex)' as B8, 'Video DSCP (Hex)' as FC, 'DSCP Mask (Hex)' as 70, and 'SIG DSCP (Hex)' as 46. Below this, the decimal values are shown: DSCP is 46, Video DSCP is 63, DSCP Mask is 28, and SIG DSCP is 28. The 'DHCP Settings' section shows 'Primary Site Specific Option Number (SSON)' as 176, 'Secondary Site Specific Option Number (SSON)' as 242, 'VLAN' as 'Not Present', '1100 Voice VLAN Site Specific Option Number (SSON)' as 232, and '1100 Voice VLAN IDs' as an empty field.

Select the **Network Topology** tab as shown in the following screen. The **Firewall/NAT Type** was 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. **Binding Refresh Time (sec)** was set to **60** seconds. This is used to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages, to periodically check the status of the SIP trunk connection to Verizon Business. The **Public IP Address** and **Public Port** sections are not used for the Verizon Business SIP trunk service connection.

The screenshot shows the 'IPOSE-Primary' configuration window with the 'Network Topology' tab selected. The 'Network Topology Discovery' section contains the following fields and controls:

- STUN Server Address:** An empty text input field.
- STUN Port:** A dropdown menu set to '3478'.
- Firewall/NAT Type:** A dropdown menu set to 'Open Internet'.
- Binding Refresh Time (sec):** A numeric input field set to '60'.
- Public IP Address:** A numeric input field showing '0 . 0 . 0 . 0'.
- Public Port:** A section with three sub-fields: 'UDP' (0), 'TCP' (0), and 'TLS' (0).
- Run STUN on startup:** An unchecked checkbox.
- Buttons:** 'Run STUN' and 'Cancel' buttons are located to the right of the Public IP Address field.

### 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 reference configuration and are not intended to be prescriptive. The **Voicemail Type** in the reference configuration is **Voicemail Lite/Pro**. The **Voicemail IP Address** in the reference configuration is **10.64.19.170**, the IP address of the Primary server running the Voicemail Pro software.

The screenshot shows the 'IPOSE-Primary' configuration window with the 'Voicemail' tab selected. The configuration fields are as follows:

- Voicemail Type:** A dropdown menu set to 'Voicemail Lite/Pro'.
- Voicemail Destination:** A dropdown menu.
- Voicemail IP Address:** A numeric input field set to '10 . 64 . 19 . 170'.
- Backup Voicemail IP Address:** A numeric input field set to '10 . 64 . 19 . 175'.
- Voicemail Channel Reservation:** A section containing:
  - Unreserved Channels:** A numeric input field set to '152'.
  - Auto-Attendant:** A numeric input field set to '0'.
  - Announcements:** A numeric input field set to '0'.
  - Voice Recording:** A numeric input field set to '0'.
  - Mandatory Voice Recording:** A numeric input field set to '0'.
  - Mailbox Access:** A numeric input field set to '0'.
- Checkboxes:** 'Messages Button Goes To Visual Voice' and 'Outcalling Control' are both checked.

## 5.2.4. System Telephony Settings

To view or change telephony settings, select the **Telephony** tab and **Telephony** sub-tab as shown in the following screen. The settings presented here simply illustrate the reference configuration and are not intended to be prescriptive. In the reference configuration, the **Inhibit Off-Switch Forward/Transfer** box is unchecked so that call forwarding and call transfer to PSTN destinations via the Verizon Business IP Trunk service can be tested. That is, a call can arrive to IP Office via the Verizon IP Contact Center service and be forwarded or transferred back to the PSTN with the outbound leg of the call using the Verizon IP Trunk service. The **Companding 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.

The screenshot displays the 'IPOSE-Primary' configuration window, specifically the 'Telephony' tab. The window is divided into several sections for configuring telephony parameters.

**Telephony Settings:**

- Dial Delay Time (sec): 4
- Dial Delay Count: 0
- Default No Answer Time (sec): 15
- Hold Timeout (sec): 0
- Park Timeout (sec): 0
- Ring Delay (sec): 5
- Call Priority Promotion Time (sec): Disabled
- Default Currency: USD
- Default Name Priority: Favor Trunk
- Media Connection Preservation: Enabled
- Phone Failback: Automatic

**Login Code Complexity:**

- ☒ Enforcement
- Minimum length: 6
- ☒ Complexity

**RTP Collector Configuration:**

- ☐ Send RTP to an RTP Collector
- Server Address: 0 . 0 . 0 . 0
- UDP Port Number: 5005
- RTP reporting interval (sec): 5

**Companding Law:**

- Switch:**
  - ☒ U-Law
  - ☐ A-Law
- Line:**
  - ☒ U-Law Line
  - ☐ A-Law Line

**Other Settings:**

- ☐ DSS Status
- ☒ Auto Hold
- ☒ Dial By Name
- ☒ Show Account Code
- ☐ Inhibit Off-Switch Forward/Transfer
- ☐ Restrict Network Interconnect
- ☐ Include location specific information
- ☐ Drop External Only Impromptu Conference
- ☒ Visually Differentiate External Call
- ☒ High Quality Conferencing
- ☒ Directory Overrides Barring
- ☒ Advertise Callee State To Internal Callers
- ☐ Internal Ring on Transfer

### 5.2.5. System VoIP Settings

To view or change system codec settings, select the **VoIP → VoIP** tab. The **RFC2833 Default Payload** field allows for the manual configuration of the payload type used on SIP calls that are initiated by the IP Office. During the compliance test, this was set to **101**, the value preferred by Verizon Business. For codec selection, 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 5.4.5**). 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 screenshot displays the IPOSE-Primary configuration window with the VoIP tab selected. The interface includes a top navigation bar with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VoIP, Contact Center, and Avaya Cloud. The VoIP tab is active, and the sub-tab 'VoIP Security' is selected. Below the sub-tab, there are two checkboxes: 'Ignore DTMF Mismatch For Phones' and 'Allow Direct Media Within NAT Location', both of which are unchecked. The 'RFC2833 Default Payload' is set to '101'. The 'Default Codec Selection' section contains three lists: 'Available Codecs', 'Unused', and 'Selected'. The 'Available Codecs' list has four items, all checked: G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, and G.729(a) 8K CS-AC. The 'Unused' list contains one item: G.711 ALAW 64K. The 'Selected' list contains three items: G.722 64K, G.711 ULAW 64K, and G.729(a) 8K CS-A. Between the 'Unused' and 'Selected' lists are four buttons: '>>>', '<<<', '<<<', and '>>>', along with up and down arrow buttons.



During the compliance test, SRTP was used internal to the enterprise wherever possible. To view or configure the media encryption settings, select the **VoIP → VoIP Security** tab on the Details pane. The **Media Security** drop-down menu is set to **Preferred** to have IP Office attempt use encrypted RTP for devices that support it and fall back to RTP for devices that do not support encryption. Under **Media Security Options**, **RTP** is selected for the **Encryptions** and **Authentication** fields. Under **Crypto Suites**, **SRTP\_AES\_CM\_128\_SHA1\_80** is selected.

**IPPOSE-Primary**

System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events SMTP SMDR VoIP Contact Center Avaya Cloud

VoIP VoIP Security Access Control Lists

Default Extension Password

Confirm Default Extension Password

Media Preferred ☐ Strict SIPs

Media Security Options

Encryptions ☒ RTP ☐ RTCP

Authentication ☒ RTP ☒ RTCP

Replay Protection ☒

SRTP Window Size

Crypto Suites

☒ SRTP\_AES\_CM\_128\_SHA1\_80 ☐ SRTP\_AES\_CM\_128\_SHA1\_32

### 5.3. IP Route

In the reference configuration, the Primary server 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 the Primary server, right-click **IP Route** from the Navigation pane, and select **New** (not shown). 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 **LAN2** as **Destination**.

IP Address	IP Mask	Gateway	Destination	Metric
0.0.0.0	0.0.0.0	10.64.19.1	LAN1	0
172.30.0.0	255.255.0.0	1.1.1.1	LAN2	0

**IP Route**

IP Address

IP Mask

Gateway IP Address

Destination

Metric

## 5.4. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 11.1. 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** Error! Reference source not found. 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 5.4.2 – 5.4.6**.

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

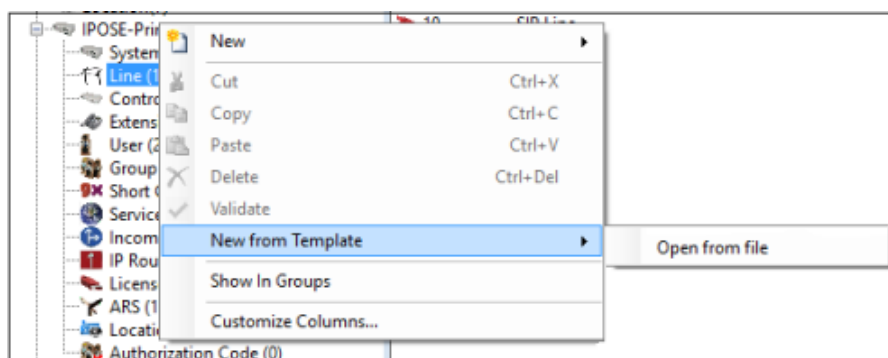
- SIP Line – Originator number for forwarded and twinning calls.
- Transport – Second Explicit DNS Server.
- SIP Credentials – Registration Requirement.
- SIP Advanced Engineering.

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 5.4.2 – 5.4.6**.

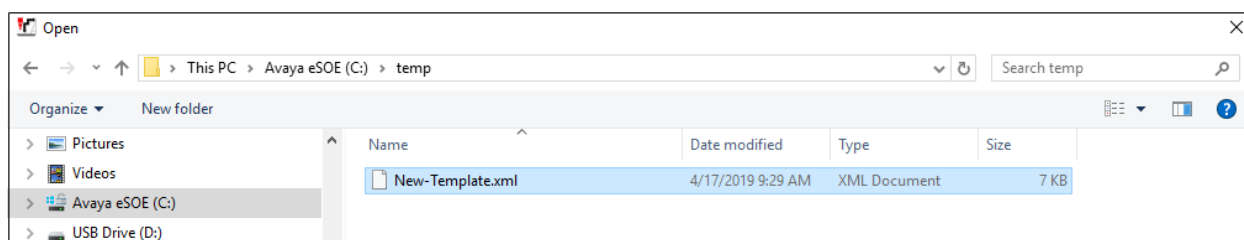
### 5.4.1. Creating a SIP Line from an XML Template

**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.

Copy a previously created template file to the computer where IP Office Manager is installed. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New → New from Template**. Select **Open from file**.



Navigate to the directory where the template was copied on the local computer (e.g., `\temp`) and select it. Click **Open** (not shown).



The new SIP Line is created, and it will appear on the **Navigation** pane (e.g., SIP Line 22). The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.2 to 5.4.6**.

Line		
Line Number	Line Type	Line SubType
1	IP Office Line	WebSocket Server SCN
8	IP Office Line	WebSocket Server SCN
16	SIP Line	
21	SIP Line	
22	SIP Line	

### 5.4.2. SIP Line – SIP Line Tab

The **SIP Line** tab in the Details pane is shown below for Line Number 22, used for the Verizon Business IPCC service. The **ITSP Domain Name** is left blank. With this setting, IP Office will use the **ITSP Proxy Address** (see **Section Error! Reference source not found.**) as the host part in the SIP URIs of the destination headers (Request URI and To headers) for outbound calls. **Local Domain Name** is set to the IP address of the Avaya IP Office LAN2 interface (e.g., **1.1.1.2**). IP Office will use this IP address as the host part in origination headers (From, Contact, P-Asserted Identity and Diversion headers). By default, the **In Service** and **Check OOS** boxes are checked. With these settings, IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office is determined by the **Binding Refresh Time** for LAN2, as shown in **Section 5.2.2**.

In the reference configuration, the IP Office **Country Code** is set to **1**. The From and PAI headers received from Verizon for calls from U.S. PSTN numbers contain “+1” before the calling PSTN number. By configuring the IP Office **Country Code** to 1, the caller ID display presented to IP Office users will be the PSTN number without any codes or prefixes. For example, a call from 3035387006 would display 3035387006. If the **Country Code** does not match the value following the “+” from Verizon, the IP Office user display would show the contents of the **International Prefix** field, followed by the value following the “+”, followed by the PSTN number. For example, if the Country Code parameter were left blank, the IP Office user would see a display such as “01113035387006”. Aside from display implications, if the **Country Code** is not configured, other patterns may also fail to match as expected, such as a match on the **Incoming CLI** field of the Incoming Call Route. See **Section 5.8.3** for configuration of incoming call routing based on the calling number.

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.

The screenshot displays the 'SIP Line - Line 22' configuration window. The 'SIP Line' tab is active, showing fields for Line Number (22), ITSP Domain Name (blank), Local Domain Name (1.1.1.2), URI Type (SIP URI), Location (Cloud), Prefix (blank), National Prefix (blank), International Prefix (011), Country Code (1), Name Priority (System Default), and Description (Vz IPCC). The 'In Service' and 'Check OOS' checkboxes are checked. The 'Session Timers' section shows 'Refresh Method' set to 'Re-invite' and 'Timer (sec)' set to 1800. The 'Redirect and Transfer' section shows 'Incoming Supervised REFER' and 'Outgoing Supervised REFER' both set to 'Auto', 'Send 302 Moved Temporarily' unchecked, and 'Outgoing Blind REFER' checked.

Under **Redirect and Transfer**, the default automatic determination of **Incoming Supervised REFER** and **Outgoing Supervised REFER** is **Auto**. A supervised transfer occurs when a consultation call is made and the REFER contains a Replaces header. The Verizon Business IPCC service does not support supervised REFER, and with this setting, IP Office will not use REFER for supervised transfers. **Send 302 Moved Temporarily** is unchecked, as Verizon does not support receiving a 302 Moved Temporarily message. The **Outgoing Blind REFER** box can be optionally checked to enable use of REFER for blind transfers. In the reference configuration, this parameter is checked. See **Section 2.2** for limitations.

### 5.4.3. SIP Line – Transport Tab

Select the **Transport** tab. The **ITSP Proxy Address** is set to the IP address provided by Verizon (**172.30.205.55**), as shown in **Figure 1**. In the **Network Configuration** area, **UDP** is selected as the **Layer 4 Protocol**. The **Send Port** is set to the port number provided by Verizon Business. As shown in **Figure 1**, this port is **5072** in the sample configuration. The **Listen Port** can retain the default value **5060**. The **Use Network Topology Info** parameter is set to **None**.

SIP Line	Transport	Call Details	VoIP	SIP Credentials	SIP Advanced	Engineering
ITSP Proxy Address: 172.30.205.55						
Network Configuration						
Layer 4 Protocol		UDP		Send Port		5072
Use Network Topology Info		None		Listen Port		5060
Explicit DNS Server(s)		0 . 0 . 0 . 0		0 . 0 . 0 . 0		
Calls Route via Registrar		<input checked="" type="checkbox"/>				
Separate Registrar						

### 5.4.4. SIP Line – Call Details Tab

Select the **Call Details** tab. To add a new SIP URI, click the **Add...** button. A New URI area will be opened. To edit an existing entry, click an entry in the list at the top, and click the **Edit...** button.

SIP Line	Transport	Call Details	VoIP	SIP Credentials	SIP Advanced	Engineering		
SIP URIs								
URI	Groups	Credential	Local URI	Contact	P Asserted ID	P Preferred ID	Diversion Header	Remote Party ID
1	22 22	0: <None>	Auto	Auto				
<div>Add...</div> <div>Remove</div> <div>Edit...</div>								

In the example screen below, a previously configured entry is edited. The **Incoming Group** parameter, set here to **22**, 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, also set to **22**, is relevant when using the SIP REFER method to transfer an inbound toll-free call back out the Verizon Business IPCC service. The **Max Sessions** parameter was set to **10**. This value sets the maximum number of simultaneous calls that can use the URI before IP Office returns busy to any further calls.

**Auto** is selected for the **Local URI** and **Contact** parameters. With this configuration, information in the Incoming Call Route (**Section 5.88**) is used to determine what call is accepted on the SIP Line. Set the **Field meaning** section to the values shown in the screenshot below.

**SIP Line - 22 | Call Details | SIP URI**

New URI

Incoming Group: 22 Max Sessions: 10

Outgoing Group: 22

Credentials: 0: <None>

	Display	Content	Field meaning		
			Outgoing Calls	Forwarding/Twinning	Incoming Calls
Local URI	Auto	Auto	Caller	Caller	Called
Contact	Auto	Auto	Caller	Caller	Called
P Asserted ID	<input type="checkbox"/> None	None	None	None	None
P Preferred ID	<input type="checkbox"/> None	None	None	None	None
Diversion Header	<input type="checkbox"/> None	None	None	None	None
Remote Party ID	<input type="checkbox"/> None	None	None	None	None

OK Cancel Help

### 5.4.5. 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 reference configuration, **Custom** is selected and the codecs **G729(a) 8K CS-ACELP** and **G.711 ULAW 64K** preferred by Verizon are specified. This will cause IP Office to include G.729a and G.711MU in the Session Description Protocol (SDP) offer, in that order. The **Fax Transport Support** drop-down is set to **T38 Fallback**. This enables T.38 to be used if supported, and 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 is 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.

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.

SIP Line: Transport SIP URI VoIP SIP Credentials SIP Advanced Engineering

Codec Selection: Custom

Unused: G.711 ALAW 64K, G.722 64K

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

Fax Transport Support: T38 Fallback

DTMF Support: RFC2833/RFC4733

Media Security: Disabled

Local Hold Music: ☐

Re-invite Supported: ☒

Codec Lockdown: ☐

Allow Direct Media Path: ☐

Force direct media with phones: ☐

PRACK/100rel Supported: ☐

### 5.4.6. SIP Line – SIP Advanced Tab

Select the **SIP Advanced** tab. In the **Media** area, the **Allow Empty INVITE** box is checked to allow IP Office to be the recipient of a Verizon Business IPCC enhanced transfer where the initial INVITE may not have SDP information. The **Indicate HOLD** box is checked to have IP Office send an INVITE with media attribute “sendonly”, indicating to Verizon when a call is placed on hold. Other parameters may be left at default values.

The screenshot displays the 'SIP Advanced' configuration tab within a software application. The interface is divided into several sections: Addressing, Identity, Media, and Call Control. The 'Addressing' section includes 'Association Method' (By Source IP address), 'Call Routing Method' (Request URI), 'Use P-Called-Party' (unchecked), and 'Suppress DNS SRV Lookups' (unchecked). The 'Identity' section lists various options like 'Use phone-context', 'Add user=phone', 'Use + for International', 'Use PAI for Privacy', 'Use Domain for PAI', 'Caller ID from From header', 'Send From In Clear', 'Cache Auth Credentials' (checked), 'User-Agent and Server Headers', 'Send Location Info' (Never), 'Add UUI header', and 'Add UUI header to redirected calls'. The 'Media' section features 'Allow Empty INVITE' (checked), 'Send Empty re-INVITE' (unchecked), 'Allow To Tag Change' (unchecked), 'P-Early-Media Support' (None), 'Send SilenceSupp=Off' (unchecked), 'Force Early Direct Media' (unchecked), 'Media Connection Preservation' (Disabled), and 'Indicate HOLD' (checked). The 'Call Control' section 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' (unchecked), and 'No REFER if using Diversion' (unchecked).

Section	Parameter	Value	
Addressing	Association Method	By Source IP address	
	Call Routing Method	Request URI	
	Use P-Called-Party	<input type="checkbox"/>	
	Suppress DNS SRV Lookups	<input type="checkbox"/>	
Identity	Use "phone-context"	<input type="checkbox"/>	
	Add user=phone	<input type="checkbox"/>	
	Use + for International	<input type="checkbox"/>	
	Use PAI for Privacy	<input type="checkbox"/>	
	Use Domain for PAI	<input type="checkbox"/>	
	Caller ID from From header	<input type="checkbox"/>	
	Send From In Clear	<input type="checkbox"/>	
	Cache Auth Credentials	<input checked="" type="checkbox"/>	
	User-Agent and Server Headers		
	Send Location Info	Never	
Media	Allow Empty INVITE	<input checked="" type="checkbox"/>	
	Send Empty re-INVITE	<input type="checkbox"/>	
	Allow To Tag Change	<input type="checkbox"/>	
	P-Early-Media Support	None	
	Send SilenceSupp=Off	<input type="checkbox"/>	
	Force Early Direct Media	<input type="checkbox"/>	
	Media Connection Preservation	Disabled	
	Indicate HOLD	<input checked="" type="checkbox"/>	
	Call Control	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		<input type="checkbox"/>	
Emulate NOTIFY for REFER		<input type="checkbox"/>	
No REFER if using Diversion		<input type="checkbox"/>	



## 5.5. IP Office Line

In IP Office Server Edition systems, IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. To edit an existing IP Office Line, select **Line** in the Navigation pane, and select the appropriate line to be configured in the Group pane. The screen below shows the IP Office Line to the IP500 V2 Expansion System.

The screenshot displays the IP Office configuration interface. On the left is a navigation pane with a tree structure including BOOTP (3), Operator (3), Solution, User(49), Group(7), Short Code(50), Directory(2), Time Profile(0), Account Code(1), User Rights(1), Location(7), IPOSE-Primary, System (1), Line (14), Control Unit (9), Extension (18), User (22), Short Code (4), Service (0), Incoming Call Route (6), IP Route (3), and License (10). The 'Line' tab is selected, showing a list of lines. Line 8 is highlighted. The main pane shows the configuration for 'IP Office Line - Line 8'. The 'Line' sub-tab is active, displaying fields for Line Number (8), Telephone Number, Transport Type (WebSocket Server), Prefix, Networking Level (SCN), Outgoing Group ID (99007), Security (Medium), Number of Channels (500), and Outgoing Channels (500). The 'Gateway' section shows Address (10.5.5.180), Location (8: Miami), Password, and Confirm Password. The 'SCN Resiliency Options' section includes checkboxes for Supports Resiliency, Backs up my IP phones, Backs up my hunt groups, and Backs up my IP DECT phones.

The screen below shows the IP Office Line, **VoIP Settings** tab. In the reference configuration, a fax machine is connected to one of the analog ports on the Expansion System. **Fax Transport Support** is set to **T.38 Fallback**. The **Media Security** is set to **Same as System (Preferred)** to have IP Office use the system setting for media security set in **Section 5.2.5** to encrypted RTP. Default values were used for all other parameters.

The screenshot displays the 'VoIP Settings' tab for the IP Office Line configuration. The 'Codec Selection' section shows a dropdown menu set to 'System Default'. Below it are two lists: 'Unused' containing 'G.711 ALAW 64K' and 'Selected' containing 'G.722 64K', 'G.711 ULAW 64K', and 'G.729(a) 8K CS-ACELP'. Navigation buttons (right arrow, up arrow, down arrow, left arrow) are between the lists. The 'Fax Transport Support' is set to 'T38 Fallback'. The 'Call Initiation Timeout (s)' is set to 4. The 'Media Security' is set to 'Same as System (Preferred)'. The 'Advanced Media Security Options' section includes a checkbox for 'Same As System' which is checked. At the top right, there are checkboxes for 'Out Of Band DTMF' and 'Allow Direct Media Path', both of which are checked.

## 5.6. 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.

### 5.6.1. SIP User 6241

The following screen shows the **User** tab for user 6241. This user corresponds to an Avaya J169 SIP endpoint.

The screenshot shows the 'User' configuration window for user 'aj169: 6241'. The window has a title bar with the user name and a standard toolbar. Below the title bar is a tabbed interface with the 'User' tab selected. The 'User' tab contains various configuration fields and a list of checkboxes for user profile settings.

Field	Value
Name	aj169
Password	••••••••
Confirm Password	••••••••
Unique Identity	
Conference PIN	••••
Confirm Audio Conference PIN	••••
Account Status	Enabled
Full Name	Avaya J169
Extension	6241
Email Address	aj169@customera.com
Locale	
Priority	5
System Phone Rights	None
Profile	Basic User

- ☐ Receptionist
- ☐ Enable Softphone
- ☐ Enable one-X Portal Services
- ☐ Enable one-X TeleCommuter
- ☐ Enable Remote Worker
- ☐ Enable Desktop/Tablet VoIP client
- ☐ Enable Mobile VoIP Client
- ☐ Send Mobility Email

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.

**SIP Extension: 11210 6241**

Extension ID: 11210


Base Extension: 6241

Phone Password: •••••

Confirm Phone Password: •••••

Caller Display Type: On

Reset Volume After Calls: ☐

Device Type:  Avaya J169 (SIP Feature)

Location: Automatic

Fallback As Remote Worker: Auto

Module: 0

Port: 0

Disable Speakerphone: ☐

The following screen shows the **VoIP** tab for the extension. The **IP Address** field may be left blank. Check the **Reserve Avaya IP endpoint license** box. The **Codec Selection** parameter may retain the default setting **System Default** to follow the system configuration shown in **Section 5.2.5**. The Media Security parameter may also retain the default setting **Same as System (Preferred)** to follow the system configuring shown in **Section 5.2.5**.

IP Address: 0 . 0 . 0 . 0

Codec Selection: System Default

Unused: G.711 ALAW 64K

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

Reserve License: Reserve Avaya IP endpoint license

Fax Transport Support: None

DTMF Support: RFC2833/RFC4733

3rd Party Auto Answer: None

Media Security: Same as System (Preferred)

Advanced Media Security Options: ☒ Same As System

☐ Local Hold Music

☒ Re-invite Supported

☐ Codec Lockdown

☒ Allow Direct Media Path

## 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** (not shown) 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 a hunt group with **Extension 401** and **Name Call Center**. This hunt group was configured to contain various telephones from **Figure 1**. The **Ring Mode** was set to **Longest Waiting** (i.e., “longest waiting”, most idle user receives next call). Click the **Edit** button to change the **User List**.

The screenshot displays the configuration window for a hunt group titled "Longest Waiting Group Call Center: 401". The window is divided into several sections:

- Group Tab:** Contains settings for the hunt group.
  - Name:** Call Center
  - Extension:** 401
  - Ring Mode:** Longest Waiting
  - Hold Music Source:** No Change
  - Ring Tone Override:** None
  - Agent's Status on No-Answer Applies To:** None
  - Central System:** IPOSE-Primary
  - Profile:** Standard Hunt Group
  - Exclude From Directory:** ☐
  - No Answer Time (sec):** System Default (15)
  - Advertise Group:** ☒
- User List:** A table listing the members of the hunt group.

Extension	Name	System
<input checked="" type="checkbox"/> 6242	Avaya 9508	IP500 Expansion
<input checked="" type="checkbox"/> 6237	Avaya 9611	IPOSE-Primary
- Buttons:** Edit... and Remove

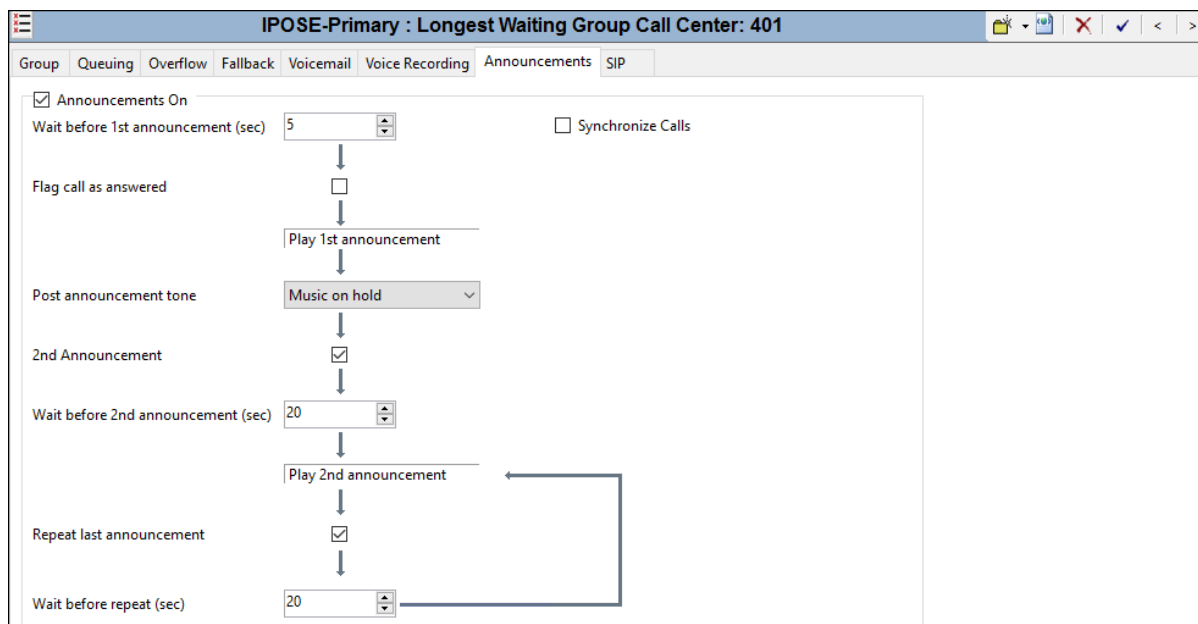
The following screen shows the **Queuing** tab for hunt group 401. In the reference configuration, the hunt group was configured to allow queuing so that incoming Verizon toll-free calls could be queued when all the members of the hunt group were busy on calls. In the testing associated with these Application Notes, the **Queue Length** was varied using both “No Limit” and specifically sized queues. For example, if the **Queue Length** is configured to 2, and if two calls are already in queue, a third call to the Verizon toll-free number corresponding to this hunt group will get busy tone because IP Office will send a 486 Busy Here (i.e., if there is no Voicemail for the hunt group). As another example, if the **Queue Length** has a fixed limit of 2, and if two calls are already in queue, a third call to the Verizon toll-free number destined for this hunt group from a priority caller (see **Section 5.8.3**) will be queued ahead of non-priority callers, temporarily expanding the queue.

IP Office supports priority for queuing. For example, if low priority calls are waiting in queue, a higher priority call entering queue can be moved to the front of the queue and serviced before lower priority callers. For an inbound SIP trunk call, the priority can be specified on the Incoming Call Route as shown in **Section 5.8.3**.

The screenshot displays the 'IPOSE-Primary : Longest Waiting Group Call Center: 401' configuration window. The 'Queuing' tab is selected, showing the following settings:

- ☒ Queuing On
- Queue Length: No Limit (dropdown menu)
- ☒ Normalize Queue Length
- Queue Type: Assign Call On Agent Answer (dropdown menu)
- Calls In Queue Alarm: ☐ (checkbox)
- Calls In Queue Threshold: 1 (spin box)
- Analog Extension to Notify: <None> (dropdown menu)

The following screen shows the **Announcements** tab for hunt group 401. In the reference configuration, when a call arrives, when all members of the hunt group are busy on calls, the caller will first hear ring back tone. If a member of the hunt group does not become available after 10 seconds, the call will be answered by IP Office (i.e., 200 OK will be sent to Verizon), and the toll-free caller will hear a first announcement. Note that the **Flag call as answered** box is relevant for reporting applications but does not change the fact that IP Office will answer the call when the first announcement is played. If the call is still not answered after the first announcement completes, the caller will hear music, a repeating second announcement, music, and so on until the call is answered by a member of the hunt group or answered by voicemail for the hunt group (if configured). If a member of the hunt group becomes available while the caller is listening to ring back, music, or an announcement, the call is de-queued and delivered to the available member.



## 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.

Verizon Business IPCC service allows for blind transfers out their service using the SIP REFER Method. In the screen shown below, the short code **7N** is illustrated. The **Code** parameter is set to **7N**. 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 **53: Vz IPCC**, configurable via ARS. See **Section 5.10** for example ARS route configuration for “53:Vz IPCC”.

The screenshot shows a configuration window titled "7N;: Dial". It contains the following fields and values:

Field	Value
Code	7N;
Feature	Dial
Telephone Number	N
Line Group ID	53: Vz IPCC
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

The following screen illustrates a short code that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code** “\*17” is defined for **Feature** “**Voicemail Collect**”. This short code will be used as one means to allow a Verizon toll-free number to be programmed to route directly to voice messaging, via inclusion of this short code as the destination of an Incoming Call Route. See **Section 5.8** for configuration of Incoming Call Routes.

The screenshot shows a configuration window titled "\*17: Voicemail Collect". It contains the following fields and values:

Field	Value
Code	*17
* This Short Code is common to all systems.	
Feature	Voicemail Collect
Telephone Number	*?U
Line Group ID	0
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

The following screen illustrates another short code. In this case, the **Code** “55N;” is defined for **Feature** “**Conference Meet Me**”. The **Telephone Number** parameter is set to **N**. In the verification of these Application Notes, this short code was used in conjunction with a Voicemail Pro module named “MeetMeConf”. Although the Voicemail Pro configuration is beyond the

scope of these Application Notes, the module enabled a PSTN caller to dial a Verizon toll-free number, be prompted to enter a conference ID and PIN by Voicemail Pro, and then be transferred to the appropriate meet-me conference based on the ID entered by the caller. Local IP Office callers could also dial 556xxx to join the corresponding conference ID.

## 5.8. Incoming Call Routes

In this section, IP Office Incoming Call Routes are illustrated. Each Incoming Call Route will map a Verizon Business toll-free number to a destination user, group, or function on IP Office. In some cases, the destination will be chosen based on the combination of the toll-free number and the caller id of the caller. To add an incoming call route, right click on **Incoming Call Route** (not shown) 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.

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

In the screen shown below, the incoming call route for **Incoming Number 8668502380** is illustrated. The **Line Group Id** is **22**, matching the **Incoming Group** field configured in the **Call Details** tab for the SIP Line to the Verizon Business IPCC service, in **Section 5.4.4**.



Select the **Destinations** tab. From the **Destination** drop-down, select an extension to receive the call when a PSTN user dials 8668502380. This number is associated with IP Office user extension 6241. (The **Destination** was changed in the course of testing to associate different destinations with the toll-free numbers.)

TimeProfile	Destination	Fallback Extension
Default Value	6241 aj169	

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

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

In the screen shown below, an incoming call route for **Incoming Number 8668523221** is illustrated. The **Line Group Id** is **22**, matching the Incoming Group field configured in the Call Details tab for the SIP Line to Verizon Business in **Section 5.4.4**. Optionally, the **Tag** parameter can be populated with a descriptive name that will associate the call with this incoming call route.

Bearer Capability	Any Voice
Line Group ID	22
Incoming Number	8668523221
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	IPCC-4
Hold Music Source	System Source
Ring Tone Override	None

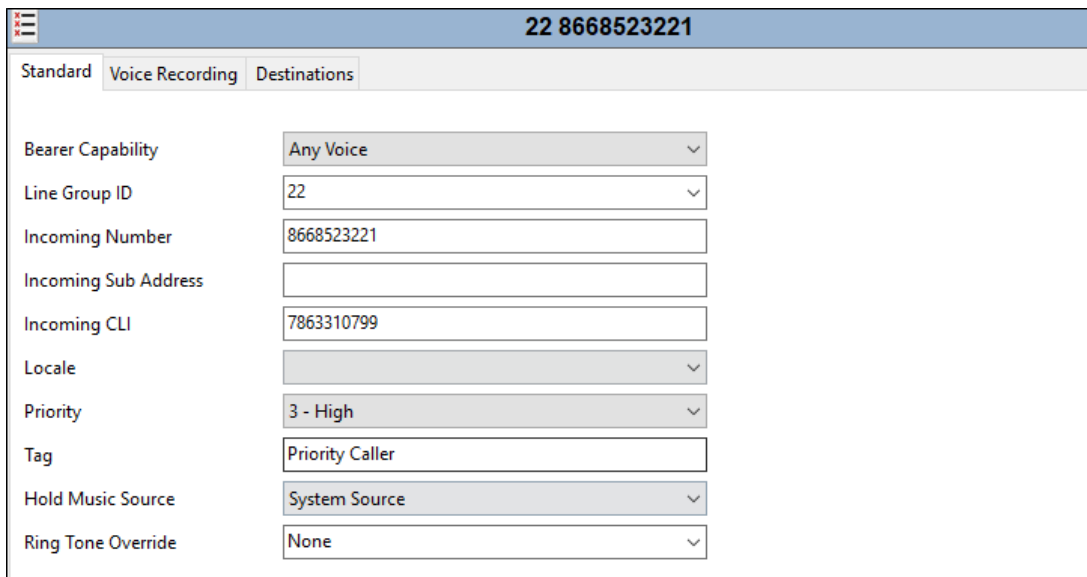
Select the **Destinations** tab. From the **Destination** drop-down, select the destination to receive the call when an arbitrary PSTN user dials 8668523221. This toll-free number is associated with IP Office hunt group extension 401, the “Call Center” hunt group.

TimeProfile	Destination	Fallback Extension
Default Value	401 Call Center	

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

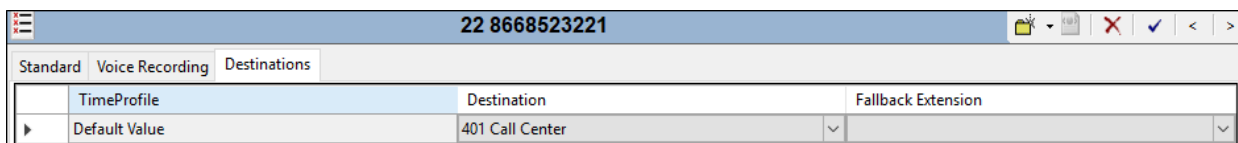
This section presents a simple example showing that IP Office can use the calling party number to distinguish call priority or call destination, for calls to the same toll-free number. Although the matching shown here is based on the full calling number, partial matching is also possible (e.g., to match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

In the screen shown below, the incoming call route for **Incoming Number 8668523221** and **Incoming CLI 7863310799** is illustrated. The **Line Group Id** is **22**, matching the Incoming Group field configured in the Call Details tab for the SIP Line to Verizon Business in **Section 5.4.4**. Note that the **Incoming Number** is the same as the toll-free number configured in the previous section. This route will be used for calls to the toll-free number specifically from a caller with caller ID “7863310799”. In this case, to allow this caller to be treated with priority when calling in, the **Priority** field is set to **3 - High**. Optionally, the **Tag** parameter can be populated with a descriptive name that will associate the call with this incoming call route.



22 8668523221	
Standard Voice Recording Destinations	
Bearer Capability	Any Voice
Line Group ID	22
Incoming Number	8668523221
Incoming Sub Address	
Incoming CLI	7863310799
Locale	
Priority	3 - High
Tag	Priority Caller
Hold Music Source	System Source
Ring Tone Override	None

Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when PSTN user 7863310799 dials 8668523221. In this case, the **Destination** is also the hunt group **401 Call Center**, but since high priority has been configured via the **Standard** tab, incoming calls from this caller will move to the front of the queue and be serviced before calls waiting in queue from other non-priority callers.



22 8668523221		
Standard Voice Recording Destinations		
TimeProfile	Destination	Fallback Extension
Default Value	401 Call Center	

#### 5.8.4. Incoming Call Routes to Various IP Office Features

In the reference configuration, the incoming call route for **Incoming Number 8668506850** was varied to test different destination features, such as Voice Mail, Mobile Call Control, Refer Call Redirection, and Conference Bridge. The screen showing the **Standard** tab for this toll-free number is shown below.

The screenshot shows the configuration for the incoming call route 22 8668506850. The 'Standard' tab is selected. The configuration fields are as follows:

Field	Value
Bearer Capability	Any Voice
Line Group ID	22
Incoming Number	8668506850
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	IPCC-5
Hold Music Source	System Source
Ring Tone Override	None

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

- **\*17** (short code “Voicemail Collect”, as shown in **Section 5.77**). With this destination, an incoming call to 8668506850 will be delivered directly to voice mail, allowing the caller to log-in to voice mail and access messages.
- **VM:MeetMe**. With this destination, an incoming call to 8668506850 will be delivered directly to the Voicemail Pro module “MeetMe” created for use as a conference bridge.
- **VM:Refer**. With this destination, an incoming call to 8668506850 will be delivered directly to the Voicemail Pro module “Refer” created for use as a Refer Call Redirection example. See **Section 5.9** for an illustration of this Voicemail Pro module.

An example screen showing the short code configured for a Voicemail Pro module is shown below.

The screenshot shows the configuration for the incoming call route 22 8668506850, with the 'Destinations' tab selected. The configuration table is as follows:

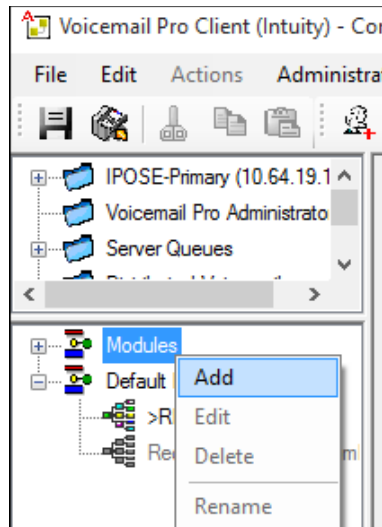
TimeProfile	Destination	Fallback Extension
Default Value	VM:MeetMe	

## 5.9. Voicemail Pro Refer Module



**Note** - While Avaya Voicemail Pro provisioning and programming is beyond the scope of this document, a sample module used in the reference configuration is described below.

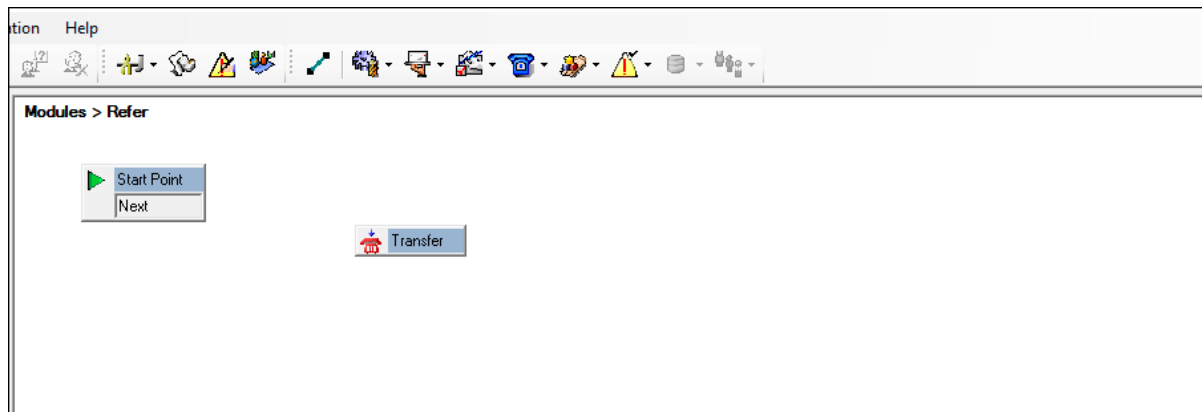
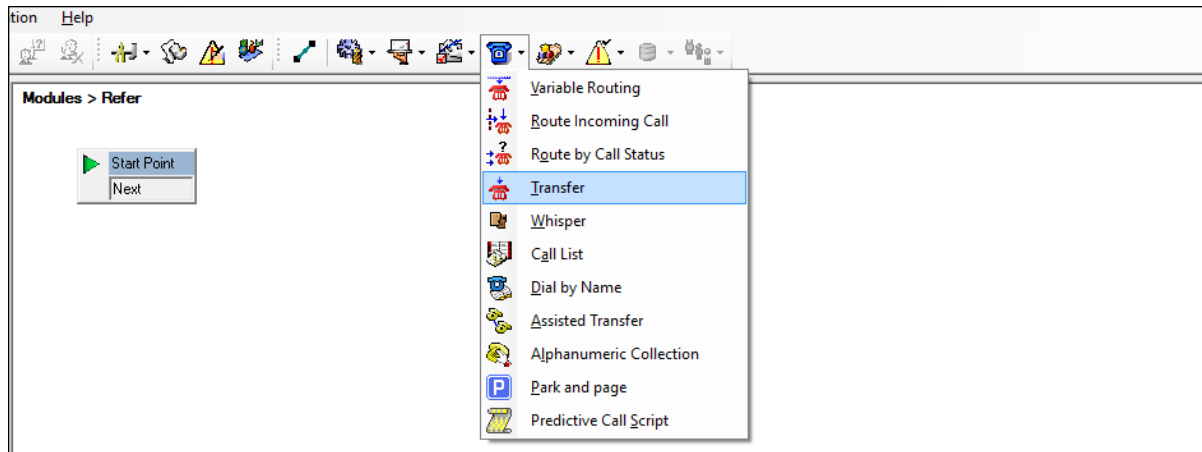
The Refer module is provisioned to play an announcement to the caller, and then generate a Refer (without Replaces) back to the Verizon Business IPCC service. This is accomplished via the Voicemail Pro Client interface.


From the IP Office Manager PC, select **Start → All Apps → IP Office → Voicemail Pro Client** to launch the Voicemail Pro Client interface. Navigate to **File → Login**, select the proper Voicemail Pro system, and log in using the appropriate credentials (not shown). Create a **Start Point** by right clicking on **Modules** and selecting **Add**.

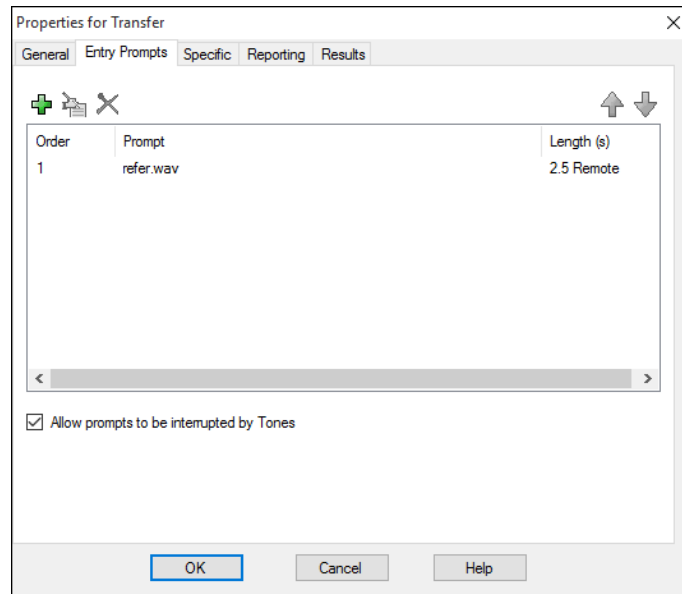


Enter a name (e.g., **Refer**) and click on **OK** (not shown). The new module “Refer” will appear, and a **Start Point** icon will appear in the work area.

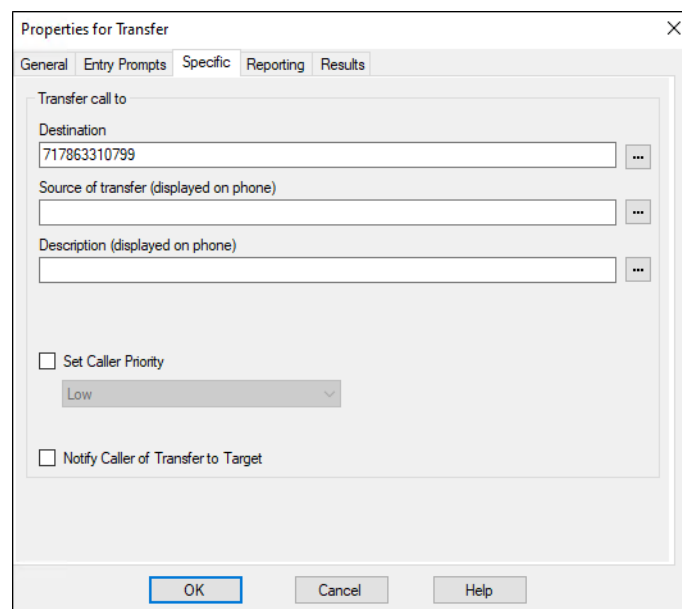
Click on the **Telephony Actions** icon , select the **Transfer** icon , and click on the work area to place the **Transfer** icon in the work area.




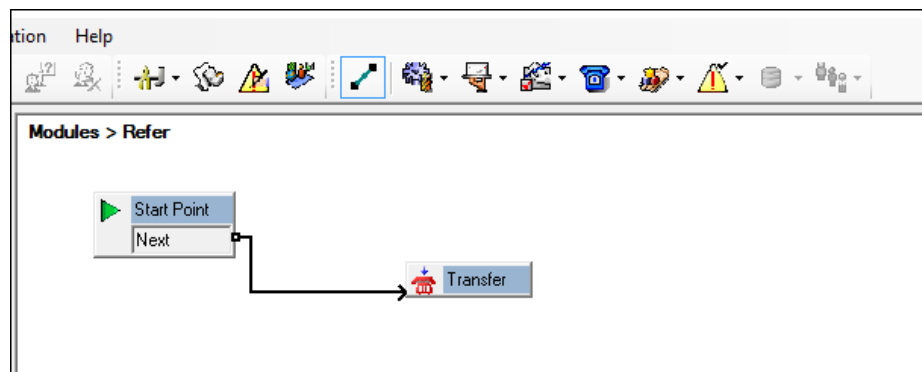
Double click on the **Transfer** icon, select the **Entry Prompts** tab and select or create an announcement to be played to the caller prior to the Refer (e.g., **refer.wav**). To modify an existing recording, double click on the .wav file and rerecord. If no .wav files exist, click on the  icon to open the .wav file editor.




On the **Specific** tab enter the destination, including the outbound Short Code (e.g., **717863310799**). Click on **OK**.



From the options bar, select the **Connector** icon  and drag a connecting flow line from the **Start Point** box to the **Transfer** box.



From the top menu select **File → Save & Make Live** or select the  icon.

When the associated Verizon Business Toll-Free number is received, IP Office will send the call to Voicemail Pro (see **Section 5.8.4**). The caller will hear the announcement (e.g., **refer.wav**), and Voicemail Pro/Avaya IP Office will send a REFER back to the Verizon Business IPCC service, specifying “17863310799” in the Refer-To header. The Verizon Business IPCC service will then send a new Invite to the 1-786-331-0799 destination on the PSTN.

## 5.10. Alternate Route Selection (ARS)

Alternate Route Selection (ARS) is used to route outbound traffic to the SIP line. To define a new ARS route, right-click **ARS** in the Navigation pane and select **New**. In the Details pane that appears, a collection of matching patterns (similar to short codes) can be entered to route calls as shown below.

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 the **ARS Route ID 53**, created in the sample configuration. Verizon Business IPCC service allows for blind transfers using the SIP REFER Method. The sequence of **Xs** used in the **Code** column of the entries to specify the exact number of digits to be expected following the access code. The entry below shows that for calls to area codes in the North American Numbering Plan, the user dials 7, followed by 11 digits. The **Telephone Number** is set to “+.”. This prepends a plus sign (+) to the beginning of the number dialed, denoting a global E.164 number. This is the format preferred by Verizon Business IPCC service for the destination number specified in the Refer-To header. The **Line Group ID** is set to **22** matching the number of the **Outgoing Group** configured on the **Call Details** tab of SIP Line 22 to Verizon Business (Section 5.4.4).

**Vz IPCC**

ARS

ARS Route ID: 53

Route Name: Vz IPCC

Dial Delay Time: System Default (4)

Description:

☒ Secondary Dial tone: SystemTone

☒ Check User Call Barring

In Service: ☒ Out of Service Route: <None>

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

Code	Telephone Number	Feature	Line Group ID
1xxxxxxx	+. .	Dial	22

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

Alternate Route Priority Level: 1

Alternate Route Wait Time: 5

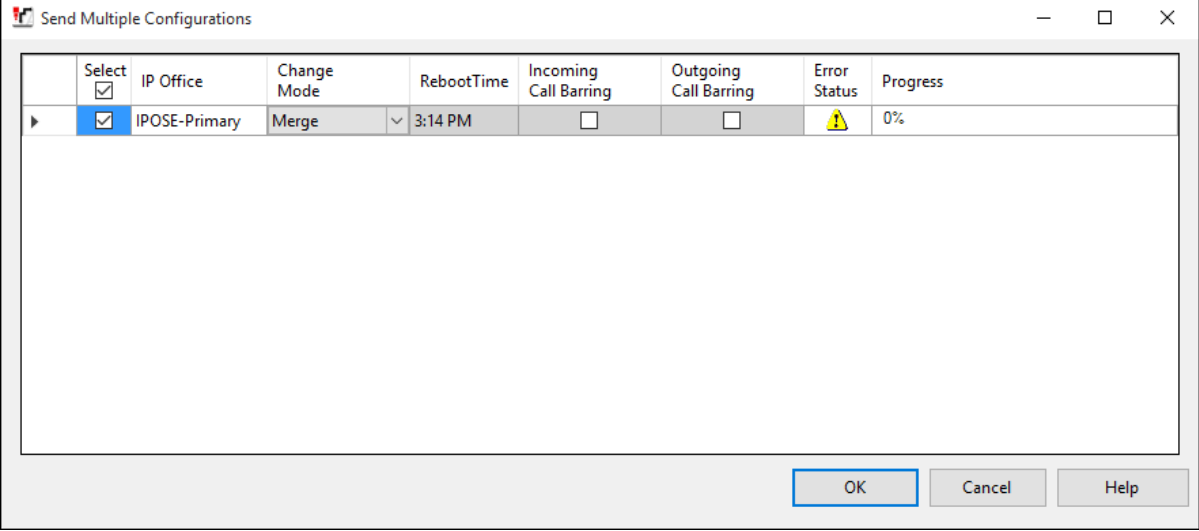
Alternate Route: <None>




## 5.11. 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 **Reboot** 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.



The dialog box titled "Send Multiple Configurations" contains a table with the following data:

Select	IP Office	Change Mode	RebootTime	Incoming Call Barring	Outgoing Call Barring	Error Status	Progress
<input checked="" type="checkbox"/>	IPOSE-Primary	Merge	3:14 PM	<input type="checkbox"/>	<input type="checkbox"/>		0%

At the bottom right of the dialog box are three buttons: **OK**, **Cancel**, and **Help**.

## 6. Avaya IP Office Expansion Configuration

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. 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. On the left is a navigation pane with a tree structure. The 'IP500 Expansion' item is selected and expanded. The main pane is titled 'System Inventory' and shows the configuration for the 'Server Edition Expansion System'. It is divided into three sections: 'Hardware Installed', 'System Settings', and 'Features Configured'.

**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. Expansion System - Physical Hardware

In the sample configuration, the IP500 V2 Expansion System contained a PHONE8 analog card and a COMBO6210 card, for the support of analog and digital stations. Also included is a VCM64 (Voice Compression Module). Both the VCM64 and the COMBO6210 cards provide voice compression channels to the control unit. Voice compression channels are needed to support VoIP calls, including IP extensions and or IP trunks.

The screenshot displays the Avaya IP Office configuration interface, specifically the 'IP 500 V2' hardware configuration page. The left navigation pane shows the 'Control Unit' item selected. The main pane is divided into two sections: 'Control Unit' and 'IP 500 V2'.

**Control Unit**

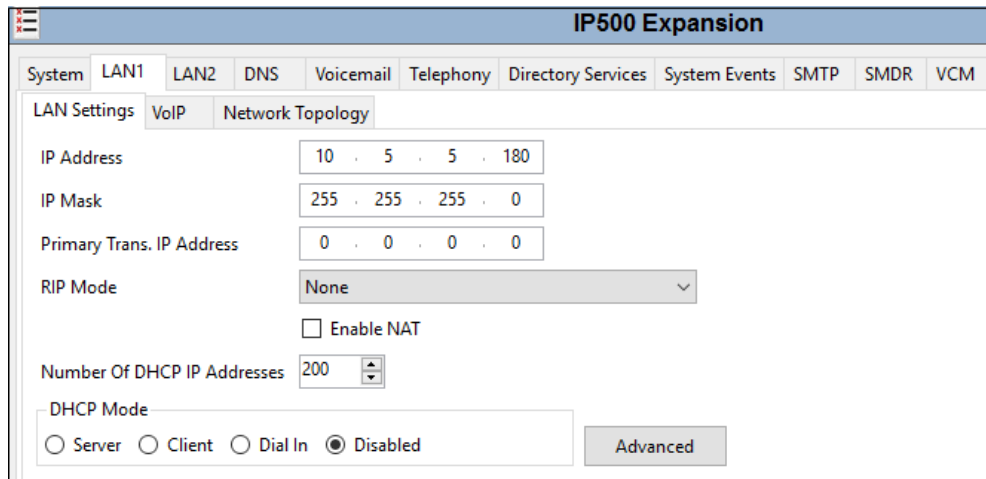
Dev No.	Dev Type	Version
1	IP 500 V2	11.1.0.0 build 237
2	VCM64/PRID U	11.1.0.0.0 build 237
3	PHONE8	11.1.0.0.0 build 237
4	COMBO6210/ATM4	11.1.0.0.0 build 237

**IP 500 V2**

Unit	
Device Number	1
Unit Type	IP 500 V2
Version	11.1.0.0 build 237
Serial Number	00e00706ebf2
Unit IP Address	10.5.5.180
Interconnect Number	0
Module Number	Control Unit

## 6.2. Expansion System - LAN Settings

In the sample configuration, LAN1 is used to connect the Expansion System to the enterprise network. To view or configure the LAN1 IP address, select **System** on the Navigation pane. Select the **LAN1 → LAN Settings** tab on the Details pane. As shown in **Figure 1**, the IP Address of the Expansion System is **10.5.5.180**. Other parameters on this screen may be set according to customer requirements.



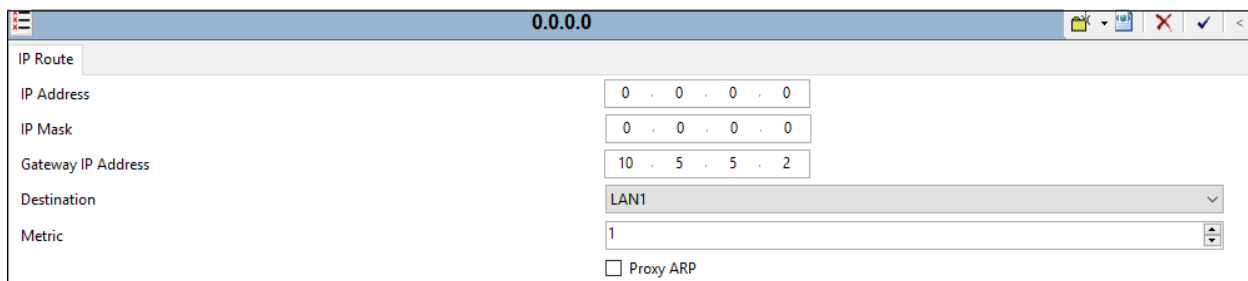
The screenshot shows the 'IP500 Expansion' configuration window. The 'System' tab is selected, and the 'LAN1' sub-tab is active. The 'LAN Settings' section is expanded, showing the following fields:

- IP Address: 10 . 5 . 5 . 180
- IP Mask: 255 . 255 . 255 . 0
- Primary Trans. IP Address: 0 . 0 . 0 . 0
- RIP Mode: None (dropdown menu)
- ☐ Enable NAT
- Number Of DHCP IP Addresses: 200 (spinner)
- DHCP Mode: ☐ Server ☐ Client ☐ Dial In ☒ Disabled
- Advanced button

Defaults were used on the **VoIP** and **Network Topology** tabs (not shown).

## 6.3. Expansion System - IP Route

To create an IP route for the Expansion System, right-click on **IP Route** on the left Navigation pane. Select **New** (not shown). The configuration is similar to the one on the Primary server (**Section 5.33**), with the difference that in the reference configuration, the default gateway for the Expansion System is **10.5.5.2**.



The screenshot shows the 'IP Route' configuration window. The 'IP Route' tab is selected, and the 'New' sub-tab is active. The 'IP Route' section is expanded, showing the following fields:

- IP Address: 0 . 0 . 0 . 0
- IP Mask: 0 . 0 . 0 . 0
- Gateway IP Address: 10 . 5 . 5 . 2
- Destination: LAN1 (dropdown menu)
- Metric: 1 (spinner)
- ☐ Proxy ARP

## 6.4. Expansion System - IP Office Line

The IP Office Lines are automatically created on each server when the Expansion System is added to the solution. Below is the IP Office Line (**Line Number 17**) to the Primary server.

The screenshot shows the configuration page for 'IP Office Line - Line 17'. The 'Line' tab is selected. The configuration includes:

- Line Number: 17
- Transport Type: WebSocket Client
- Networking Level: SCN
- Security: Medium
- Telephone Number: (empty)
- Prefix: (empty)
- Outgoing Group ID: 99999
- Number of Channels: 250
- Outgoing Channels: 250
- Gateway Address: 10 . 64 . 19 . 170
- Port: 443
- Location: 2: Denver
- Password: (masked)
- Confirm Password: (masked)
- SCN Resiliency Options:
  - ☐ Supports Resiliency
  - ☐ Backs up my IP phones
  - ☐ Backs up my hunt groups
  - ☐ Backs up my IP DECT phones
- Description: (empty)

In the reference configuration, a fax machine is connected to one of the analog ports on the Expansion System. To accommodate T.38 fax, select the **VoIP Settings** tab and set **Fax Transport Support** to **T38 Fallback**. The **Media Security** drop-down menu is set to “**Same as System (Preferred)**” to have IP Office use the system setting for media security set in **Section 5.2.5** to encrypted RTP.

The screenshot shows the configuration page for 'IP Office Line - Line 17' with the 'VoIP Settings' tab selected. The configuration includes:

- Codec Selection: System Default
- Unused codecs: G.711 ALAW 64K, G.723.1 6K3 MP-MLQ
- Selected codecs: G.722 64K, G.711 ULAW 64K, G.729(a) 8K CS-ACELP
- Fax Transport Support: T38 Fallback
- Call Initiation Timeout (s): 4
- Media Security: Same as System (Preferred)
- Advanced Media Security Options: ☒ Same As System
- Other options:
  - ☐ VoIP Silence Suppression
  - ☒ Out Of Band DTMF
  - ☒ Allow Direct Media Path

Select the **T38 Fax** tab. The **Use Default Values** box is unchecked, and 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.

The screenshot shows the 'IP Office Line - Line 17' configuration window with the 'T38 Fax' tab selected. The 'Use Default Values' checkbox is unchecked. The 'T38 Fax Version' is set to 0, and the 'Transport' is set to UDPTL. In the 'Redundancy' section, both 'Low Speed' and 'High Speed' are set to 2. The 'TCF Method' is set to Trans TCF, 'Max Bit Rate (bps)' is 14400, 'EFlag Start Timer (ms)' is 2600, 'EFlag Stop Timer (ms)' is 2300, and 'Tx Network Timeout (sec)' is 150. On the right, 'Scan Line Fix-up' and 'TFOP Enhancement' are checked, while 'Disable T30 ECM', 'Disable EFlags For First DIS', 'Disable T30 MR Compression', and 'NSF Override' are unchecked. 'Country Code' and 'Vendor Code' are both set to 0.

## 6.5. Save IP Office Expansion System 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 **Reboot** 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** to proceed.

The screenshot shows the 'Send Multiple Configurations' dialog box. It contains a table with the following data:

Select	IP Office	Change Mode	RebootTime	Incoming Call Barring	Outgoing Call Barring	Error Status	Progress
<input checked="" type="checkbox"/>	IP500 Expansion	Merge	12:12 PM	<input type="checkbox"/>	<input type="checkbox"/>		0%

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

## 7. Verizon Business Configuration

Information regarding Verizon Business IP Contact Center service offer can be found by contacting a Verizon Business sales representative, or by visiting <https://enterprise.verizon.com/products/customer-experience-services/transport-and-intelligent-routing/ip-contact-center/>

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

For service provisioning, Verizon will require the customer IP address used to reach the Avaya IP Office. For the compliance testing, Verizon provided the IP address and port used by the Verizon SBC, as well as the toll-free numbers. This information was used to complete the configuration of the Avaya IP Office shown in the previous sections.

## 8. Verification Steps

This section provides example verifications of the Avaya configuration with Verizon Business Private IP (PIP) Trunk service.

### 8.1. Avaya IP Office

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

#### 8.1.1. System Status Application

The Avaya IP Office System Status application can be used to verify the service state of the SIP line. From the IP Office Manager application, select **File → Advanced → System Status**. Under **Control Unit IP Address** select the IP address of the IP Office system under verification. Log in using the appropriate credentials.

The screenshot shows the 'Login' dialog box of the Avaya IP Office System Status application. It has a blue header with 'Online' and 'Offline' tabs. The 'Control Unit Address' is set to '10.64.19.170'. 'Proxy Server Address' is '<None>'. 'Services Base TCP Port' is '50804'. 'Local IP Address' is 'Automatic'. 'User Name' is 'Administrator'. 'Password' is masked with dots. There are checkboxes for 'Auto reconnect' (unchecked), 'Secure connection' (checked), and 'Websocket connection' (unchecked). A 'Logon' button is at the bottom right.

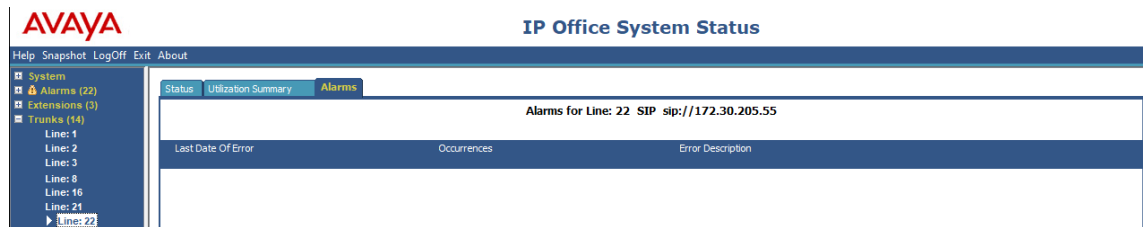
Select the SIP line from the left pane (**Line 22** in the reference configuration). On the **Status** tab in the right pane, verify that the **Current State** is **Idle** for each channel (assuming no active calls at present time).

The screenshot shows the 'IP Office System Status' application. The left pane shows a tree view with 'System', 'Alarms (22)', 'Extensions (3)', 'Trunks (14)', and 'Lines'. 'Line: 22' is selected. The right pane shows the 'Status' tab with a 'SIP Trunk Summary' section. Below this is a table with columns: Channel Number, URI, G..., Call Ref, Current State, Time in State, Remote Media Address, Codec, Connection Type, Caller ID or Dialed Digits, Other Party on Call, Direction of Call, Round Trip Delay, Receive Jitter, Receive Packet Los..., Transmit Jitter, and Transmit Packet Los... The table shows 6 channels, all with 'Current State' as 'Idle' and 'Time in State' as '14:37:51'.

Channel Number	URI	G...	Call Ref	Current State	Time in State	Remote Media Address	Codec	Connection Type	Caller ID or Dialed Digits	Other Party on Call	Direction of Call	Round Trip Delay	Receive Jitter	Receive Packet Los...	Transmit Jitter	Transmit Packet Los...
1				Idle	14:37:51											
2				Idle	23:00:33											
3				Idle	23:00:33											
4				Idle	23:00:33											
5				Idle	23:00:33											
6				Idle	23:00:33											

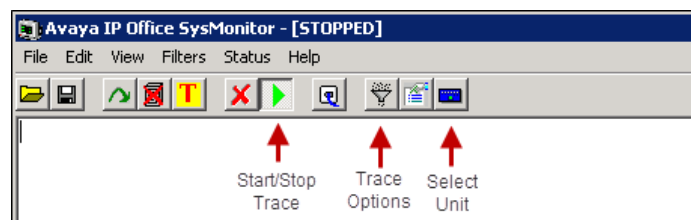
In the lower part of the screen, the **Trace All** button may be pressed to display real time tracing information as calls are made using this SIP Line.

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

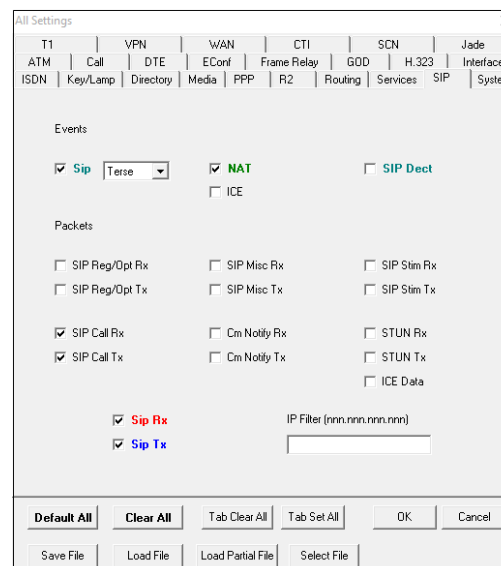


### 8.1.2. System Monitor

The Avaya IP Office Monitor application can be used to monitor and troubleshoot signaling messaging on the SIP trunk. Launch the application from **Start → Programs → IP Office → Monitor** on the PC where IP Office Manager was installed. Click the **Select Unit** icon on the taskbar and Select the IP address of the IP Office system under verification.



Clicking the **Trace Options** icon on the taskbar and selecting the **SIP** tab allows modifying the threshold used for capturing events, types of packets to be captured, filters, etc. Additionally, the color used to represent the packets in the trace can be customized by right clicking on the type of packet and selecting to the desired color.





As an example, the following shows a portion of the monitoring window for an inbound call to Verizon IP Toll Free number 1-866-852-3221. Details of the SIP INVITE message sent by Verizon are shown below. This information matches the configuration in these Application Notes and is not intended to be prescriptive. The intent is to illustrate the INVITE sent by Verizon in the reference configuration, along with the means to retrieve this type of trace information from IP Office.

```
Avaya IP Office SysMonitor - [STOPPED] Monitoring 10.64.19.170 (IPOSE-Primary (Server Edition(P)) (Select)); Log Settings - C:\Users\...\sysmonitorsettings.ini
File Edit View Filters Status Help

13:44:32 18845381mS SIP Rx: UDP 172.30.205.55:5072 -> 1.1.1.2:5060
INVITE sip:8668523221@adavc.avaya.globalipcom.com:5060 SIP/2.0
Via: SIP/2.0/UDP 172.30.205.55:5072;branch=z9hG4bKbnpj55108olrr9k9c0u0.1
Allow: INVITE, ACK, BYE, OPTIONS, CANCEL, SUBSCRIBE, REFER
Content-Length: 318
Max-Breadth: 60
CSeq: 1 INVITE
Contact: <sip:+17863310799@172.30.205.55:5072;sipappsessionid=app-lwjthdqb1reg;wlsfcid=sip-tgau2ca4rnrx;transport=udp>
Call-ID: wls-f5753617-3cd632a034b7bc9813c4f37603b7050aa52ff657cfd2f6a0-0024-5747
Max-Forwards: 69
From: <sip:+17863310799@152.188.183.52:5060;user=phone>;tag=49b4a7aa
c: application/sdp
To: sip:18668523221@1.1.1.2
P-Asserted-Identity: "AVAYA INC" <sip:+17863310799@152.188.183.52;user=phone>
User-Agent: CS2000_NGSS/8.0

v=0
o=genband 268688384 1591386273 IN IP4 172.30.205.164
s=-
c=IN IP4 172.30.205.164
t=0 0
m=audio 14232 RTP/AVP 18 0 8 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sgn: 0
a=csrc:1 image udpt1 t38

13:44:32 18845381mS Sip: Association found trunk: SIP Line (22)
13:44:32 18845381mS CMCallEvt: 0000000000000000 0.1471.0 -1 BaseEP: NEW CMEndpoint f6641de0 TOTAL NOW=1 CALL_LIST=0
13:44:32 18845381mS Sip: SIP Line (22): sip_trunk_config_items 50020009, sip_trunk_config_items_2 00000000, voip.flags 81000949
13:44:32 18845381mS Sip: SIPDialog f6629ac8 created, dialogs 2 txn_keys 1 video 1 presentation 1 camera 1 unsupp audio 0
13:44:32 18845381mS Sip: 0000000000000000 0.1471.0 -1 SIPTrunk Endpoint(f6629ac8) SetUnintTransactionCondition to Unint_None
13:44:32 18845381mS CMap: IP::SetCodec pcp[418]b0r0 0 -> f66536c0
13:44:32 18845381mS Sip: SIP Line (22): License, Valid 1, Available 100, Consumed 0
13:44:32 18845381mS Sip: SIPTrunkEndpointDialogOwner::SetRemoteAddressForRequest from 172.30.205.55:5072 to 172.30.205.55:5072
13:44:32 18845381mS Sip: FindContactParameters mFarDisplayString <AVAYA INC> mFarDisplayNumber <+17863310799> restricted 0
13:44:32 18845381mS Sip: 0a4013aa000005bf 22.1471.1 -1 SIPTrunk Endpoint(f6629ac8) Cloned
13:44:32 18845381mS Sip: SIPDialog::ExtractResponseParamsFromViaHeader remote sent by: 172.30.205.55:5072 trunk
13:44:32 18845381mS Sip: SIPDialog::ExtractResponseParamsFromViaHeader remote sent by transport: SIP/2.0/UDP trunk
13:44:32 18845381mS Sip: SIPTrunkEndpointDialogOwner::SetRemoteAddressForResponse from 172.30.205.55:5072 to 172.30.205.55:5072
13:44:32 18845381mS Sip: 0a4013aa000005bf 22.1471.1 -1 SIPTrunk Endpoint(f6629ac8) SendSIPResponse: INVITE code 100 SENT TO 172.30.205.55 5072
13:44:32 18845381mS SIP Tx: UDP 1.1.1.2:5060 -> 172.30.205.55:5072
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.30.205.55:5072;branch=z9hG4bKbnpj55108olrr9k9c0u0.1
From: <sip:+17863310799@152.188.183.52:5060;user=phone>;tag=49b4a7aa
Call-ID: wls-f5753617-3cd632a034b7bc9813c4f37603b7050aa52ff657cfd2f6a0-0024-5747
CSeq: 1 INVITE
Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,NOTIFY
Supported: timer
Server: IP Office 11.1.0.0.0 build 237
Content-Length: 0
To: <sip:18668523221@1.1.1.2>;tag=b1995c50bdfbde0

13:44:32 18845381mS Sip: 0a4013aa000005bf 22.1471.1 -1 SIPTrunk Endpoint(f6629ac8) INVITE Received ep 0a4013aa000005bf 22.1471.1 -1 SIPTrunk Endpoint(f6641de0), dialo
13:44:32 18845381mS Sip: 0a4013aa000005bf 22.1471.1 -1 SIPTrunk Endpoint(f6629ac8) UpdateSIPCallState SIPDialog::INITIAL(0) -> SIPDialog::INVITE_RCVD(9)
```

## 9. Conclusion

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

These Application Notes demonstrated how IP Office Release 11.1 can be successfully combined with a Verizon Business IP Contact Center VoIP Inbound Service connection to enable a business to receive toll-free calls. 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, Deploying IP Office Server Edition Solution*, Release 11.1, Issue 14, April 2020
- [2] *IP Office™ Platform 11.1, Deploying Avaya IP Office Servers as Virtual Machines*, April 2020
- [3] *IP Office™ Platform 11.1, Deploying an IP500 V2 IP Office Essential Edition System*, April 2020.
- [4] *Administering Avaya IP Office™ Platform with Manager*, Release 11.1, Issue 1, April 2020.
- [5] *Administering Avaya IP Office™ Platform with Web Manager*, Release 11.1, Issue 1, April 2020.
- [6] *Planning for and Administering Avaya IX™ Workplace Client for Android, iOS, Mac and Windows*, Release 3.8, Issue 1, March 2020
- [7] *Using Avaya IX™ Workplace Client for Android, iOS, Mac and Windows*, Release 3.8, Issue 1, March 2020
- [8] *RFC 3261 SIP: Session Initiation Protocol*. <https://www.ietf.org/rfc/rfc3261.txt>

Additional IP Office documentation can be found at:

<https://ipofficekb.avaya.com/>

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 11.1 with Verizon IP Trunk Service Suite:

[VzIPT-IPO11\_1] Application Notes for Configuring SIP Trunking Using Verizon Business IP Trunking Service with Avaya IP Office Release 11.1, Issue 1.0

The following documents may be obtained by contacting a Verizon Business Account Representative.

- *Retail VoIP Interoperability Test Plan*
- *Network Interface Specification Retail VoIP Trunk Interface (for non-registering devices)*

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