



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Level 3 SIP Trunking with Avaya IP Office Release 8.1 – Issue 1.0

Abstract

These Application Notes describes the steps to configure Session Initiation Protocol (SIP) Trunking between Level 3 and Avaya IP Office Release 8.1.

Level 3 SIP Trunking provides PSTN access via a SIP trunk between the enterprise and the Level 3 network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted in the Avaya Solution and Interoperability Test Lab, utilizing Level 3 SIP Trunk Services.

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Level 3 and Avaya IP Office Release 8.1.

The Level 3 SIP Trunking service referenced within these Application Notes is positioned for customers that have an IP-PBX or IP-based network equipment with SIP functionality, but need a form of IP transport and local services to complete their solution.

Level 3 SIP Trunking will enable delivery of origination and termination of local, long-distance and toll-free traffic across a single broadband connection. A SIP signaling interface will be enabled to the Customer Premises Equipment (CPE).

The Level 3 SIP Trunking service uses Digest Authentication for outbound calls from the enterprise, using challenge-response authentication for each call to the Level 3 network based on a configured user name and password (provided by Level 3 and configured in IP Office). This call authentication scheme as specified in SIP RFC 3261 provides security and integrity protection for SIP signaling.

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the Level 3 SIP Trunking service via the public Internet and exercise the features and functionality listed in **Section 2.1**. The simulated enterprise site was comprised of Avaya IP Office and various Avaya endpoints.

The Level 3 SIP Trunk Service passed compliance testing with any observations or limitations described in **Section 2.2**.

2.1. Interoperability Compliance Testing

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

- Response to SIP OPTIONS queries from the provider
- Incoming PSTN calls to various phone types including H.323, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types including H.323, digital, and analog telephones at the enterprise. All outgoing PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from soft clients. Avaya IP Office supports two soft clients: Avaya IP Office Phone Manager and Avaya IP Office Softphone. Avaya IP Office Phone Manager supports two modes (PC softphone and telecommuter). Both clients in each supported mode were tested.
- Various call types including: local, long distance, outbound toll-free, operator services and local directory assistance.

- Codec G.711MU and G.729A.
- T.38 Fax
- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- Voicemail navigation using DTMF for inbound and outbound calls.
- User features such as hold and resume, transfer, and conference.
- Off-net call forwarding and twinning.
- REFER method for call redirection

2.2. Test Results

Interoperability testing of Level 3 SIP Trunking was completed with successful results for all test cases with the exception of the observations/limitations described below.

- Inbound toll-free and outbound emergency calls (911) are supported but were not tested as part of the compliance test.
- **SIP OPTIONS from IP Office:** During the compliance test, no SIP OPTIONS messages were sent from IP Office to Level 3 even though IP Office was configured to send them. The reason for this is still under investigation. SIP OPTIONS messages were tested in the opposite direction from Level 3 to IP Office. Level 3 does not require IP Office to initiate OPTIONS so this behavior had no user impact. The configuration relating to the sending of OPTIONS is described in **Sections 5.2, 5.6.1 and 5.11.**
- **Ring No Answer:** Outbound call from IP Office is allowed to ring without answer until the call times out (approximately three minutes). At this time, Level 3 sends a “401 Unauthorized” response with parameter stale=yes. IP Office resends the INVITE with proper credentials. Level 3 sends a “481 Unknown Dialog” response. Even with the error response, the call is properly disconnected. There is no user impact to this behavior.
- **No Matching Codec Offered:** If the IP Office SIP trunk is improperly configured to have no matching codec with the service provider and an outbound call is placed, Level 3 returns a “480 Temporarily Unavailable” response instead of a “488 Not Acceptable Here” response. The user hears fast busy. There is no user impact to this behavior.
- **Off-net Call Forward with REFER:** If REFER is enabled, IP Office will send a REFER message to Level 3 when an inbound call is forwarded back to the PSTN. Level 3 responds with a 603 Decline response even though the call is properly forwarded. There is no user impact to this behavior.
- **Blind Transfer with REFER:** If REFER is enabled, IP Office will send a REFER message to Level 3 when an inbound call is transferred to the PSTN. If the transfer is completed before the transfer destination answers (blind transfer), then IP Office sends an unexpected re-INVITE to Level 3 after the call has been transferred resulting in a “481 Unknown Dialog” response from Level 3. However, the call is properly transferred so there is no user impact to this behavior.

2.3. Support

For technical support on Level 3 SIP Trunking, contact Level 3 using the Customer Service links at www.Level3.com or by calling 1-877-4LEVEL3.

3. Reference Configuration

Figure 1 illustrates the sample configuration used for the DevConnect compliance testing. The sample configuration shows an enterprise site connected to Level 3 SIP Trunking.

Located at the enterprise site is an Avaya IP Office 500 V2. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public network. Endpoints include an Avaya 1608 IP Telephone (with H.323 firmware), an Avaya 9641G IP Telephone (with H.323 firmware), an Avaya 9640G IP Telephone (with H.323 firmware), an Avaya IP Office Phone Manager, an Avaya IP Office Softphone, an Avaya 5420 Digital Telephone, and an Avaya 6210 Analog Telephone. The site also has a Windows 2003 Server running Avaya Voicemail Pro for voicemail and running Avaya IP Office Manager to configure the Avaya IP Office.

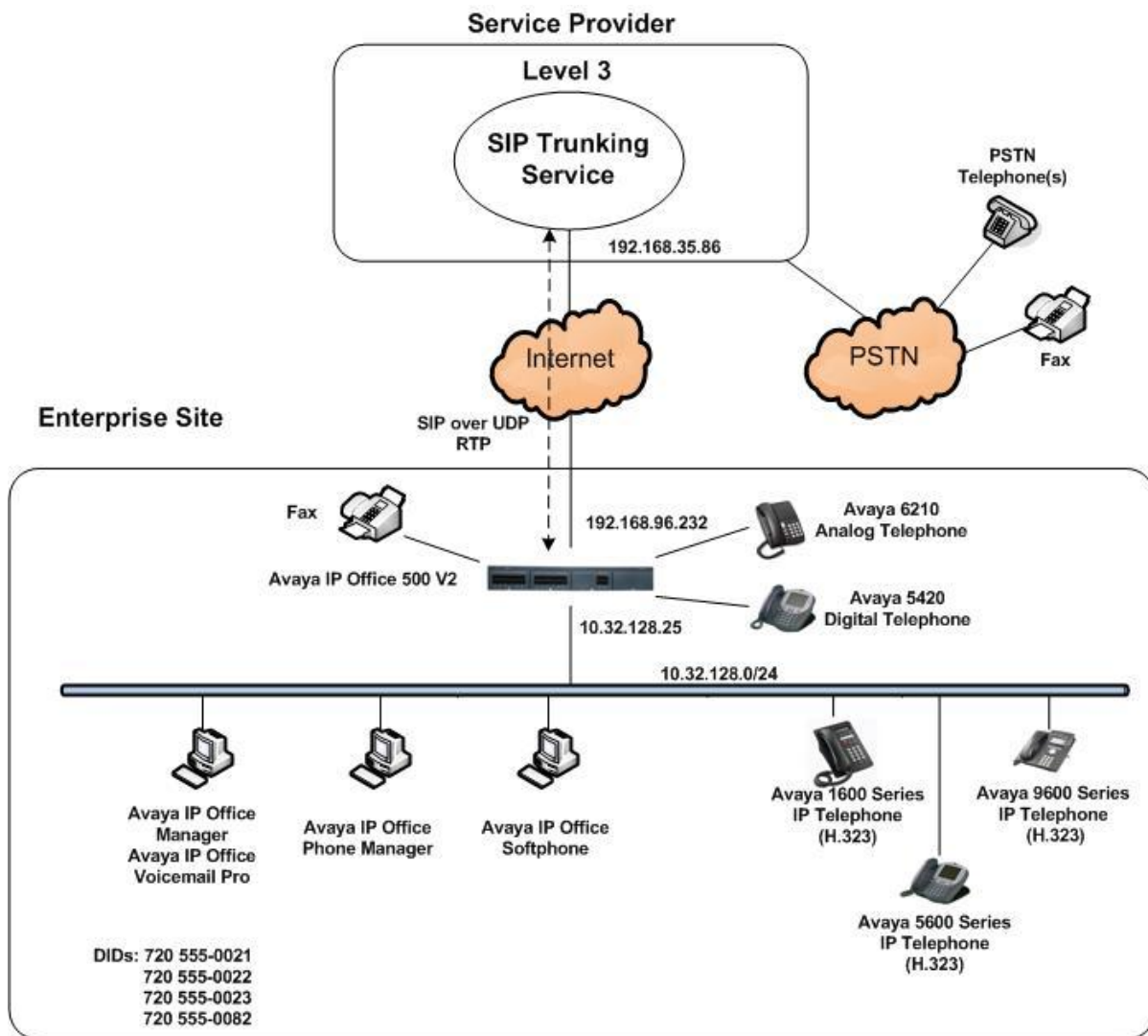


Figure 1: Avaya Interoperability Test Lab Configuration

For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been replaced with private addresses and all phone numbers have been replaced with numbers that cannot be routed over the PSTN.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to Level 3. The short code of 9 is stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Level 3. For calls within the North American Numbering Plan (NANP), the user dialed 11 (1 + 10) digits for long distance calls and 10 digits for local calls. Avaya IP Office sent either 11 digits or 10 digits, depending on the type of NANP call, in the Request URI and the To field of an outbound SIP INVITE message. It was configured to send 10 digits in the From field. For inbound calls, Level 3 SIP Trunking sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and the Avaya IP Office such as a session border controller or data firewall. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes. However, it should be noted that SIP and RTP traffic between the service provider and the Avaya IP Office must be allowed to pass through these devices.

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Avaya Telephony Components	
Equipment	Software
Avaya IP Office 500 v2	8.1 (63)
Avaya Voicemail Pro	8.1 (Build 8.1.9016.0)
Avaya IP Office Manager	10.1 (63)
Avaya 16xx Series IP Telephones (H.323) running Avaya one-X® Deskphone Value Edition	1.3 SP2 (1.3.2)
Avaya 56xx Series IP Telephones (H.323)	2.9.1
Avaya 96x0 Series IP Telephones (H.323) running Avaya one-X® Deskphone Edition	3.1 SP5 (3.1.05S)
Avaya 96x1 Series IP Telephones (H.323) running Avaya one-X® Deskphone Edition	6.2 SP2 (6.2209)
Avaya IP Office Video Softphone (SIP)	3.2.3 (64595)
Avaya IP Office Phone Manager (H.323)	4.2.36
Avaya Digital Telephones (5420)	-
Avaya Analog Telephones	-

Level 3 Components	
Equipment	Software
Level 3 Enterprise Edge	Version 1

5. Configure Avaya IP Office

Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the Avaya IP Office Manager PC, select **Start** → **Programs** → **IP Office** → **Manager** to launch the application. A screen that includes the following in the center may be displayed:

WELCOME to IP Office Administration

What would you like to do ?

[Create an Offline Configuration](#)

[Open Configuration from System](#)

[Read a Configuration from File](#)

Select **Open Configuration from System**. If the above screen does not appear, the configuration may be alternatively opened by navigating to **File** → **Open Configuration** at the top of the Avaya IP Office Manager window. Select the proper Avaya IP Office system from the pop-up window and log in with the appropriate credentials. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, omit the Group pane in the center, and the Details pane on the right side. Since the Group Pane has been omitted, its content is shown as submenus in the Navigation pane. These panes (Navigation, Group and Details) will be referenced throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the service provider (such as twinning and IP Office Softphone support) is assumed to already be in place.

5.1. Licensing and Physical Hardware

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

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

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane, and on the right is the 'SIP Trunk Channels' details pane.

IP Offices Navigation Pane:

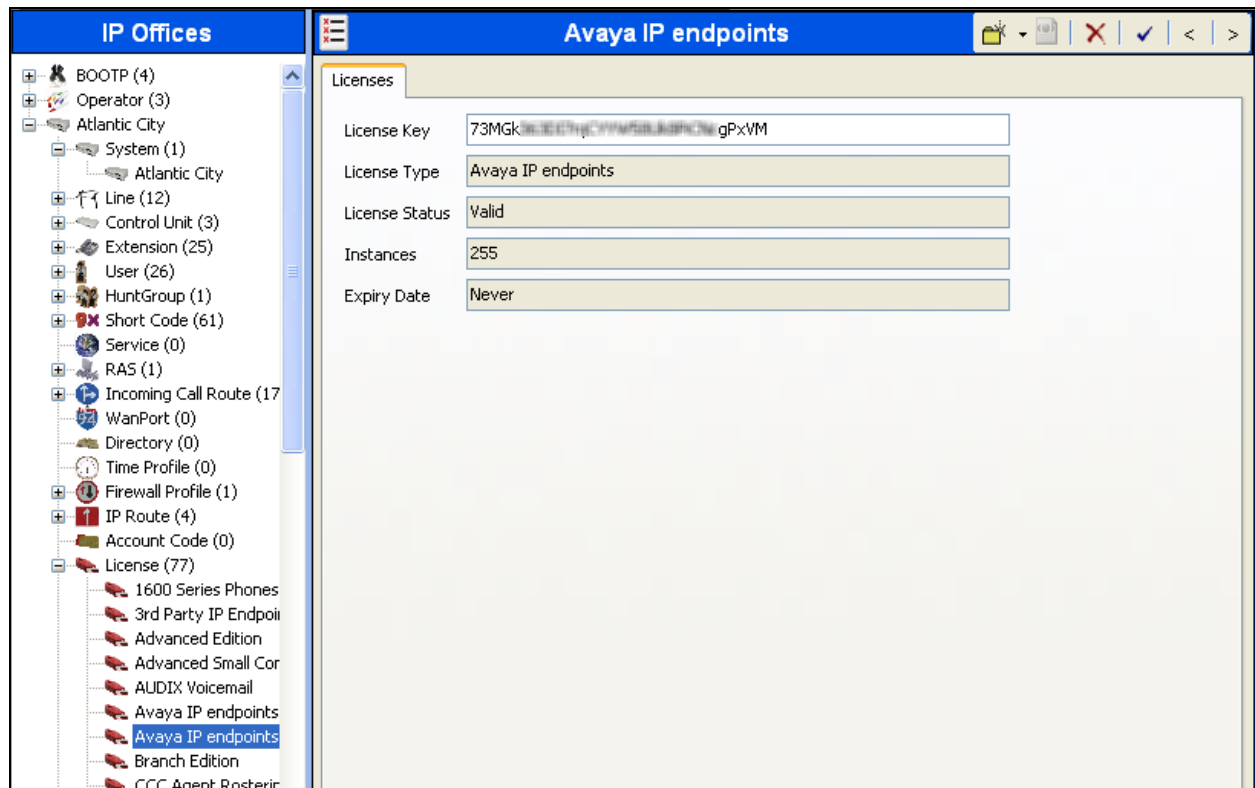
- BOOTP (4)
- Operator (3)
- Atlantic City
 - System (1)
 - Line (12)
 - Control Unit (3)
 - Extension (25)
 - User (26)
 - HuntGroup (1)
 - Short Code (61)
 - Service (0)
 - RAS (1)
 - Incoming Call Route (19)
 - WanPort (0)
 - Directory (0)
 - Time Profile (0)
 - Firewall Profile (1)
 - IP Route (4)
 - Account Code (0)
 - License (77)
 - 1600 Series Phones
 - 3rd Party IP Endpoi
 - Advanced Edition
 - Advanced Small Cor
 - AUDIX Voicemail
 - Avaya IP endpoints
 - Avaya IP endpoints
 - Branch Edition

SIP Trunk Channels Details Pane:

The 'Licenses' tab is selected, showing the following details:

License Key	TyKof[redacted]3wc7x
License Type	SIP Trunk Channels
License Status	Valid
Instances	255
Expiry Date	Never

If Avaya IP Telephones will be used as is the case in these Application Notes, verify the Avaya IP endpoints license. Click **License** → **Avaya IP endpoints** in the Navigation pane. Confirm a valid license with sufficient **Instances** in the Details pane.



To view the physical hardware comprising the IP Office system, expand the components under the **Control Unit** in the Navigation pane. In the sample configuration, the second component listed is a Combination Card. This module has 6 digital stations ports, two analog extension ports, 4 analog trunk ports and 10 VCM channels. The VCM is a Voice Compression Module supporting VoIP codecs. An IP Office hardware configuration with a VCM component is necessary to support SIP trunking.

To view the details of the component, select the component in the Navigation pane. The following screen shows the details of the **IP 500 V2**.

The screenshot displays the IP Office configuration interface. On the left is the 'IP Offices' navigation pane, and on the right is the 'IP 500 V2' details pane.

IP Offices Navigation Pane:

- BOOTP (4)
- Operator (3)
- Atlantic City
 - System (1)
 - Atlantic City
 - Line (12)
 - Control Unit (3)
 - 1 IP 500 V2**
 - 2 COMBO6210/ATM4
 - 3 DIGSTA8/ATM4
 - Extension (25)
 - User (26)
 - HuntGroup (1)
 - Short Code (61)

IP 500 V2 Details Pane:

Unit	
Device Number	1
Unit Type	IP 500 V2
Version	8.1 (63)
Serial Number	00000000e3
Unit IP Address	10.32.128.25
Interconnect Number	0
Module Number	Control Unit

5.2. LAN2 Settings

In the sample configuration, the WAN port was used to connect the Avaya IP Office to the public network. The LAN2 settings correspond to the WAN port on the Avaya IP Office 500. To access the LAN2 settings, first navigate to **System** → <Name>, where <Name> is the system name assigned to the IP Office. In the case of the compliance test, the system name is **Atlantic City**. Next, navigate to the **LAN2** → **LAN Settings** tab in the Details Pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office WAN port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements.

The screenshot displays the Avaya IP Office configuration interface. On the left, the 'IP Offices' tree shows a hierarchy starting with 'Atlantic City' under 'System (1)'. The main pane is titled 'Atlantic City' and contains several tabs: 'System', 'LAN1', 'LAN2', 'DNS', 'Voicemail', 'Telephony', 'Directory Services', 'System Events', 'SMTP', and 'SMDR'. The 'LAN2' tab is selected, and within it, the 'LAN Settings' sub-tab is active. The configuration fields are as follows:

- IP Address:** 192 . 168 . 96 . 232
- IP Mask:** 255 . 255 . 255 . 224
- Primary Trans. IP Address:** 0 . 0 . 0 . 0
- Firewall Profile:** <None> (dropdown menu)
- RIP Mode:** None (dropdown menu)
- Enable NAT:** ☐
- Number Of DHCP IP Addresses:** 200 (spinner)
- DHCP Mode:** Server, Client, Dialin, Disabled (radio buttons, with 'Disabled' selected)
- Advanced:** A button to expand further settings.

On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2. Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the example below and are also the default values. For a customer installation, if the default values are not sufficient, appropriate values will be provided by Level 3. All other parameters should be set according to customer requirements.

The screenshot shows the Avaya IP Office configuration interface for 'Atlantic City'. The 'VoIP' tab is selected, and the 'SIP Registrar' sub-tab is active. The interface includes several sections for configuration:

- General Settings:**
 - ☒ H.323 Gatekeeper Enable
 - ☒ SIP Trunks Enable
 - ☒ SIP Registrar Enable
- H.323 Settings:**
 - ☐ H.323 Auto-create Extn
 - ☐ H.323 Auto-create User
 - ☐ H.323 Remote Extn Enable
 - ☒ Enable RTCP Monitoring On Port 5005
- RTP Port Number Range:**
 - Port Range (Minimum): 49152
 - Port Range (Maximum): 53246
- DiffServ Settings:**
 - DSCP(Hex): B8, DSCP Mask (Hex): FC, SIG DSCP (Hex): 88
 - DSCP: 46, DSCP Mask: 63, SIG DSCP: 34
- DHCP Settings:**
 - Primary Site Specific Option Number (SSON): 176
 - Secondary Site Specific Option Number (SSON): 242
 - VLAN: Not Present
 - 1100 Voice VLAN Site Specific Option Number (SSON): 232
 - 1100 Voice VLAN IDs: (empty field)
- RTP Keepalives:**
 - Scope: Disabled, Periodic timeout: 30
 - Initial keepalives: Enabled

On the **Network Topology** tab in the Details Pane, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to **Open Internet**.
- Set **Binding Refresh Time (seconds)** to **30**. This value is used to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- Set the **Public Port** to the port Avaya IP Office will listen on.
- All other parameters should be set according to customer requirements.

The screenshot shows the 'Atlantic City' configuration window. The 'Network Topology' tab is selected. The 'Network Topology Discovery' section contains the following fields and values:

Field	Value
STUN Server IP Address	10 . 90 . 168 . 13
STUN Port	3478
Firewall/NAT Type	Open Internet
Binding Refresh Time (seconds)	30
Public IP Address	192 . 168 . 96 . 232
Public Port	5060

At the bottom right of the configuration area, there are two buttons: 'Run STUN' and 'Cancel'. Below these buttons is a checkbox labeled 'Run STUN on startup' which is currently unchecked.

5.3. System Telephony Settings

To access the System Telephony settings, first navigate to **System** → **Atlantic City** in the Navigation Pane and then navigate to the **Telephony** → **Telephony** tab in the Details Pane. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the service provider across the SIP trunk. If for security reasons incoming calls should not be allowed to transfer back to the PSTN then leave this setting checked.

The screenshot displays the 'Atlantic City' configuration window, specifically the 'Telephony' tab. The interface is divided into several sections:

- Navigation Tabs:** System, LAN1, LAN2, DNS, Voicemail, **Telephony**, Directory Services, System Events, SMTP, SMDR, Twinning, VCM, CCR, and Codecs.
- Sub-Tabs:** **Telephony**, Tones & Music, and Call Log.
- Analogue Extensions:**
 - Default Outside Call Sequence: Normal (dropdown)
 - Default Inside Call Sequence: Ring Type 1 (dropdown)
 - Default Ring Back Sequence: Ring Type 2 (dropdown)
 - Restrict Analogue Extension Ringer Voltage: ☐
- Timing Settings:**
 - Dial Delay Time (secs): 4 (spinner)
 - Dial Delay Count: 0 (spinner)
 - Default No Answer Time (secs): 25 (spinner)
 - Hold Timeout (secs): 120 (spinner)
 - Park Timeout (secs): 300 (spinner)
 - Ring Delay (secs): 5 (spinner)
 - Call Priority Promotion Time (secs): Disabled (dropdown)
 - Default Currency: USD (dropdown)
 - Default Name Priority: Favor Trunk (dropdown)
- Companding Law:**
 - Switch:** U-Law (selected), A-Law (radio button)
 - Line:** U-Law Line (selected), A-Law Line (radio button)
- Advanced Settings:**
 - ☐ DSS Status
 - ☒ Auto Hold
 - ☒ Dial By Name
 - ☒ Show Account Code
 - ☐ Inhibit Off-Switch Forward/Transfer
 - ☐ Restrict Network Interconnect
 - ☐ Drop External Only Impromptu Conference
 - ☐ Visually Differentiate External Call
 - ☐ Unsupervised Analog Trunk Disconnect Handling
 - ☒ High Quality Conferencing

5.4. Twinning Calling Party Settings

To view or change the System Twinning settings, first navigate to **System** → **Atlantic City** in the Navigation Pane and then navigate to the **Twining** tab in the Details Pane as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank. Click the **OK** button at the bottom of the page (not shown).



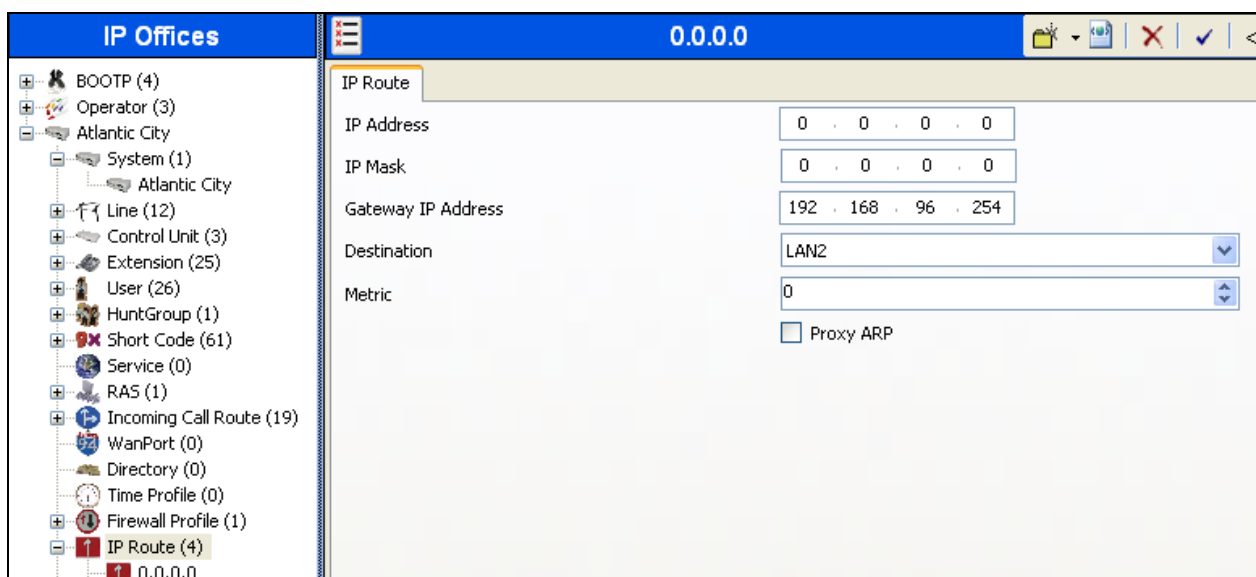
The screenshot shows the 'Atlantic City' configuration window with the 'Twining' tab selected. The 'Send original calling party information for Mobile Twinning' checkbox is unchecked. Below it, the 'Calling party information for Mobile Twinning' field is empty.

5.5. IP Route

Navigate to **IP Route** → **0.0.0.0** in the left Navigation Pane if a default route already exists. Otherwise, to create the default route, right-click on **IP Route** and select **New**. Create/verify a default route with the following parameters:

- Set **IP Address** and **IP Mask** to **0.0.0.0**.
- Set **Gateway IP Address** to the IP Address of the default router to reach Level 3.
- Set **Destination** to **LAN2** from the drop-down list.

Click the **OK** button at the bottom of the page (not shown).



The screenshot shows the 'IP Route' configuration window for the route '0.0.0.0'. The 'IP Address' and 'IP Mask' fields are both set to '0 . 0 . 0 . 0'. The 'Gateway IP Address' field is set to '192 . 168 . 96 . 254'. The 'Destination' dropdown menu is set to 'LAN2'. The 'Metric' field is set to '0'. The 'Proxy ARP' checkbox is unchecked.

5.6. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and Level 3 SIP Trunking. To create a SIP line, right-click **Line** in the Navigation Pane and select **New** → **SIP Line**.

5.6.1. SIP Line – SIP Line Tab

On the **SIP Line** tab in the Details Pane, configure the parameters as shown below.

- Set **ITSP Domain Name** to the IP address of the Level 3 SIP proxy.
- Set **Send Caller ID** to **Diversion Header**. With this setting and the related configuration in **Section 5.4**, IP Office will include the Diversion Header for calls that are directed via Mobile Twinning out the SIP Line to Level 3. It will also include the Diversion Header for calls that are call forwarded out the SIP Line.
- Check **REFER Support**.
- Set **Incoming** and **Outgoing** to **Always**. In the compliance test, this feature was enabled to test transfers of a call between a PSTN phone and an enterprise phone to a second PSTN phone.
- Check the **In Service** box. This makes the trunk available to incoming and outgoing calls.
- Check the **Check OOS** box. IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office will use the **Binding Refresh Time** for LAN2, as shown in **Section 5.2**.
- Default values may be used for all other parameters.

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane with a tree structure including: BOOTP (4), Operator (3), Atlantic City, System (1), Line (12) (selected), Control Unit (3), Extension (25), User (26), HuntGroup (1), Short Code (61), Service (0), RAS (1), Incoming Call Route (19), WanPort (0), Directory (0), Time Profile (0), Firewall Profile (1), IP Route (4), Account Code (0), License (77), Tunnel (0), User Rights (8), ARS (1), RAS Location Request (0), and E911 System (1). The main pane is titled 'SIP Line - Line 20' and contains several tabs: SIP Line, Transport, SIP URI, VoIP, T38 Fax, and SIP Credentials. The 'SIP Line' tab is active, showing the following configuration fields:

- Line Number: 20
- ITSP Domain Name: 192.168.35.86
- In Service: ☒
- Use Tel URI: ☐
- Prefix: (empty)
- Check OOS: ☒
- National Prefix: 0
- Call Routing Method: Request URI
- Country Code: (empty)
- Originator number for forwarded and twinning calls: (empty)
- International Prefix: 00
- Name Priority: System Default
- Caller ID from From header: ☐
- Send From In Clear: ☐
- User-Agent and Server Headers: (empty)
- Send Caller ID: Diversion Header
- Association Method: By Source IP address
- REFER Support: ☒
- Incoming: Always
- Outgoing: Always
- UPDATE Supported: Never

5.6.2. SIP Line - Transport Tab

Select the **Transport** tab. Set the parameters as shown below.

- Set **ITSP Proxy Address** to the IP address of the Level 3 SIP proxy.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to the network port configured in **Section 5.2**.
- Set the **Send Port** to **5070**.
- Default values may be used for all other parameters.

The screenshot shows the 'SIP Line - Line 20' configuration window with the 'Transport' tab selected. The window has a blue title bar and a toolbar with icons for save, delete, add, and navigation. Below the title bar are tabs for 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'Transport' tab is active, showing the following configuration fields:

- ITSP Proxy Address:** 192.168.35.86
- Network Configuration:**
 - Layer 4 Protocol:** UDP (dropdown menu)
 - Send Port:** 5070 (spin box)
 - Use Network Topology Info:** LAN 2 (dropdown menu)
 - Listen Port:** 5060 (spin box)
- Explicit DNS Server(s):** Two groups of IP address fields, each containing '0 . 0 . 0 . 0'.
- Calls Route via Registrar:** ☒
- Separate Registrar:** An empty text field.

5.6.3. SIP Line - SIP Credentials Tab

A SIP Credentials entry must be created for Digest Authentication used by Level 3 SIP Trunking service to authenticate calls from the enterprise to the PSTN. To create a SIP Credentials entry, first select the **SIP Credentials** tab. Click the **Add** button and the **New SIP Credentials** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit** button. In the bottom of the screen, the Edit Channel area will be opened. In the example screen below, a new entry was added. The entry was created with the parameters shown below:

- Set **User name** and **Authentication Name** to the value provided by the service provider.
- Set **Password** to the value provided by the service provider.
- Uncheck the **Registration required** option. Level 3 does not require registration for Digest Authentication.

Click **OK**.

The screenshot shows the 'SIP Line - Line 20*' configuration window with the 'SIP Credentials' tab selected. The window has a toolbar with icons for adding, removing, editing, and navigating. Below the toolbar is a list of existing credentials with columns: Index, UserName, Authentication Name, Contact, Expiry (mins), and Register. To the right of the list are buttons for 'Add...', 'Remove', and 'Edit...'. Below the list is a 'New SIP Credentials' section with the following fields:

User name	1-23Q-3413
Authentication Name	1-23Q-3413
Contact	
Password	*****
Expiry (mins)	60
Registration required	☐

To the right of the 'New SIP Credentials' section are 'OK' and 'Cancel' buttons.

5.6.4. SIP Line - SIP URI Tab

A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. Select the **SIP URI** tab, then click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit** button. In the example screen below, a new entry is created. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact** and **Display Name** to *Use Internal Data*. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any User as shown in **Section 5.8**.
- For **Registration**, select the account credentials previously configured on the line's **SIP Credentials** tab in **Section 5.6.3**.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group **20** was defined that only contains this line (line 20).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

Click **OK**.

The screenshot shows the 'SIP Line - Line 20*' configuration window. The 'SIP URI' tab is selected. The main area contains a table with the following columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. To the right of the table are buttons for 'Add...', 'Remove', and 'Edit...'. Below the table is the 'New Channel' section, which contains the following fields and values:

Field	Value
Via	192.168.96.232
Local URI	Use Internal Data
Contact	Use Internal Data
Display Name	Use Internal Data
PAI	None
Registration	1: 1-23Q-3413
Incoming Group	20
Outgoing Group	20
Max Calls per Channel	10

At the bottom right of the 'New Channel' section are 'OK' and 'Cancel' buttons.

In the sample configuration, the single SIP URI shown above was sufficient to allow incoming calls for Level 3 DID numbers destined for specific IP Office users or IP Office hunt groups. The calls are accepted by IP Office since the incoming number will match the **SIP Name** configured for the user or hunt group that is the destination for the call. For service numbers, such as a DID number routed directly to voicemail, or DID numbers routed to Short Codes, the DID numbers that IP Office should admit can be entered into the **Local URI** and **Contact** fields instead of *Use Internal Data*.

The following shows the SIP URI tab for SIP Line 20 after the SIP URIs corresponding to Voice mail Auto Attendant (720-555-0014) and FNE00 (720-555-0082) has been added.

SIP Line - Line 20									
SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials									
Channel	Groups		Via	Local URI	Contact	Display Name	PAI	Credential	M
1	20	20	192.168.96.232				None	1: 1-23Q-3...	11
2	20	20	192.168.96.232	7205550014	7205550014		None	1: 1-23Q-3...	11
3	20	20	192.168.96.232	7205550082	7205550082		None	1: 1-23Q-3...	11

Add...
Remove
Edit...

5.6.5. SIP Line - VoIP Tab

Select the VoIP tab, to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below.

- For **Codec Selection**, select **System Default** from the pull-down menu. A list of the codecs in their current order of preference is shown on the right in the Selected column. The compliance test used the default codec list. To use a custom list of codecs, select **Custom** for **Codec Selection**. Next, move unwanted codecs from the Selected Column to the Unused column. Lastly, move the codecs up or down the list in the Selected column to achieve the desired order of preference.
- Uncheck the **VoIP Silence Suppression** box.
- Set the **Fax Transport Support** to **T.38**.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Check the **Re-invite Supported** box.
- Default values may be used for all other parameters.

Click the **OK** button at the bottom of the page (not shown).

The screenshot shows the 'SIP Line - Line 20' configuration window with the 'VoIP' tab selected. The window has a blue title bar and a toolbar with icons for help, save, delete, confirm, and navigation. Below the title bar are tabs for 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'VoIP' tab is active, showing the 'Codec Selection' section with a dropdown menu set to 'System Default'. Below this are two columns: 'Unused' (empty) and 'Selected' (containing G.711 ULAW 64K, G.711 ALAW 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MQ). Between the columns are buttons for moving items (>>, <<, up, down) and a 'Reset' button. To the right of the codec columns are four checkboxes: 'VoIP Silence Suppression' (unchecked), 'Re-invite Supported' (checked), 'Use Offerer's Preferred Codec' (unchecked), and 'Codec Lockdown' (unchecked). Below these are two more checkboxes: 'PRACK/100rel Supported' (unchecked). At the bottom, there are three fields: 'Fax Transport Support' set to 'T38', 'Call Initiation Timeout (s)' set to '4', and 'DTMF Support' set to 'RFC2833'.

5.7. Short Codes

Define a short code to route outbound traffic to the SIP line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **9N;**. This short code will be invoked when the user dials 9 followed by any number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N"@192.168.35.86"**. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value **N** represents the number dialed by the user. The IP address 192.168.35.86 is the IP address of the Level 3 SIP proxy.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.6.4**. This short code will use this line group when placing the outbound call.

Click the OK button (not shown).

The screenshot displays the Avaya SIP Office configuration interface. On the left is the 'IP Offices' navigation pane, which includes a tree view with categories like BOOTP (4), Operator (3), Atlantic City, System (1), Line (12), Control Unit (3), Extension (25), User (26), HuntGroup (1), and Short Code (61). The 'Short Code (61)' category is expanded, showing sub-items *00, *01, and *02. The main pane on the right is titled '9N;; Dial' and contains a 'Short Code' tab. This tab has several configuration fields: 'Code' is set to '9N;;', 'Feature' is set to 'Dial' (via a dropdown), 'Telephone Number' is set to 'N"@192.168.35.86"', 'Line Group ID' is set to '20' (via a dropdown), 'Locale' is set to an empty dropdown, and 'Force Account Code' is an unchecked checkbox.

Optionally, add or edit a short code that can be used to access the SIP Line anonymously. In the screen shown below, the short code ***67N;** is illustrated. This short code is similar to the **9N;** short code except that the **Telephone Number** field begins with the letter **W**, which means “withhold the outgoing calling line identification”. In the case of the SIP Line to Level 3 documented in these Application Notes, when a user dials *67 plus the number, IP Office will include the user’s telephone number in the P-Asserted-Identity (PAI) header along with “Privacy: Id”. Level 3 will allow the call due to the presence of a valid DID in the PAI header, but will prevent presentation of the caller id to the called PSTN destination.

The screenshot shows a configuration window titled ***67N;: Dial**. It contains the following fields:

- Code:** *67N;
- Feature:** Dial (selected from a dropdown menu)
- Telephone Number:** WN"@192.168.35.86"
- Line Group ID:** 20 (selected from a dropdown menu)
- Locale:** (empty dropdown menu)
- Force Account Code:** ☐

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 FNE00** is defined for **Feature FNE Service** to **Telephone Number 00**. This short code will be used as means to allow a Level 3 DID to be programmed to route directly to this feature, via inclusion of this short code as the destination of an Incoming Call Route. See **Section 5.9**. This feature is used to provide dial tone to twinned mobile devices (e.g. cell phone) directly from IP Office; once dial tone is received the user can perform dialing actions including making calls and activating Short Codes.

The screenshot shows a configuration window titled **FNE00: FNE Service**. It contains the following fields:

- Code:** FNE00
- Feature:** FNE Service (selected from a dropdown menu)
- Telephone Number:** 00
- Line Group Id:** 0 (selected from a dropdown menu)
- Locale:** (empty dropdown menu)
- Force Account Code:** ☐

5.8. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.6**. To configure these settings, first navigate to **User** → *Name* in the Navigation Pane where *Name* is the name of the user to be modified. In the example below, the name of the user is **Extn243**. Select the **SIP** tab in the Details Pane. The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From and Contact headers for outgoing SIP trunk calls and allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.6.4**). The example below shows the settings for User **Extn243**. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from Level 3. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network. Click the **OK** button (not shown).

The screenshot shows a web-based configuration interface for a user named "Extn243: 243". The interface has a blue header bar with the user name and a search icon. Below the header is a row of tabs: User, Voicemail, DND, ShortCodes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, and Bu. The "SIP" tab is selected. Below the tabs are three text input fields: "SIP Name" with the value "7205550023", "SIP Display Name (Alias)" with the value "Extn243", and "Contact" with the value "7205550023". At the bottom, there is an unchecked checkbox labeled "Anonymous".

The following screen shows the **Mobility** tab for User Extn243. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case **917325551234**. Other options can be set according to customer requirements.

The screenshot displays the 'Extn243: 243' configuration window. The 'Mobility' tab is selected, showing various settings. The 'Internal Twinning' section is collapsed. The 'Mobility Features' section is expanded, and the 'Mobile Twinning' checkbox is checked. Below this, the 'Twinned Mobile Number (including dial access code)' is set to '917325551234'. Other settings include 'Twinned Time Profile' set to '<None>', 'Mobile Dial Delay (secs)' set to '2', and 'Mobile Answer Guard (secs)' set to '0'. Several other options like 'Twin Bridge Appearances', 'Twin Coverage Appearances', 'Twin Line Appearances', 'Hunt group calls eligible for mobile twinning', 'Forwarded calls eligible for mobile twinning', 'Twin When Logged Out', 'one-X Mobile Client', 'Mobile Call Control', and 'Mobile Callback' are also visible, with 'Mobile Call Control' being checked.

Section	Option	Value
Internal Twinning	Internal Twinning	☐
	Twinned Handset	<None>
	Maximum Number of Calls	1
	Twin Bridge Appearances	☐
Mobility Features	Mobility Features	☒
	Mobile Twinning	☒
	Twinned Mobile Number (including dial access code)	917325551234
	Twinned Time Profile	<None>
	Mobile Dial Delay (secs)	2
	Mobile Answer Guard (secs)	0
	Hunt group calls eligible for mobile twinning	☐
	Forwarded calls eligible for mobile twinning	☐
	Twin When Logged Out	☐
	one-X Mobile Client	☐
Mobile Call Control	☒	
Mobile Callback	☐	

5.9. Incoming Call Route

An incoming call route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by the service provider. To create an incoming call route, right-click **Incoming Call Routes** in the Navigation Pane and select **New**. On the **Standard** tab of the Details Pane, enter the parameters as shown below.

- Set the **Bearer Capacity** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.6.4**.
- Set the **Incoming Number** to the incoming number on which this route should match.
- Default values can be used for all other fields.

The screenshot shows the 'Incoming Call Route' configuration window for the number '20 7205550023'. The 'Standard' tab is active, showing the following fields:

Field	Value
Bearer Capacity	Any Voice
Line Group ID	20
Incoming Number	7205550023
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

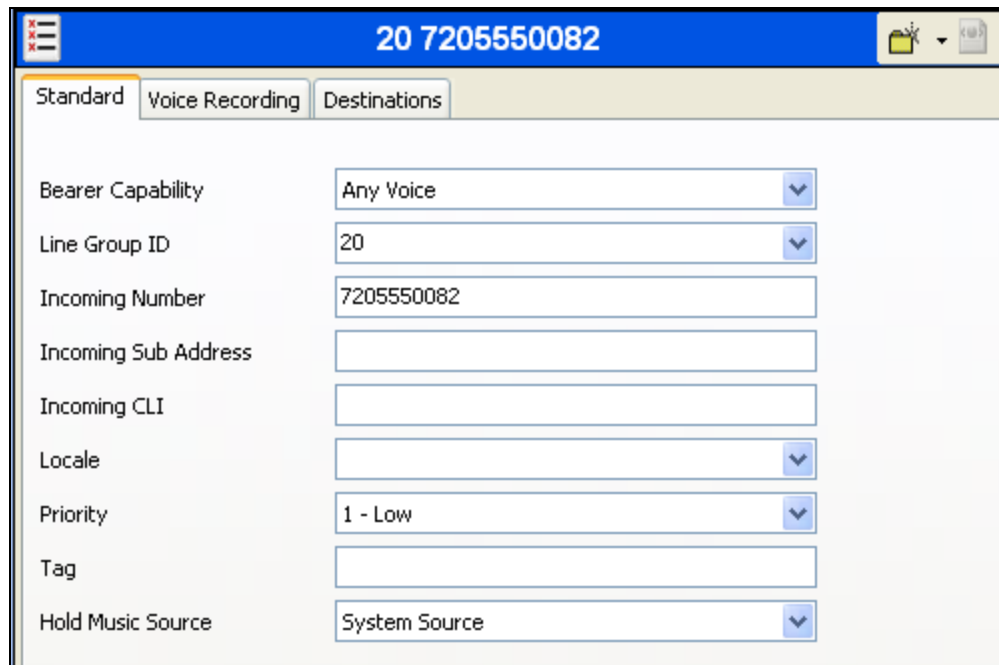
On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 7205550023 on line 20 are routed to extension 243.

The screenshot shows the 'Destinations' tab of the 'Incoming Call Route' configuration window. It displays a table with the following data:

TimeProfile	Destination	Fallback Extension
Default Value	243 Extn243	

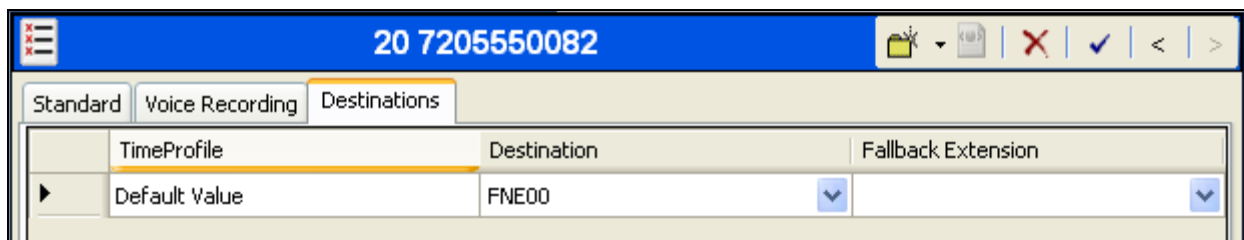
Incoming Call Routes for other direct mappings of DID numbers to IP Office users listed in **Figure 1** are omitted here, but can be configured in the same fashion.

In the screen shown below, the incoming call route for **Incoming Number 7205550082** is illustrated. The **Line Group Id** is **20**, matching the Incoming Group field configured in the SIP URI tab in **Section 5.6.4**.



20 7205550082	
Standard Voice Recording Destinations	
Bearer Capability	Any Voice
Line Group ID	20
Incoming Number	7205550082
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

When configuring an Incoming Call Route, the **Destination** field can be manually configured with a number such as a short code, or certain keywords available from the drop-down list. For example, the following **Destinations** tab for an incoming call route contains the **Destination FNE00** entered manually. **FNE00** is the short code for **FNE Service**, as shown in **Section 5.7**. An incoming call to 720-555-0082 will be delivered directly to internal dial tone from the IP Office, allowing the caller to perform dialing actions including making calls and activating Short Codes. The incoming caller ID must match the Twinned Mobile Number entered in the User Mobility tab (**Section 5.8**); otherwise, the IP Office responds with a 486 Busy Here and the caller will hear a busy tone.

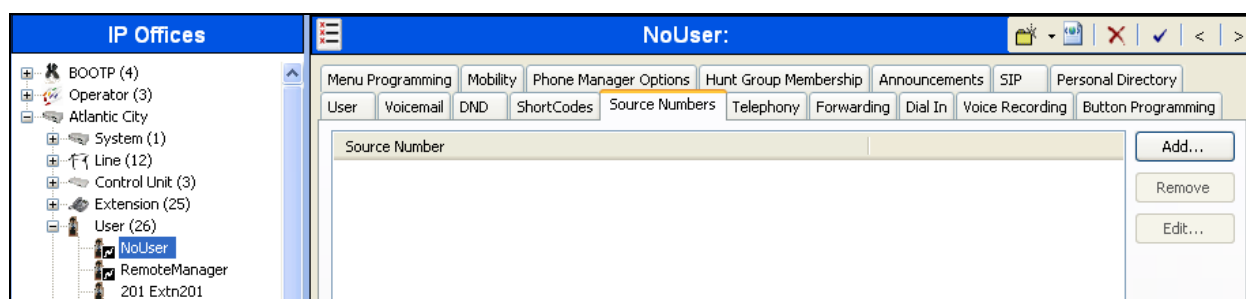


20 7205550082			
Standard Voice Recording Destinations			
	TimeProfile	Destination	Fallback Extension
▶	Default Value	FNE00	

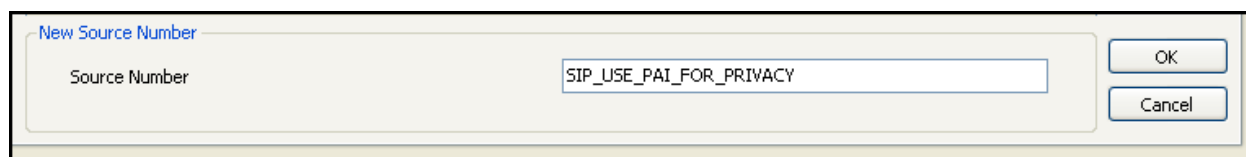
5.10. Privacy/Anonymous Calls

For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with “restricted” and “anonymous” respectively. Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing. By default, Avaya IP Office will use PPI for privacy. For the compliance test, PAI was used for the purposes of privacy.

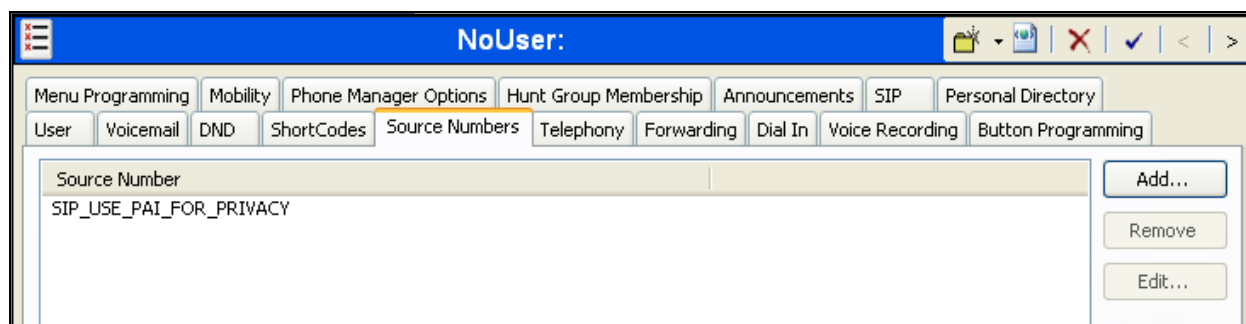
To configure Avaya IP Office to use PAI for privacy calls, navigate to **User** → **NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the Details Pane, the **Source Number** field will appear. Enter **SIP_USE_PAI_FOR_PRIVACY**. Click **OK**.



The **SIP_USE_PAI_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below. Click **OK** at the bottom of the screen (not shown).



5.11. SIP Options Frequency

Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. Avaya IP Office will send a SIP OPTIONS message only if an OPTIONS request is not received from the far-end during the defined interval. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.2** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **NoUser** user. The OPTIONS period is determined in the following manner:

- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 300 seconds is used.
- To establish a period less than 300 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 seconds. The OPTIONS message period will be equal to the **Binding Refresh Time**.
- To establish a period greater than 300 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 300 seconds. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

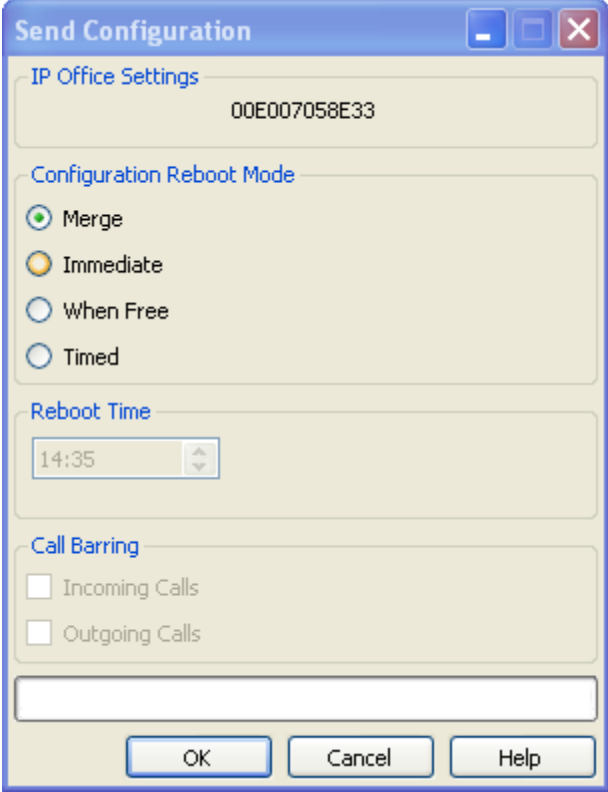
For the compliance test, a SIP OPTIONS frequency of 30 seconds was defined by setting the **Binding Refresh Time** to **30** seconds in **Section 5.2** and not defining a **SIP_OPTIONS_PERIOD** parameter.

If a **SIP_OPTIONS_PERIOD** parameter is needed to define a longer interval, the required NoUser Source Number can be defined in the same manner as the **SIP_USE_PAI_FOR_PRIVACY** parameter in **Section 5.10**.

5.12. Save Configuration

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

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



The image shows a Windows-style dialog box titled "Send Configuration". It has a standard title bar with minimize, maximize, and close buttons. The dialog is divided into several sections:

- IP Office Settings:** A text field containing the value "00E007058E33".
- Configuration Reboot Mode:** A group box containing four radio buttons: "Merge" (selected), "Immediate", "When Free", and "Timed".
- Reboot Time:** A time selection control showing "14:35" with up and down arrows.
- Call Barring:** A group box containing two checkboxes: "Incoming Calls" and "Outgoing Calls", both of which are currently unchecked.

At the bottom of the dialog is an empty text input field and three buttons: "OK", "Cancel", and "Help".

6. Level 3 SIP Trunking Configuration

Level 3 is responsible for the configuration of Level 3 SIP Trunking. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. Level 3 will provide the customer the necessary information to configure the Avaya IP Office SIP connection to Level 3 including:

- IP address of the Level 3 SIP proxy
- Supported codecs
- DID numbers
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices
- Username and Password for Digest Authentication.

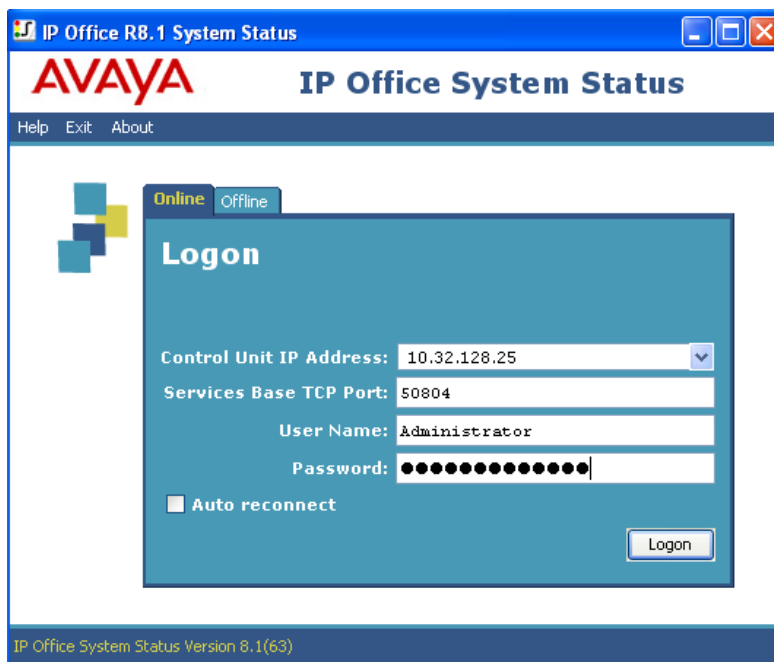
7. Verification Steps

This section provides verification steps that may be performed in the field to verify that the solution is configured properly.

7.1. System Status

The System Status application is used to monitor and troubleshoot IP Office. Use the System Status application to verify the state of the SIP trunk. System Status can be accessed from **Start** → **Programs** → **IP Office** → **System Status**.

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



The screenshot displays the 'IP Office System Status' application window. The title bar reads 'IP Office R8.1 System Status'. The window features the Avaya logo and the text 'IP Office System Status'. Below this is a menu bar with 'Help', 'Exit', and 'About'. The main content area has a status indicator showing 'Online' in yellow and 'Offline' in blue. A 'Logon' dialog box is centered, containing the following fields: 'Control Unit IP Address' (with a dropdown menu showing '10.32.128.25'), 'Services Base TCP Port' (with the value '50804'), 'User Name' (with the value 'Administrator'), and 'Password' (masked with dots). There is an 'Auto reconnect' checkbox and a 'Logon' button at the bottom right of the dialog. The footer of the application window states 'IP Office System Status Version 8.1(63)'.

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

AVAYA IP Office System Status

Help Snapshot LogOff Exit About

System
Alarms (4)
Extensions (17)
Trunks (12)
 Lines: 1 - 4
 Lines: 5 - 8
 Line: 17
 Line: 18
 Line: 19
 ▶ Line: 20
 Active Calls
 Resources
 Voicemail
 IP Networking

Status Utilization Summary Alarms Registration

SIP Trunk Summary

Peer Domain Name: 192.168.35.86
 Resolved Address: 192.168.35.86
 Line Number: 20
 Number of Administered Channels: 20
 Number of Channels in Use: 0
 Administered Compression: G711 Mu, G711 A, G729 A, G7231
 Silence Suppression: Off
 SIP Trunk Channel Licenses: Unlimited 0%
 SIP Trunk Channel Licenses in Use: 0
 SIP Device Features: REFER (Incoming and Outgoing)

Channel Number	URI G... Ref	Current State	Time in State	Remote Media A...	Co...	Conne...	Caller ID or Dial...	Other Party on Call	Direction of Call	Round Trip D...	Receive Jitter	Receive Packe...	Transmit Jitter	Transmit Packe...
1		Idle	02:35:53											
2		Idle	02:35:53											
3		Idle	02:35:53											
4		Idle	02:35:53											
5		Idle	02:35:53											

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

Status Utilization Summary **Alarms** Registration

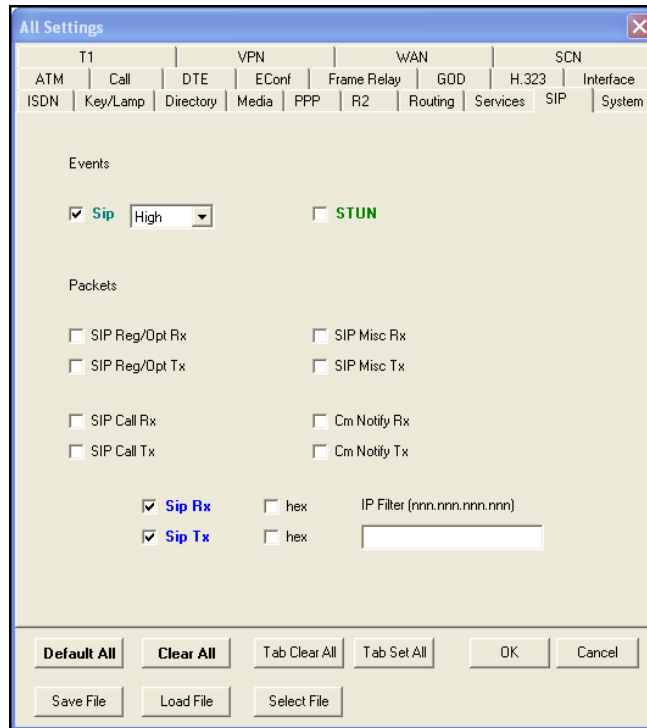
Alarms for Line: 20 SIP 192.168.35.86

Last Date Of Error	Occurrences	Error Description
--------------------	-------------	-------------------

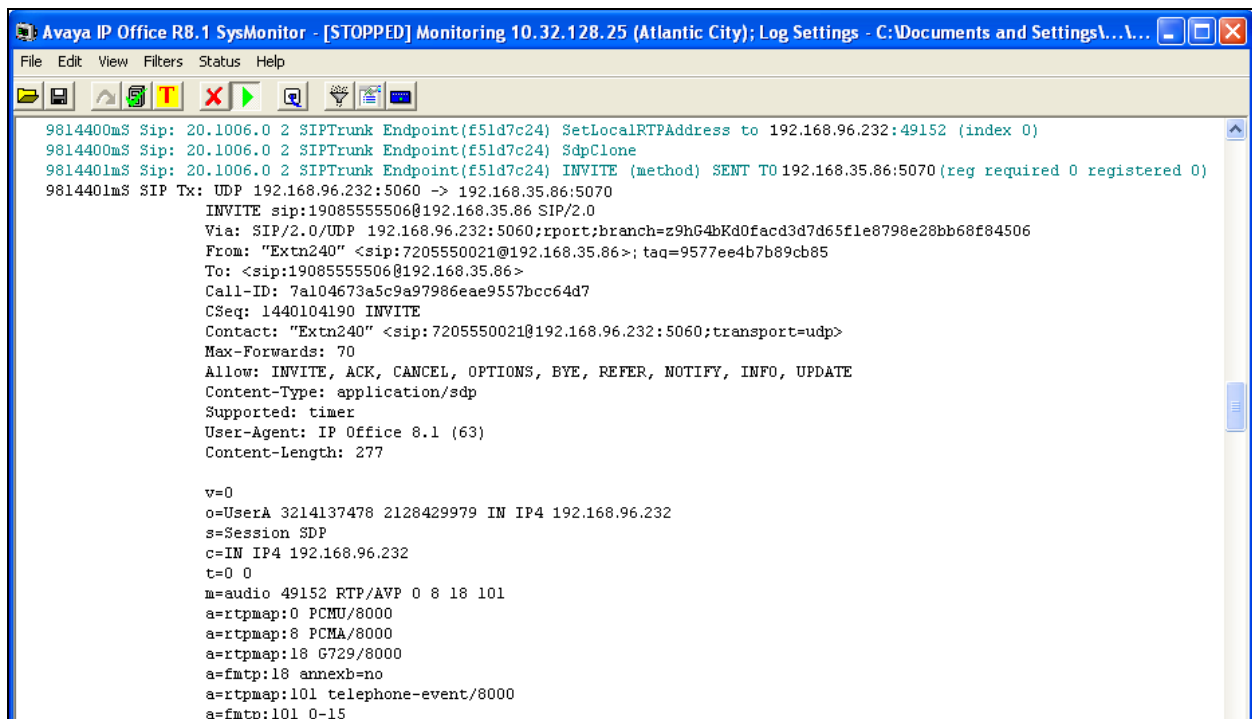
7.2. Monitor

The Monitor application can also be used to monitor and troubleshoot IP Office. Monitor can be accessed from **Start** → **Programs** → **IP Office** → **Monitor**. The application allows the monitored information to be customized. To customize, select **Filters** → **Trace Options**.

The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked.



As an example, the following shows a portion of the monitoring window for an outbound call from extension 240, whose DID is 720-555-0021, calling out to the PSTN via the Level 3 IP Trunking Service. The telephone user dialed 9-1-908-555-5506.



8. Conclusion

These Application Notes describe the configuration necessary to connect Avaya IP Office 8.1 to Level 3 SIP Trunking service. Level 3 SIP Trunking is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. It provides a flexible, cost-saving alternative to traditional hardwired telephony trunks. Level 3 SIP Trunking passed compliance testing. Please refer to **Section 2.2** for any exceptions.

9. Additional References

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

[1] IP Office 8.1 IP500/IP500 V2 Installation, Document Number 15-601042, Issue 27f, March 04, 2013

[2] IP Office Release 8.1 Manager 10.1, Document Number 15-601011, Issue 29t, February 20, 2013

[3] IP Office System Status Application, Document Number 15-601758 Issue 07a, November 26, 2012

[4] IP Office Release 8.1 Administering Voicemail Pro, Document Number 15-601063, Issue 8b, December 11, 2012

[5] IP Office System Monitor, Document Number 15-601019, Issue 03b, February 28, 2013

Additional IP Office documentation can be found at:
<http://marketingtools.avaya.com/knowledgebase/>

10. Appendix A: SIP Line Template

Avaya IP Office supports a SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

Note that not all of the configuration information, particularly items relevant to a specific installation environment, is included in the SIP Line Template. Therefore, it is critical that the SIP Line configuration be verified/updated after a template has been imported and additional configuration be supplemented using **Section 5.6** in these Application Notes as a reference.

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

```
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20130307</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>Level3</DescriptiveName>
  <ITSPDomainName>192.168.35.86</ITSPDomainName>
  <SendCallerID>CallerIDDIV</SendCallerID>
  <ReferSupport>true</ReferSupport>
  <ReferSupportIncoming>1</ReferSupportIncoming>
  <ReferSupportOutgoing>1</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <UpdateSupport>UpdateNever</UpdateSupport>
  <UserAgentServerHeader />
  <CallerIDfromFromheader>false</CallerIDfromFromheader>
  <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
  <ITSPProxy>192.168.35.86</ITSPProxy>
  <LayerFourProtocol>SipUDP</LayerFourProtocol>
  <SendPort>5070</SendPort>
  <ListenPort>5060</ListenPort>
  <DNSServerOne>0.0.0.0</DNSServerOne>
  <DNSServerTwo>0.0.0.0</DNSServerTwo>
  <CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
  <SeparateRegistrar />
  <CompressionMode>AUTOSELECT</CompressionMode>
  <UseAdvVoiceCodecPrefs>false</UseAdvVoiceCodecPrefs>
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
  <ReinviteSupported>true</ReinviteSupported>
  <FaxTransportSupport>FOIP_T38</FaxTransportSupport>
  <UseOffererPreferredCodec>false</UseOffererPreferredCodec>
```

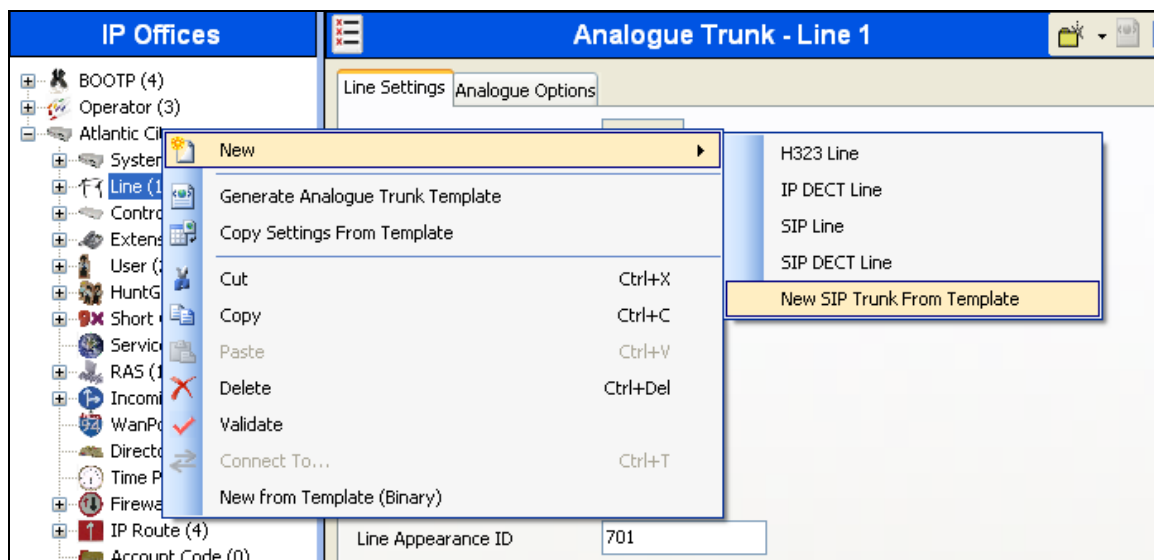
```

<CodecLockdown>false</CodecLockdown>
<Rel100Supported>false</Rel100Supported>
<T38FaxVersion>3</T38FaxVersion>
<Transport>UDPTL</Transport>
<LowSpeed>0</LowSpeed>
<HighSpeed>0</HighSpeed>
<TCFMethod>Trans_TCF</TCFMethod>
<MaxBitRate>FaxRate_14400</MaxBitRate>
<EflagStartTimer>2600</EflagStartTimer>
<EflagStopTimer>2300</EflagStopTimer>
<UseDefaultValues>true</UseDefaultValues>
<ScanLineFixup>true</ScanLineFixup>
<TFOPENenhancement>true</TFOPENenhancement>
<DisableT30ECM>false</DisableT30ECM>
<DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
<DisableT30MRCompression>false</DisableT30MRCompression>
<NSFOVERRIDE>false</NSFOVERRIDE>
<SIPCredentials>
  <Expiry>60</Expiry>
  <RegistrationRequired>false</RegistrationRequired>
</SIPCredentials>
</Template>

```

To import the above template into a new installation:

1. On the PC where IP Office Manager was installed, copy and paste the above template into a text document named **US_Level3_SIPTrunk.xml**. Move the .xml file to the IP Office Manager template directory (C:\Program Files\Avaya\IP Office\Manager\Templates). It may be necessary to create this directory.
2. Import the template into an IP Office installation by creating a new SIP Line as shown in the screenshot below. In the Navigation Pane on the left, right-click on **Line** then navigate to **New → New SIP Trunk From Template**:



3. Verify that **United States** is automatically populated for **Country** and **Level3** is automatically populated for **Service Provider** in the resulting Template Type Selection screen as shown below. Click **Create new SIP Trunk** to finish the importing process.



Template Type Selection

Locale: United States (US English)

Country: United States

Service Provider: Level3 ☐ Display All

Create new SIP Trunk Cancel

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