Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Accelerated Connections SIP Trunking Service 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 Accelerated Connections and Avaya IP Office Release 8.1.

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

Accelerated Connections is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

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

The Accelerated Connections 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.

The Accelerated Connections SIP Trunking Service 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).

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the Accelerated Connections 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 Accelerated Connections SIP Trunking 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:

- Sending SIP OPTIONS queries to the provider
- Incoming PSTN calls to SIP and H.323 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 SIP and H.323 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 Video Softphone and Avaya Flare® Experience for Windows were tested.
- Various call types including: local, long distance, outbound toll-free, international 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 input for inbound and outbound calls.
- User features such as hold and resume, transfer, and conference.
- Off-net call forwarding and twinning.

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Inbound toll-free calls and 911 emergency calls are supported but were not tested as part of the compliance test.

The following items are not supported:

- Operator (dial 0) and operator-assisted (dial 0 + 10 digits) services
- REFER method for call redirection

2.2. Test Results

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

- **Calling Party Number on inbound calls**: On inbound calls, sometimes the last digit of the inbound calling party 11-digit number is truncated in the SIP From header. This leads to an incorrect calling party number of 1 + 9 digits being displayed at the enterprise. Upon investigation, Accelerated Connections determined that the incorrect number was being received from the PSTN carrier. A ticket was raised with the PSTN carrier.
- **Calling Party Number on outbound twinned calls**: Calls to IP Office extensions with Twinning enabled, result in a new call being forked to another destination, (which is typically on the PSTN), allowing both endpoints to ring simultaneously. The calling party number on the forked call should show the original caller to the IP Office extension as the calling party. However, sometimes no calling party number is displayed. Upon investigation, it was determined that the correct calling party number was passed from IP Office to Accelerated Connections to the PSTN carrier. A ticket was raised with the PSTN carrier.
- **SIP OPTIONS**: During the compliance test, Accelerated Connections did not send SIP OPTIONS to the enterprise. Sending of OPTIONS is not required and is listed here simply as an observation. OPTIONS were sent in the opposite direction.
- **Codec not locked down on outbound calls**: On outbound calls, Accelerated Connections responds to the INVITE request with multiple codecs, instead of selecting one from the INVITE SDP list. IP Office uses the first compatible codec in the list. This behavior has no user impact. Calls were successful.
- **Call Forwarding calling party number**: Inbound PSTN calls to an IP Office extension, which is call forwarded to another PSTN endpoint, displays the incorrect calling party number. The forwarding extension is displayed as the calling party instead of the original PSTN caller. This is an IP Office issue which is corrected in release 8.1(69).
- **Blind transfer with Avaya 1100 IP Deskphones calling party number**: An Avaya 1100 Series IP Deskphone (SIP) places an outbound call to the PSTN, then performs a blind transfer of this call to another PSTN endpoint. In this scenario, the wrong calling party number is displayed on the second PSTN endpoint. The display shows the number dialed by the Avaya 1100 Series IP Deskphone for the initial outbound call including the

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short code (9) + 1 + 10 digits. The behavior is expected to be the same as with IP Office H.323 endpoints which is to display the transferring party number. This is an IP Office issue that was fixed and tested in a development release. The fix will be available in a future IP Office generally available (GA) release.

- **T.38 Fax**: If IP Office is configured to use T.38/Fallback for fax, outbound fax calls use T.38 but on inbound calls IP Office never attempts to negotiate use of T.38. Instead, IP Office always uses G.711 fax. To ensure use of T.38 fax for both inbound and outbound calls, IP Office must be set to use T.38 only. This is under investigation by the IP Office development team.

2.3. Support

For technical support on the Accelerated Connections SIP Trunking Service, contact Accelerated Connections using the Support links at www.acceleratedconnections.com or by calling 1-866-814-1635.

3. Reference Configuration

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

Located at the enterprise site is an Avaya IP Office 500 V2 with various endpoints. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public network. The site also has Avaya Preferred Edition for voicemail and Avaya IP Office Manager to configure the Avaya IP Office running on a single Windows 2003 Server.

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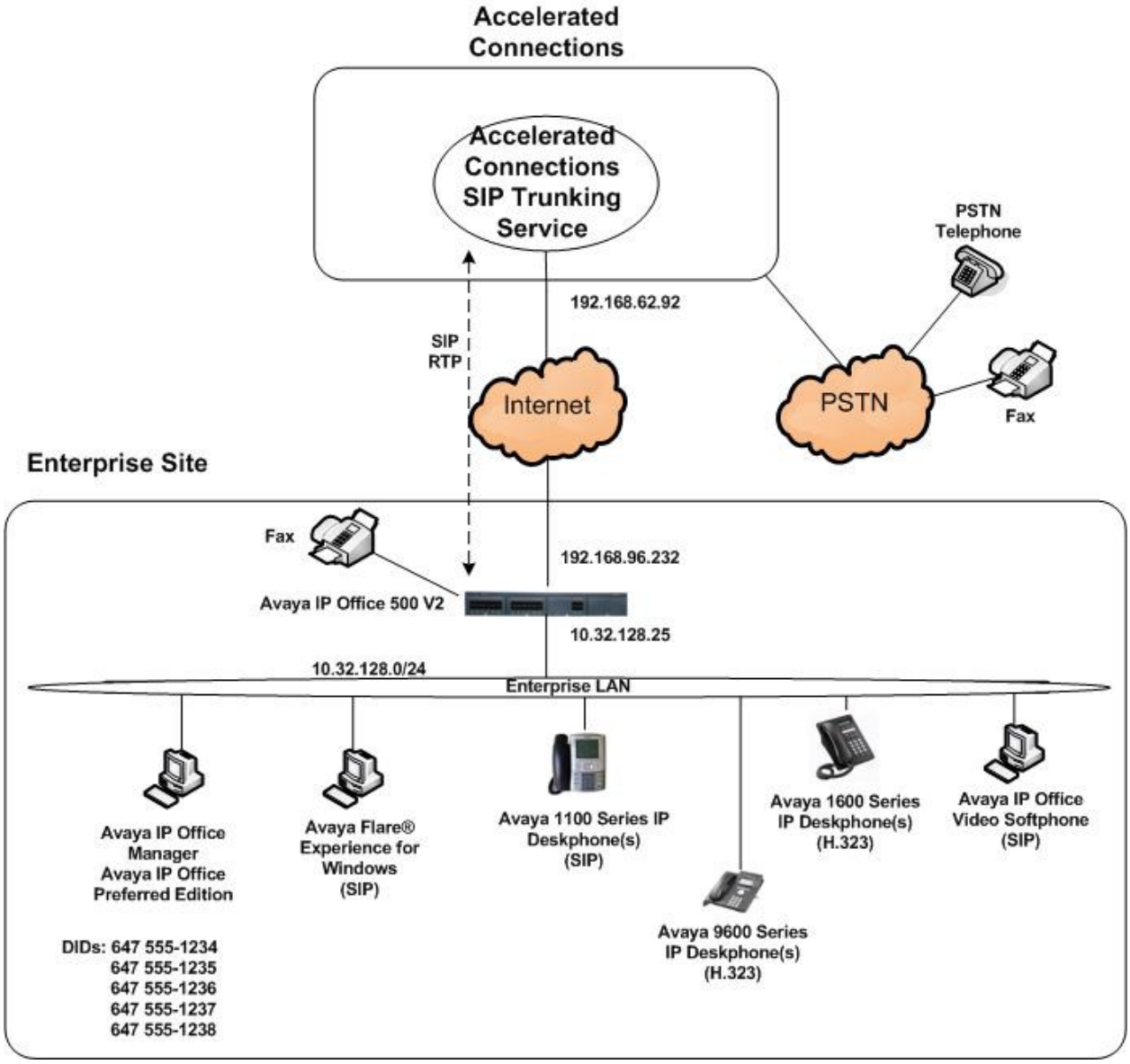

Figure 1: Avaya Interoperability Test Lab Configuration

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 Accelerated Connections. The short code of 9 and any preceding 1 is stripped off by Avaya IP Office and the remaining 10 digits were sent unaltered to Accelerated Connections. For outbound calls, Avaya IP Office sent 10 digits in all headers. For inbound calls, the Accelerated Connections SIP Trunking Service sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

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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

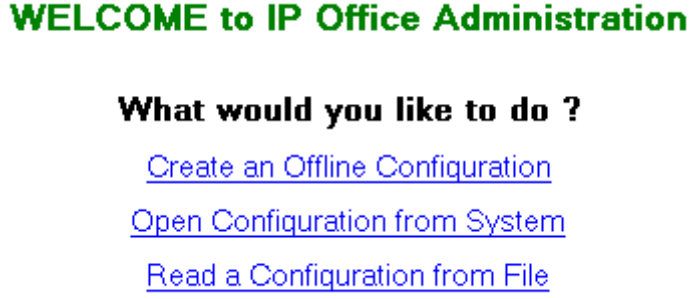
The following equipment and software were used for the sample configuration provided:

Avaya Telephony Components	
Equipment	**Software**
Avaya IP Office 500 v2	8.1 (67)
Avaya IP Office Manager	10.1 (67)
Avaya IP Office Preferred Edition (Voicemail)	8.1 (Build 8.1.9102.0)
Avaya 1140E IP Deskphone (SIP)	4.3 SP1 (04.03.12.00)
Avaya 1608 IP Deskphone (H.323) running Avaya one-X® Deskphone Value Edition	1.3 SP2 (1.3.2)
Avaya 9641G IP Deskphone (H.323) running Avaya one-X® Deskphone Edition	6.2 SP2 (6.2209)
Avaya IP Office Video Softphone (SIP)	3.2.3.48 (67009)
Avaya Flare® Experience for Windows	1.1.1.7

Accelerated Connections Components	
Equipment	**Software**
Genband S3 Session Border Controller	7.1.13.0

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:



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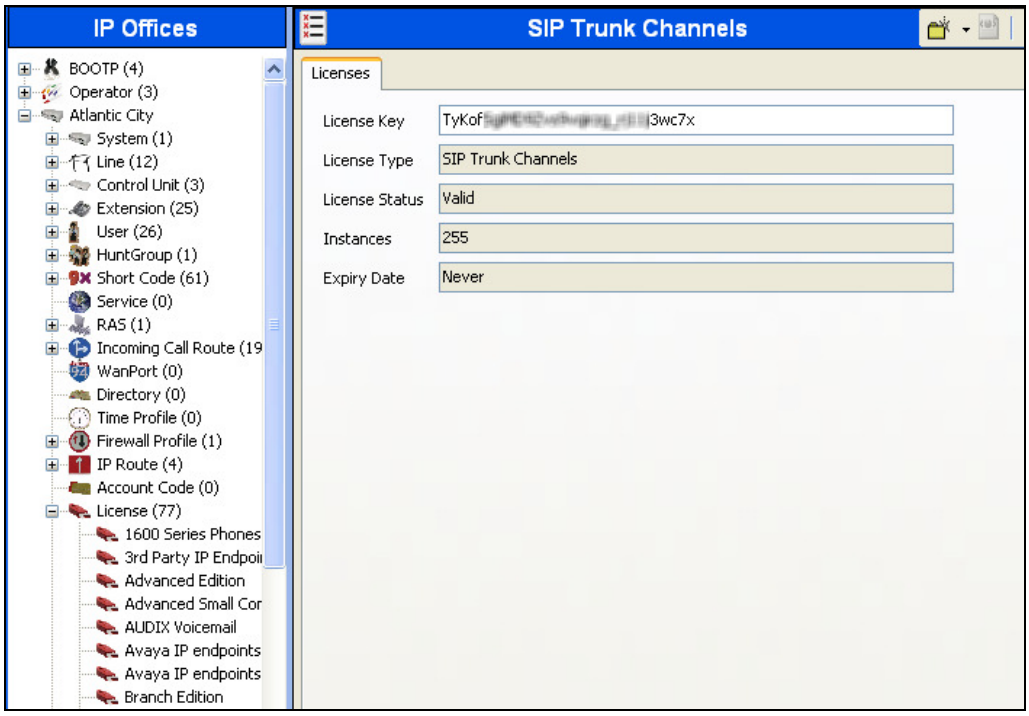
Select **Open Configuration from System**. If the above screen does not appear, the configuration may be alternatively opened by navigating to **File → Open Configuration** at the top of the Avaya IP Office Manager window. Select the proper Avaya IP Office system from the pop-up window and log in with the appropriate credentials. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this document, the **View** menu was configured to show the Navigation pane on the left side, omit the Group pane in the center, and show 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 Video Softphone support) is assumed to already be in place.

In the sample configuration, **Atlantic City** was used as the system name. All navigation described in the following sections (e.g., **License → SIP Trunk Channels)** appears as submenus underneath the system name **Atlantic City** in the Navigation Pane.

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.



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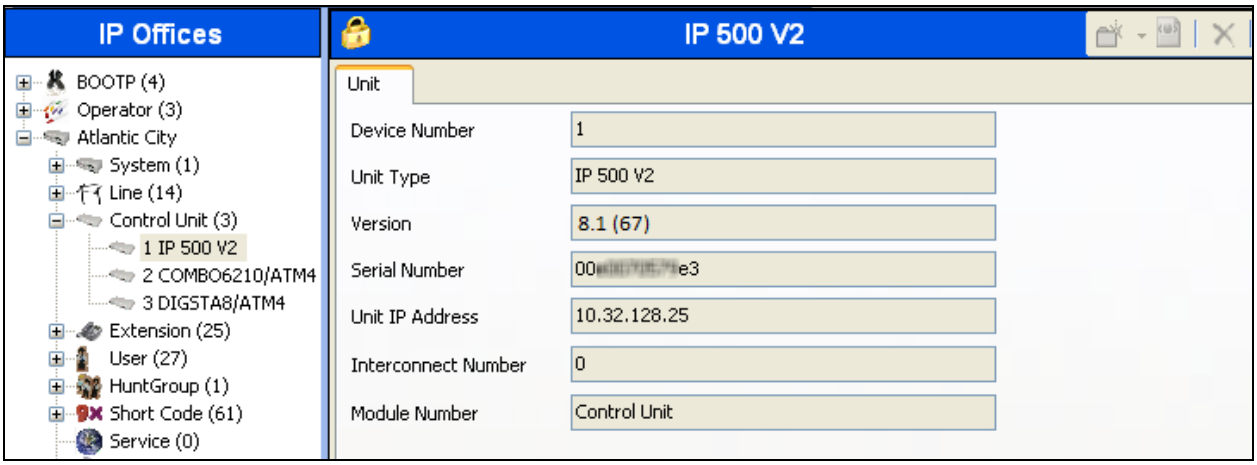
To view the physical hardware comprising the IP Office system, expand the components under the **Control Unit** in the Navigation pane. In the sample configuration, the second component listed is a Combination Card. This module has 6 digital stations ports, two analog extension ports, 4 analog trunk 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**.



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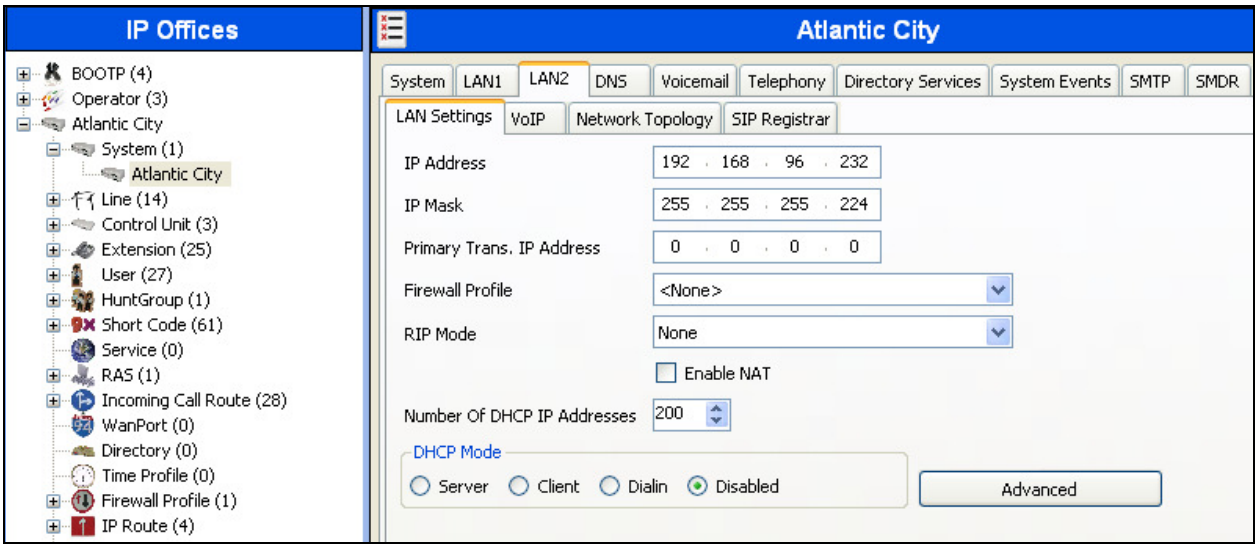
5.2. System

Configure the necessary system settings.

5.2.1. System - LAN2 Tab

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.



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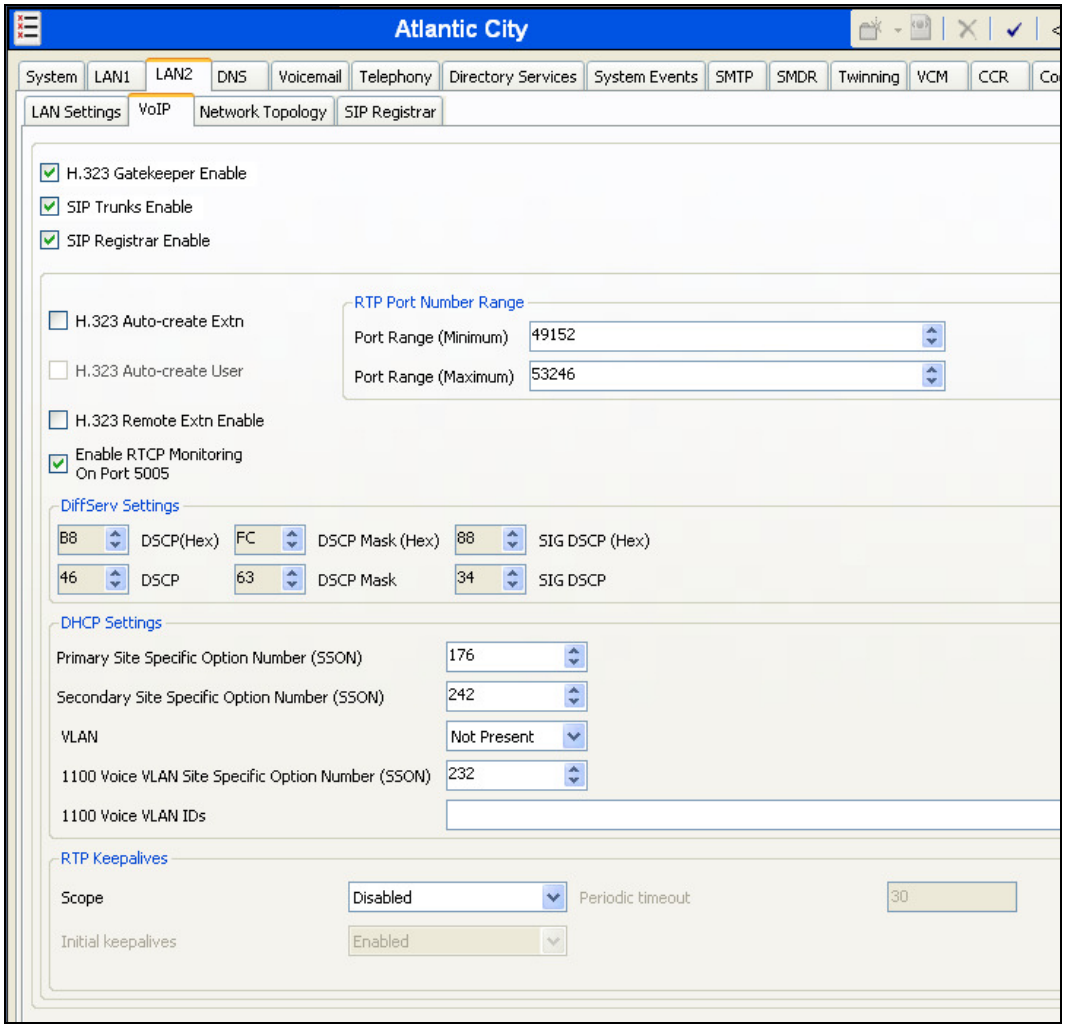
On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP 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 Accelerated Connections. All other parameters should be set according to customer requirements.



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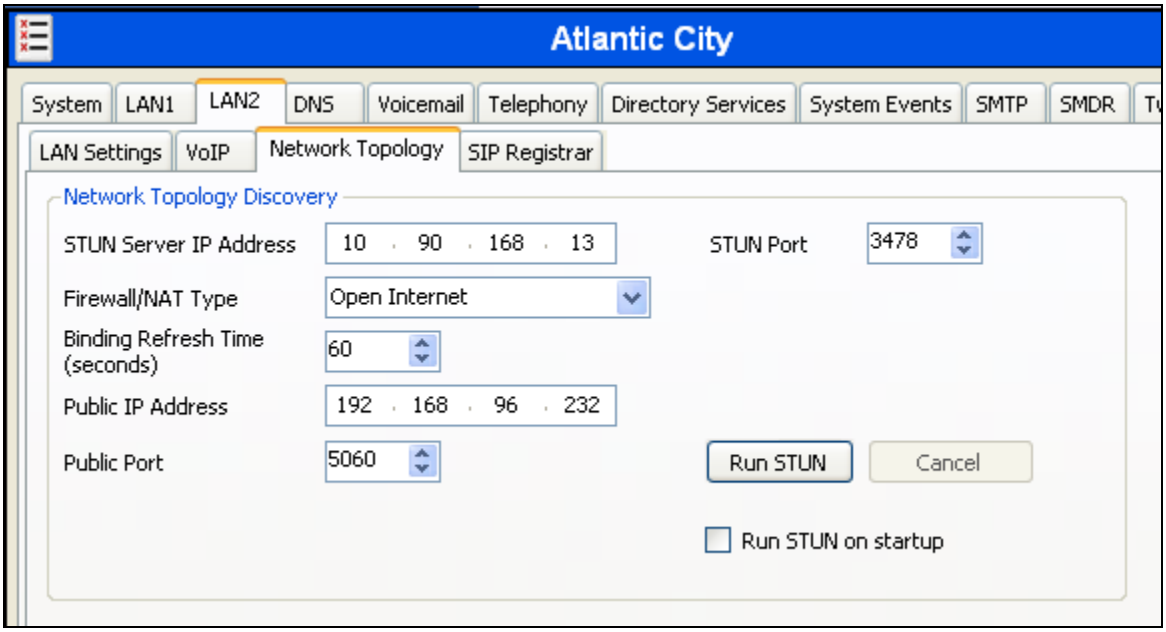
On the **Network Topology** tab in the Details Pane, configure the following parameters:

- 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 **60**. 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.



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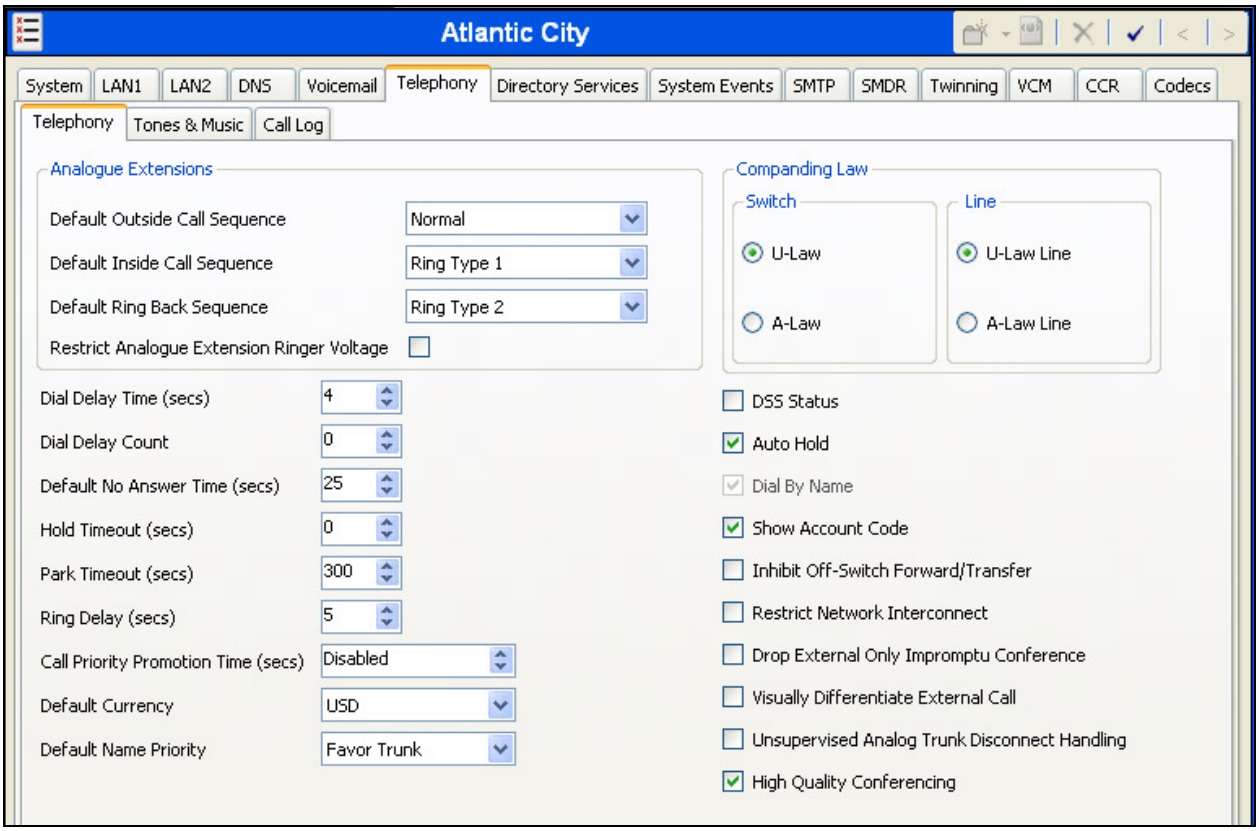
5.2.2. System - Telephony Tab

To access the System Telephony settings, 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. If for security reasons incoming calls should not be allowed to transfer back to the PSTN then leave this setting checked.



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5.2.3. System - Twinning Tab

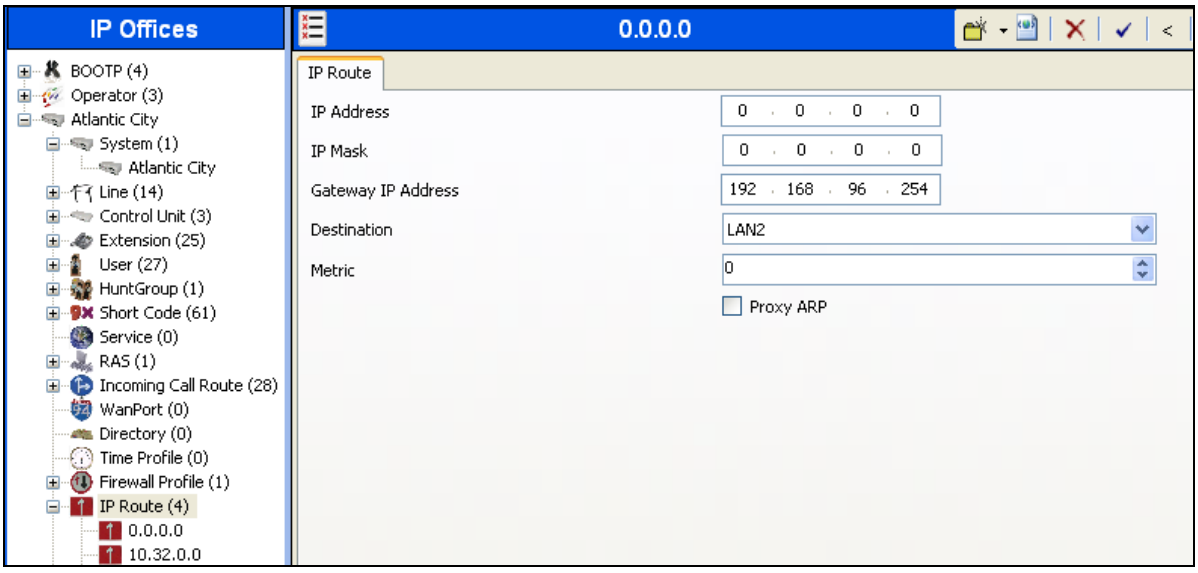
To view or change the System Twinning settings, navigate to the **Twinning** tab in the Details Pane as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning**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

5.3. 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 Accelerated Connections.
- Set **Destination** to **LAN2** from the drop-down list.

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



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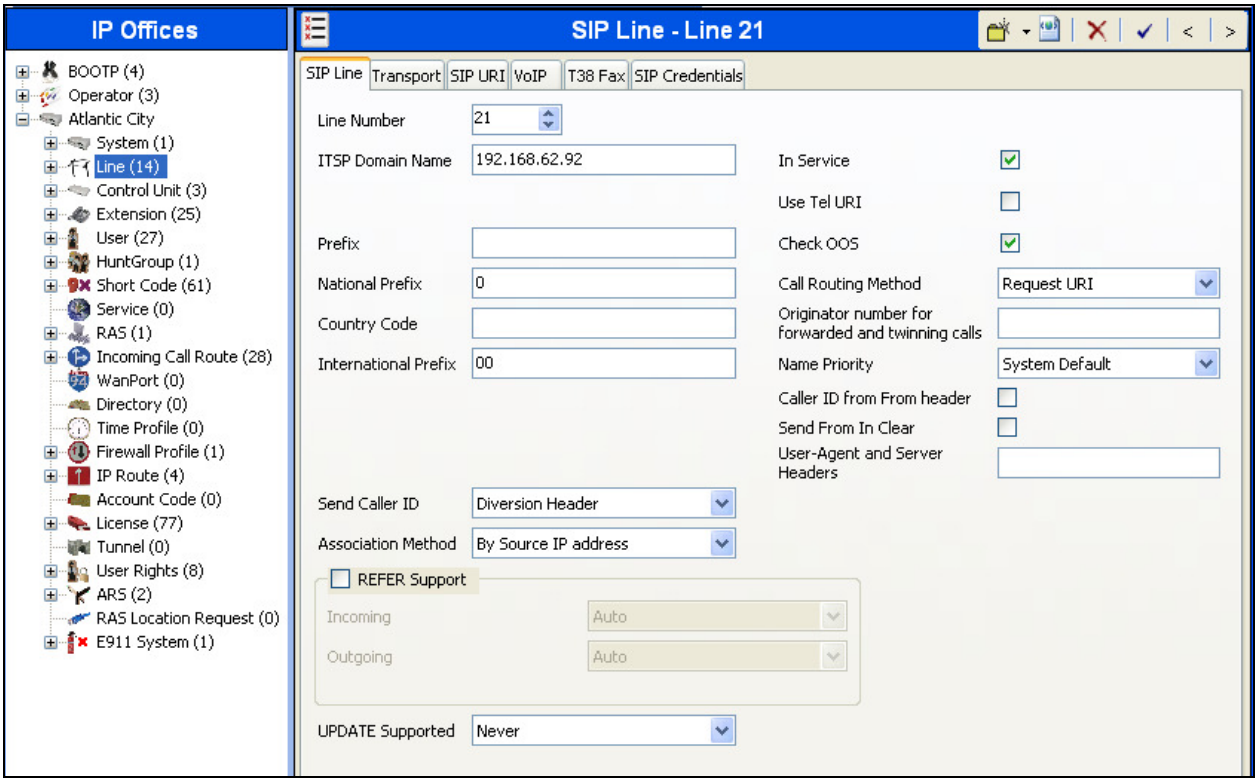
5.4. SIP Line

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

5.4.1. 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 Accelerated Connections SIP proxy.
- Set **Send Caller ID** to **Diversion Header**. With this setting and the related configuration in **Section 5.2.3**, IP Office will include the Diversion Header for calls that are directed via Mobile Twinning out the SIP Line to Accelerated Connections. It will also include the Diversion Header for calls that are call forwarded out the SIP Line.
- Uncheck **REFER Support**.
- 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 5.2.1**.
- Default values may be used for all other parameters.



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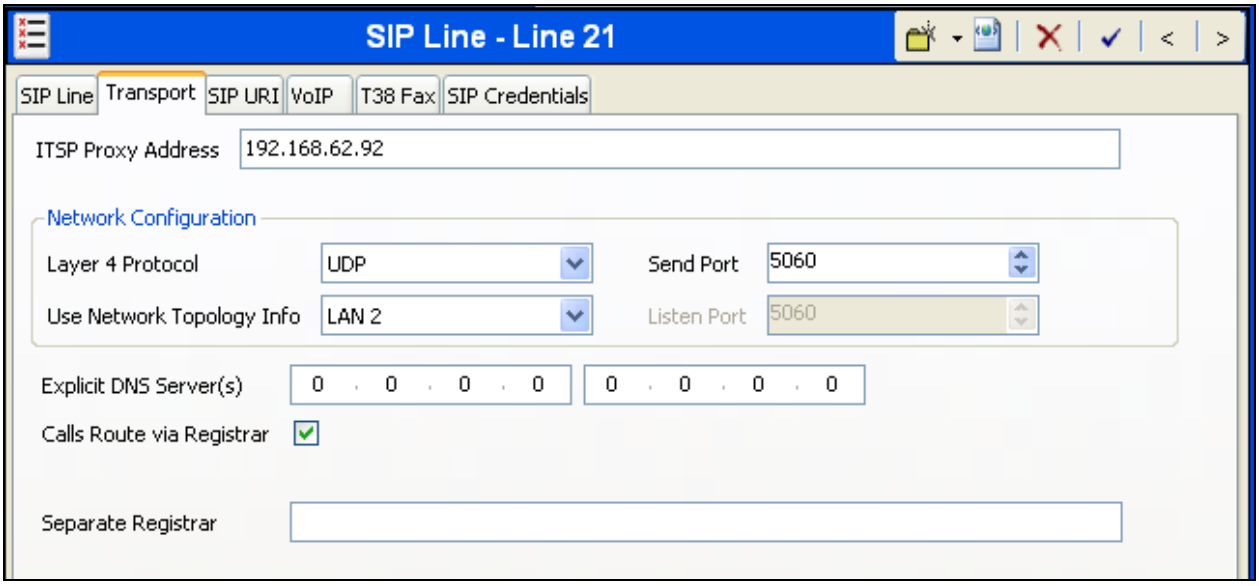
5.4.2. SIP Line - Transport Tab

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

- Set **ITSP Proxy Address** to the IP address of the Accelerated Connections SIP proxy.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to the network port used by the SIP line to access the far-end and configured in **Section 5.2.1**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.



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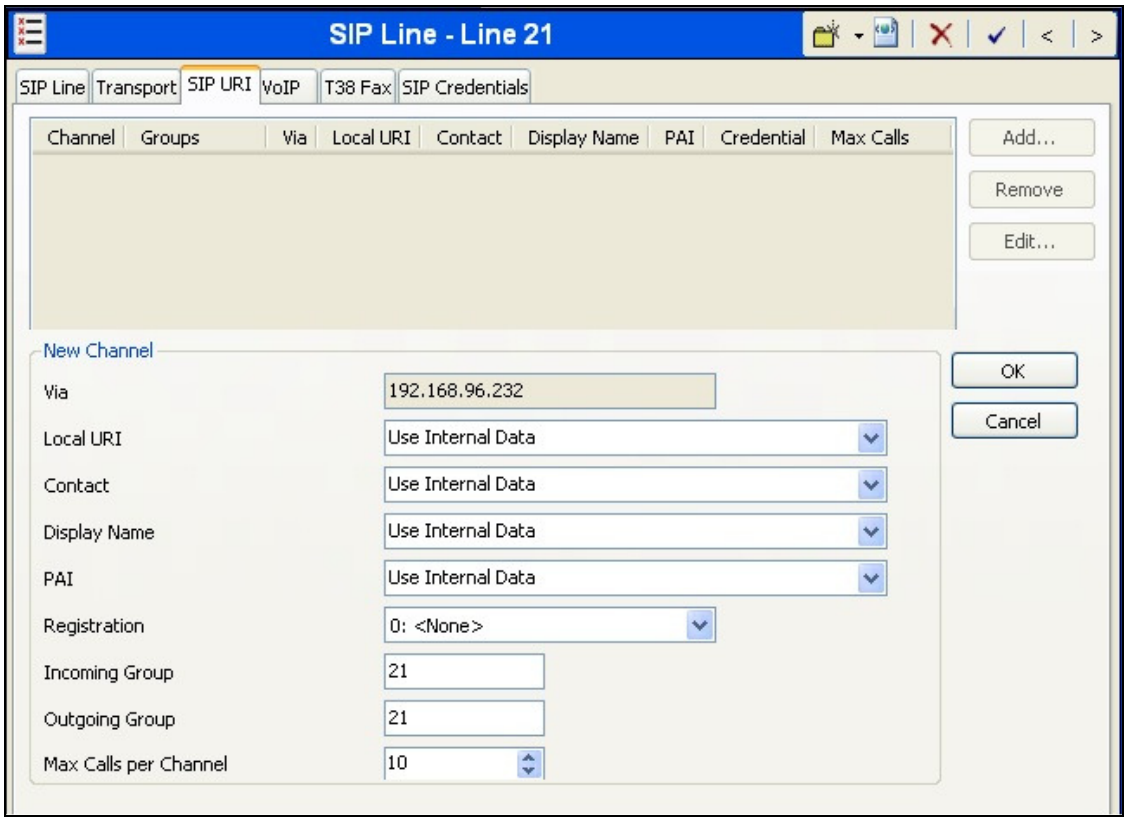
5.4.3. SIP Line - SIP URI Tab

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**, **Display Name** and **PAI** 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 5.7**.
- For **Registration**, select **0: <None>** from the pull-down menu.
- 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 in **Section 5.8.2**. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining ARS entries for routing outbound traffic to this line in **Section 5.6**. For the compliance test, a new incoming and outgoing group **21** was defined that only contained this line (line 21).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

Click **OK**.



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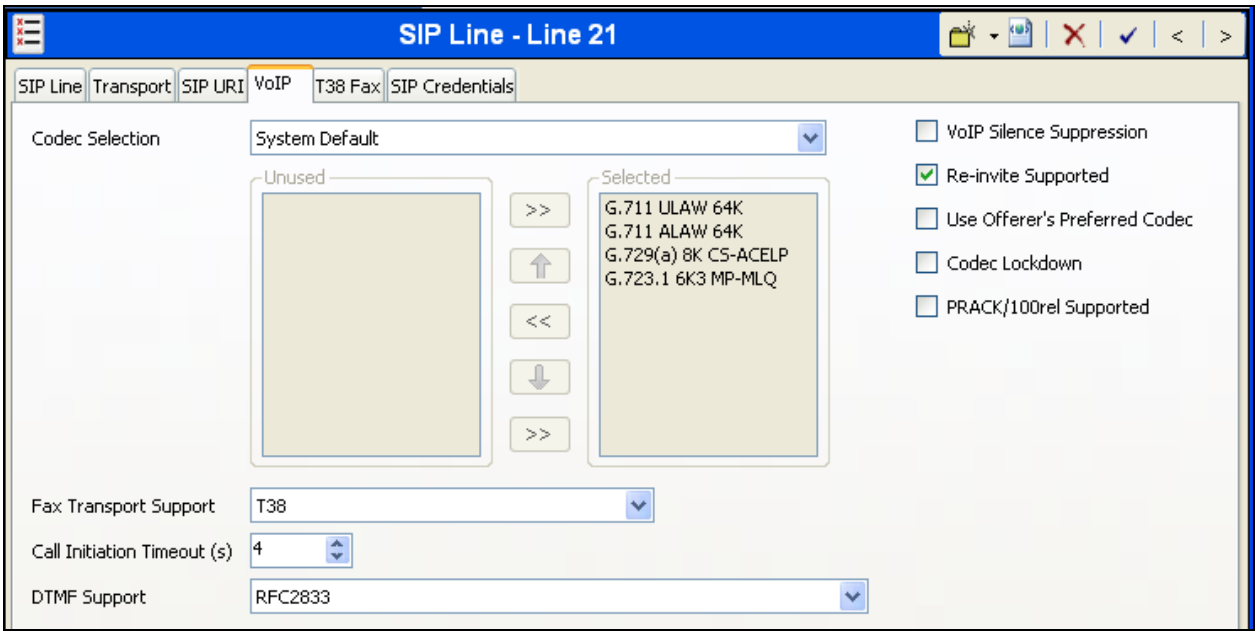
Additional SIP URIs may be required to allow inbound calls to numbers not associated with a user, such as a short code. These URIs are created in the same manner as shown above with the exception that the incoming DID number is entered directly in the **Local URI**, **Contact**, **Display Name** and **PAI** fields.

5.4.4. 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.
- Check the **Re-invite Supported** 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.
- Default values may be used for all other parameters.

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



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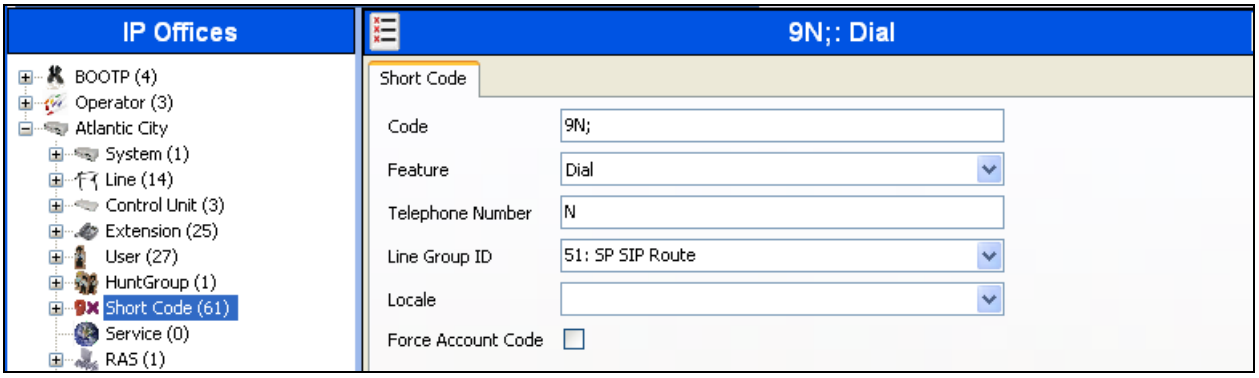
5.5. Short Codes

ARS is used to route outbound traffic to the SIP line. A short code is used to route outbound traffic to ARS. 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**. The value **N** represents the number dialed by the user after removing the **9** prefix. This value is passed to ARS.
- Set the **Line Group Id** to the ARS route to be used which is defined in **Section 5.6**.

Click the **OK** button (not shown).



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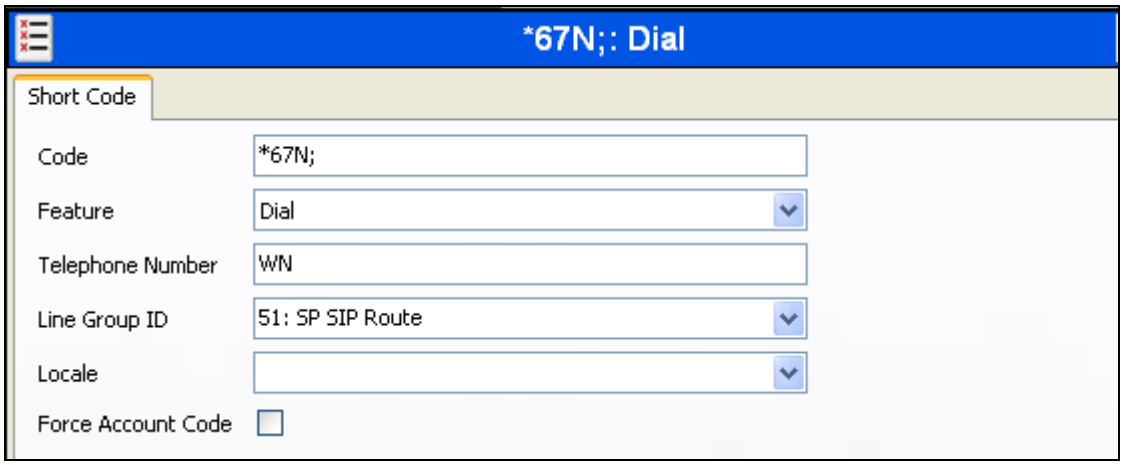
Optionally, add or edit a short code that can be used to access the SIP Line anonymously. In the screen shown below, the short code ***67N;*** is illustrated. This short code is similar to the **9N;** short code except that the **Telephone Number** field begins with the letter **W**, which means "withhold the outgoing calling line identification". In the case of the SIP Line to Accelerated Connections 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 and will include the Privacy: Id header. Accelerated Connections 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.



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5.6. ARS

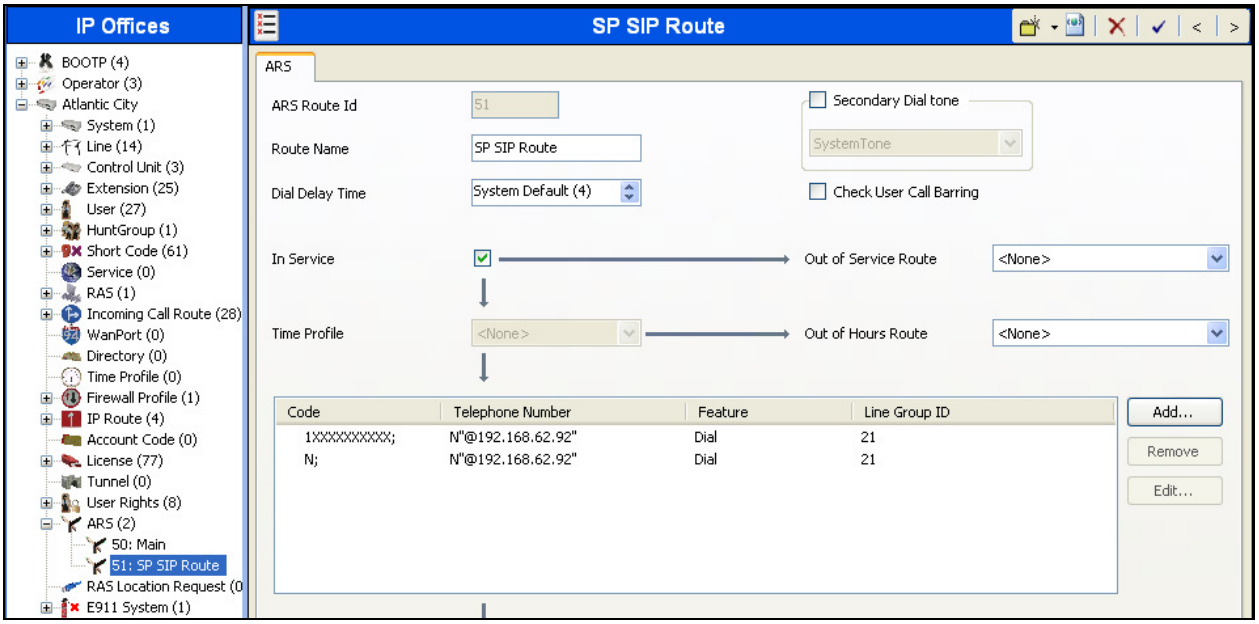
ARS is used to route outbound traffic to the SIP line. To define a new ARS route, right-click **ARS** in the Navigation pane and select **New**. In the Details pane that appears, a collection of matching patterns (similar to short codes) can be entered to route calls as shown below.

For the compliance test, two entries were created. The first entry matches on any 1 + 10 digit number (**1XXXXXXXXXX**) and then sends only 9 digits (**N**) in the SIP INVITE message on the line group defined in **Section 5.4.3** (e.g., line group 21). The second entry matches on any other number (**N**) and passes it unaltered to the line group.

To create an entry, click the **Add** button and enter the following in the pop-up window.

- In the **Code** field, enter the pattern to match the number passed to ARS from the short code in **Section 5.5** followed by a semi-colon.
- Set **Feature** to **Dial.** This is the action that the short code will perform.
- Set **Telephone Number** to **N"@192.168.62.92"**. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. In the first entry, **N** represents the remaining 9 digits after removing the preceding 1. In the second entry, the value **N** represents the complete number passed to ARS. The IP address 192.168.62.92 is the IP address of the Accelerated Connections 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 5.4.3**. This short code will use this line group when placing the outbound call.

Click the **OK** button (not shown).



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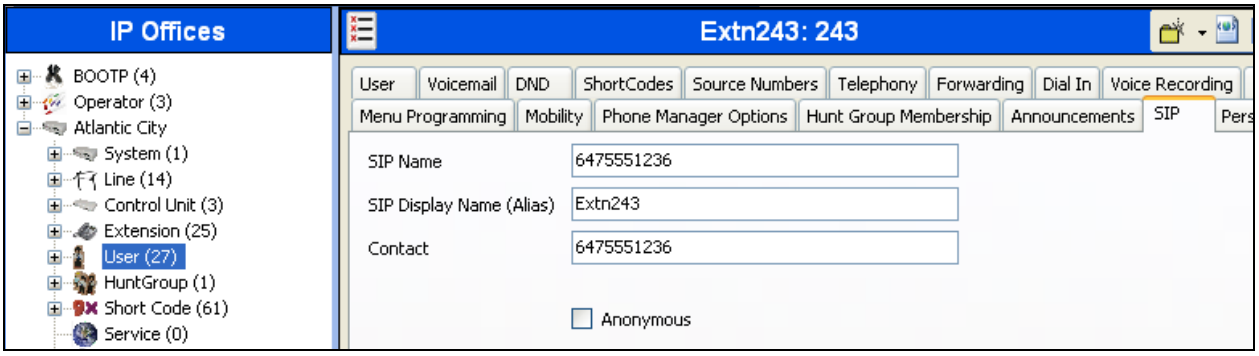
5.7. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.4**. 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 5.4.3**). 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 Accelerated Connections. 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).



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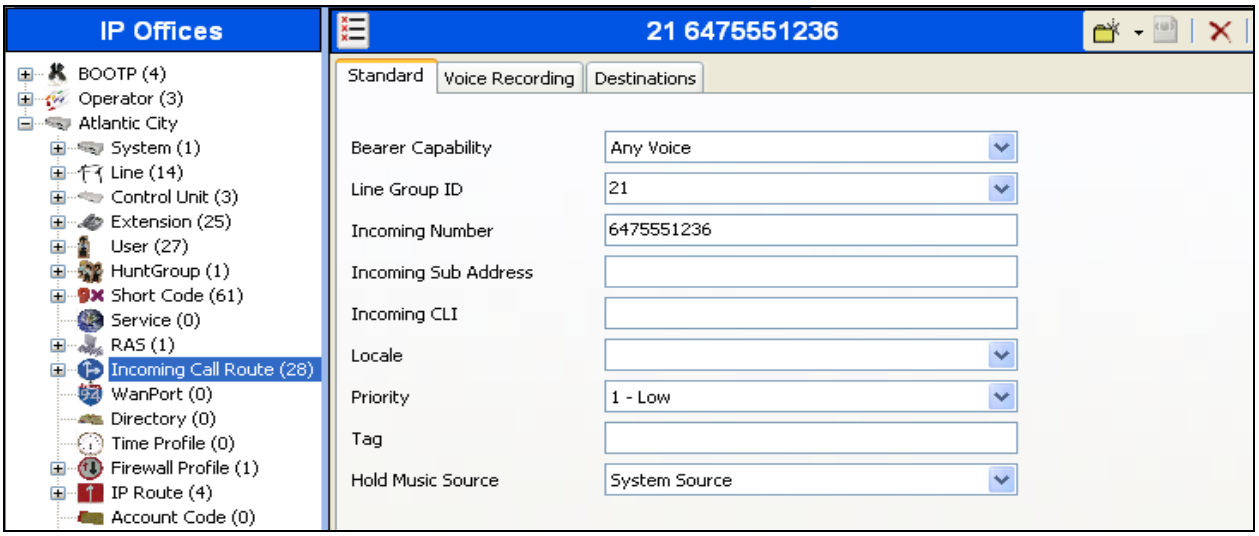
5.8. Incoming Call Route

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**.

5.8.1. Incoming Call Route – Standard Tab

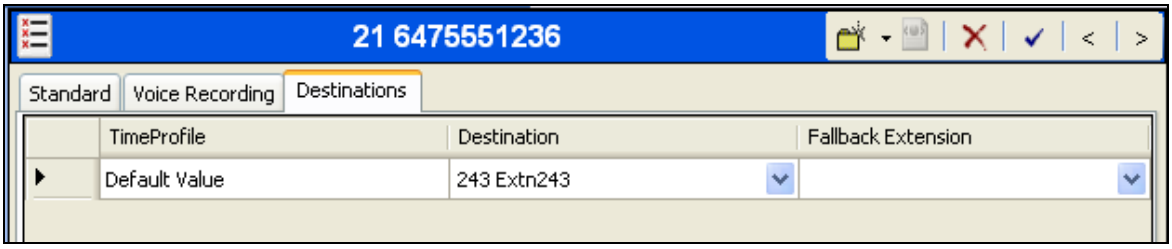
On the **Standard** tab of the Details Pane, enter the parameters as shown below.

- Set the **Bearer Capacity** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.4.3**.
- Set the **Incoming Number** to the incoming number on which this route should match.
- Default values can be used for all other fields.



5.8.2. Incoming Call Route – Destinations Tab

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 6475551236 on line 21 are routed to extension 243.



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.

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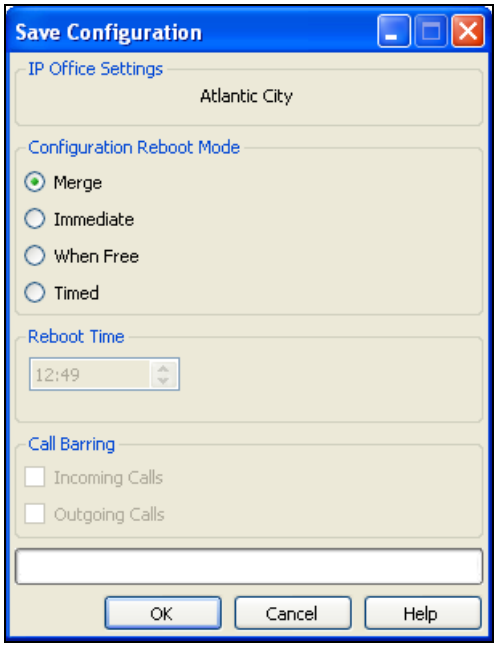
5.9. Save Configuration

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** to proceed.



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6. Accelerated Connections SIP Trunking Configuration

Accelerated Connections is responsible for the configuration of the Accelerated Connections SIP Trunking Service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. In the case of the compliance test , this is the IP Office LAN2 address. Accelerated Connections will provide the customer the necessary information to configure the Avaya IP Office SIP connection to Accelerated Connections including:

- IP address of the Accelerated Connections 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

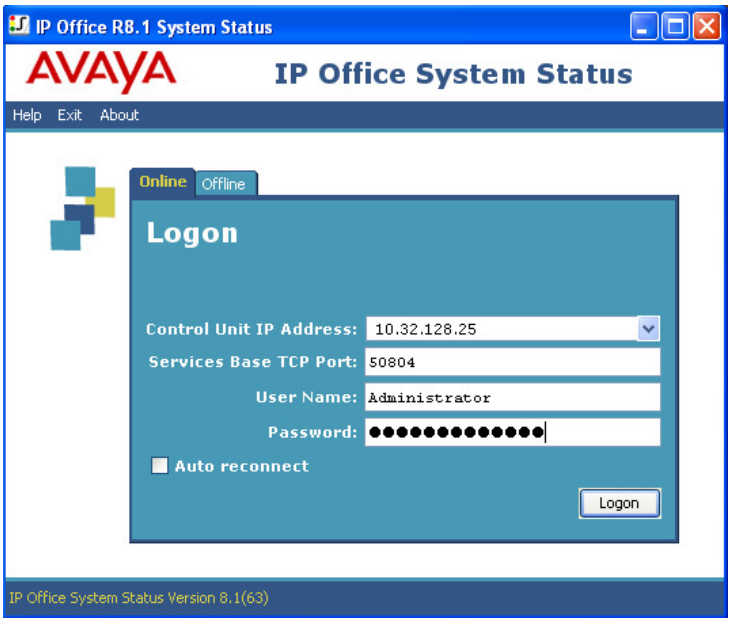
7. Verification Steps

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**.



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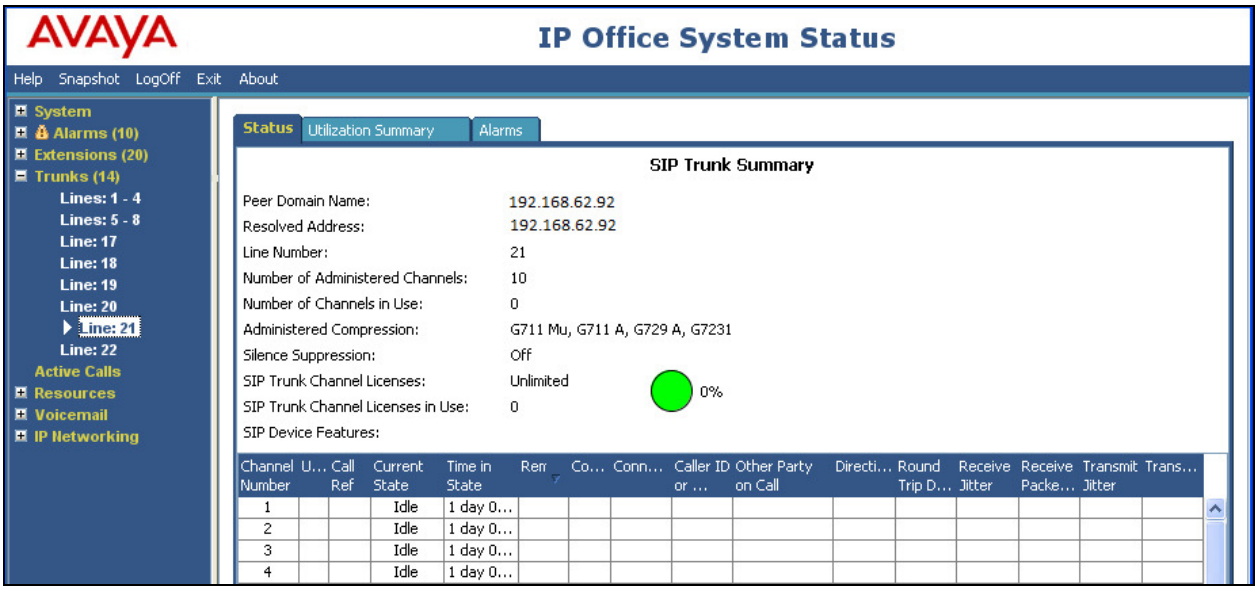
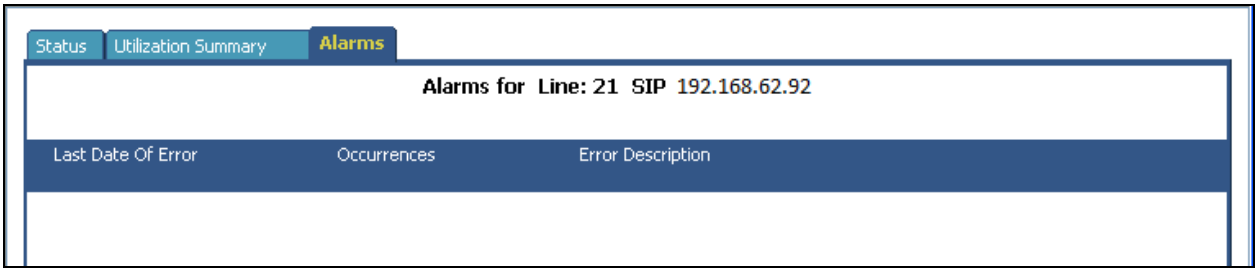
Select the SIP line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current Stat**e is **Idle** for each channel.



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



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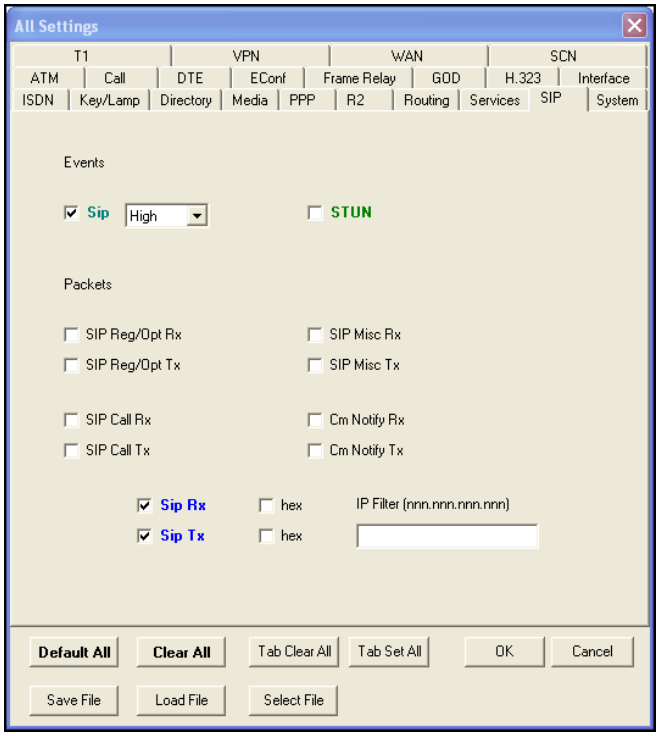
7.2. Monitor

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.



8. Conclusion

These Application Notes describe the configuration necessary to connect Avaya IP Office 8.1 to the Accelerated Connections SIP Trunking Service. The Accelerated Connections SIP Trunking Service 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. The Accelerated Connections SIP Trunking Service passed compliance testing. Please refer to **Section 2.2** for any exceptions.

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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 27m, July 2, 2013.
[2] *IP Office Release 8.1Manager 10.1*, Document Number 15-601011, Issue 29u, April 5, 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 03c, March 1, 2013.

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

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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 5.4** in these Application Notes as a reference.

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

```
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20130710</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>Accelerated Connections</DescriptiveName>
  <ITSPDomainName>192.168.62.92</ITSPDomainName>
  <SendCallerID>CallerIDDIV</SendCallerID>
  <ReferSupport>false</ReferSupport>
  <ReferSupportIncoming>2</ReferSupportIncoming>
  <ReferSupportOutgoing>2</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <UpdateSupport>UpdateNever</UpdateSupport>
  <UserAgentServerHeader />
  <CallerIDfromFromheader>false</CallerIDfromFromheader>
  <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
  <ITSPProxy>192.168.62.92</ITSPProxy>
  <LayerFourProtocol>SipUDP</LayerFourProtocol>
  <SendPort>5060</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>
  <UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
```


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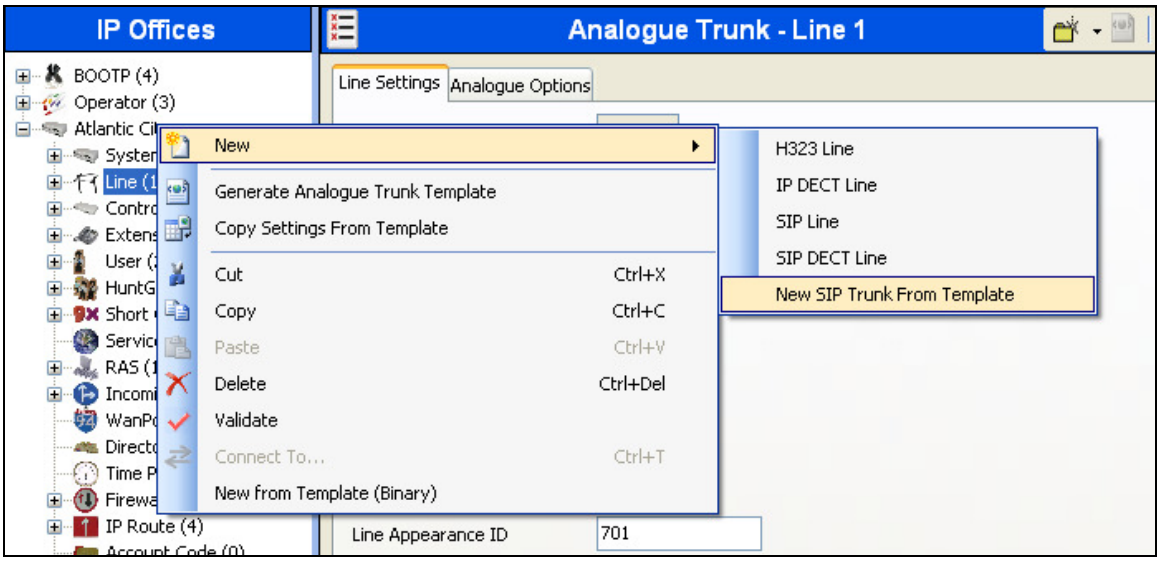
```
  <CodecLockdown>false</CodecLockdown>
  <Rel100Supported>false</Rel100Supported>
  <T38FaxVersion>3</T38FaxVersion>
  <Transport>UDPTL</Transport>
  <LowSpeed>0</LowSpeed>
  <HighSpeed>0</HighSpeed>
  <TCFMethod>Trans_TCF</TCFMethod>
  <MaxBitRate>FaxRate_14400</MaxBitRate>
  <EflagStartTimer>2600</EflagStartTimer>
  <EflagStopTimer>2300</EflagStopTimer>
  <UseDefaultValues>true</UseDefaultValues>
  <ScanLineFixup>true</ScanLineFixup>
  <TFOPEnhancement>true</TFOPEnhancement>
  <DisableT30ECM>false</DisableT30ECM>
  <DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
  <DisableT30MRCompression>false</DisableT30MRCompression>
  <NSFOverride>false</NSFOverride>
</Template>
```

To import the above template into a new installation:

1. On the PC where IP Office Manager was installed, copy and paste the above template into a text document named **US_AcceleratedConnections_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**:



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3. Verify that **United States** is automatically populated for **Country** and **AcceleratedConnections** 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.

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