Avaya Solution & Interoperability Test Lab

Application Notes for Configuring WorldNet SIP Trunking Service with Avaya IP Office R9.1 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Avaya IP Office Release 9.1 to interoperate with the SIP Trunking Service offered by WorldNet Telecommunications.

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

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

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the procedures for configuring an enterprise solution using Avaya IP Office Release 9.1 to interoperate with the SIP Trunking Service offered by WorldNet Telecommunications.

The WorldNet Telecommunications SIP Trunking Service referenced within these Application Notes is positioned for customers who have an IP-PBX or IP-based network equipment with SIP functionality, but need a network service to access the PSTN from the enterprise using IP transport to complete their solution.

The WorldNet Telecommunications SIP Trunking Service will enable delivery of origination and termination of local, long-distance, toll-free, international, and other types of calls across a single broadband IP connection. A SIP signaling interface will be enabled to the Customer Premises Equipment (CPE).

For brevity, the remainder of this document refers to the WorldNet Telecommunications SIP Trunking Service as WorldNet SIP Trunking Service or simply WorldNet.

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the WorldNet SIP Trunking Service via the public Internet and exercise the features and functionality listed in **Section 2.1**. The simulated enterprise site comprised of an Avaya IP Office 500 V2 running Release 9.1 software, Avaya Preferred Edition messaging application (a.k.a. Voicemail Pro), Avaya H.323 and SIP deskphones, and the SIP-based Avaya Communicator softphone.

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

2.1. Interoperability Compliance Testing

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

- SIP OPTIONS queries and responses.
- Incoming calls from the PSTN to H.323 and SIP telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing calls to the PSTN from H.323 and SIP telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Various call types including: local, long distance, outbound toll-free, international, and directory assistance calls.

- G.711u and G.729a codecs.
- Caller ID presentation and Caller ID restriction.
- DTMF transmission using RFC 2833.
- Voicemail access and navigation for inbound and outbound calls.
- Telephony supplementary features such as hold and resume, transfer, and conference.
- Off-net call forwarding and call transfer/conference.
- Twinning on inbound calls to PSTN mobile phones.
- Use of the SIP INVITE message for transferring calls off-net to the PSTN.
- T.38 and G.711 pass-through fax.
- Inbound and outbound long-duration call stability.
- Inbound and outbound long hold time call stability.
- Response to incomplete call attempts and trunk busy or error conditions.

2.2. Test Results

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

- **Internal IP Exposure –** An internal private IP was exposed in messages from WorldNet:
    - The Call-ID header of inbound INVITE embedded a private IP.
    - The tag field added to the From or To headers in the inbound messages from WorldNet (OPTIONS, INVITE, 183 Session Progress, 200 OK, BYE, etc.) contained a private IP.
    - For privacy-enabled inbound calls, the Remote-Party-ID header of inbound INVITE used a private IP as URI-host.

  After investigation, WorldNet support informed that no configuration could be used to address this issue, and the risk of exposing the WorldNet private IP over the public Internet was acceptable.

- **G.729a Codec –** The G.729a codec could not be used on inbound calls. This was because WorldNet configured the inbound INVITE to contain G.711u only. Consequently:
    - If IP Office was configured to use G.729a only, it would reject the inbound INVITE with "488 Not Acceptable Here".
    - If IP Office was configured to use G.729a and G.711u, the inbound call would negotiate to use G.711u.

  Outbound calls using G.729a were accepted by WorldNet and would connect with effective 2-way audio.

- **Direct Media –** The Direct Media capability on IP Office allows RTP media directly between IP endpoints rather than having all the media flow through the IP Office, using up VoIP and relay resources. This capability is not supported on an IP Office SIP Line (SIP trunk) that is configured to carry both voice and T.38 fax calls. Consequently, Direct Media was disabled for the test circuit.

- **Remote Worker –** Remote Worker (phones connected directly to the public Internet functioning as enterprise local extensions) is not supported by the combined Avaya/WorldNet solution as documented in these Application Notes. This is because its setup requires IP Office to be front-ended by the Avaya Session Border Controller for Enterprise which was not used in this compliance test.

Items not supported or not tested include the following:

- **Operator Service** – WorldNet does not support Operator (0) and Operator-Assisted (0 + 10-digits) calls.

- **REFER** – WorldNet does not support use of the SIP REFER method for transferring calls off-net to the PSTN. In the compliance test, off-net call transfer was tested using the SIP INVITE method.

- **Call Types Not Tested –** Emergency 911 and inbound toll-free calls were not tested.

2.3. Support

Contact information for technical support on the WorldNet SIP Trunking Service:

- Web: http://www.WorldNetpr.com/contact-us
- Phone: 787-705-9000

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

3. Reference Configuration

Figure 1 illustrates the test configuration showing an enterprise site connected to the WorldNet SIP Trunking Service.

Within the enterprise site is an Avaya IP Office 500 V2 running the Release 9.1 software. Endpoints include various Avaya IP Telephones (with H.323 and SIP firmware) and SIP-based Avaya Communicator softphone. The site also has a Windows PC running Voicemail Pro for providing a voice messaging service to the Avaya IP Office users, and Avaya IP Office Manager for administering the Avaya IP Office.

Mobility Twinning is configured for some of the Avaya IP Office users so that calls to these user phones will also ring and can be answered at the configured mobile phones.

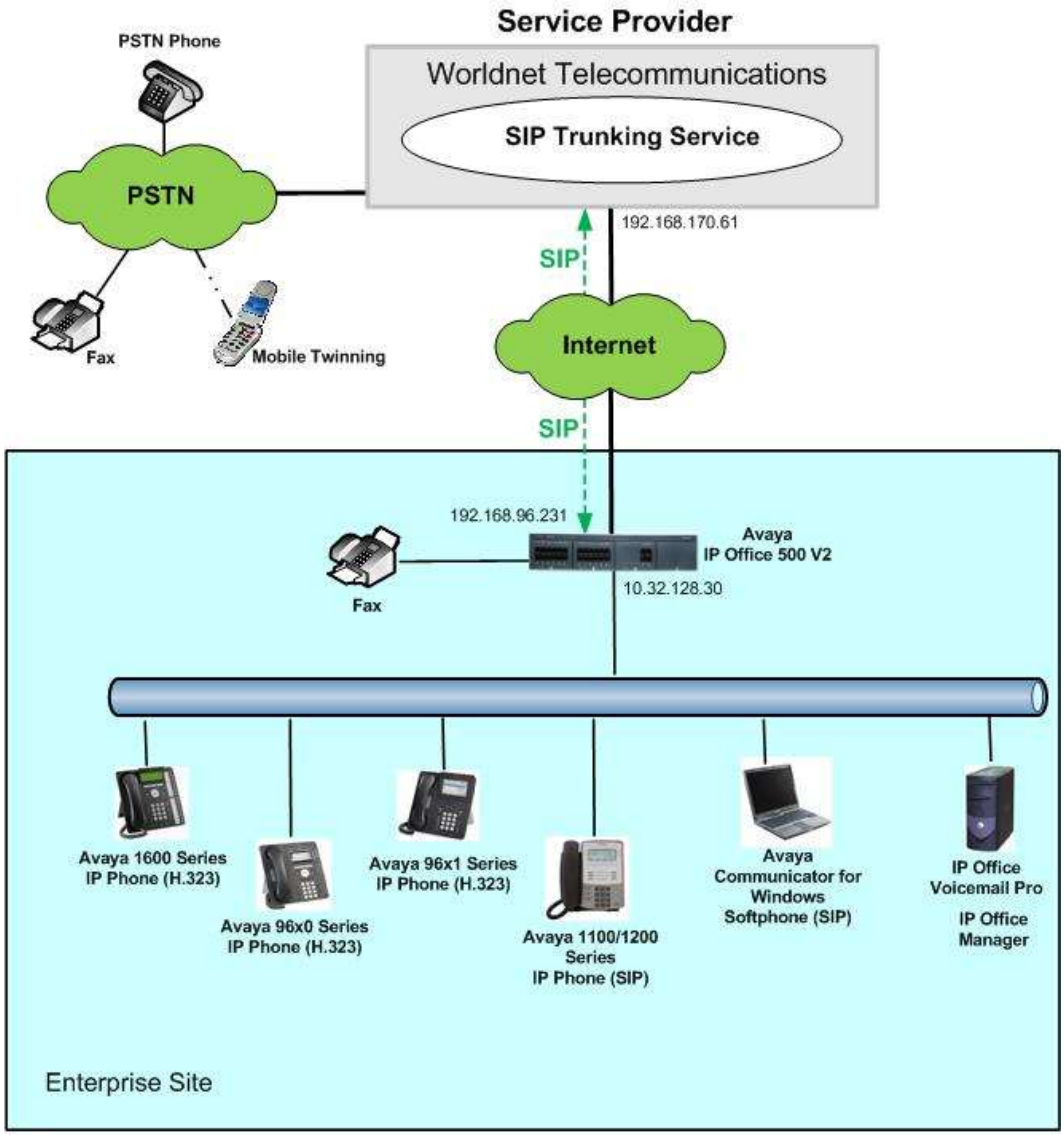


Figure 1: Avaya IP Office with WorldNet SIP Trunking Service

For security reasons, any actual public IP addresses used in the configuration have been replaced with private IP addresses in these Application Notes.

During compliance testing, Avaya IP Office users dialed a short code of 8 or 9 plus N digits to send digits across the SIP trunk to WorldNet. The short code (8 or 9) was stripped off by Avaya IP Office but the remaining N digits were sent to the service provider network. For calls within the North American Numbering Plan (NANP), the user dialed 11 (1 + 10) digits for long distance and local calls. Thus, for these NANP calls, Avaya IP Office sent 11 digits in the Request URI and the To header of an outbound SIP INVITE message. WorldNet sent 10 digits in the Request URI and the To header of inbound SIP INVITE messages.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and 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 Avaya IP Office must be allowed to pass through these devices.

The administration of the Voicemail Pro messaging service and endpoints on Avaya IP Office are standard. Since these configuration tasks are not directly related to the inter-operation with the WorldNet SIP Trunking Service, they are not included in these Application Notes.

4. Equipment and Software Validated

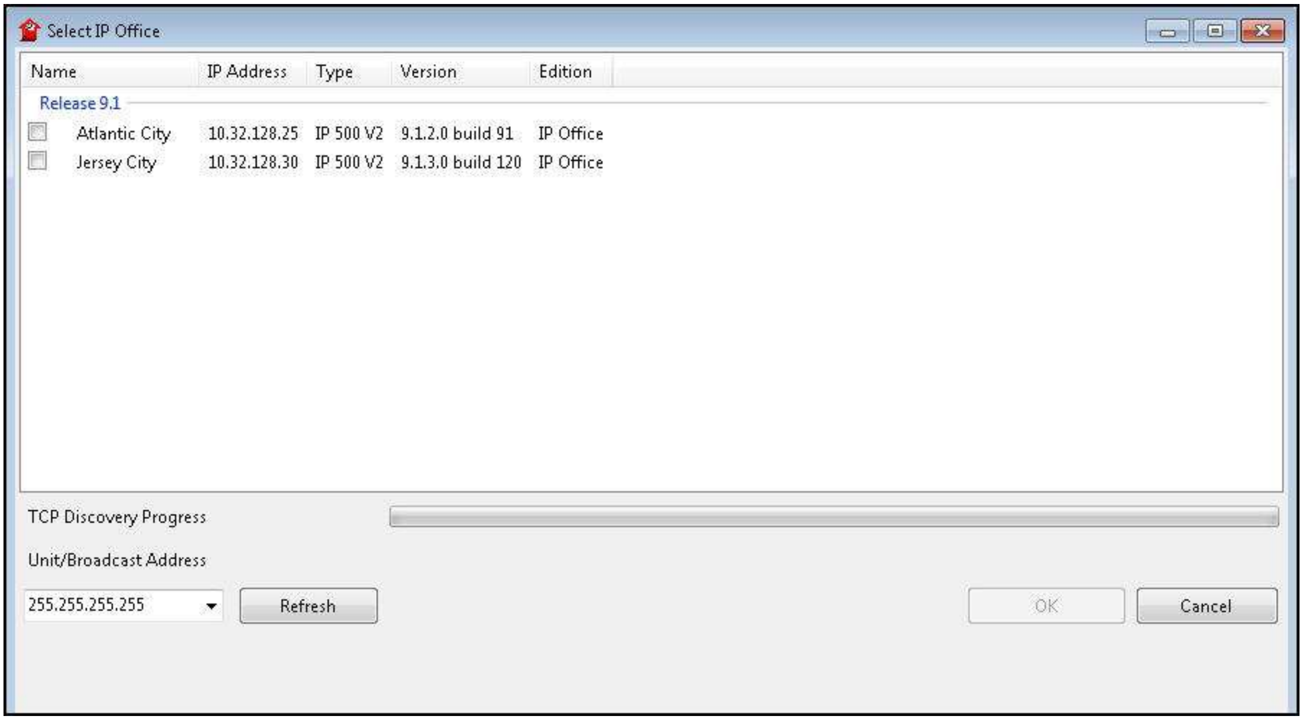
The following equipment and software/firmware were used for the sample configuration provided:

Avaya Telephony Components	
Equipment / Software	**Release / Version**
Avaya IP Office 500 V2	9.1.3.0 build 120
Avaya IP Office COMBO6210/ATM4 Module	9.1.3.0 build 120
Avaya IP Office Manager	9.1.3.0 build 120
Avaya Preferred Edition (a.k.a Voicemail Pro)	9.1.300.2
Avaya 1616 IP Telephones (H.323)	Avaya one-X® Deskphone 1.3 SP5
Avaya 9611G IP Telephones (H.323)	Avaya one-X® Deskphone 6.6.0.29_V474
Avaya 9630G IP Telephones (H.323)	Avaya one-X® Deskphone 3.2.3
Avaya 1120E IP Telephone (SIP)	4.04.18.00
Avaya Communicator for Windows	2.0.3.30
WorldNet Components	
Equipment / Software	**Release / Version**
ORACLE Session Border Controller	6.4.0
Metaswitch	V8.3.11

Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2, and also when deployed with all configurations of IP Office Server Edition without T.38 fax service (T.38 fax is not supported on IP Office Server Edition). Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog/digital endpoints or trunks.

5. Configure Avaya IP Office

Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the PC running Avaya IP Office Manager, select **Start → All Programs → IP Office → Manager** to launch the application. A **Select IP Office** pop-up window is displayed as shown below. Select the required Avaya IP Office system from the pop-up window and click **OK** to log in with the appropriate credentials (not shown). The configuration may alternatively be opened by navigating to **File → Open Configuration** at the top of the Avaya IP Office Manager window (not shown).



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 and Details) will be referenced throughout the Avaya IP Office configuration.

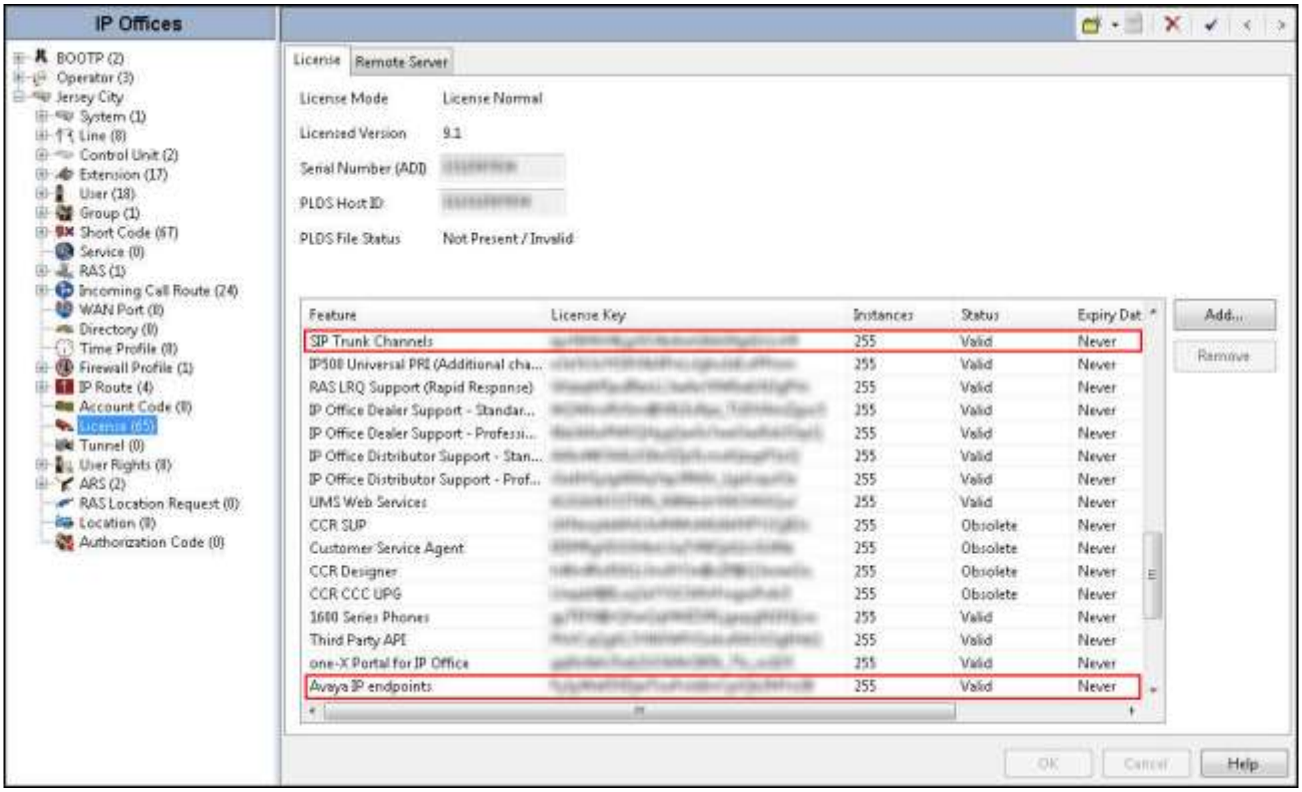
All licensing and feature configuration that is not directly related to the interface with the service provider (such as administering IP endpoints) is assumed to already be in place.

In the sample configuration, **Jersey City** was used as the system name. All navigation described in the following sections (e.g., **Control Unit → IP 500 V2)** appears as submenus underneath the system name **Jersey City** in the Navigation Pane. The configuration screens highlight values/settings configured for the compliance test. Defaults were used for other values and may be customized based upon requirements in the field.

5.1. Licensing and Physical Hardware

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

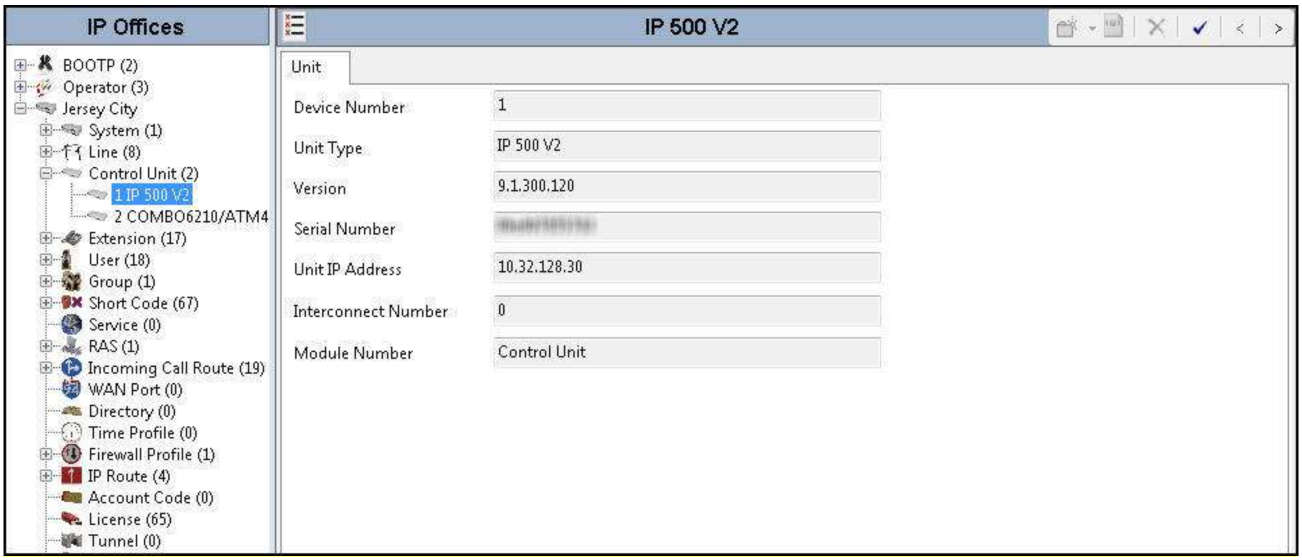
To verify that there is a **SIP Trunk Channels** License with sufficient capacity, click **License** in the Navigation Pane. Confirm a valid license with sufficient **Instances** (trunk channels) in the Details Pane. The screen below also shows the valid license for **Avaya IP endpoints**.



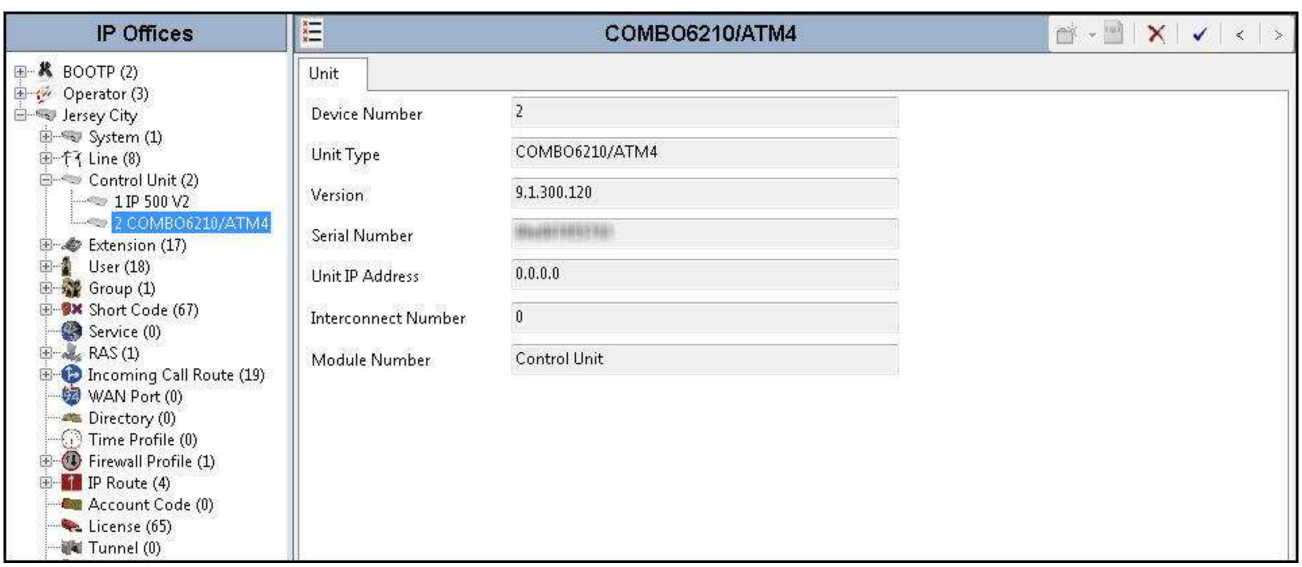
To view the physical hardware comprising the Avaya IP Office system, expand the components under **Control Unit** in the Navigation Pane. In the sample configuration, the second component listed is a Combination Card. This module contains 6 digital station ports, two analog extension ports, 4 analog trunk ports and 10 VCM channels. The VCM is a Voice Compression Module supporting VoIP codecs. An Avaya 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 screen below shows the details of the IP 500 V2.



The screen below shows the details of the Combination Card.

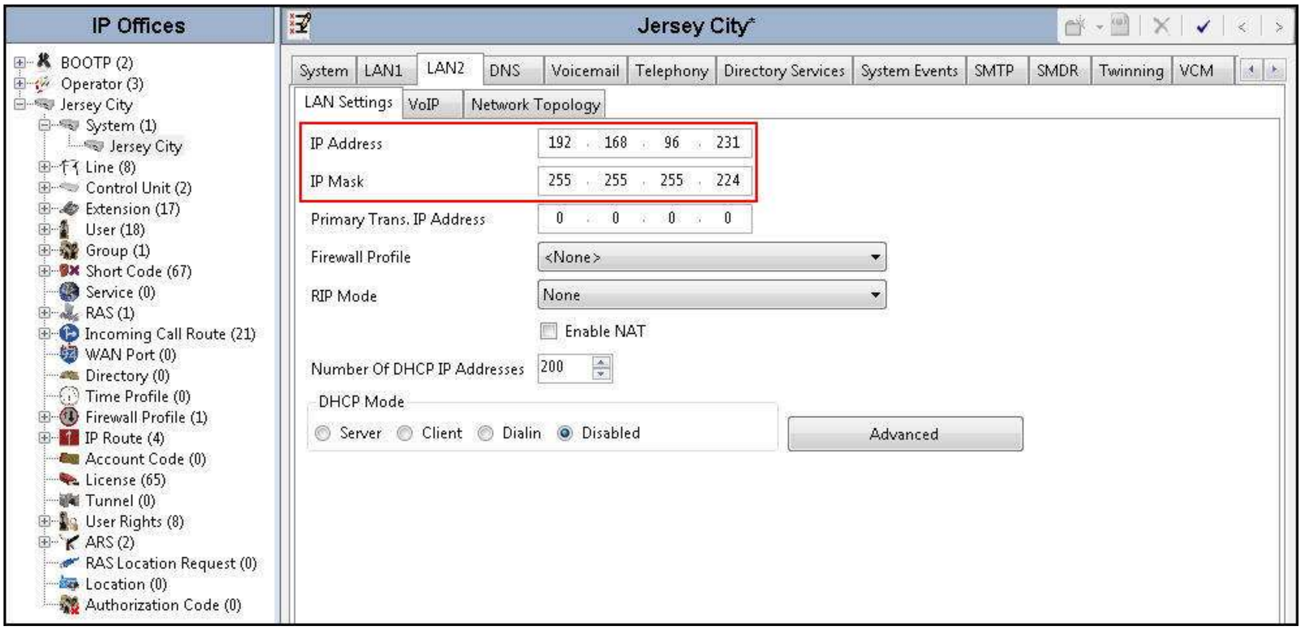


5.2. System

This section configures the necessary system settings.

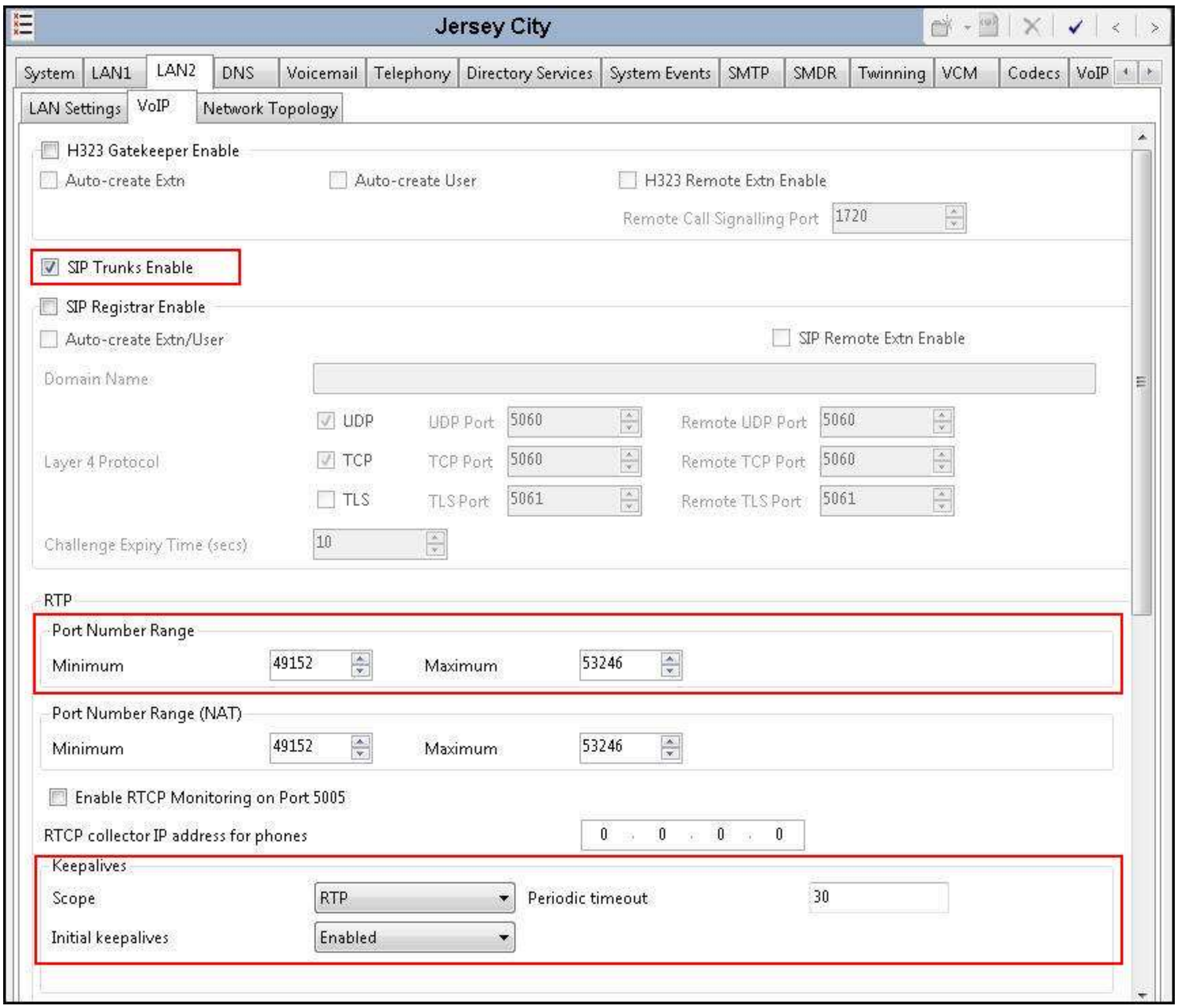
5.2.1. System - LAN2 Tab

In the sample configuration, ***Jersey City*** was used as the system name and the WAN port (LAN2 port) was used to connect Avaya IP Office to the public network. The LAN2 settings correspond to the WAN interface on Avaya IP Office. To access the LAN2 settings, first navigate to **System → Jersey City** in the Navigation Pane and then 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.

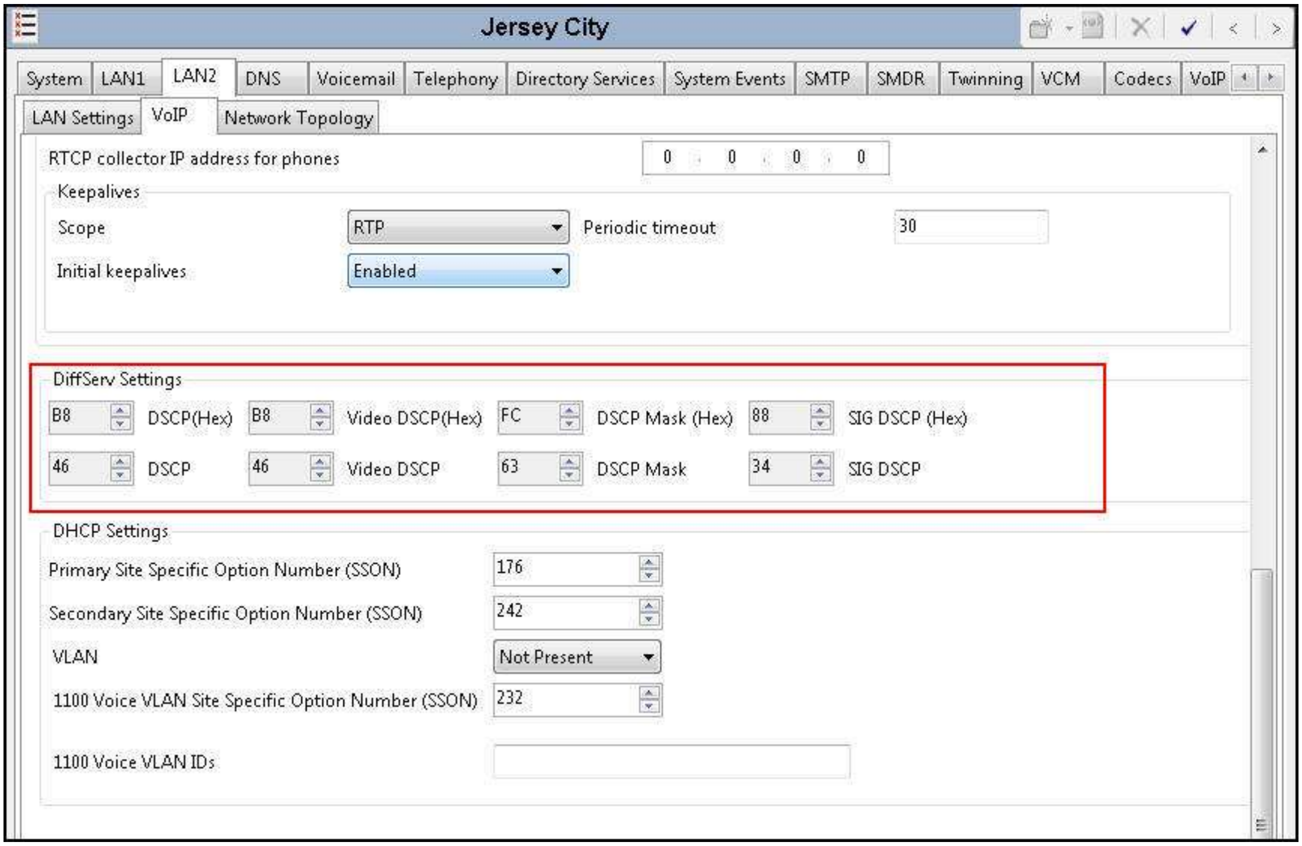


On the **VoIP** tab of LAN2 in the Details Pane, configure the following parameters:

- Check the **SIP Trunks Enable** box to enable the configuration of SIP trunks.
- In the **RTP** section, the **RTP Port Number Range** can be customized to a specific range of receiving ports for the RTP media, as agreed with the service provider. Based on this setting, Avaya IP Office would request RTP media be sent to a port in the configurable range for calls using LAN2.
- In the **Keepalives** section, select ***RTP*** for **Scope**; select ***Enabled*** for **Initial keepalives**; enter ***30*** for **Periodic timeout**. These settings direct IP Office to send a RTP keepalive packet starting at the time of initial connection and every 30 seconds thereafter if no other RTP traffic is present. This facilitates the flow of media in cases where each end of the connection is waiting for media from the other, as well as helping to keep firewall (if used) ports open for the duration of the call.

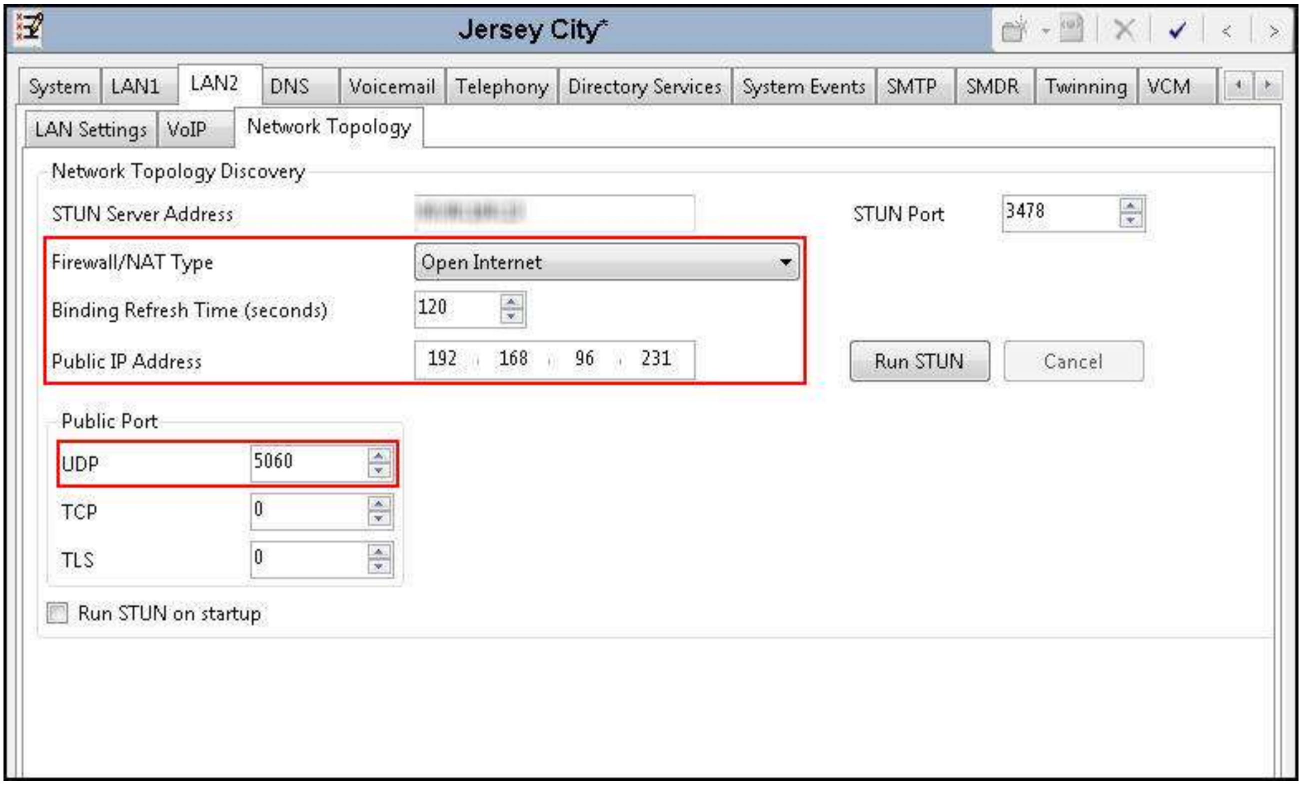


Scroll down to the **DiffServ Settings** section. Avaya IP Office can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the screen below and are also the default values. For a customer installation, if the default values are not sufficient, appropriate values should be provided by the customer.



On the **Network Topology** tab of LAN2 in the Details Pane, configure the following parameters:

- Select **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***. With the ***Open Internet*** setting, **STUN Server Address** is not used.
- Set **Binding Refresh Time (seconds)** to a desired value. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider. See **Section 5.10** for complete details.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- Set **Public Port** to ***5060*** for **UDP**.

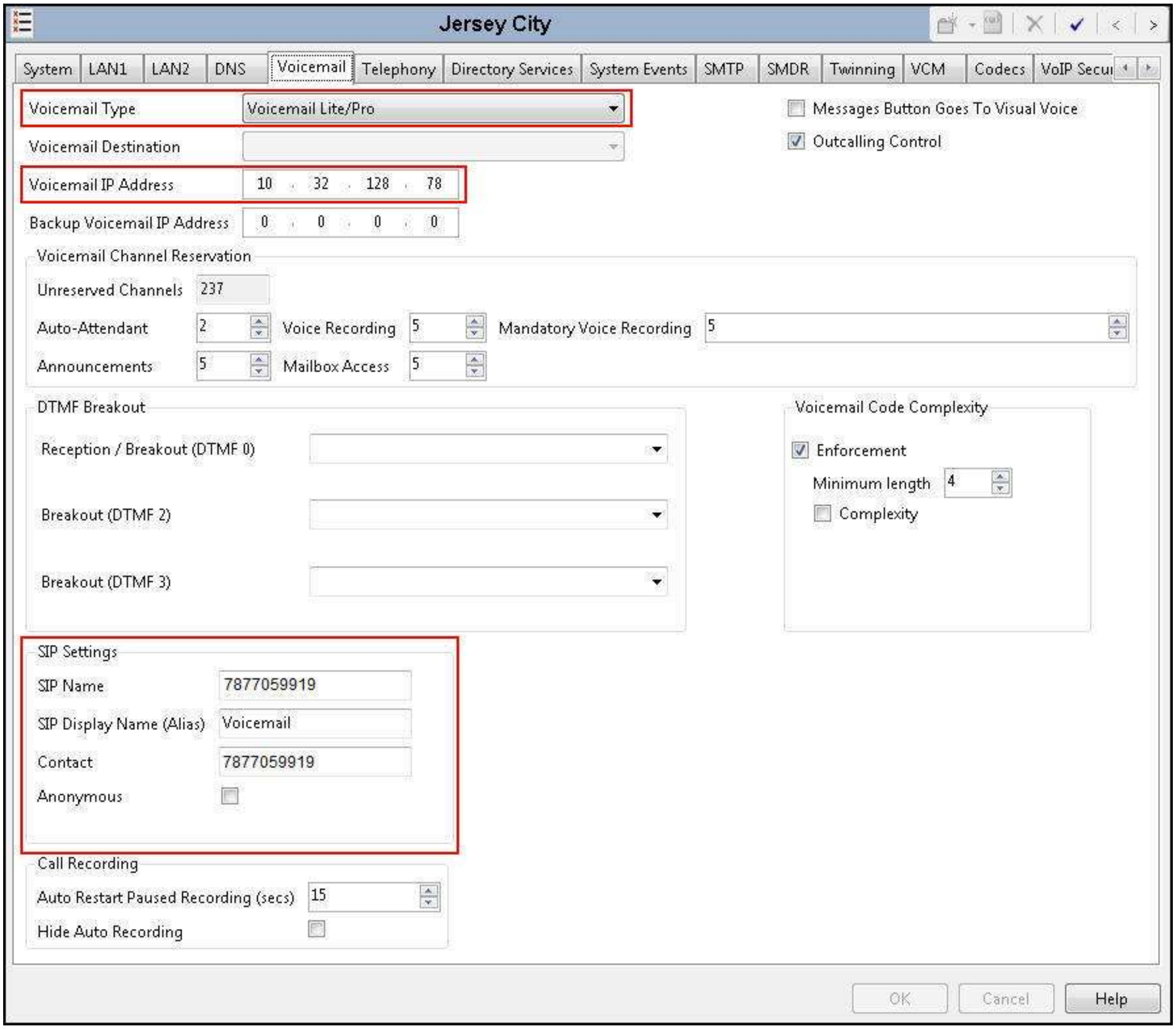


During compliance testing, the LAN1 interface was used to connect the Avaya IP Office to the enterprise site IP network. The LAN1 interface configuration is not directly relevant to the interface with WorldNet, and therefore is not described in these Application Notes.

5.2.2. System - Voicemail Tab

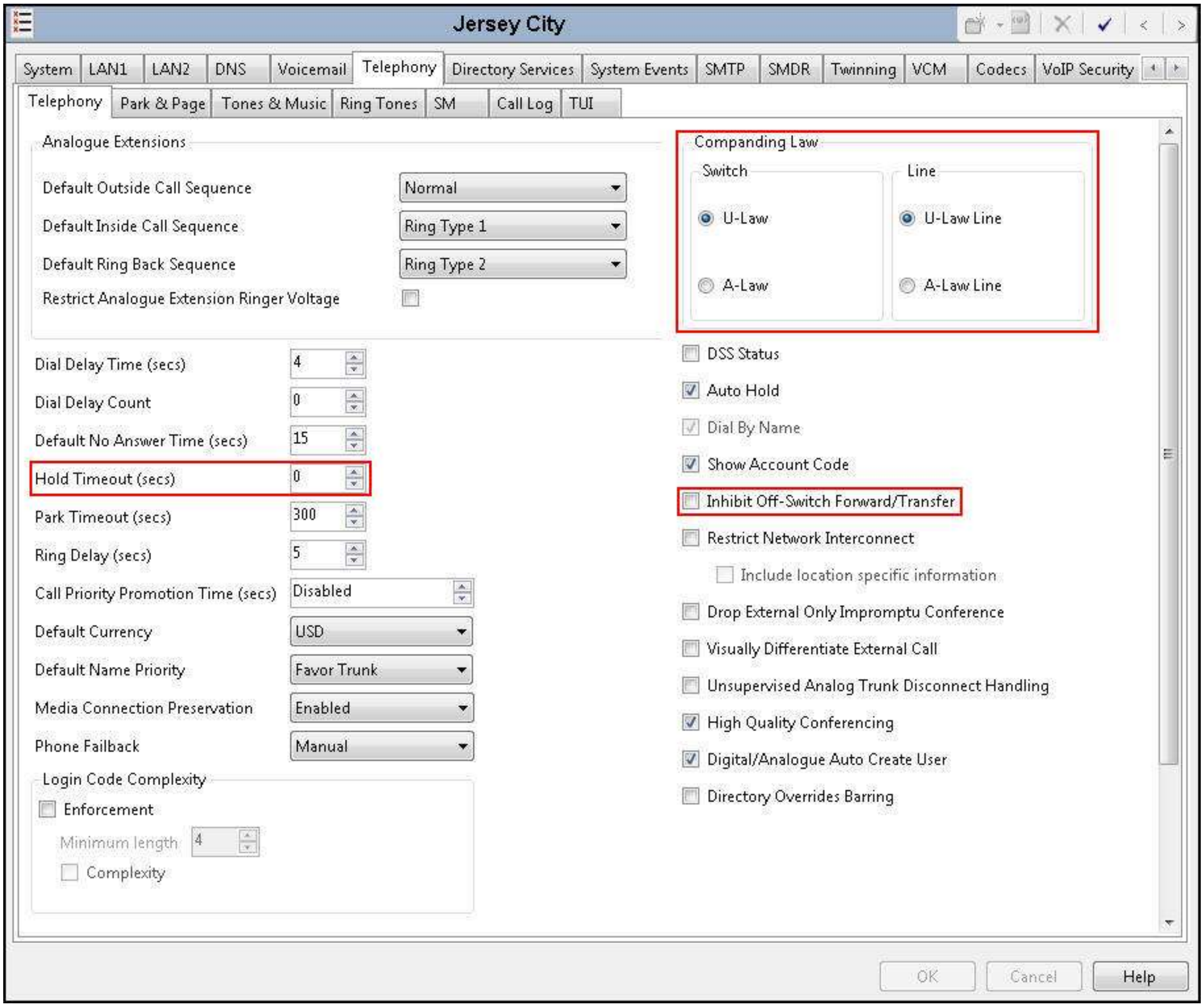
In the **Voicemail** tab of the Details Pane, configure the **SIP Settings** section. The **SIP Name** and **Contact** are set to one of the DID numbers assigned by WorldNet. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Uncheck the **Anonymous** box to allow the Voicemail Caller ID information to be sent to the network.

Note the selection for **Voicemail Type** and the IP address setting for **Voicemail IP Address**. These are for configuring Voicemail Pro as the voice messaging service for Avaya IP Office users (part of the standard IP Office setup beyond the scope of these Application Notes).



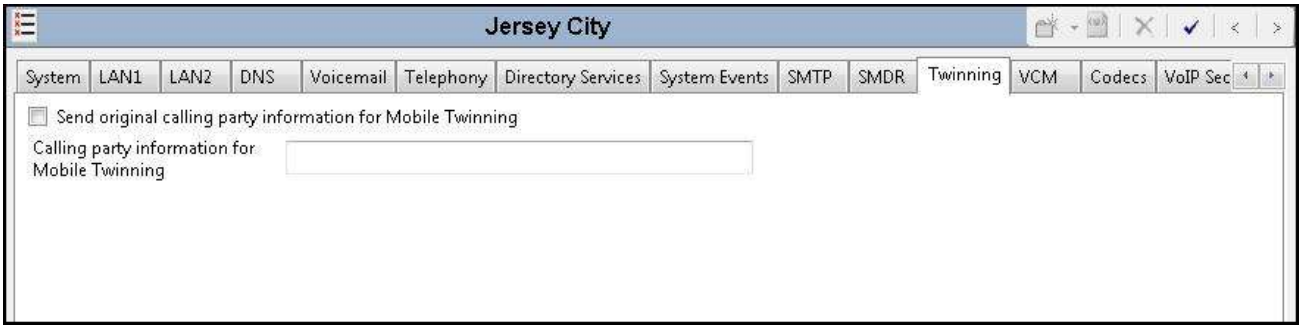
5.2.3. System - Telephony Tab

Navigate to the **Telephony → Telephony** tab on the Details Pane. Enter or select ***0*** for **Hold Timeout (secs)** so that calls on hold will not time out. Choose the **Companding Law** typical for the enterprise site. For the compliance test, ***U-LAW*** was used. 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 per customer business policies. Note that this configuration might pose a security issue (Toll Fraud). Customers should exercise caution with this configuration.



5.2.4. 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.



5.2.5. System – Codecs Tab

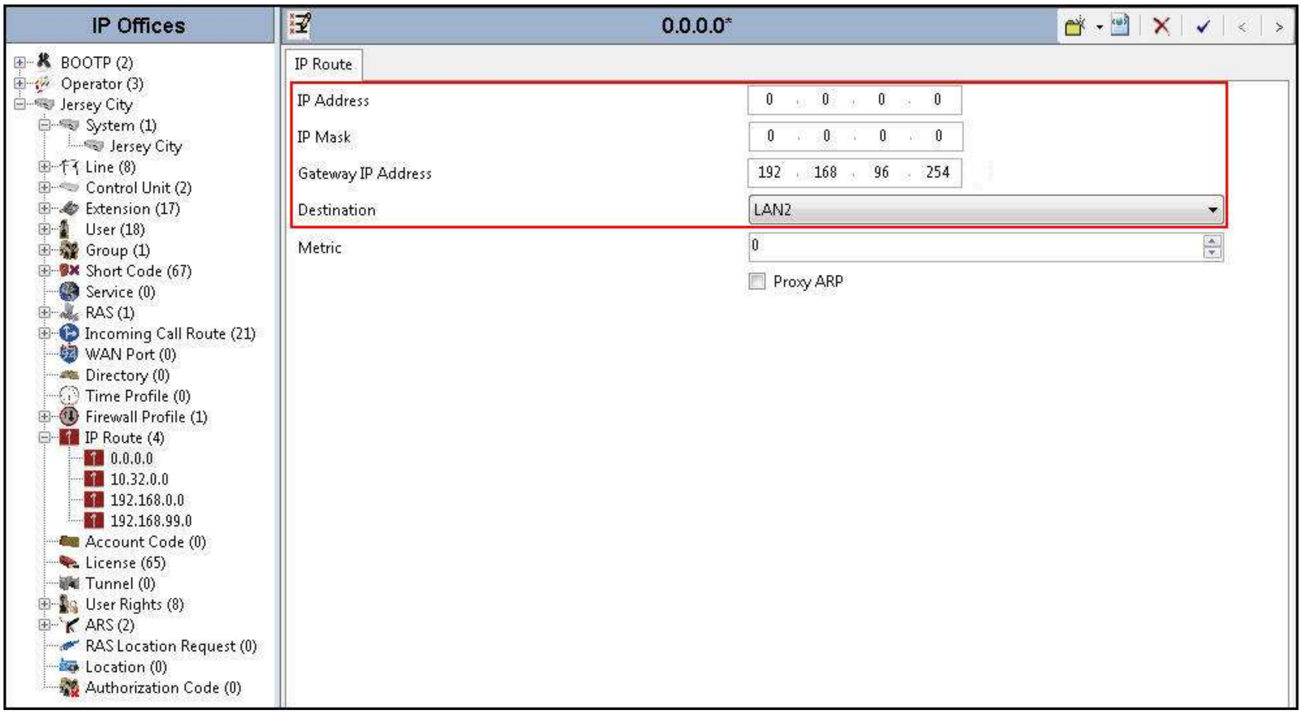
In the **Codecs** tab of the Details Pane, select or enter ***101*** for **RFC2833 Default Payload**. This setting was preferred by WorldNet for use with out-band DTMF tone transmissions.

On the left, observe the list of **Available Codecs**. In the screen below, which is not intended to be prescriptive, the box next to each codec is checked, making all the codecs available in other screens where codec configuration may be performed. The **Default Codec Selection** area enables the codec preference order on a system-wide basis. By default, all IP (SIP and H.323) lines and extensions will assume the system default codec selection unless configured otherwise for the specific line or extension.

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** (not shown)**.** Create and 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 gateway for the public Internet to which the IP Office WAN port is connected.
- Set **Destination** to ***LAN2*** from the drop-down list.



5.4. Administer SIP Line

A SIP Line is needed to establish the SIP connection between Avaya IP Office and the WorldNet network. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.4.1** to create the SIP Line from the template.

Note: DevConnect-generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML-format templates can be used to create SIP trunks on both IP Office Standard Edition (500 V2) and IP Office Server Edition systems

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

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

Therefore, it is important that the SIP Line configuration be reviewed and updated after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.2** through **5.4.8**.

Also, the following SIP Line settings are not supported on Avaya IP Office Basic Edition:

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

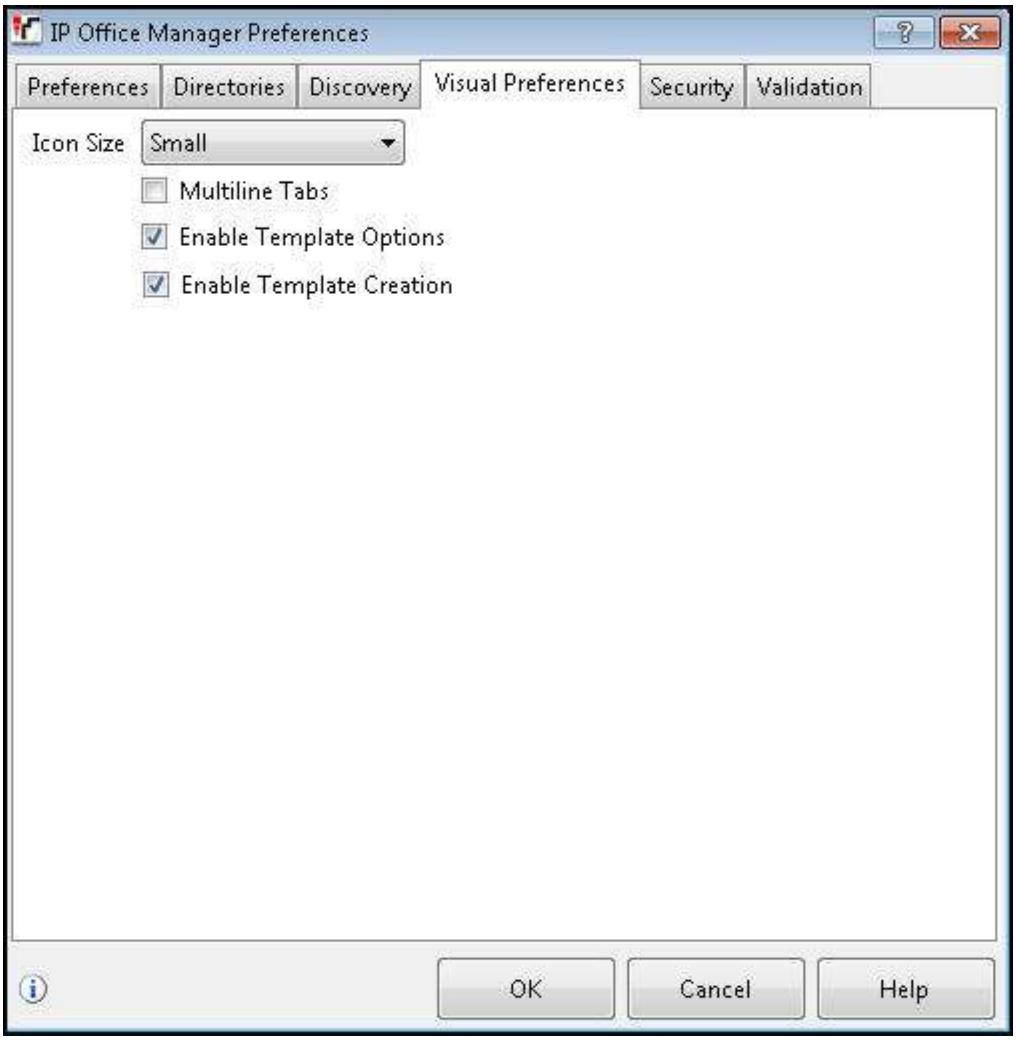
5.4.1. Create SIP Line From Template

1. Copy the template file associated with these Application Notes to a location (e.g., C:\Temp) on the computer where IP Office Manager is installed. Verify that the template file name is

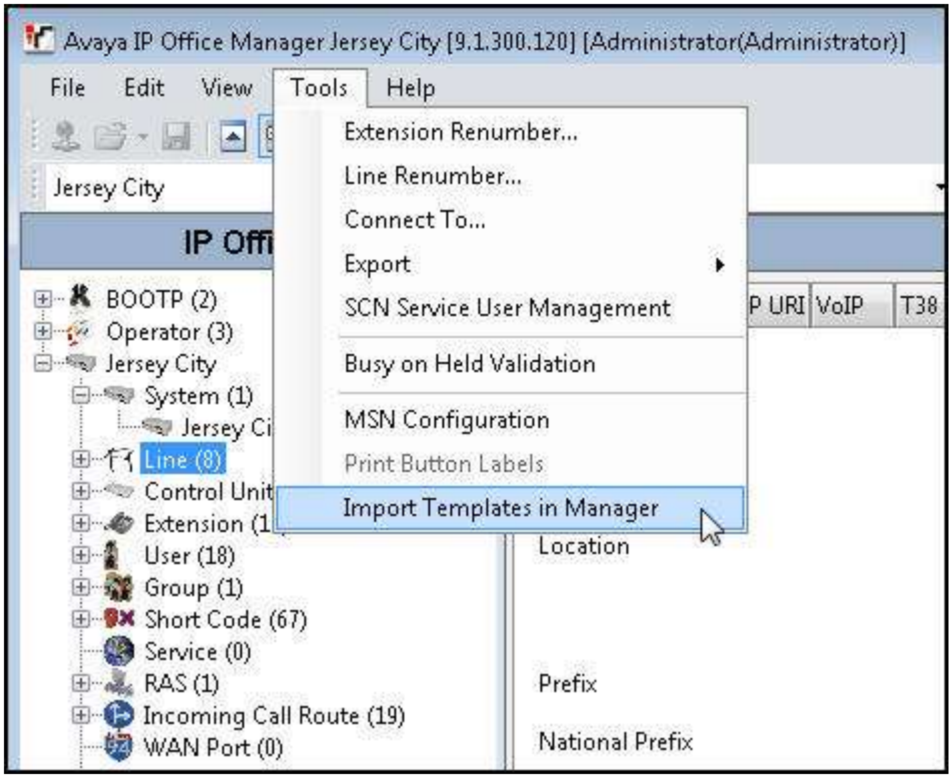
   AF_WorldNet_SIPTrunk.xml

   The file name is important in locating the proper template file in **Step 4**.

2. Verify that template options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the **IP Office Manager Preferences** window that appears, select the **Visual Preferences** tab. Verify that the option box is checked next to **Enable Template Options**. Click **OK**.



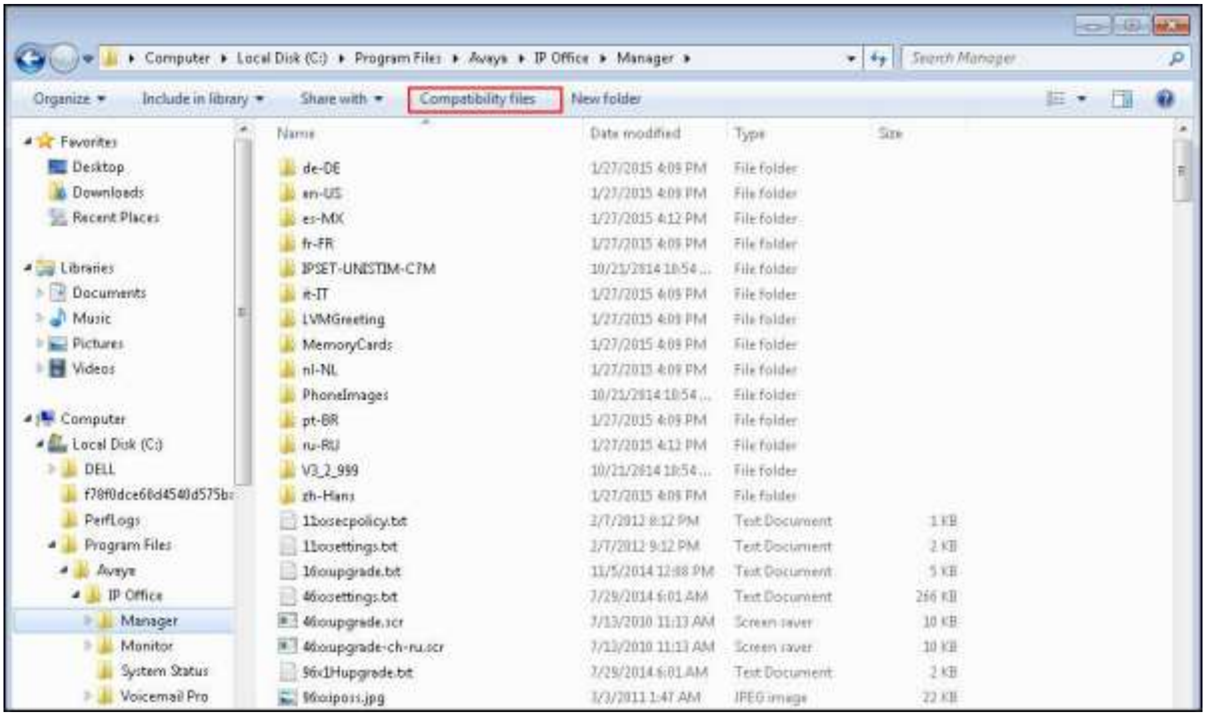
3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 4**. The default template location is **C:\Program Files\Avaya\IP Office\Manager\Templates**.



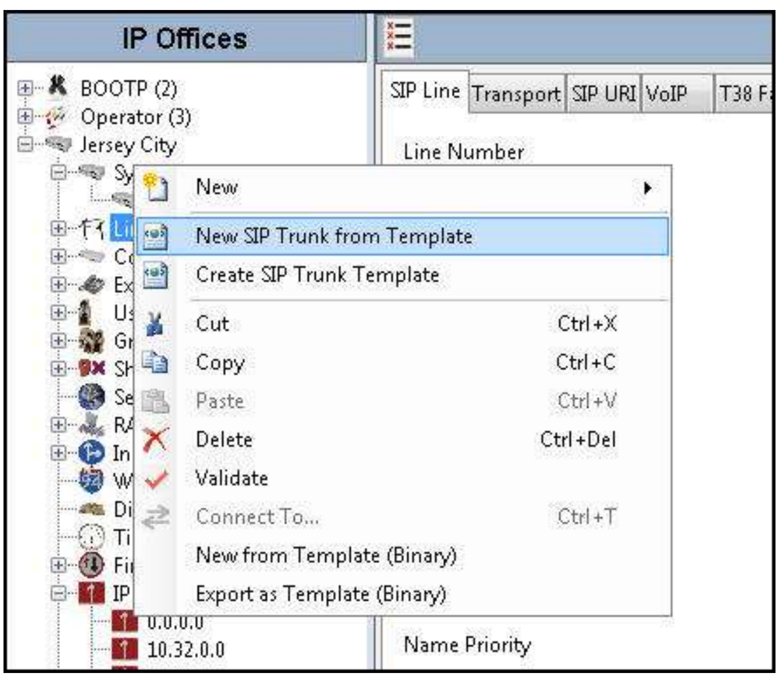
In the pop-up window that appears (not shown), select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. Click **OK** (not shown) to continue.

If preferred, this step may be skipped if the template file is copied directly to the IP Office template directory.

Note: Windows 7 (and later) locks the **Templates** directory in **C:\Program Files\Avaya\IP Office\Manager**, and it cannot be viewed. To enable browsing of the **Templates** directory, open Windows Explorer, navigate to **C:\Program Files\Avaya\IP Office\Manager** (or **C:\Program Files (x86)\Avaya\IP Office\Manager**), and then click on the **Compatibility files** option shown below. The **Templates** directory and its contents can then be viewed.



4. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then select **New SIP Trunk from Template**.



In the subsequent **Template Type Selection** pop-up window, select ***WorldNet*** from the **Service Provider** drop-down list as shown below. This selection corresponds to parts of the template file name as specified in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.

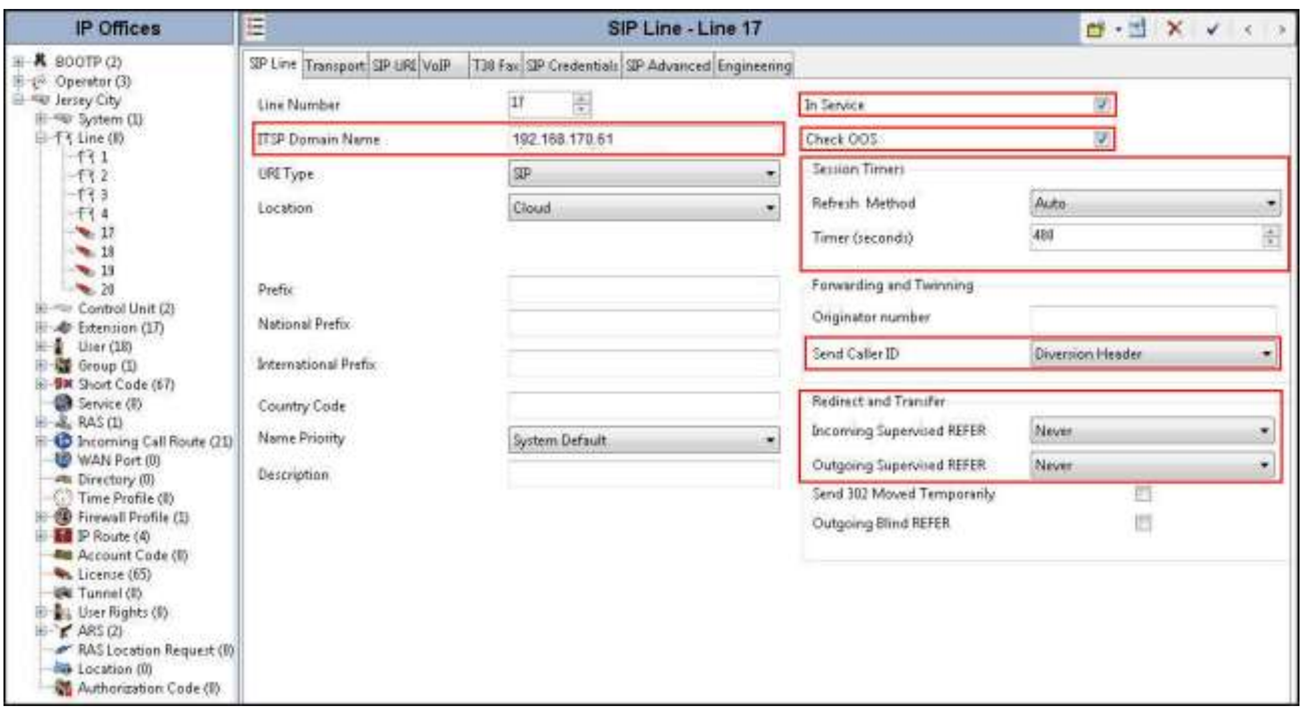
Note that the newly created SIP Line may not immediately appear in the Navigation pane until the configuration was saved, closed and reopened in IP Office Manager.

5. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.2** through **5.4.8**.

5.4.2. SIP Line – SIP Line Tab

In the **SIP Line** tab of the Details Pane, configure the parameters as shown below:

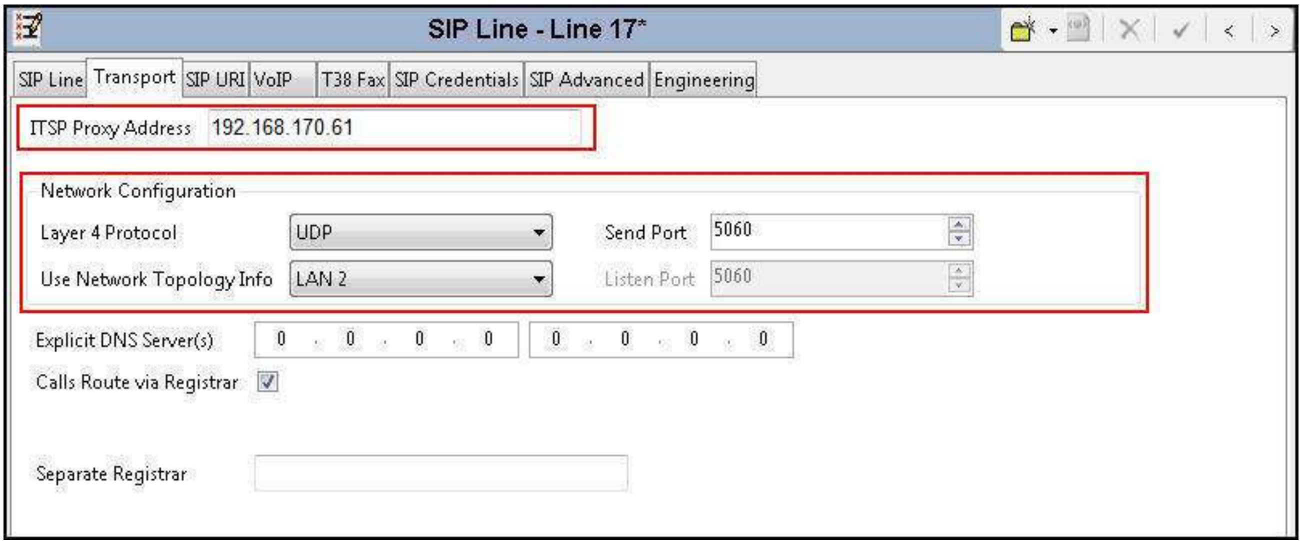
- Set **ITSP Domain Name** to the access interface IP of the WorldNet SIP Trunking Service (provided by WorldNet).
- Check the **In Service** box.
- Check **OOS** box. Avaya IP Office will check the SIP OPTIONS response from the far end to determine whether to keep the SIP Line in service.
- In the **Session Timers** section, set **Method for Session Refresh** to ***Auto***. With this setting Avaya IP Office will send UPDATE messages for session refresh if the other party supports UPDATE. If UPDATE is not supported, re-INVITE messages are sent. Set **Timer (seconds)** to a desired value. Avaya IP Office will send out session refresh UPDATE or re-INVITE at the specified intervals (half of the specified value).
- In the **Forwarding and Twinning** section, set **Send Caller ID** to ***Diversion Header***. With this setting and the related configuration in **Section 5.2.4**, Avaya IP Office will include the Diversion Header for calls that are redirected via Mobile Twinning out the SIP Line to the PSTN. It will also include the Diversion Header for calls that are forwarded out the SIP Line.
- Under **Redirect and Transfer**, select ***Never*** for **Incoming Supervised REFER** and **Outgoing Supervised REFER**. WorldNet does not support use of the REFER method for off-net call transfer.



5.4.3. SIP Line – Transport Tab

Navigate to the **Transport** tab and set the following:

- Set the **ITSP Proxy Address** to the access interface IP of the WorldNet SIP Trunking Service (provided by WorldNet).
- Set the **Layer 4 Protocol** to ***UDP***.
- Set **Use Network Topology Info** to the network port used by the SIP line to access the far-end as configured in **Section 5.2.1**.
- Set the **Send Port** to ***5060***.



5.4.4. SIP Line – SIP URI Tab

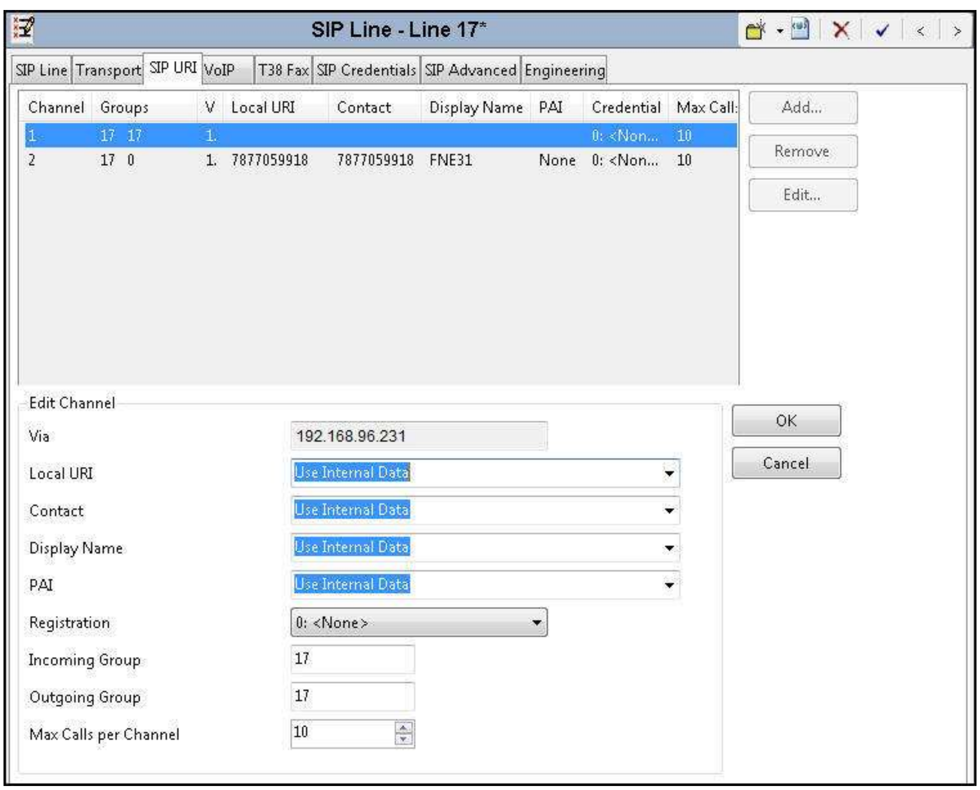
Select the **SIP URI** tab to create or edit a SIP URI entry. A SIP URI entry matches each incoming number that Avaya IP Office will accept on this line. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane.

For the compliance test, two SIP URI entries were created to match the DID number assigned to

- Each Avaya IP Office user (H.323 and SIP endpoints), the fax machine (analog endpoint), as well as the Voicemail messaging application (Voicemail Pro)
- The Mobile Call Control application (see **Section 5.9**)

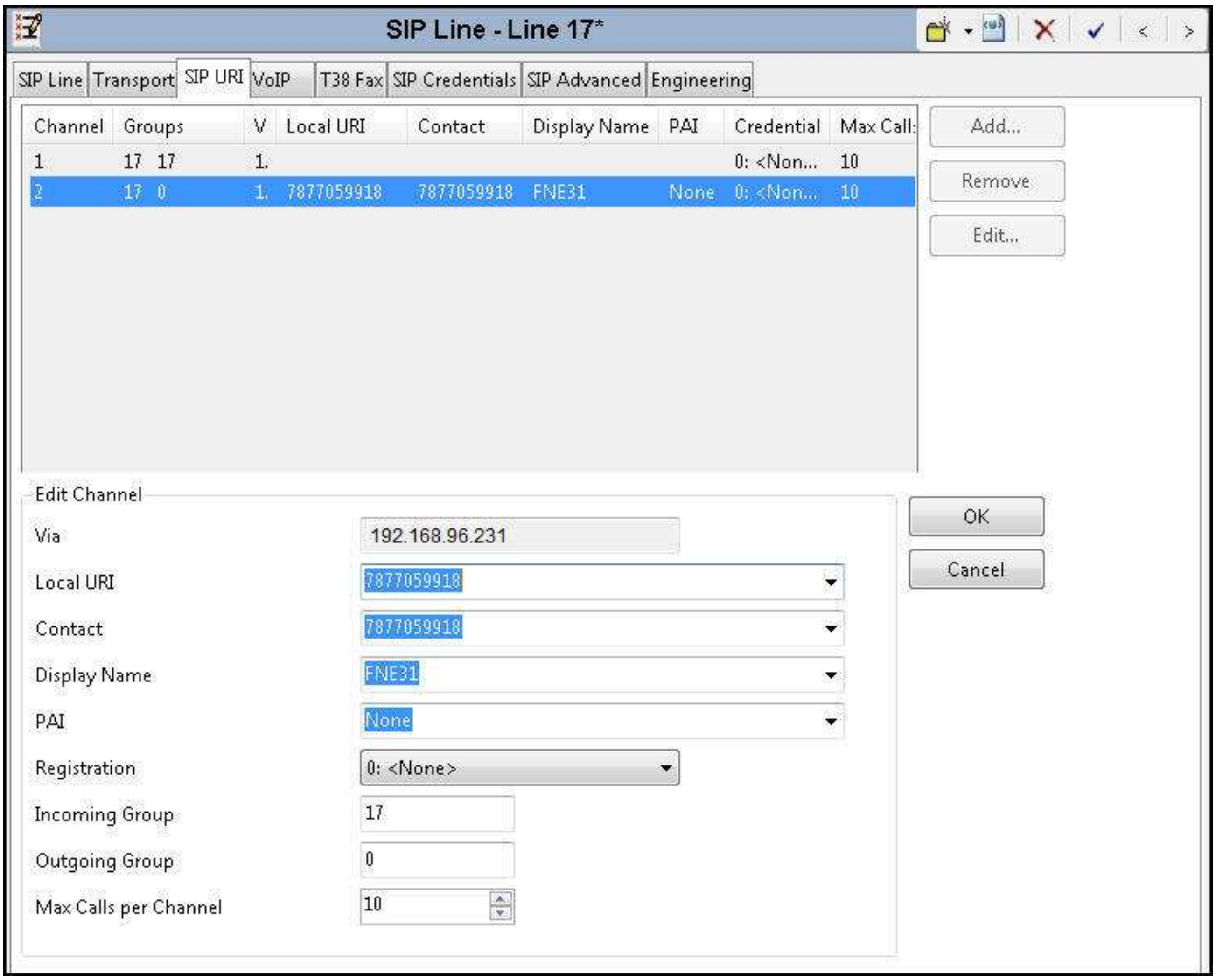
The screen below shows the edit window for the pre-configured SIP URI entry for matching inbound calls to Avaya IP Office users, fax endpoint, and the Voicemail messaging application.

- Set **Local URI** to ***Use Internal Data***. This setting allows calls on this line whose SIP URI matches the **SIP Name** set on the **SIP** tab of any **User** as shown in **Section 5.6**, or the **SIP Name** as set in the **SIP Settings** area of the System **Voicemail** tab as shown in **Section 5.2.2**.
- Set **Contact**, **Display Name** and **PAI** to ***Use Internal Data***.
- Select ***0: <None>*** for **Registration**.
- 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. For the compliance test, the same incoming and outgoing group number ***17*** was specified.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls allowed.



The screen below shows the edit window for the pre-configured SIP URI entry for matching inbound calls to the Mobile Call Control application (see **Section 5.9**). This entry was necessary since the DID number assigned to the Mobile Call Control application was not configured elsewhere for matching the incoming call Request URI. Without this SIP URI entry, the Avaya IP Office would have responded to an incoming call to the DID meant for the Mobile Call Control application with a "404 Not Found" status message and the call would have failed.

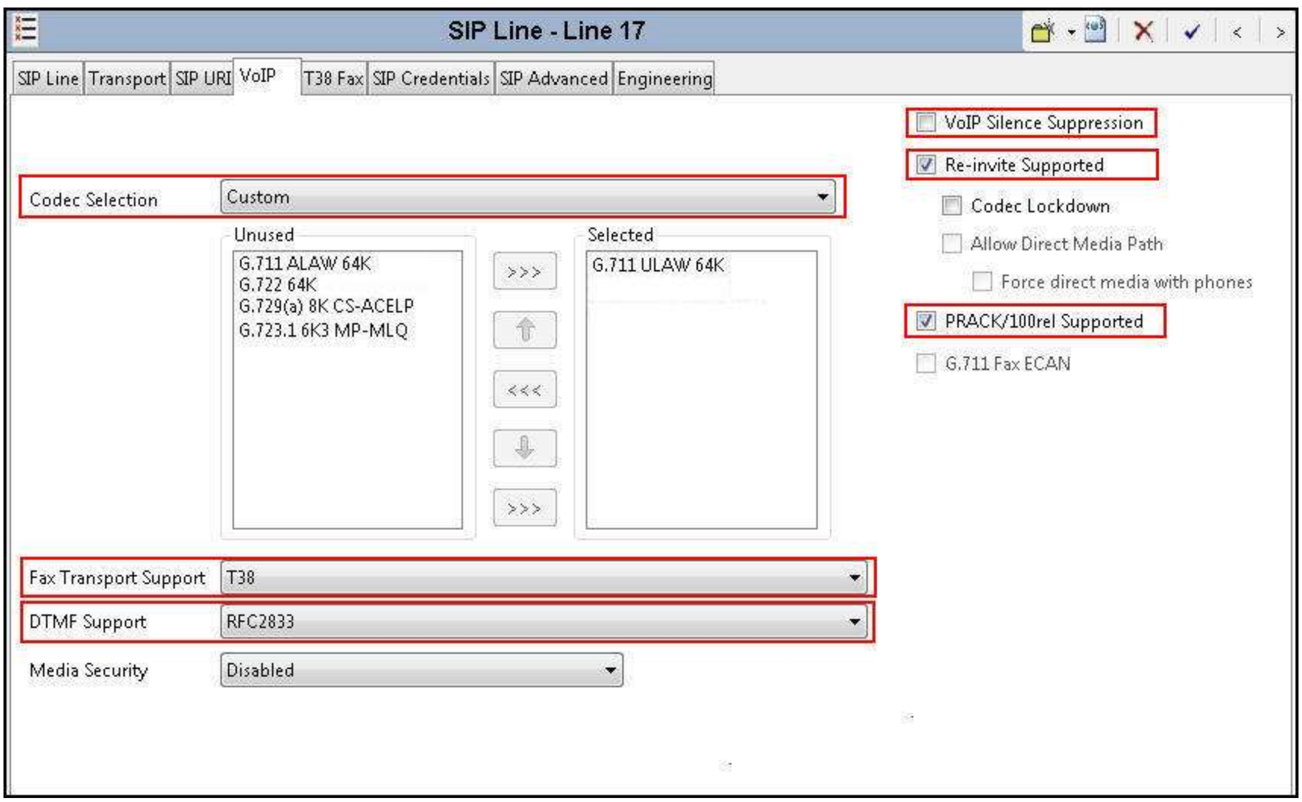
The number ***7877059918*** entered for the **Local URI** field will be configured in the Incoming Call Route in **Section** Error! Reference source not found. to deliver the call to the Mobile Call Control application. Note the settings for **Contact**, **Display Name** and **PAI** are different than the settings for these same fields in the above SIP URI entry for the Avaya IP Office users. Also note the setting for **Outgoing Group**: a setting of ***0*** means no outgoing group is configured with this SIP URI entry since it is used only for mapping incoming calls to the Mobile Control Application.



5.4.5. SIP Line – VoIP Tab

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

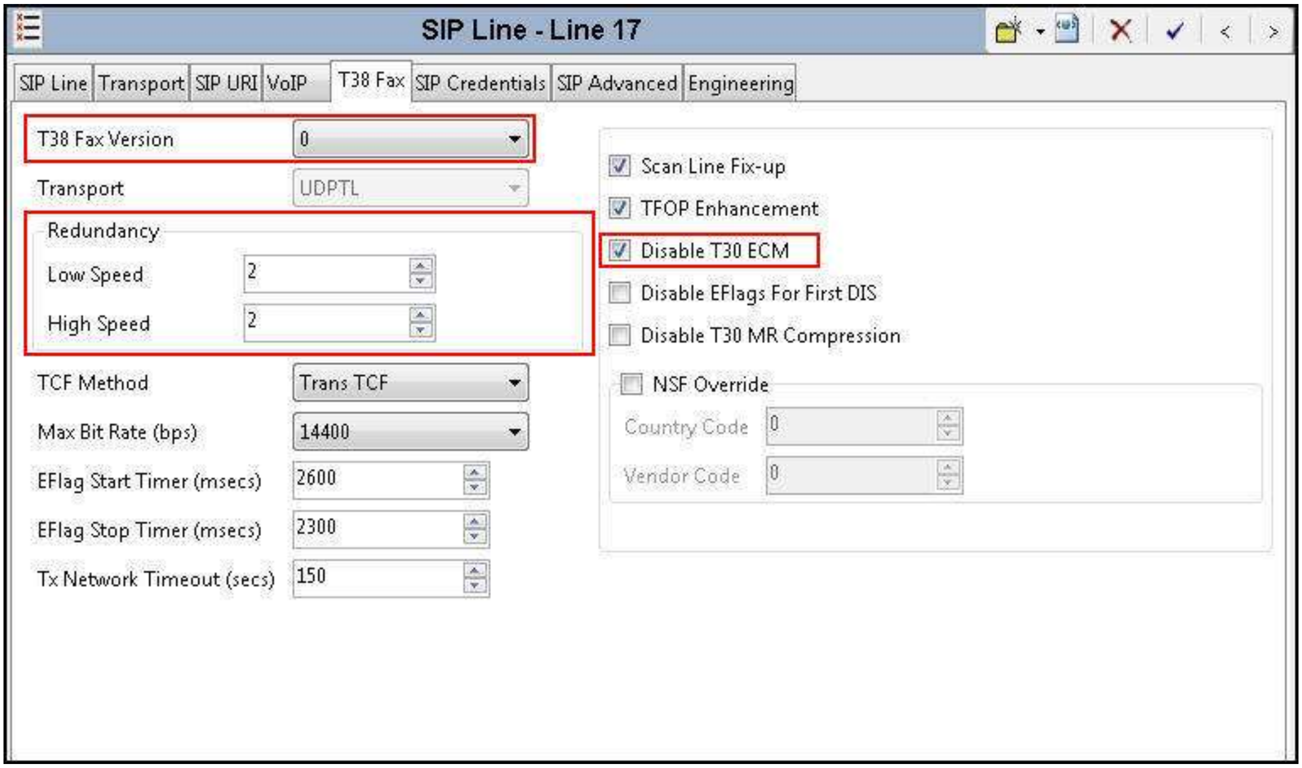
- Select ***Custom*** for **Codec Selection**.
- Choose ***G.711 ULAW 64K*** from the **Unused** box and move the selections to the **Selected** box. This selection was made to be consistent with the codec list as contained in the SDP of inbound call INVITE messages from WorldNet. With this codec setting, both inbound and outbound calls would use G.711u for media. During compliance testing, the G.729a codec was also tested. See the item **G.729a Codec** in the observation/limitation list contained in **Section 2.2** for more information on using G.729a.
- Select ***T38*** for **Fax Transport Support** to direct Avaya IP Office to use T.38 for fax. When testing fax in the G.711u pass-through mode (treating fax as regular voice call with best effort) during compliance testing, this field was set to ***G.711***.
- Select ***RFC2833*** for **DTMF Support**. This directs Avaya IP Office to send DTMF tones as out-band RTP events as per RFC2833.
- Uncheck the **VoIP Silence Suppression** option box.
- Check the **Re-invite Supported** option box. When enabled, re-INVITE can be used during a call session to change the characteristics of the session including codec re-negotiation.
- Check the **PRACK/100rel Supported** option box. This setting enables support by Avaya IP Office for the PRACK (Provisional Reliable Acknowledgement) message on SIP trunks.



5.4.6. SIP Line – T38 Fax

The settings on this tab configures T.38 fax parameters and are only accessible if **Re-invite Supported** was checked and either ***T38*** or ***T38 Fallback*** was selected for **Fax Transport Support** in the **VoIP** tab in **Section 5.4.5**.

The screen below shows the settings used for the compliance test. The **T38 Fax Version** is set to ***0***. In the **Redundancy** area, **Low Speed** and **High Speed** are set to ***2***. Check the **Disable T30 ECM** option for disabling T.30 Error Correction Mode to match the ECM configuration on the WorldNet side. All other values are left at default.



5.4.7. SIP Line – SIP Credentials Tab

SIP Credentials are used to register the SIP Trunk with a service provider that requires SIP Registration. SIP Credentials are also used to provide the required information for Digest Authentication of outbound calls. SIP Credentials are unique per customer and therefore customers must contact the service provider to obtain the proper registration and/or Digest Authentication credentials for their deployment.

For the compliance test, static IP authentication was configured by WorldNet on the test circuit, therefore this tab needs not to be visited.

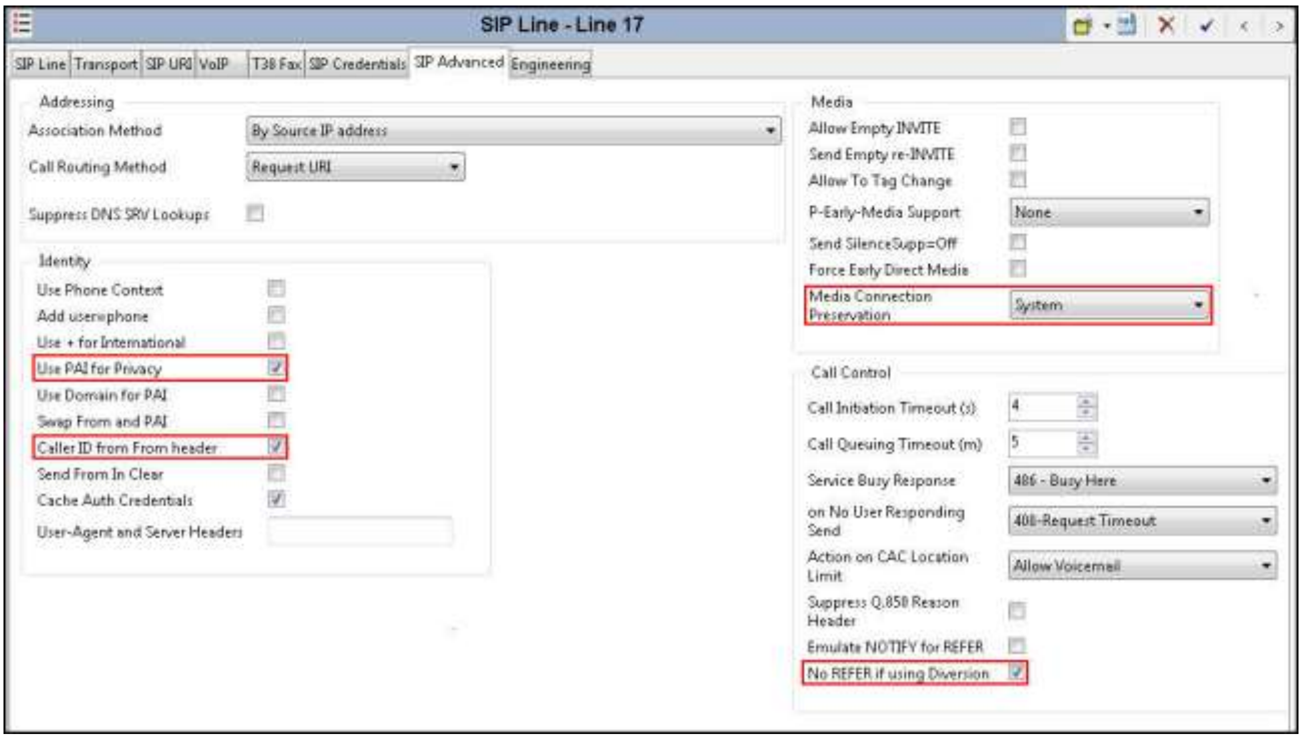
5.4.8. SIP Line – SIP Advanced Tab

Select the **SIP Advanced** tab to configure advanced SIP Line parameters.

In the **Identity** area, the **Use PAI for Privacy** box is checked for Avaya IP Office to use the P-Asserted-Identity (PAI) SIP header for privacy-requested outbound calls. With this configuration, Avaya IP Office will populate the From and Contact headers of the anonymous outbound call INVITE with "anonymous" as the URI user part, but include the normal calling user information in the PAI header. The **Caller ID from From header** box is checked for Avaya IP Office to use the Caller ID information in the From SIP header rather than the PAI or the Contact SIP header for inbound calls.

In the **Media** area, select ***System*** for **Media Connection Preservation** to allow established calls to continue despite brief network failures.

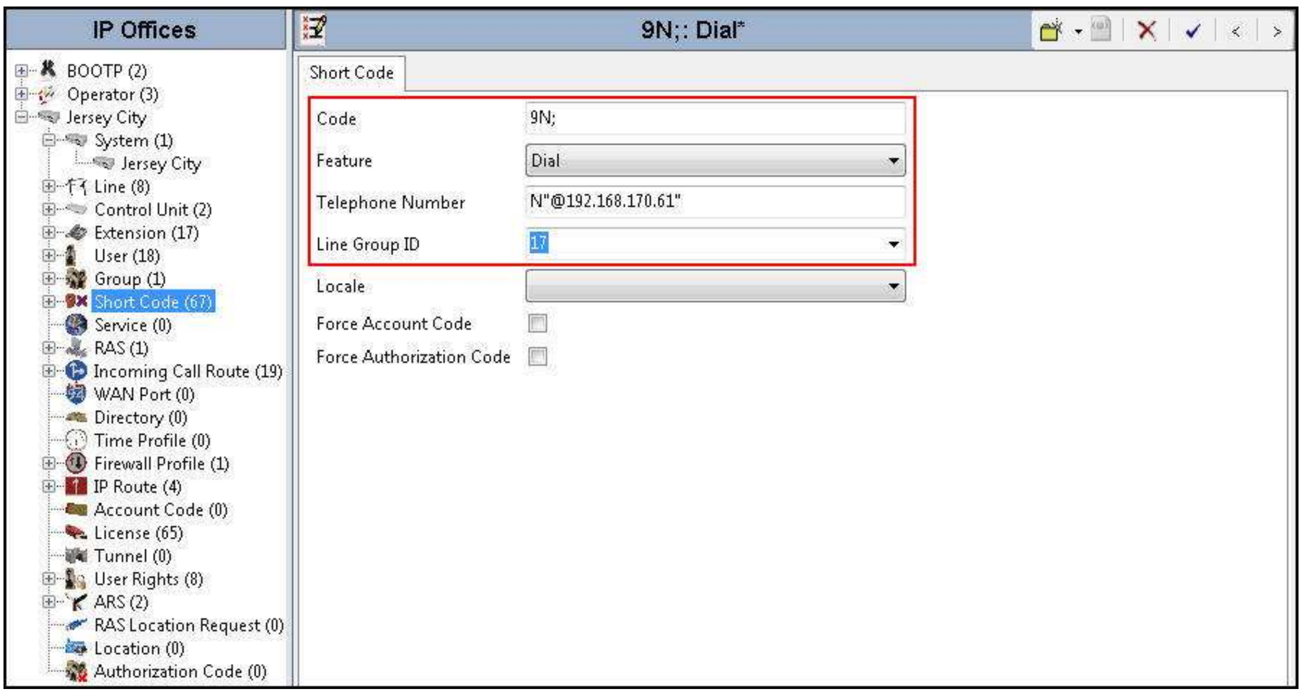
In the **Call Control** area, **No REFER if using Diversion** is checked to prevent Avaya IP Office from using the SIP REFER method on call scenarios that use the Diversion SIP header (e.g., off-net call forward or outbound call to mobile twinning number).



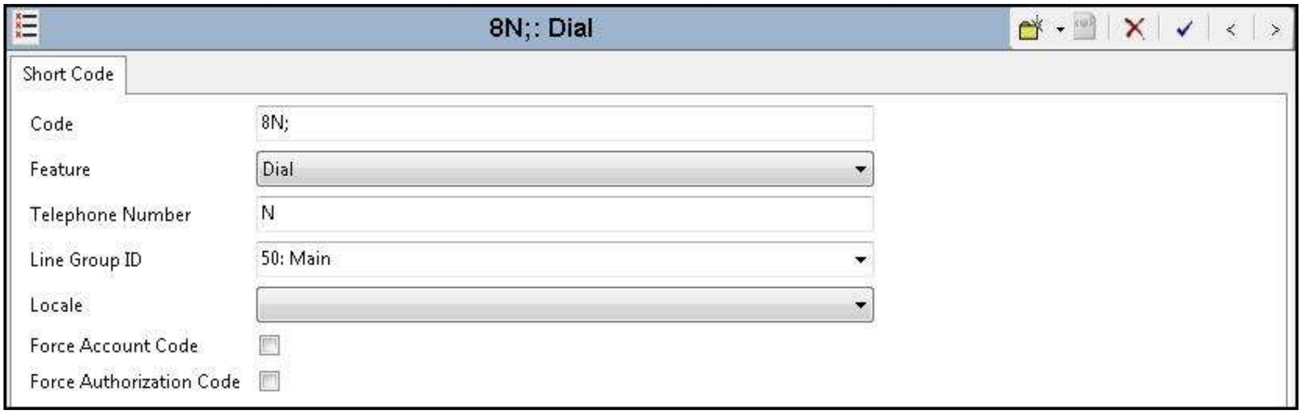
5.5. Short Code

Define a short code to route outbound calls to the SIP Line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New** (not shown). On the **Short Code** tab in the Details Pane, configure the parameters as shown below:

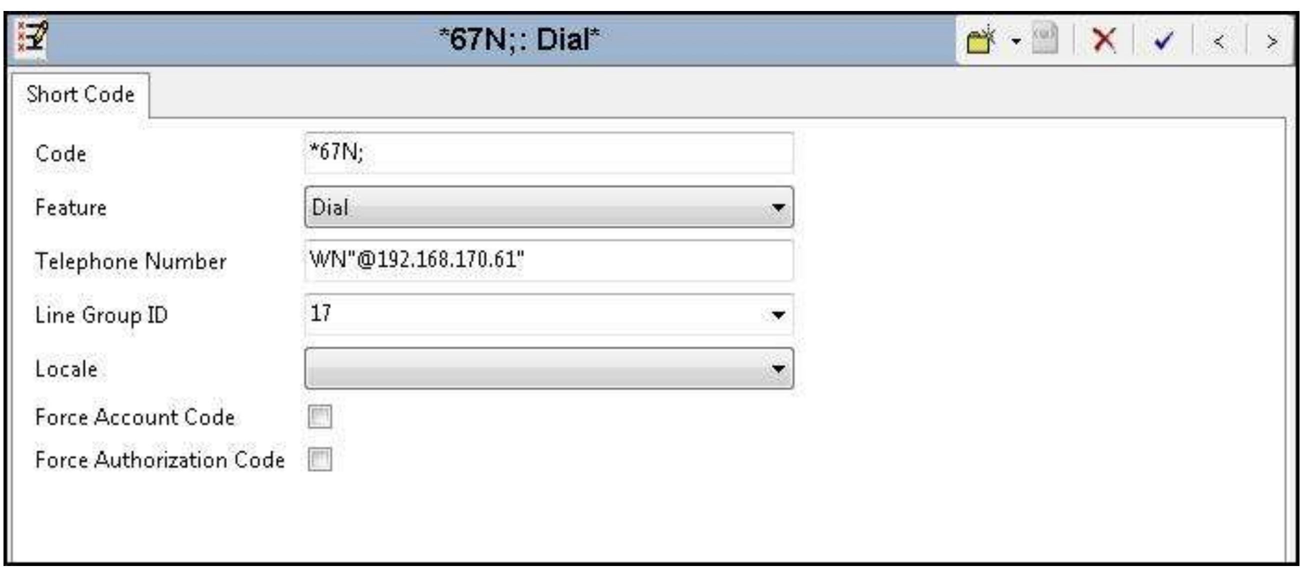
- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. The ***9N;*** short code, used for the compliance test, will be invoked when the user dials 9 followed by any number.
- Set **Feature** to ***Dial***. This is the action that the short code will perform.
- Set **Telephone Number** to ***N"@ 192.168.170.61"***. This field is used to construct the Request URI and the To header in the outgoing SIP INVITE message. The value ***N*** represents the number dialed by the user. The text string following the **@** sign is the access interface IP address of the WorldNet SIP Trunking Service (provided by WorldNet).
- Set the **Line Group Id** to the **Outgoing Group** number defined on the **SIP URI** tab on the SIP Line in **Section 5.4.4**. This short code will use this line group when placing the outbound calls.



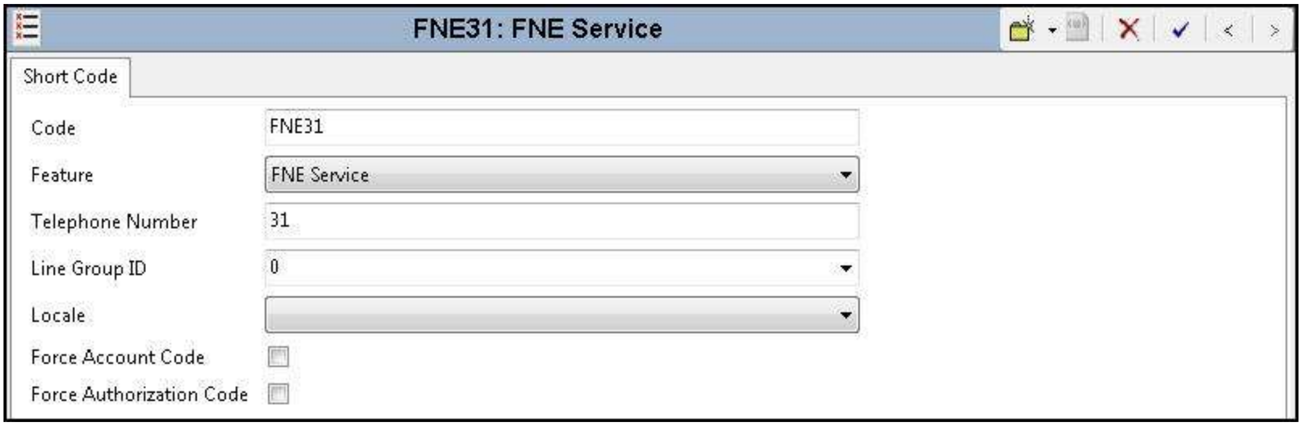
The simple **9N;** short code illustrated above does not provide a means of alternate routing if the configured SIP Line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the dialed digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. In the screen below, the short code ***8N;*** is illustrated for access to ARS. When the Avaya IP Office user dials 8 plus any number ***N***, rather than being directed to a specific **Line Group Id**, the call is directed to ***50: Main***, configurable via ARS. See **Section 5.8** for example ARS route configuration.



Optionally, add or edit a short code 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 compliance test, when a user dialed *67 plus the destination number, Avaya IP Office would include the user's telephone number (DID number assigned to the user) in the **P-Asserted-Identity** (PAI) header, populate the URI user part with "anonymous" in the From and Contact headers, and include the **Privacy: id** header in the outbound INVITE message. Consequently WorldNet would prevent presentation of the caller id to the called PSTN destination.



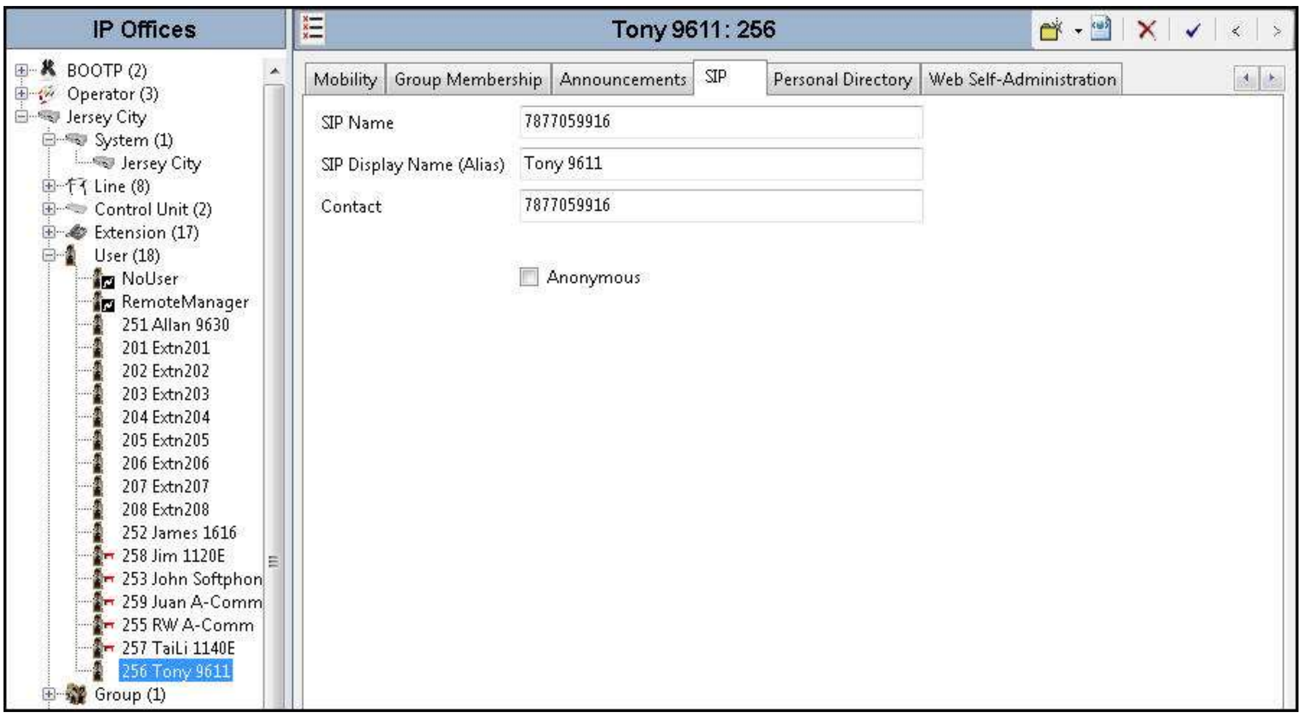
For completeness, the short code ***FNE31*** for the Mobile Call Control application is shown below. See **Section 5.7** for routing incoming calls to this application to receive internal IP Office dial tones. See **Section 5.9** for configuration to enable this mobility feature.



5.6. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP Line. 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 ***Tony 9611*** at extension 256. Select the **SIP** tab in the Details Pane. The **SIP Name** and **Contact** are set to one of the DID numbers assigned by WorldNet. The **SIP Display Name (Alias)** can optionally be configured with a descriptive text string. The value entered for the **Contact** field will be used in the Contact header for outgoing SIP INVITE to the service provider. The value entered for the **SIP Name** is used for mapping inbound calls, and as the user part of the SIP URI in the From header for outgoing SIP trunk calls.

If outbound 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 (or alternatively use the ***67N;** short code as defined in **Section 5.5**).

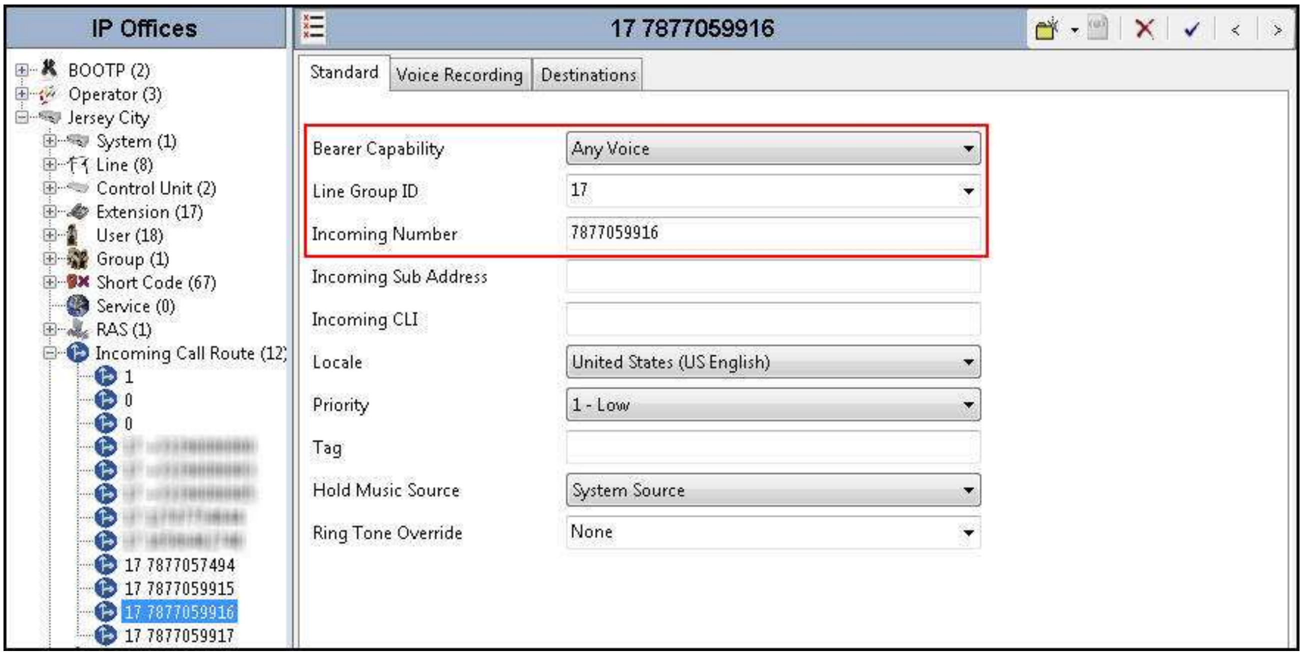


The following screen shows the similar SIP settings for an analog extension user for fax:

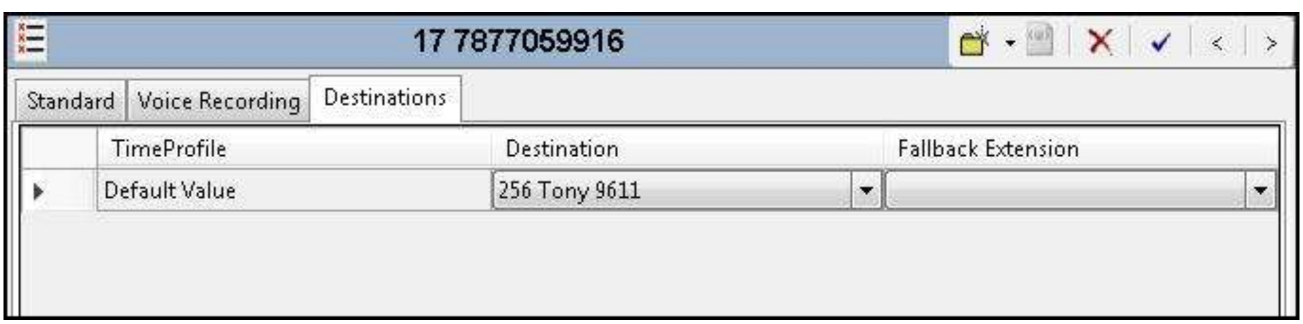
5.7. Incoming Call Route

An incoming call route maps an inbound DID number on a specific line to an internal destination. This procedure should be repeated for each DID number provided by the service provider. To create an incoming call route, right-click **Incoming Call Route** in the Navigation Pane and select **New** (not shown). On the **Standard** tab in the Details Pane, enter the parameters settings as shown below:

- Set the **Bearer Capacity** to ***Any Voice***.
- Set the **Line Group Id** to the **Incoming Group** of the SIP Line defined in **Section 5.4.4**.
- Set the **Incoming Number** to the incoming DID number on which this route should match.



On the **Destinations** tab, select the destination from the pull-down list of the **Destination** field. In this example, incoming calls to 7877059916 on Incoming Group 17 are to be routed to the user "Tony 9611" at extension 256.



The screen below shows calls routed to the IP Office fax endpoint which is an analog extension (Extn 208).

The screen below shows calls routed to IP Office Voicemail Pro for message retrieval. Note that the DID 7877059919 was assigned to Voicemail in **Section 5.2.2**.

The following **Destinations** tab for an incoming call route contains the **Destination** “FNE31” entered manually. The name “FNE31” is the short code for accessing the Mobile Call Control application. An incoming call to 7877059918 from an IP Office user’s twinned mobile phone will be delivered directly to an internal dial tone from the Avaya IP Office, allowing the caller to dial call destinations, both internal and external. See **Section 5.9** on configuration to enable the Mobile Call Control application.

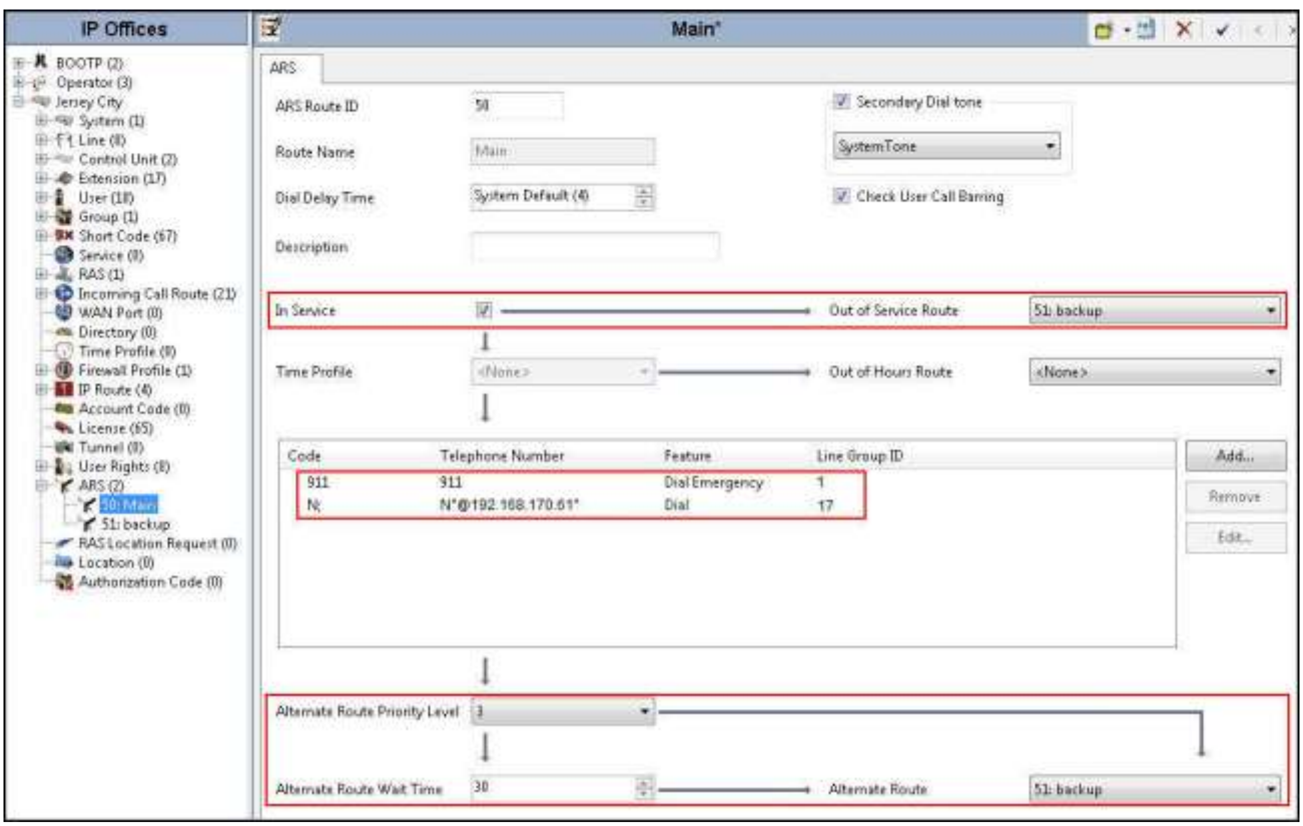
5.8. ARS and Alternate Routing

While detailed coverage of Automatic Route Selection (ARS) is beyond the scope of these Application Notes, this section includes basic ARS screen illustration and considerations.

Optionally, ARS can be used to supplement or replace the simple **9N;** short code approach documented in **Section 5.5**. With ARS, secondary dial tone can be provided after the access code, time-based routing criteria can be introduced, and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. ARS also facilitates more specific dialed telephone number matching, enabling immediate routing and alternate treatment for different types of numbers following the access code. For example, if all local and long distance calls should use the SIP Line, but service numbers should prefer a different outgoing line group, ARS can be used to distinguish between the two call patterns.

To add a new ARS route, right-click **ARS** in the Navigation Pane and select **New** (not shown). To view or edit an existing ARS route, expand ARS in the Navigation Pane and select a route name.

The following screen shows a sample ARS configuration for the route named ***50: Main***. The **In Service** parameter refers to the ARS form itself, not the Line Groups that may be referenced in the form. If the **In Service** box is un-checked, calls are routed to the ARS route name specified in the **Out of Service Route** parameter. IP Office short codes may also be defined to allow an ARS route to be disabled or enabled from a telephone. The configurable provisioning of an Out of Service Route and the means to manually activate the Out of Service Route can be helpful for scheduled maintenance or other known service-affecting events for the primary route.

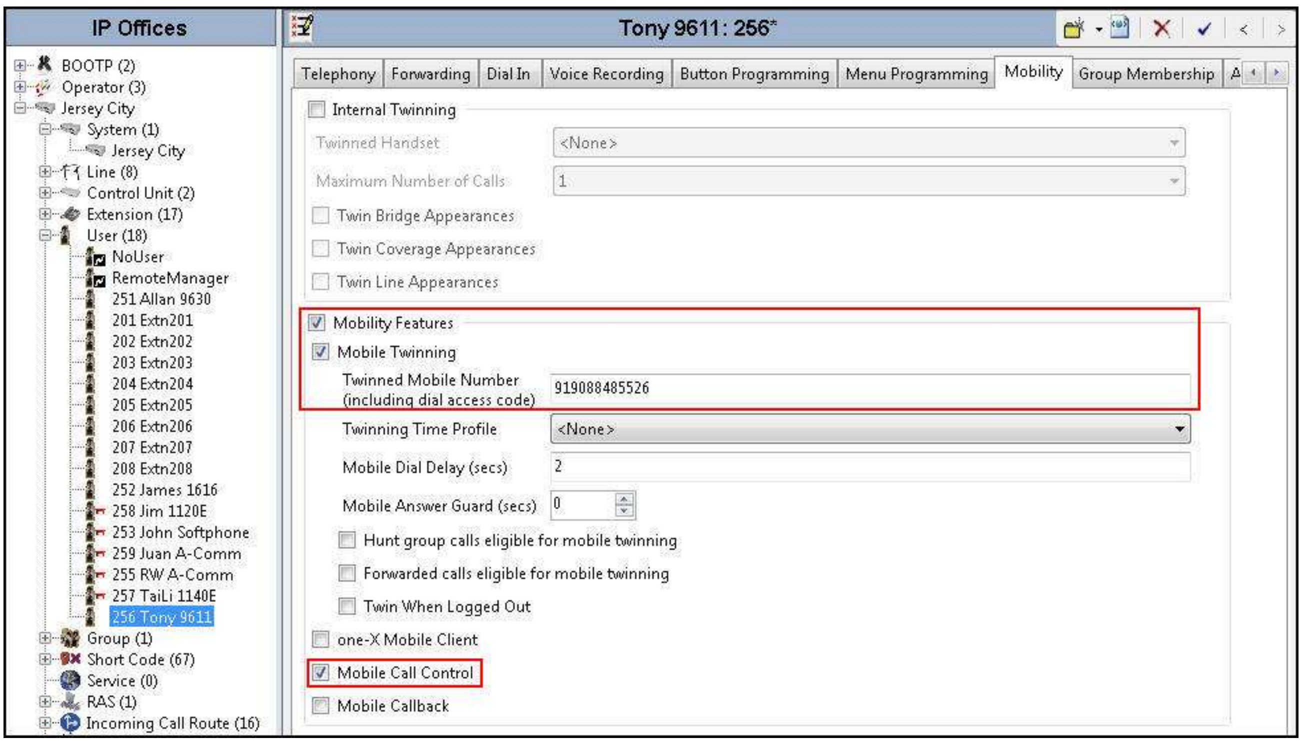


Assuming the primary route is in-service, the number passed from the short code used to access ARS (e.g., **8N;** in **Section 5.5**) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 8 plus any number, the processing for the short code **8N;** would direct the call via ARS to Line Group 17. A short code **911** can be configured to send the emergency call out using Line Group 1 when the user dials "911". If the primary route cannot be used, the call can automatically route to the route name specified in the **Alternate Route** field in the lower right of the screen (***51: Backup***). Since alternate routing is considered a privilege not available to all callers, IP Office can control access to the alternate route by comparing the calling user's priority, configured in the **User** tab of individual users, to the value in the **Alternate Route Priority Level** field.

5.9. Mobility

With Mobility configured for an Avaya IP Office user, an inbound call routed to this user automatically triggers an outbound call to the configured Mobile Twinning number for this user.

The following screen shows the **Mobility** tab for User "Tony 9611" at extension 256. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number for the twinned mobile telephone including the dial access code (short code), in this case ***919088485526*** (short code 9 plus the ensuing twinned mobile number). The **Mobile Call Control** option box is also checked so that an inbound call from the twinned mobile number (9088485526 in this example) to the Mobile Call Control application (see Incoming Call Route to "FNE31" in **Section 5.7**) will be delivered directly to an internal dial tone from the Avaya IP Office, allowing the caller to perform further dialing actions including making calls and activating Short Codes. Other options can be set according to customer requirements.



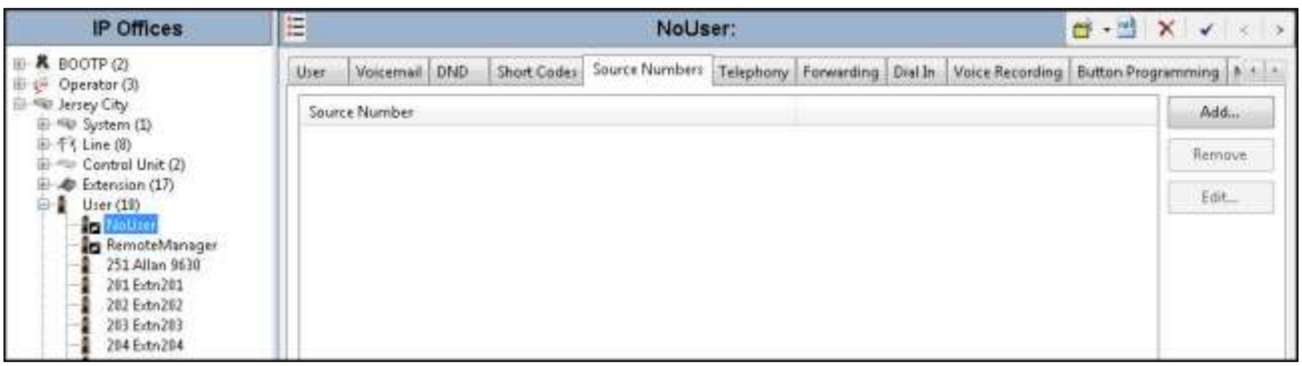
Note that when an inbound call is from the twinned mobile number to the Mobile Call Control application, the caller ID contained in the From header of the incoming INVITE must match the twinned mobile number (without the leading short code digit 9 and the PSTN access code 1), otherwise the Avaya IP Office responds with a "486 Busy Here" message and the caller will hear busy tones.

5.10. SIP Options

Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. By default, Avaya IP Office Release 9.1 sends out OPTIONS every 300 seconds. 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 5.2.1** 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:

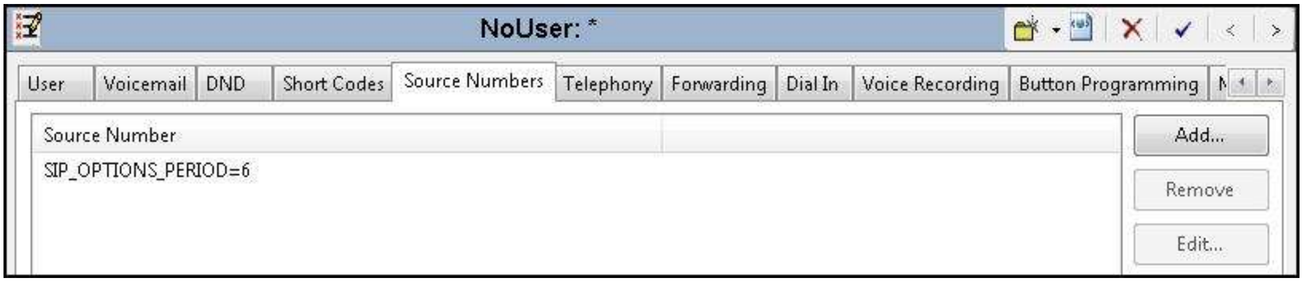
- To use the default value, set **Binding Refresh Time** to 300. OPTIONS will be sent at the 300 second frequency.
- To establish a period of less than 300 seconds, do not define the **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 seconds. The OPTIONS message period will be equal to the **Binding Refresh Time** setting.
- 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** settings.

To configure the **SIP_OPTIONS_PERIOD** parameter, 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_OPTIONS_PERIOD=X***, where ***X*** is the desired value in minutes. Click **OK**.

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

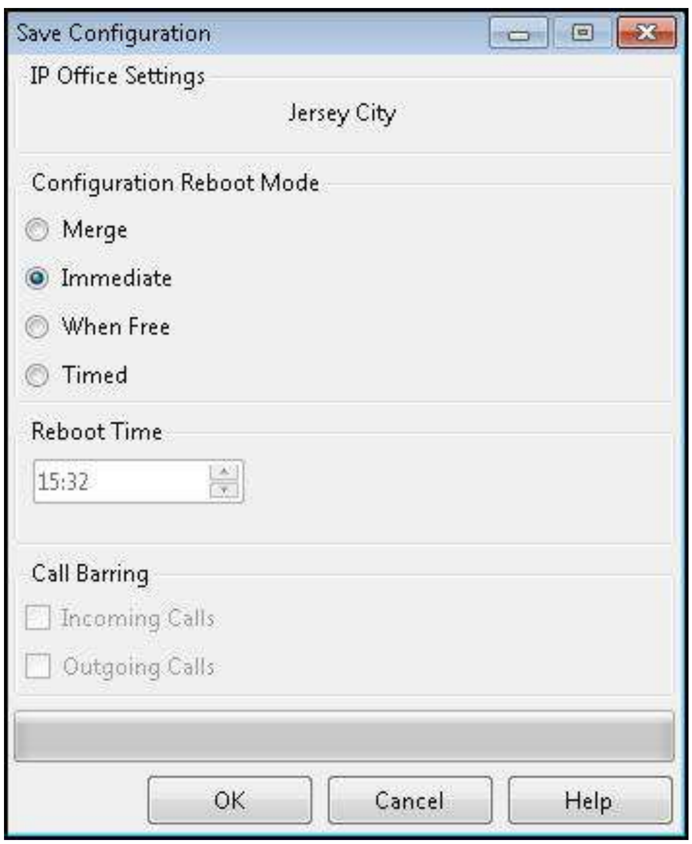


For the compliance test, an OPTIONS period of 2 minutes was desired. The **Binding Refresh Time** was set to ***120*** seconds in **Section 5.2.1**. Thus, there was no need to define **SIP_OPTIONS_PERIOD**.

5.11. Save Configuration

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

The following **Save Configuration** screen will appear, with either **Merge** or **Immediate** automatically selected based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a system reboot or a service disruption. Click **OK** to proceed.



6. WorldNet SIP Trunking Configuration

WorldNet is responsible for the configuration of its SIP Trunking Service. The customer will need to provide the IP address used to reach Avaya IP Office at the enterprise site (i.e., the IP address of the LAN2 port on Avaya IP Office). WorldNet will provide the customer the necessary information to configure Avaya IP Office including:

- The access interface IP address of the WorldNet SIP Trunking Service.
- Transport and port for the WorldNet SIP connection to the enterprise.
- DID numbers to assign to users at the enterprise.
- Supported codecs and their preference order.

7. Verification Steps

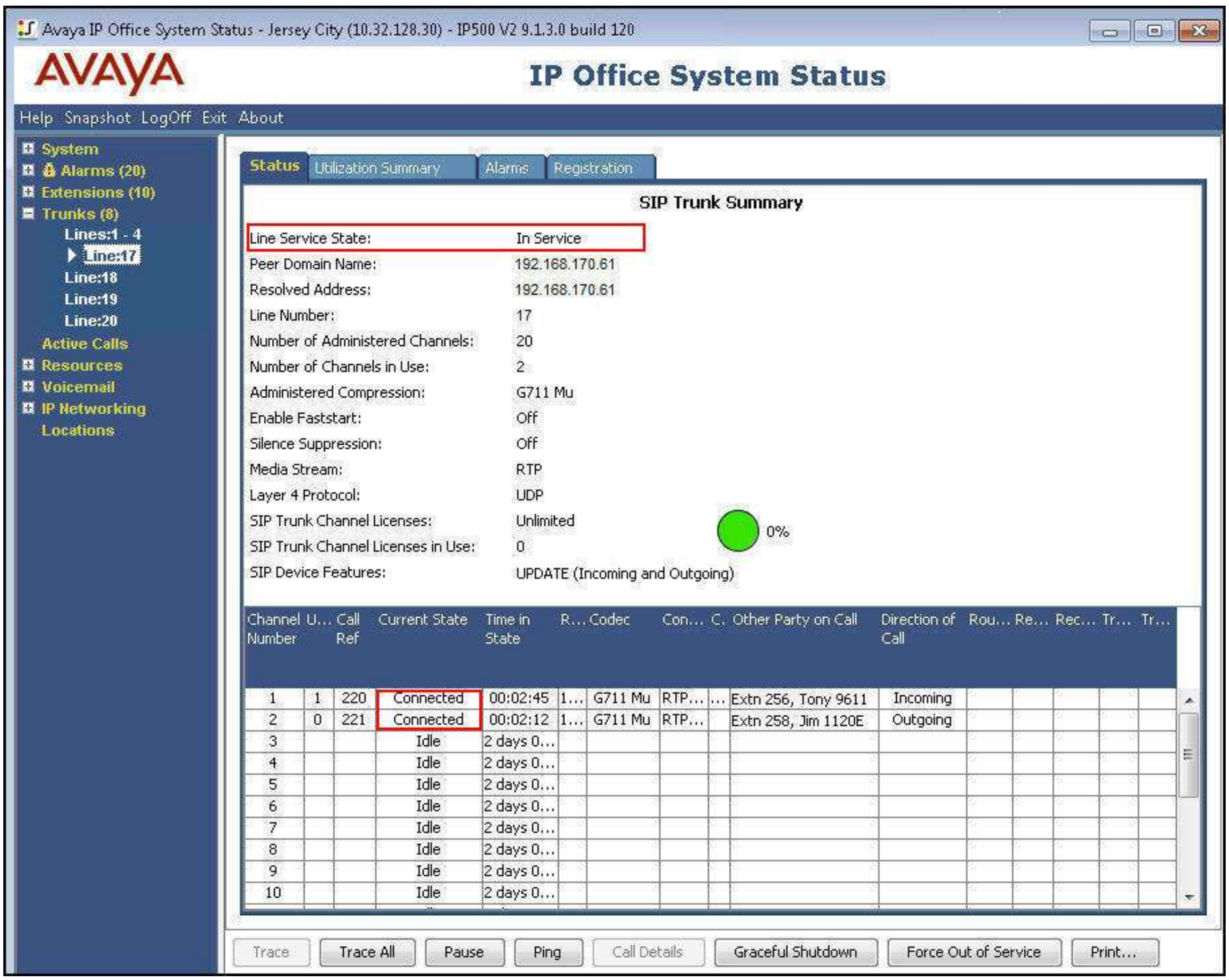
This section provides verification steps that may be performed in the field to verify that the solution is configured properly

7.1. System Status

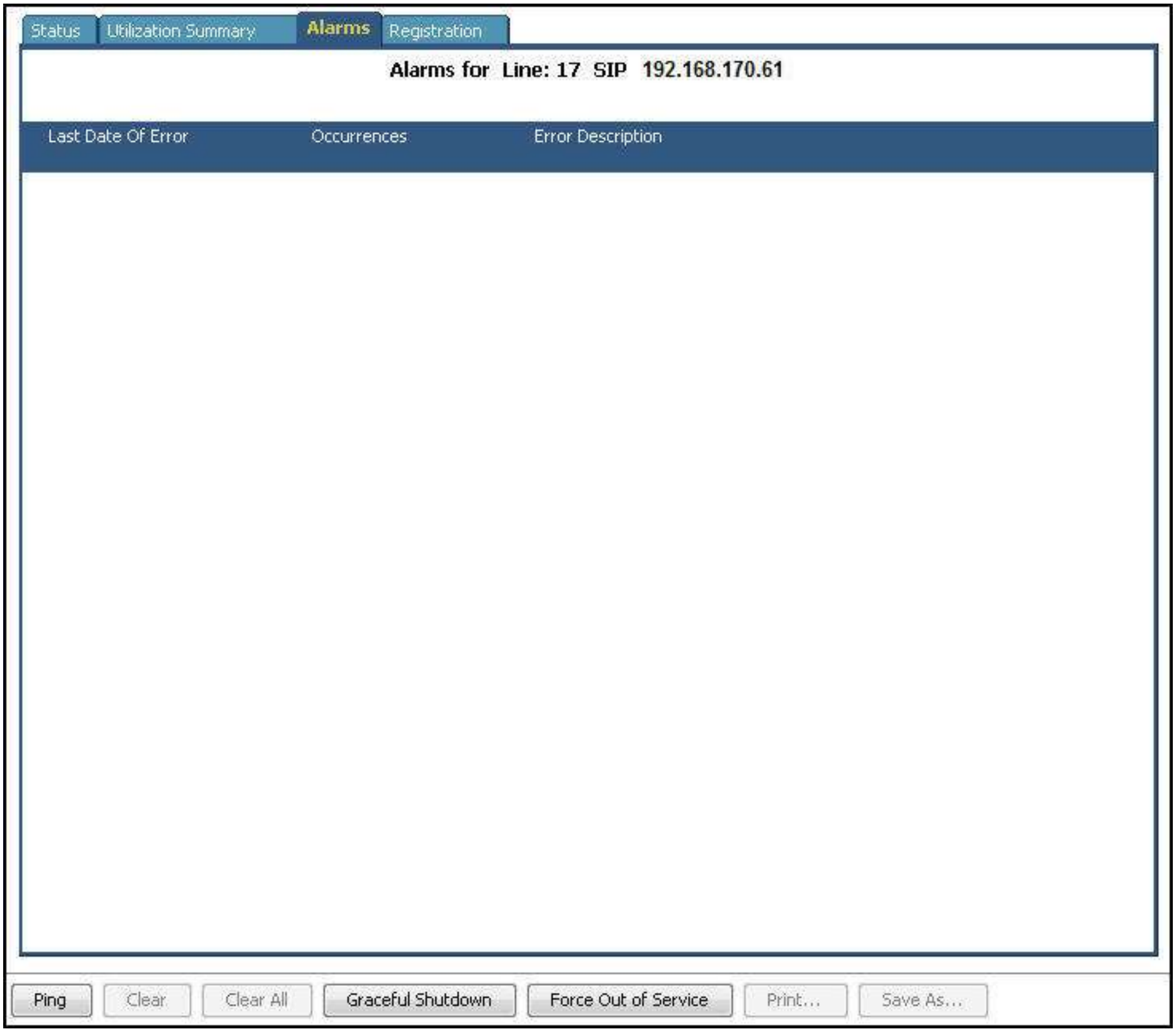
Use the Avaya IP Office System Status application to verify the SIP Line service/channels state and to check alarms:

- Launch the application from **Start → Programs → IP Office → System Status** on the Avaya IP Office Manager PC. Select the SIP Line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify

    - **Line Service State** is **In Service**.
    - **Current State** is **Idle** for channels where no active calls are currently in session; the state should be **Connected** for channels engaged in active calls.

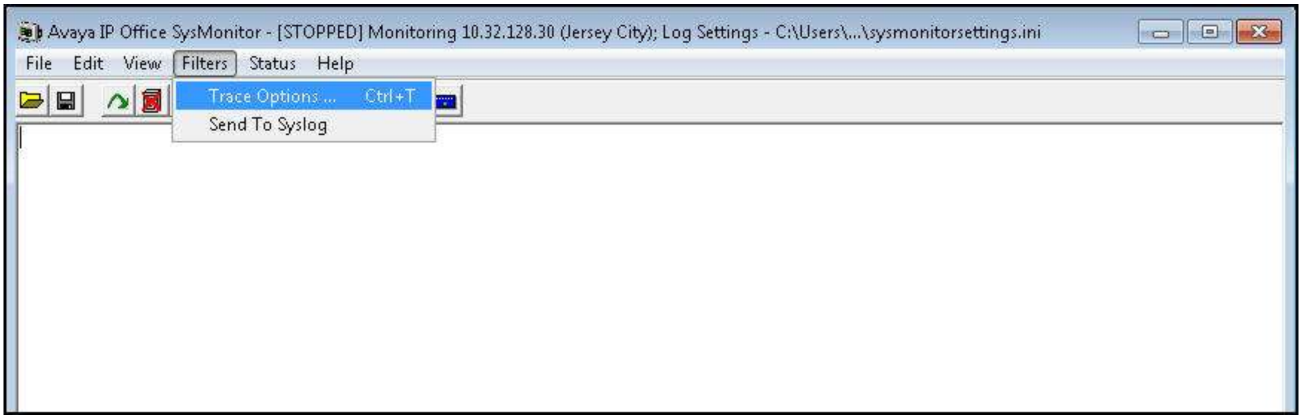


- Select the **Alarms** tab and verify that no alarms are active on the SIP Line.

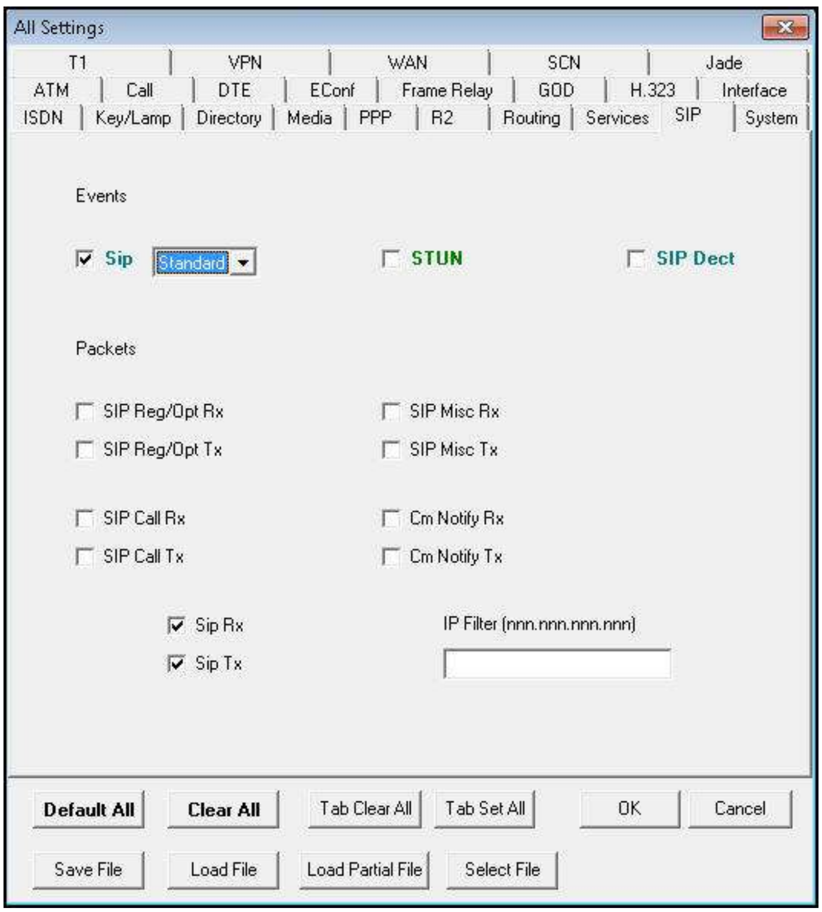


7.2. Monitor

The Monitor application can be used to monitor and troubleshoot Avaya IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor** on the Avaya IP Office Manager PC. The application allows the monitored information to be customized. To customize, select **Filters → Trace Options …** as shown below:



The trace All Settings screen will appear. The **SIP** tab allows configuration of SIP monitoring. In this example, ***Standard*** Sip **Events** and the **Sip Rx** and **Sip Tx** boxes are checked.



8. Conclusion

The WorldNet SIP Trunking Service passed compliance testing with Avaya IP Office Release 9.1. These Application Notes describe the procedures necessary to configure the connectivity between Avaya IP Office Release 9.1 and the WorldNet SIP Trunking Service as shown in **Figure 1**. Test results and observations are noted in **Section 2.2**.

9. Additional References

[1] *IP Office™ Platform 9.1, Deploying Avaya IP Office™ Platform IP500 V2,* Document Number 15-601042, Issue 30m, August 2015.

[2] *Administering Avaya IP Office™ Platform with Manager*, Release 9.1, Issue 10.23, August 2015.

[3] *IP Office ™ Platform 9.1, Administering Avaya IP Office™ Platform Voicemail Pro*, Document Number 15-601063, Issue 10f, July 2015.

[4] *IP Office™ Platform 9.1, Using IP Office System Monitor*, Document Number 15-601019, Issue 06e, May 2015.

[5] *IP Office™ Platform 9.1, Using Avaya IP Office™ Platform System Status*, Document Number 15-601758, Issue 10f, August 2015.

Additional IP Office documentation can be found at http://marketingtools.avaya.com/knowledgebase/ipoffice/general/rss2html.php?XMLFILE=manuals.xml&TEMPLATE=pdf_feed_template.html.

Product documentation for the WorldNet SIP Trunking Service is available from WorldNet. See **Section 2.3** on how to contact WorldNet.

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