Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Level 3 SIP Trunking with Avaya IP Office Release 7.0 – Issue 1.0

Abstract

These Application Notes describes the steps to configure Session Initiation Protocol (SIP) Trunking between Level 3 and Avaya IP Office Release 7.0.

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

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

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Level 3 and Avaya IP Office Release 7.0.

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

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

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

2. General Test Approach and Test Results

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

The Level 3 SIP Trunk Service passed compliance testing.

2.1. Interoperability Compliance Testing

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

- Response to SIP OPTIONS queries
- Incoming PSTN calls to various phone types including H.323, SIP, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls to various phone types including H.323, SIP, digital, and analog telephones at the enterprise. All outgoing PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Inbound and outbound PSTN calls to/from soft clients. Avaya IP Office supports two soft clients: Avaya IP Office Phone Manager and Avaya IP Office Softphone. Avaya IP Office Phone Manager supports two modes (PC softphone and telecommuter). Both clients in each supported mode were tested.
- Various call types including: local, long distance, outbound toll-free, and local directory assistance.
- Codec G.711MU and G.729A.

- T.38 Fax
- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- Voicemail navigation using DTMF for inbound and outbound calls.
- User features such as hold and resume, transfer, and conference.
- Off-net call forwarding and twinning.

2.2. Test Results

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

- Inbound toll-free and outbound emergency calls (911) are supported but were not tested as part of the compliance test.
- **Mobile Twinning Caller ID –** Level 3 sends the number in the FROM header as the caller ID to the mobile device. If this number is not one that is assigned to the SIP trunk, they require a Diversion or PAI header to have a number that is assigned. The software release of the IP Office tested sends the original caller ID in the Diversion header not the FROM header. So the mobile device shows the caller ID of the IP Office extension and not the original caller. The IP Office product team is evaluating IPOFFICE-16075 documenting this issue.
- **Off-net Call Forward Caller ID** – Similar to the limitation mentioned above about Mobile Twinning, calls that are forwarded across the SIP trunk displays the caller ID of the extension being forwarded and not that of the original caller. The IP Office product team is evaluating IPOFFICE-16076 documenting this issue.

2.3. Support

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

3. Reference Configuration

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

Located at the enterprise site is an Avaya IP Office 500. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public network. Endpoints include an Avaya 1616 IP Telephone (with H.323 firmware), an Avaya 1140E IP Telephone (with SIP firmware), an Avaya 9641 IP Telephone (with H.323 firmware), an Avaya 9611 IP Telephone (with H.323 firmware), an Avaya IP Office Phone Manager, an Avaya IP Office Softphone, an Avaya 1408 Digital Telephone, a legacy Nortel T7316E and an Avaya 6210 Analog Telephone. The site also has a Windows 2003 Server running Avaya Voicemail Pro for voicemail and running Avaya IP Office Manager to configure the Avaya IP Office.

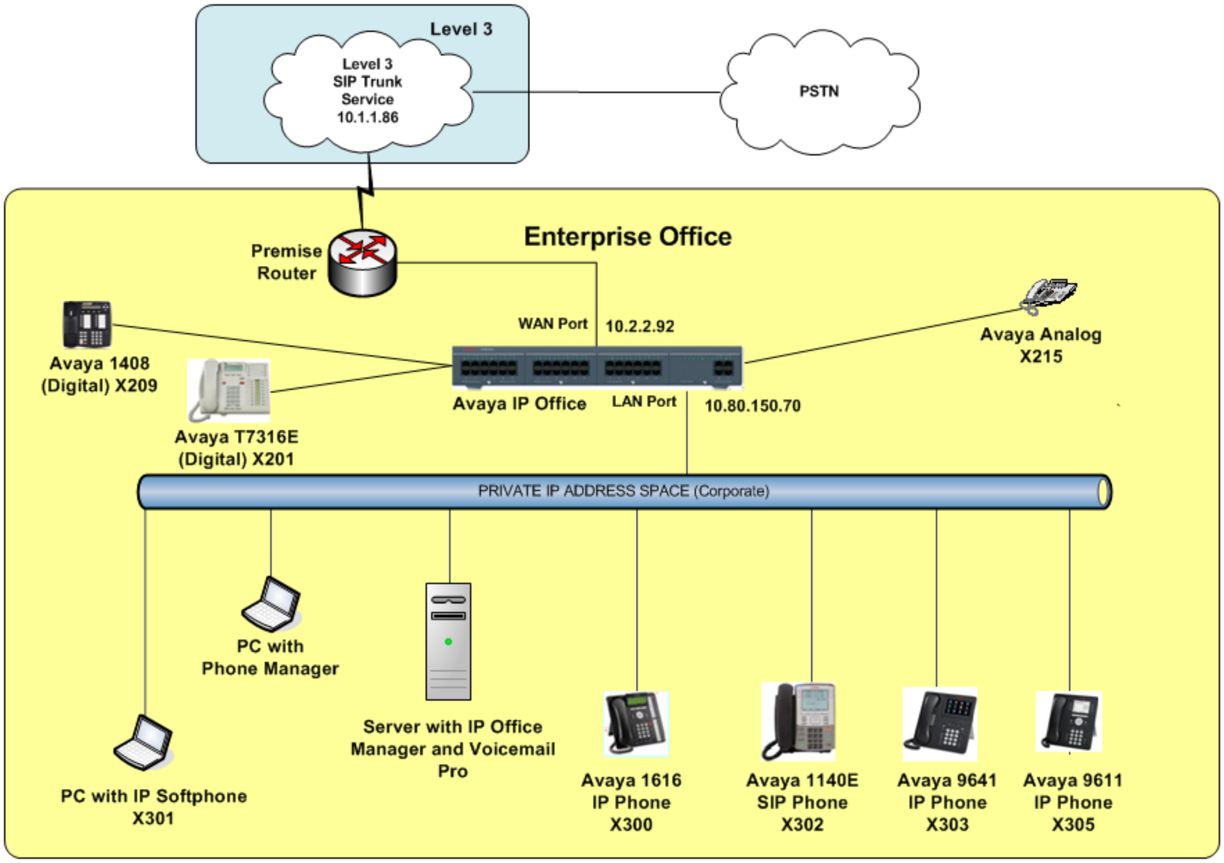


Figure 1: Avaya Interoperability Test Lab Configuration

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

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

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

4. Equipment and Software Validated

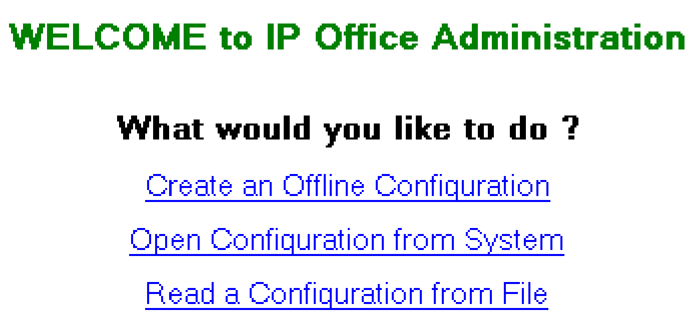
The following equipment and software were used for the sample configuration provided:

Avaya Telephony Components	
Equipment	**Software**
Avaya IP Office 500 v2	Release 7.0 (232702)
Avaya Voicemail Pro	Release 7.0 (17)
Avaya IP Office Manager	Release 9.0 (232702)
Avaya 1616SW IP Telephone (H.323)	Release 1.3
Avaya 9641SW IP Telephone (H.323)	Release 6.1
Avaya 9611G IP Telephone (H.323)	Release 6.1
Avaya 1140E IP Telephone (SIP)	SIP1140e.04.01.13.00
Avaya 2400-Series Digital Telephone	Release 6.0
Avaya IP Office Softphone	Release 3.1.2.17 59616
Avaya Phone Manager	Release 4.2.39

Level 3 Components	
Equipment	**Software**
Level 3 Enterprise Edge	Version 1

5. Configure Avaya IP Office

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

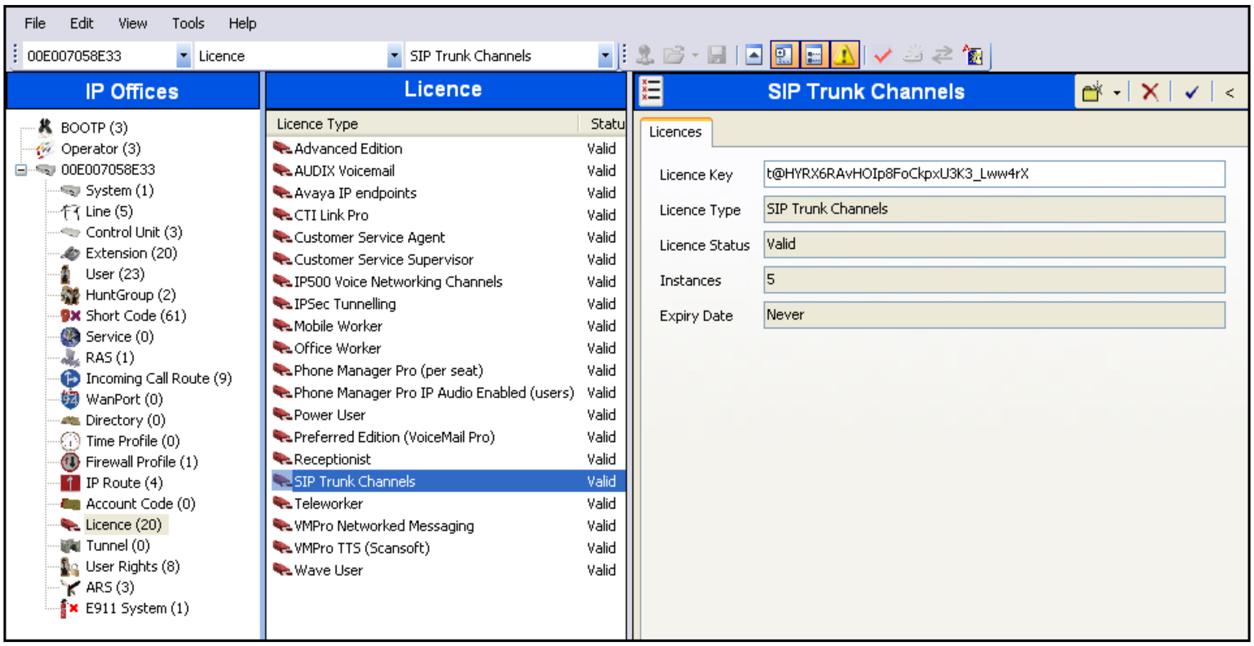


Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window and log in with the appropriate credentials. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the service provider (such as twinning and IP Office Softphone support) is assumed to already be in place.

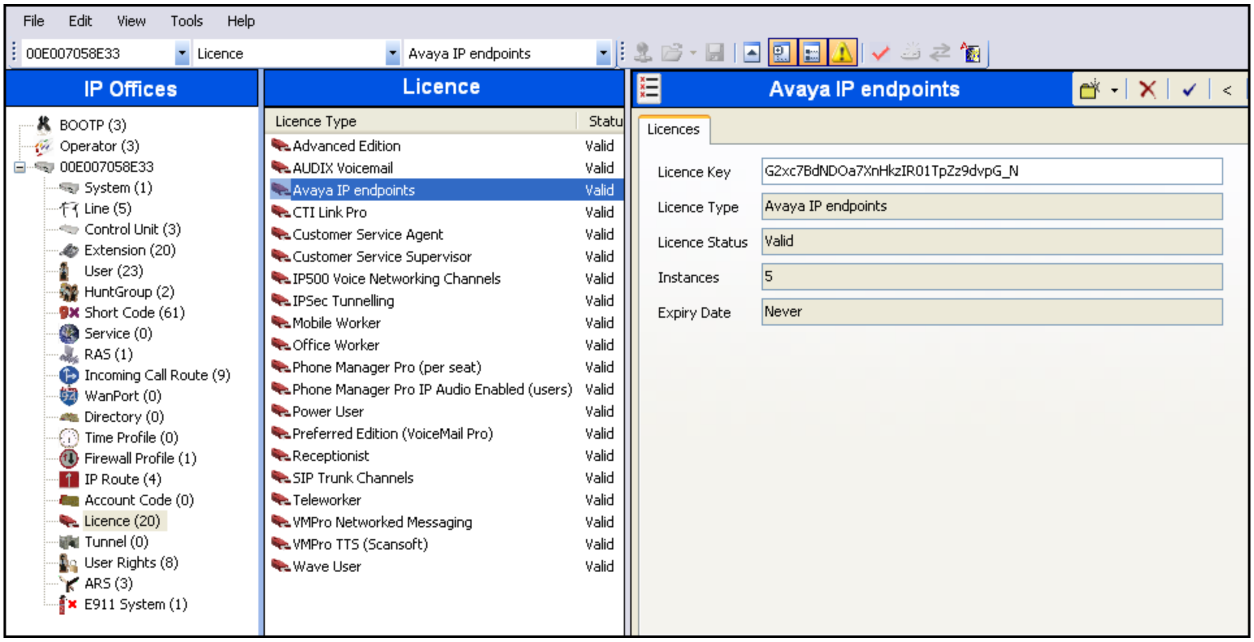
5.1. Licensing and Physical Hardware

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

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

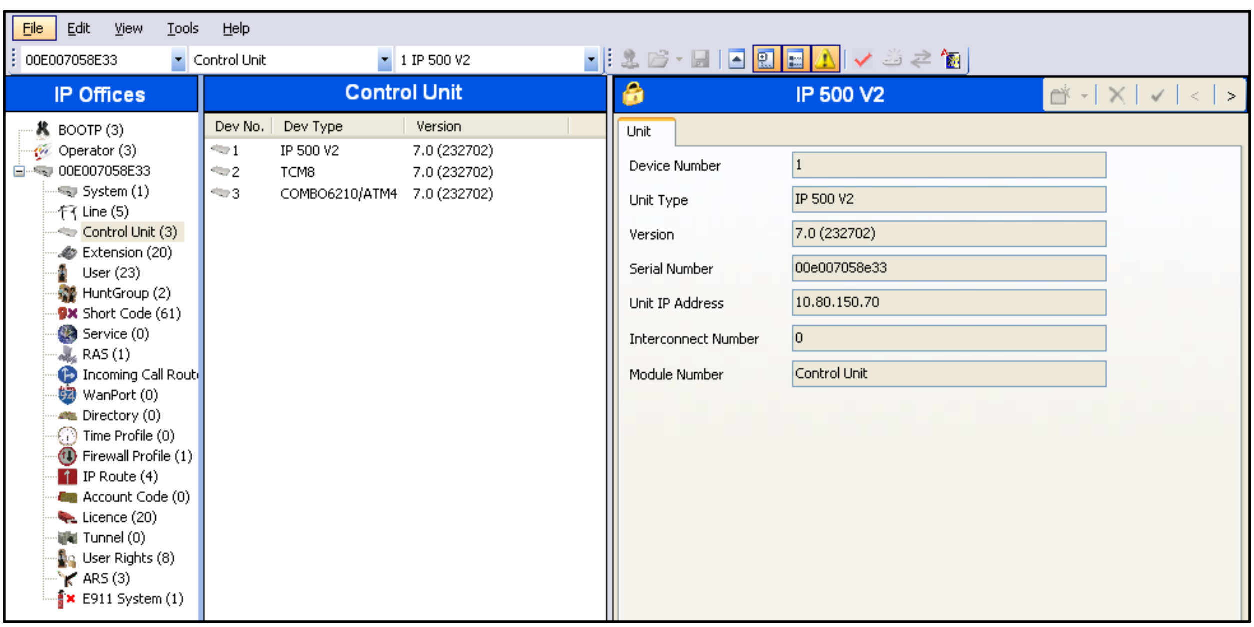


If Avaya IP Telephones will be used as is the case in these Application Notes, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints** in the Group pane. Confirm a valid license with sufficient **Instances** in the Details pane.



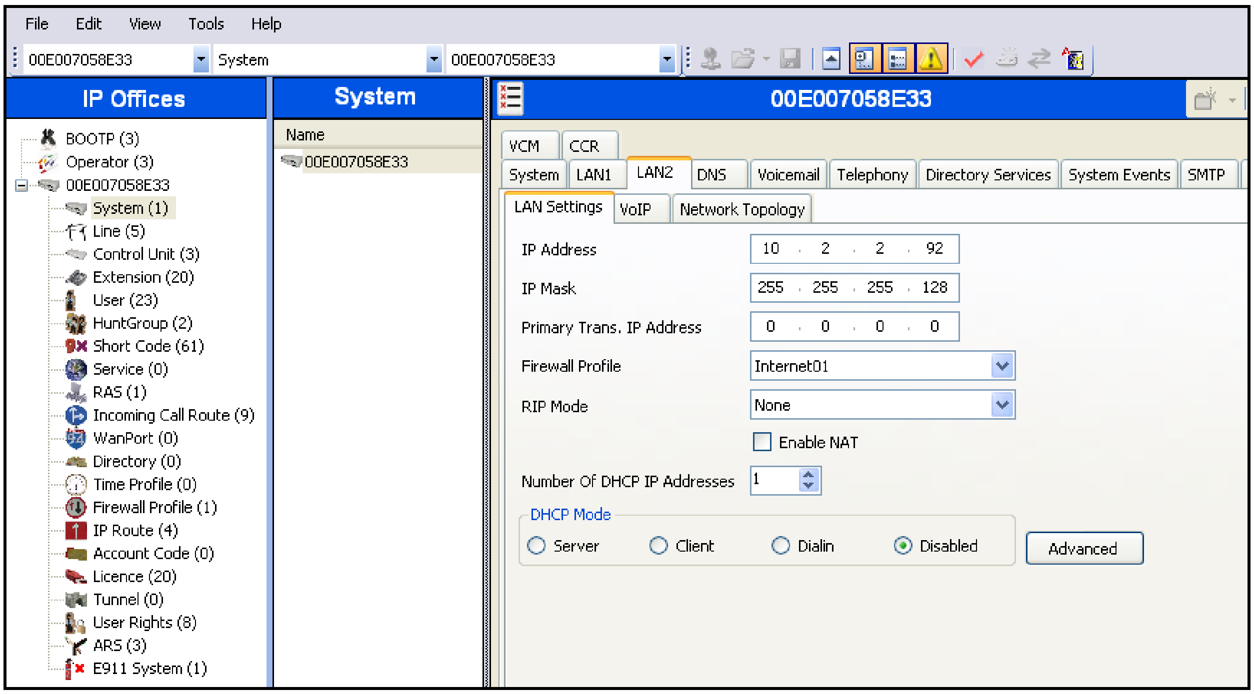
In the sample configuration, looking at the IP Office 500 from left to right, the first module is a legacy Nortel module TCM 8. The second module is a Combination Card with 4 Analog Trunks. This module has 6 Digital Stations ports, two analog extension ports, 4 analog trunk ports and 10 VCM channels. 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.

The following screen shows the modules in the IP Office used in the sample configuration. To access such a screen, select **Control Unit** in the Navigation pane. The modules appear in the Group pane. In the screen below, **IP 500 V2** is selected in the Group pane, revealing additional information about the IP 500 V2 in the Details pane.

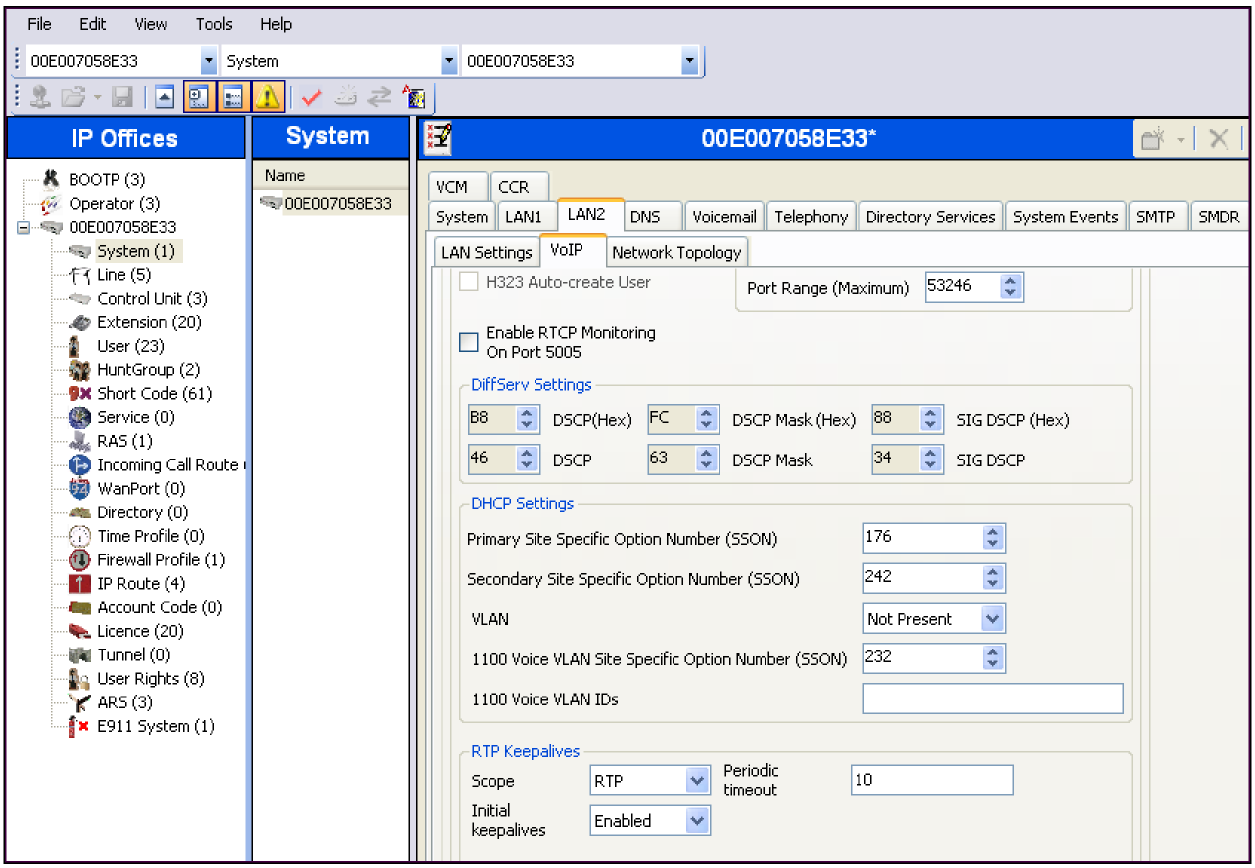


5.2. LAN2 Settings

In the sample configuration the WAN port was used to connect the Avaya IP Office to the public network. The LAN2 settings correspond to the WAN port on the Avaya IP Office 500. To access the LAN2 settings, first navigate to **System** 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. All other parameters should be set according to customer requirements.

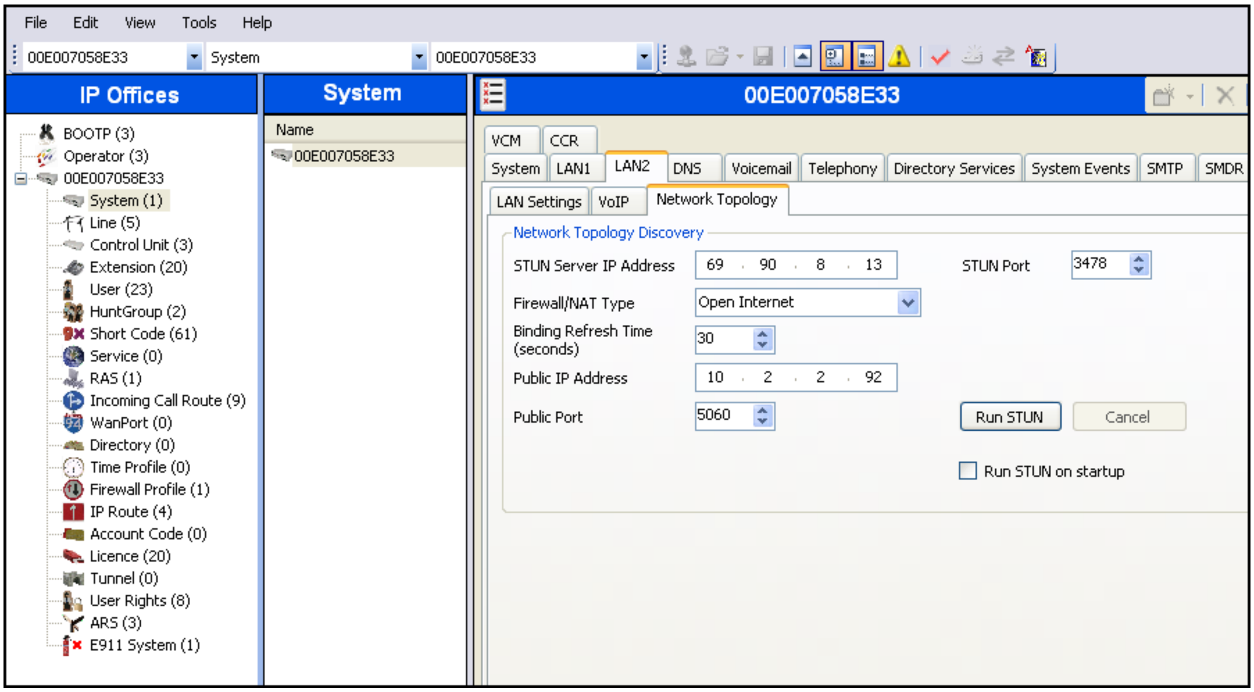


On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. Under **RTP Keepalives** set the **Scope** to ***RTP***, the **Initial keepalives** to ***Enabled*** and the **Periodic timeout** to ***10***. Enabling this will prevent the loss of speech path on calls forwarded across the SIP trunk. These settings instruct Avaya IP Office to send RTP keepalive packets every 10 seconds from the establishment of the connection. This will start media flowing from the far-end endpoint in those cases where the far-end endpoint is waiting to receive media before it starts to send media of its own. 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. All other parameters should be set according to customer requirements.



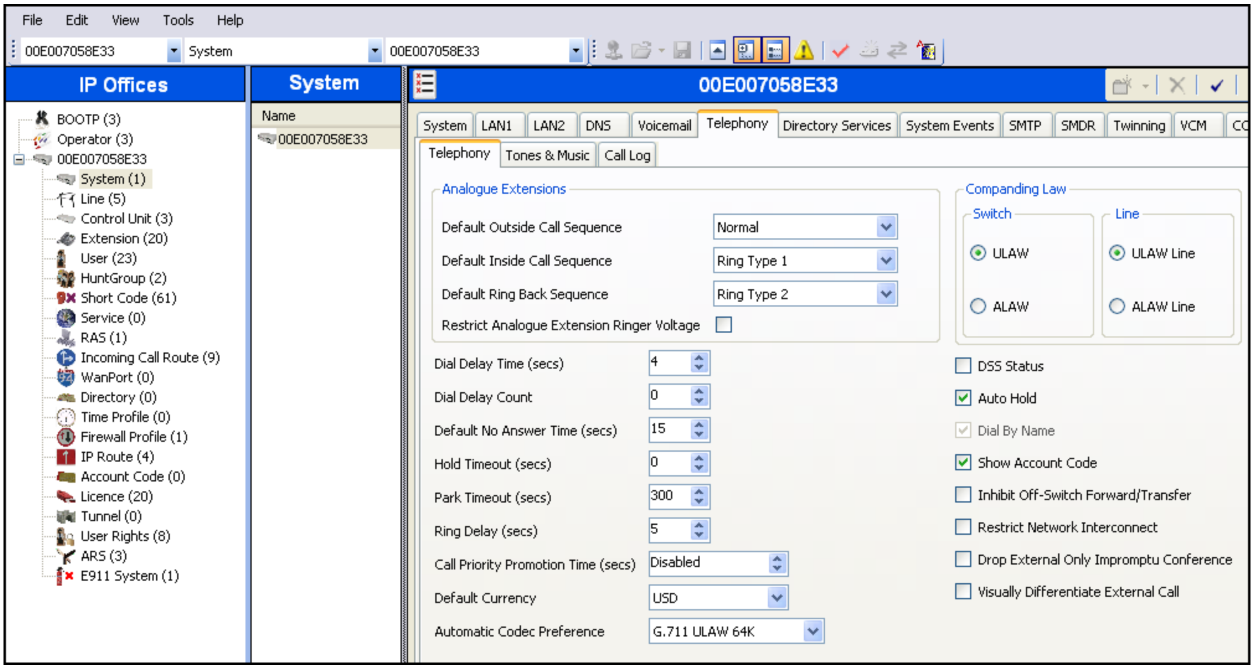
On the **Network Topology** tab in the Details Pane, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to ***Open Internet***.
- Set **Binding Refresh Time (seconds)** to ***30***. This value is used to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- Set the **Public Port** to ***5060***.
- All other parameters should be set according to customer requirements.



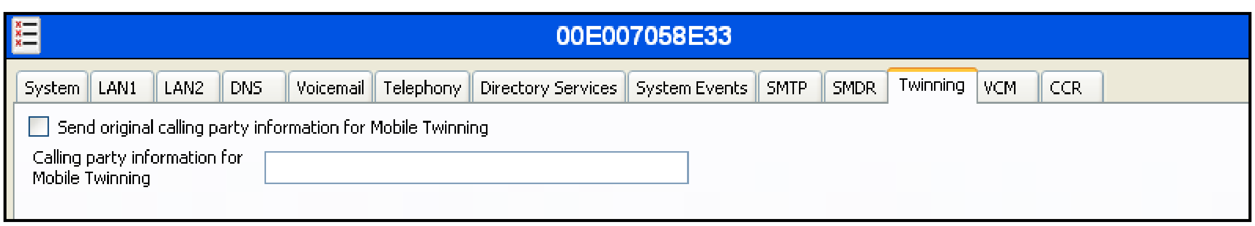
5.3. System Telephony Settings

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



5.4. Twinning Calling Party Settings

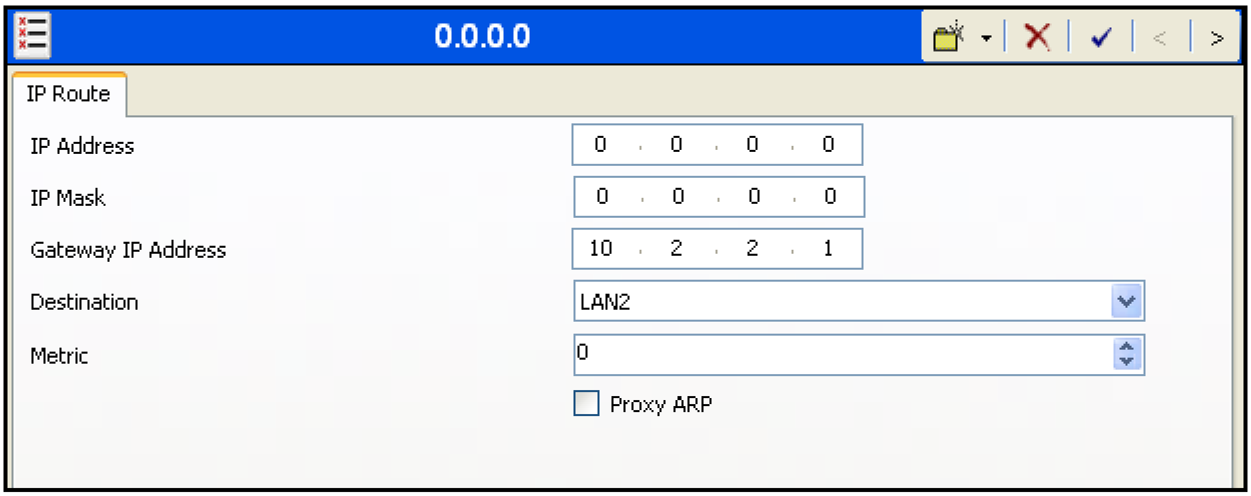
To view or change Twinning settings, select the **Twinning** tab as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank.



5.5. IP Route

Navigate to **IP Route** in the left Navigation Pane, and then right-click on the Group Pane to select **New.** Create a default route with the following parameters:

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



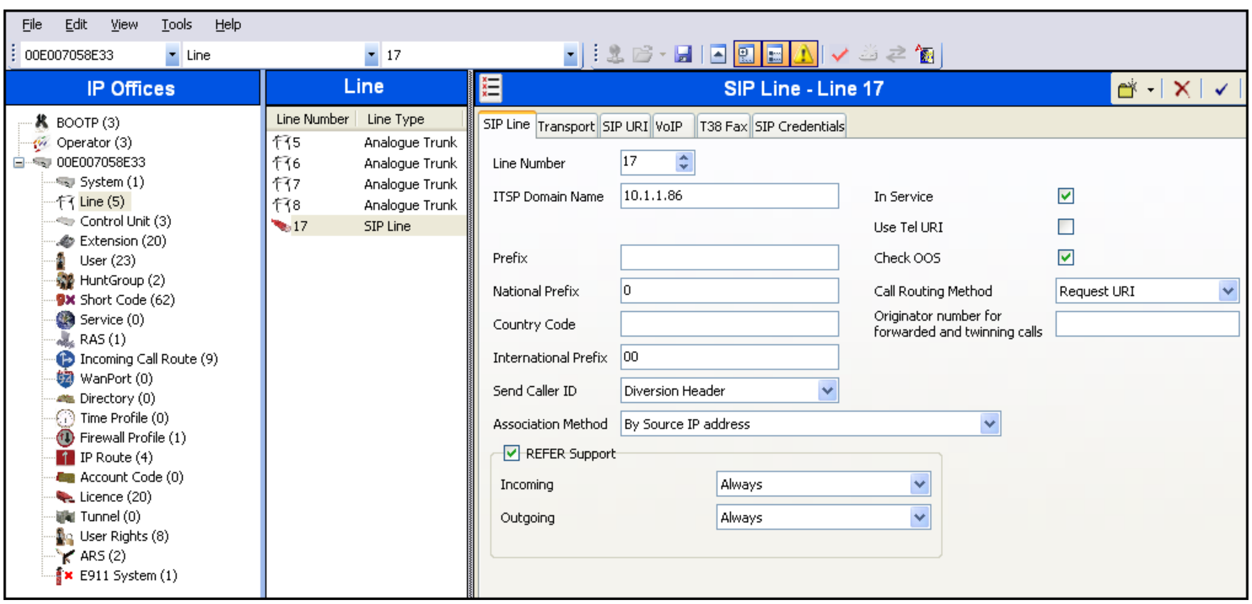
5.6. SIP Line

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

5.6.1. SIP Line – SIP Line Tab

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

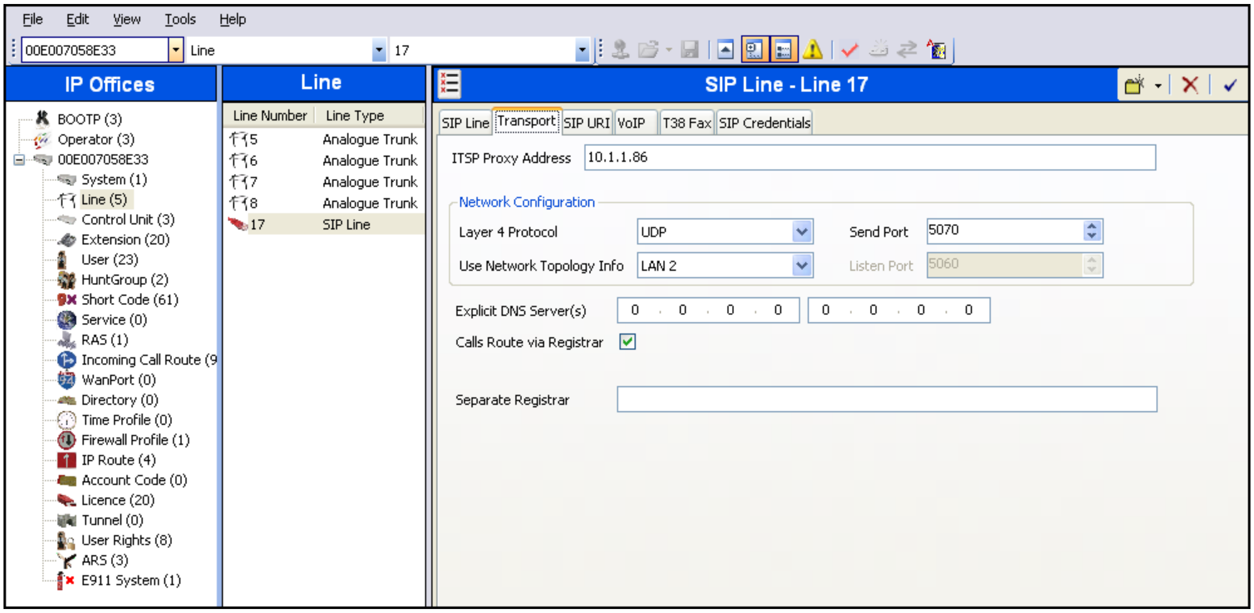
- Set **ITSP Domain Name** to the IP address of the Level 3 SIP proxy.
- Set **Send Caller ID** to ***Diversion Header***. With this setting and the related configuration in **Section 5.4**, IP Office will include the Diversion Header for calls that are directed via Mobile Twinning out the SIP Line to Level 3. It will also include the Diversion Header for calls that are call forwarded out the SIP Line. See **Section 2.2** for limitations with the software release used for compliance testing.
- Check **REFER Support**.
- Set **Incoming** and **Outgoing** to ***Always***. In the compliance test this feature was enabled to test transfers of a call between a PSTN phone and an enterprise phone to a second PSTN phone.
- Check the **In Service** box. This makes the trunk available to incoming and outgoing calls.
- Check the **Check OOS** box. IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office will use the **Binding Refresh Time** for LAN2, as shown in **Section 5.2**.
- Default values may be used for all other parameters.



5.6.2. SIP Line - Transport Tab

Select the **Transport** tab. This tab was first introduced in Release 6.1. Some information configured in this tab had been under the **SIP Line** tab in Release 6.0. Set the parameters as shown below.

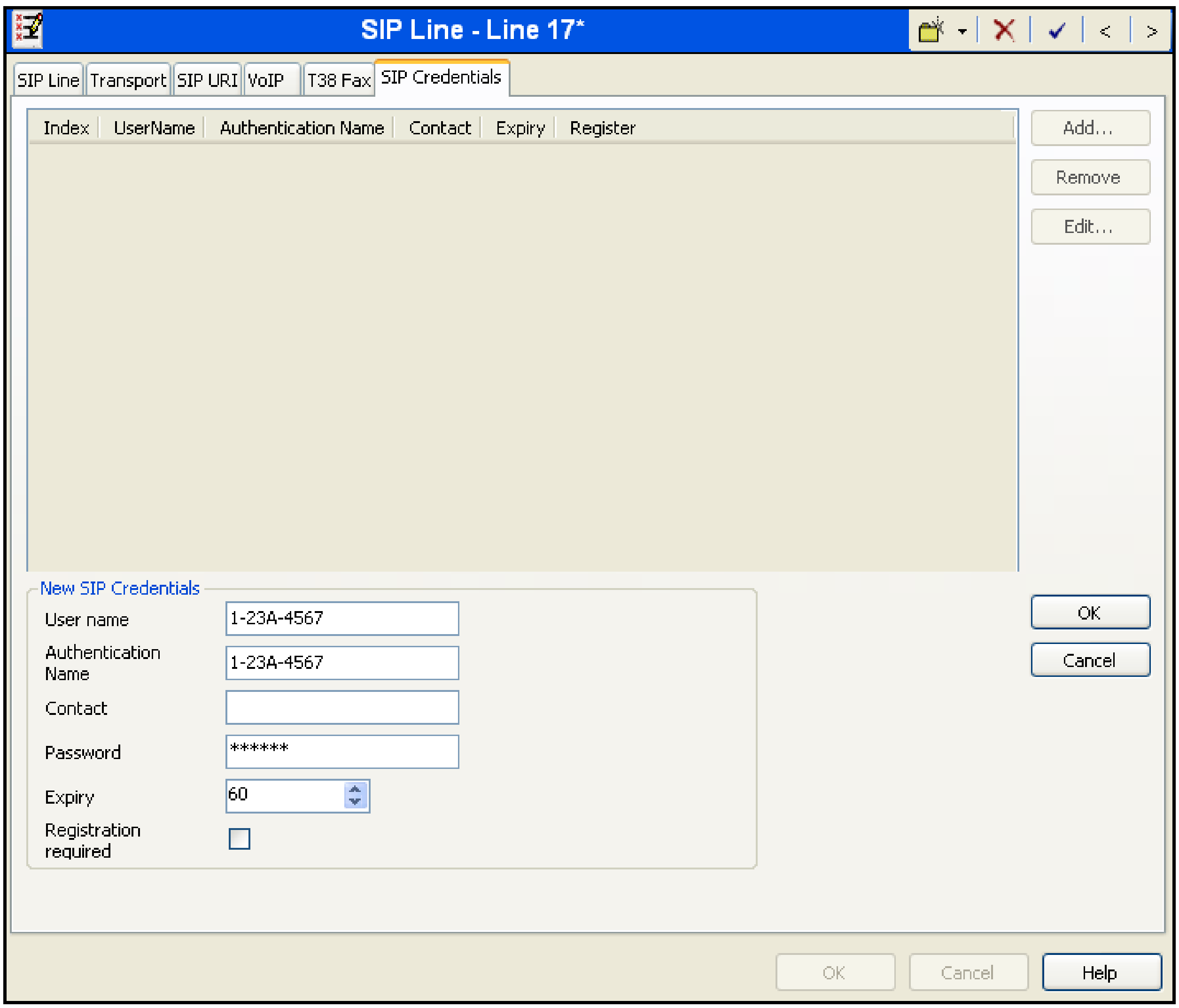
- Set **ITSP Proxy Address** to the IP address of the Level 3 SIP proxy.
- Set **Layer 4 Protocol** to ***UDP***.
- Set **Use Network Topology Info** to the network port configured in **Section 5.2**.
- Set the **Send Port** to **5070**.
- Default values may be used for all other parameters.



5.6.3. SIP Line - SIP Credentials Tab

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

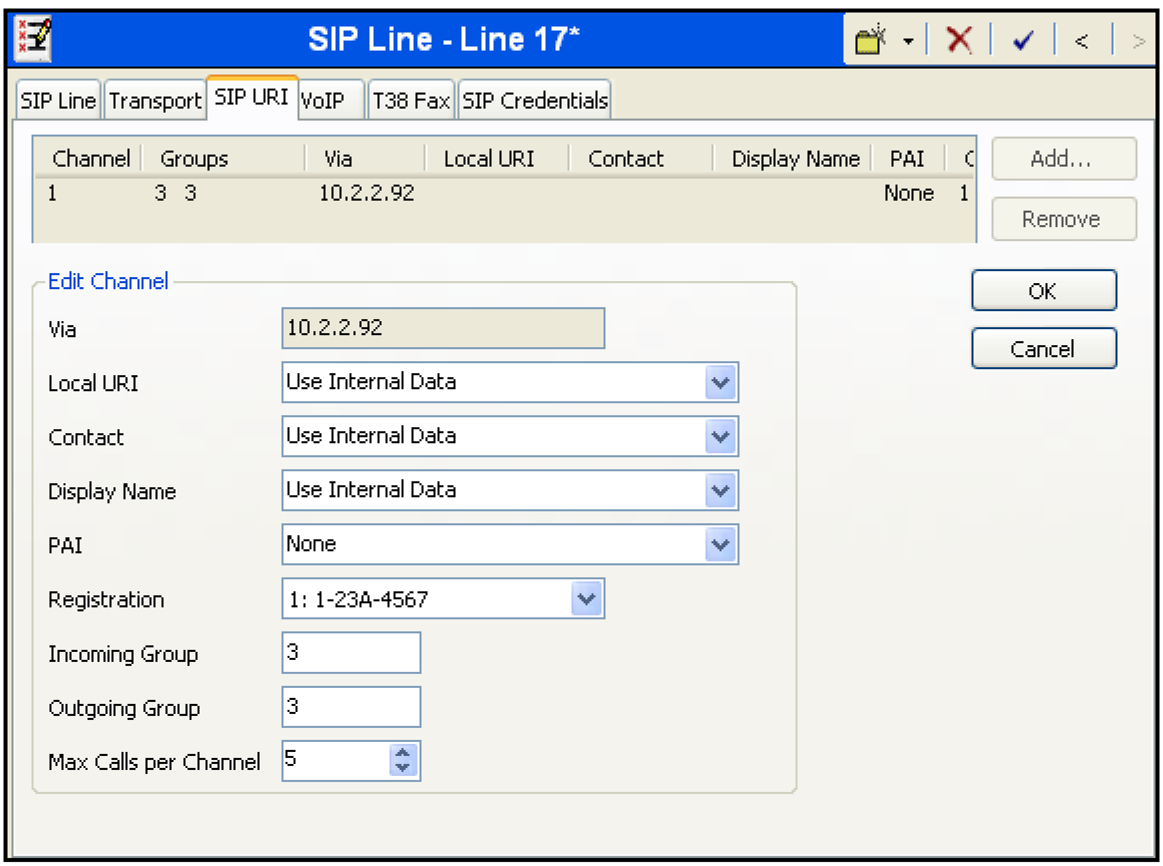
- Set **User name** and **Authentication Name** to the value provided by the service provider.
- Set **Password** to the value provided by the service provider.
- Uncheck the **Registration required** option. Level 3 does not require registration for Digest Authentication.



5.6.4. SIP Line - SIP URI Tab

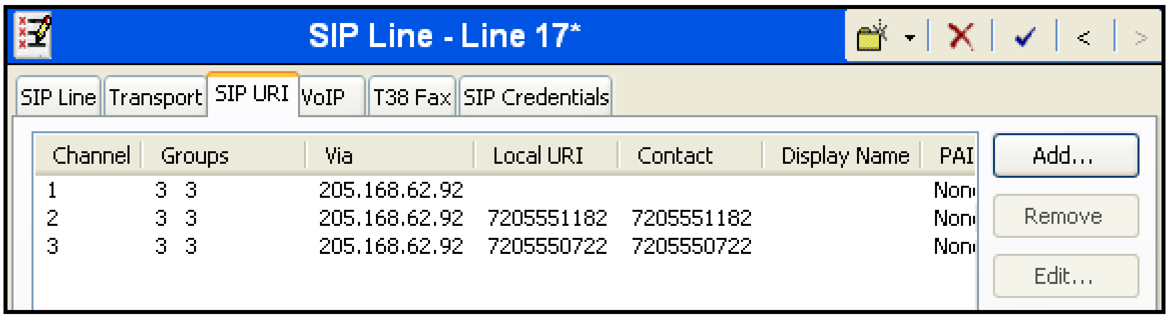
A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. Select the **SIP URI** tab, then click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top,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

- Set **Local URI**, **Contact** and **Display Name** to ***Internal Data***. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any User as shown in **Section 5.8**.
- For **Registration**, select the account credentials previously configured on the line's **SIP Credentials** tab in **Section 5.6.3.**
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group ***3*** was defined that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.



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

The following shows the SIP URI tab for SIP Line 17 after the SIP URIs corresponding to Voice mail Auto Attendant (732-555-0722) and FNE00 Short Code (720-555-1182) have been added.

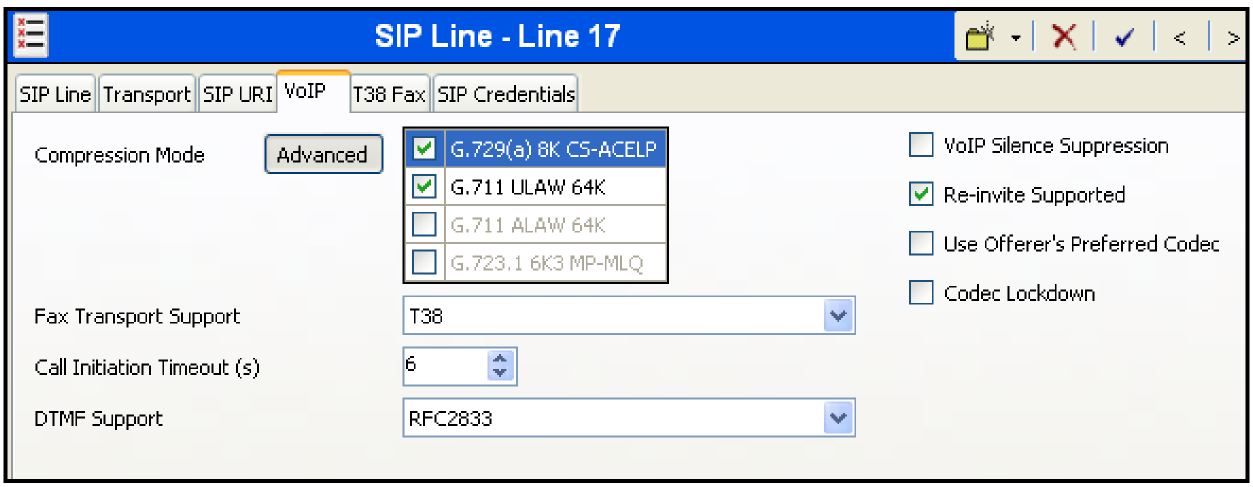


5.6.5. SIP Line - VoIP Tab

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

- Click the **Advanced** button next to **Compression Mode** to display a list of the codecs in their current order of preference. Codecs can be dragged up or down the list or deselected from the list. For compliance testing ***G.729a*** was the first choice with ***G.711ULAW*** second.
- Uncheck the **VoIP Silence Suppression** box.
- Set the **Fax Transport Support** to ***T.38***.
- Set the **DTMF Support** field to ***RFC2833***. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Check the **Re-invite Supported** box.
- Default values may be used for all other parameters.

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

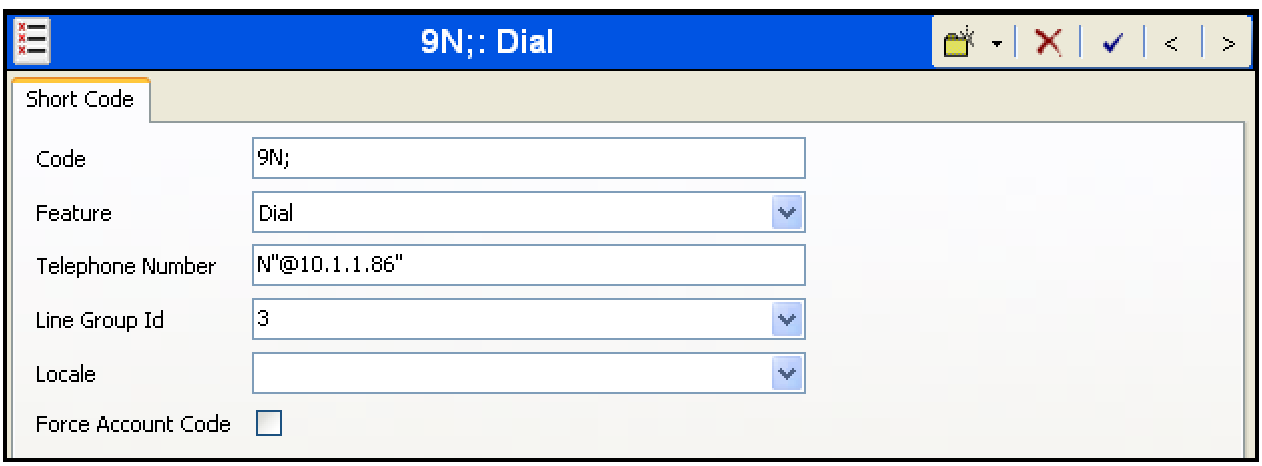


5.7. Short Codes

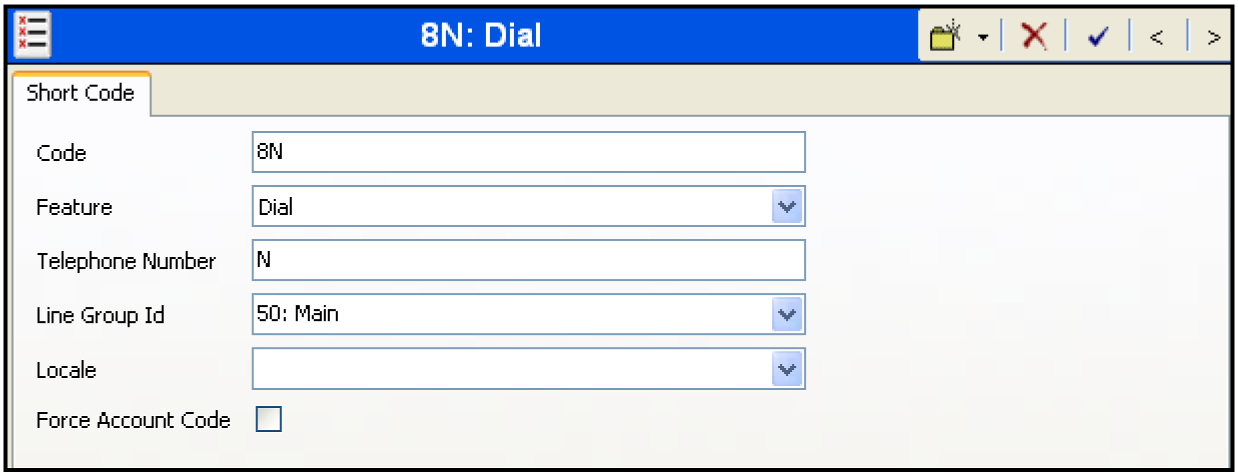
Define a short code to route outbound traffic to the SIP line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, ***9N;***. This short code will be invoked when the user dials 9 followed by any number.
- Set **Feature** to ***Dial.*** This is the action that the short code will perform.
- Set **Telephone Number** to ***N"@10.1.1.86"***. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value ***N*** represents the number dialed by the user. The IP address 10.1.1.86 is the IP address of the Level 3 SIP proxy.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.6.4**. This short code will use this line group when placing the outbound call.

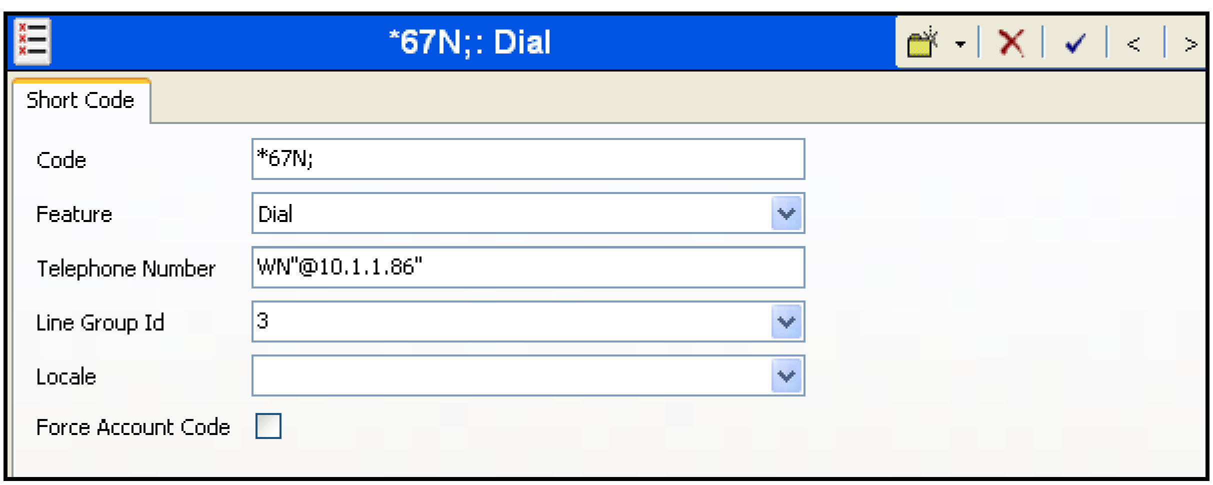
Click the OK button (not shown).



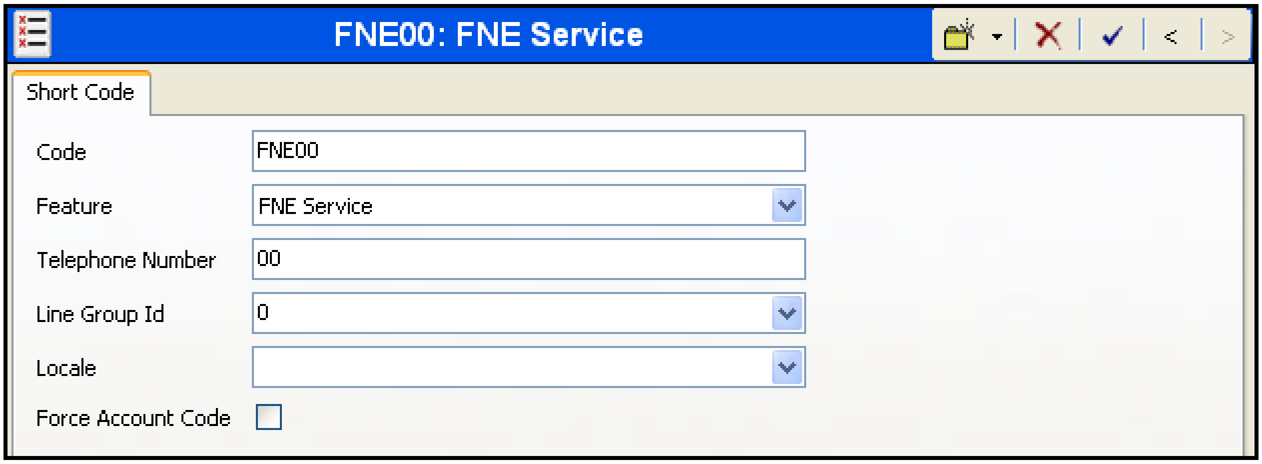
The simple "9N;" short codes illustrated above does not provide a means of alternate routing if the configured Level 3 SIP Line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. In the following example screen, 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 5.10** for example ARS route configuration for "50: Main" as well as a backup route.



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

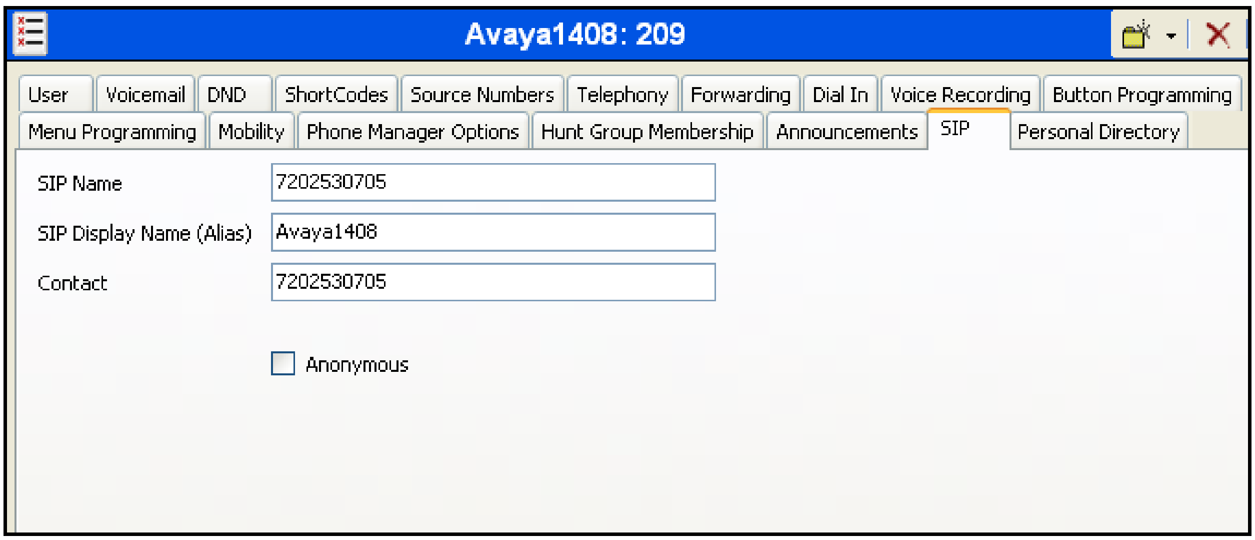


The following screen illustrates a short code that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code** ***FNE00*** is defined for **Feature** ***FNE Service*** to **Telephone Number** ***00***. This short code will be used as means to allow a Level 3 DID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

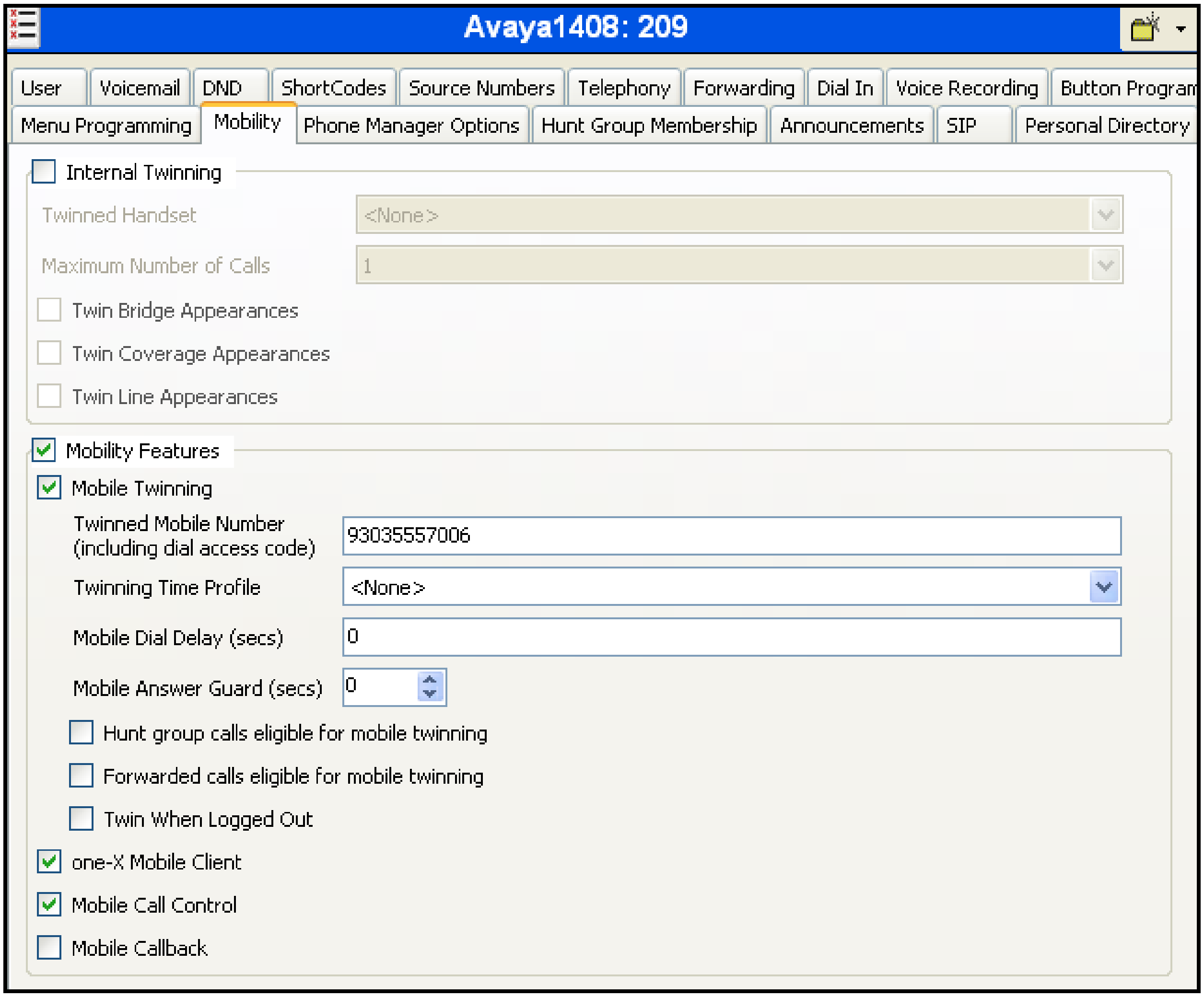


5.8. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.6**. To configure these settings, first navigate to **User** in the Navigation Pane, and then click on the user in the Group Pane to be modified. Select the **SIP** tab in the Details Pane. The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From and Contact headers for outgoing SIP trunk calls and allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.6.4**) The example below shows the settings for User 209. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from Level 3. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network. Click the **OK** button (not shown).



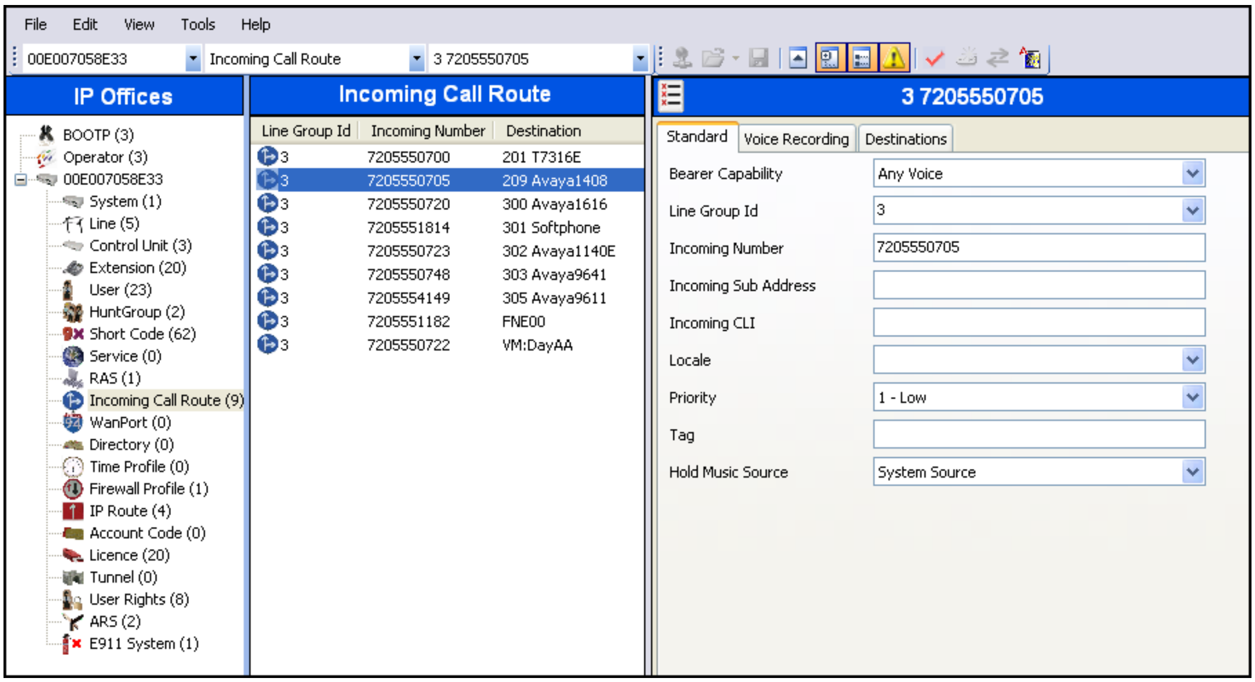
The following screen shows the **Mobility** tab for User 209. The **Mobility Features** and **Mobile Twinning boxes** are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case ***93035557006***. Other options can be set according to customer requirements.



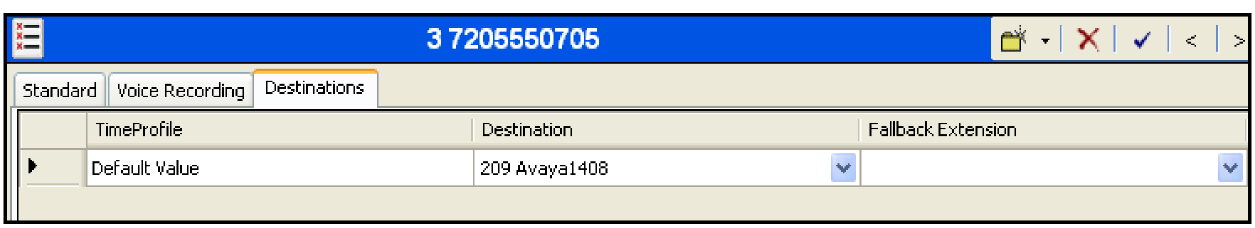
5.9. Incoming Call Route

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

- Set the **Bearer Capacity** to ***Any Voice***.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.6.3**.
- Set the **Incoming Number** to the incoming number on which this route should match. Matching is right to left.
- Default values can be used for all other fields.

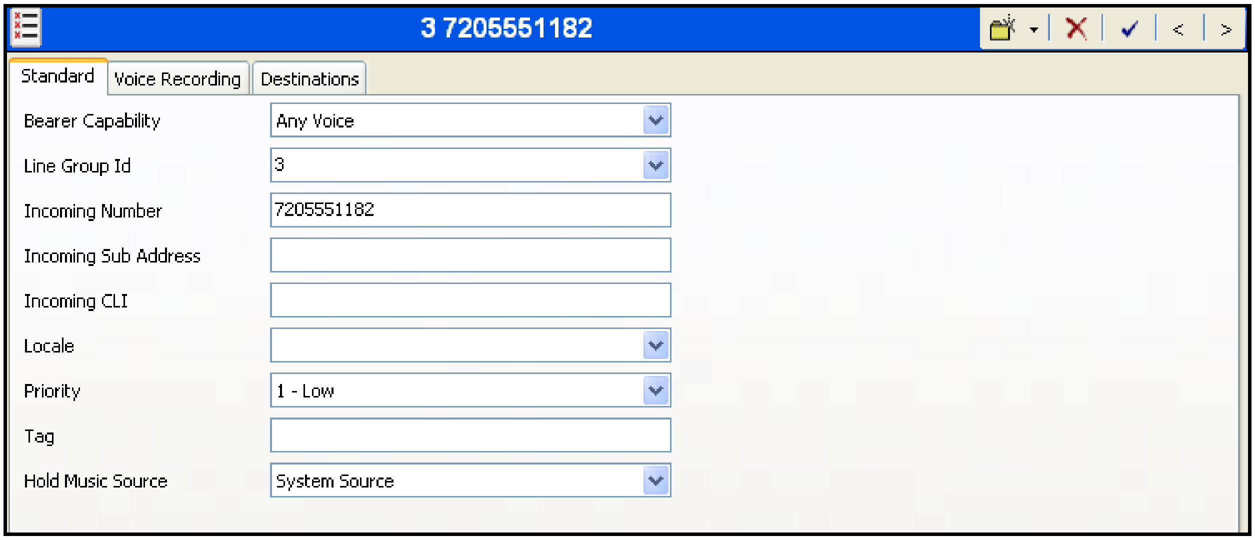


On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 7205550705 on line 3 are routed to extension 209.

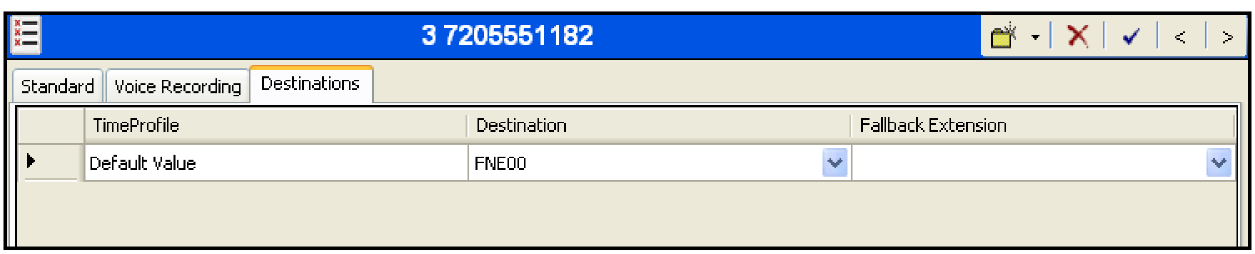


Incoming Call Routes for other direct mappings of DID numbers to IP Office users listed in **Figure 1** are omitted here, but can be configured in the same fashion.

In the screen shown below, the incoming call route for **Incoming Number** ***7205551182*** is illustrated. The **Line Group Id** is ***3***, matching the Incoming Group field configured in the SIP URI tab in **Section 5.6.4**.



When configuring an Incoming Call Route, the **Destination** field can be manually configured with a number such as a short code, or certain keywords available from the drop-down list. For example, the following **Destinations** tab for an incoming call route contains the **Destination** ***FNE00*** entered manually. ***FNE00*** is the short code for ***FNE Service***, as shown in **Section 5.7**. An incoming call to 720-555-1182 will be delivered directly to internal dial tone from the IP Office, allowing the caller to perform dialing actions including making calls and activating Short Codes. The incoming caller ID must match the Twinned Mobile Number entered in the User Mobility tab (**Section 5.8**); otherwise the IP Office responds with a 486 Busy Here and the caller will hear a busy tone.



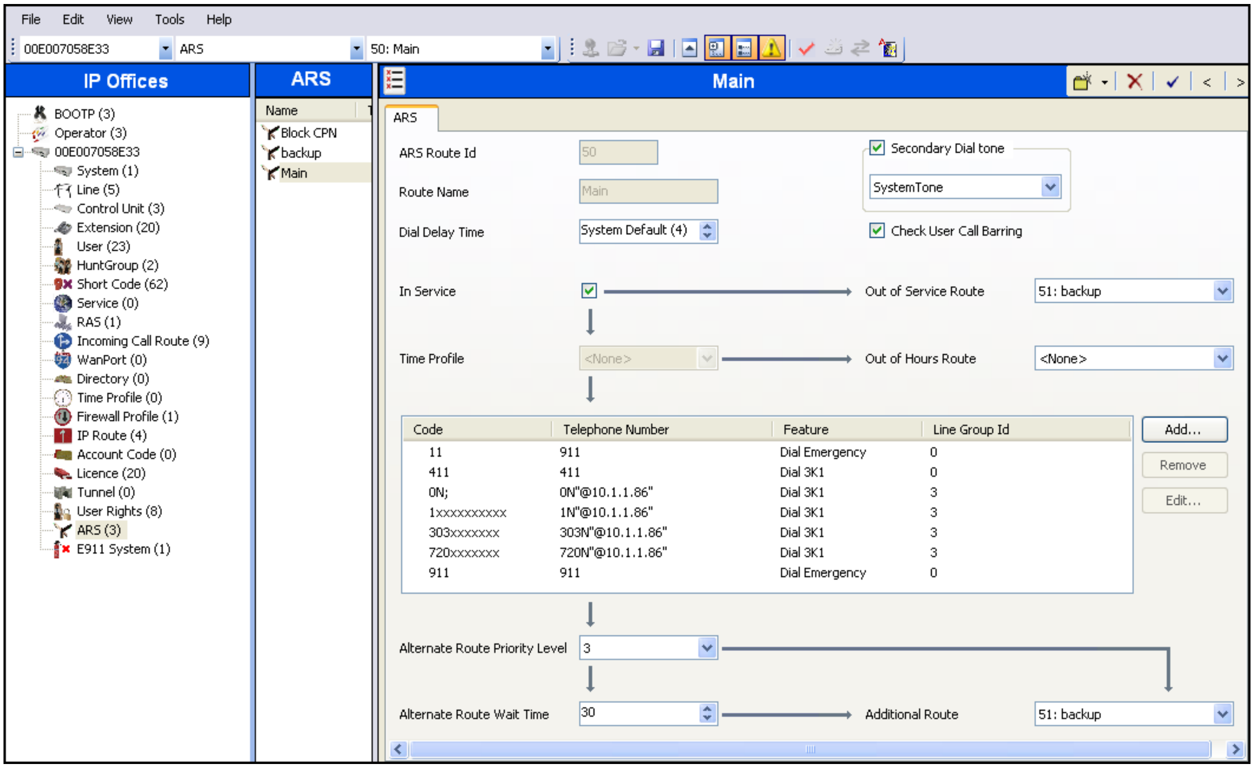
5.10. ARS and Alternate Routing

While detailed coverage of ARS is beyond the scope of these Application Notes, this section includes basic ARS screen illustrations and considerations. ARS is illustrated here mainly to illustrate alternate routing should the SIP Line be out of service or temporarily not responding.

Optionally, Automatic Route Selection (ARS) can be used rather than the simple "9N;" short code approach documented in **Section 5.7**. 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 preferentially, but service numbers should prefer a different outgoing line group, ARS can be used to distinguish the call behaviors.

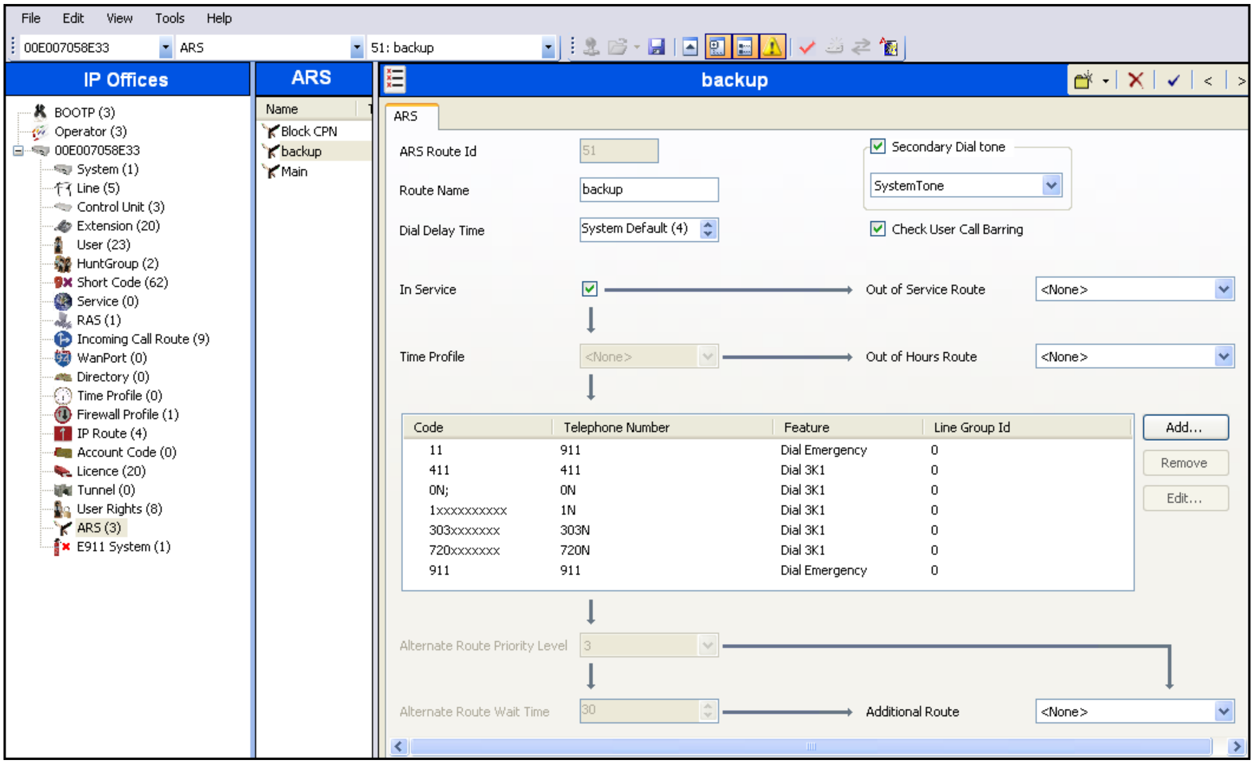
To add a new ARS route, right-click **ARS** in the Navigation pane, and select **New**. To view or edit an existing ARS route, select ARS in the Navigation pane, and select the appropriate route name in the Group pane.

The following screen shows an example ARS configuration for the route named ***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 5.7**) can be further analyzed to direct the call to a specific Line Group ID Per the example screen above, if the user dialed 8-303-555-1997, the call would be directed to Line Group 3, configured in **Section 5.6.4**. If Line Group 3 cannot be used, the call can automatically route to the route name configured in the **Additional Route** parameter in the lower right of the screen. Since alternate routing can be considered a privilege not available to all callers, IP Office can control access to the alternate route by comparing the calling user's priority to the value in the **Alternate Route Priority Level** field.

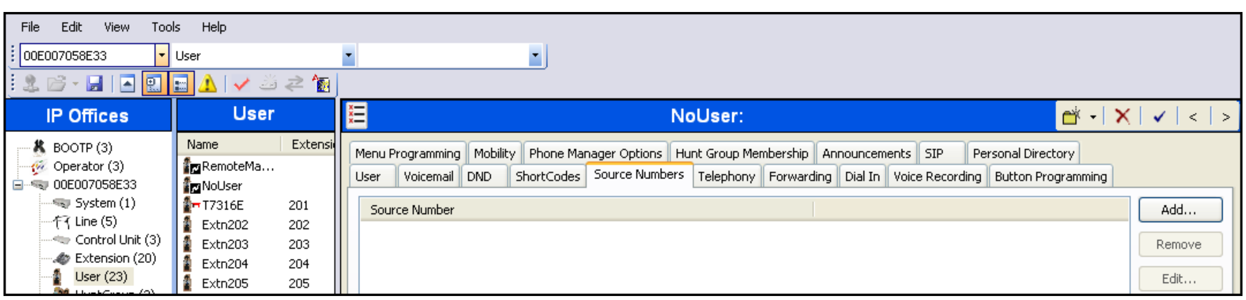
The following screen shows an example ARS configuration for the route named ***backup***, ARS Route ID 51. Continuing the example, if the user dialed 8-303-555-1997, and the call could not be routed via the primary route *50: Main* described above, the call will be delivered to this *backup* route. Per the configuration shown below, the call will be delivered to Line Group 0, using the analog lines. The configuration of the **Code, Telephone Number, Feature**, and **Line Group ID** for an ARS route is similar to the configuration already shown for short codes in **Section 5.7**.



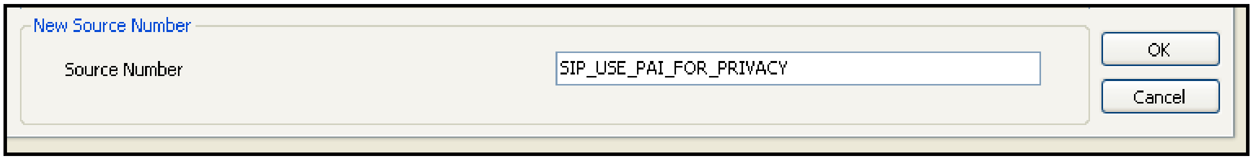
5.11. Privacy/Anonymous Calls

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

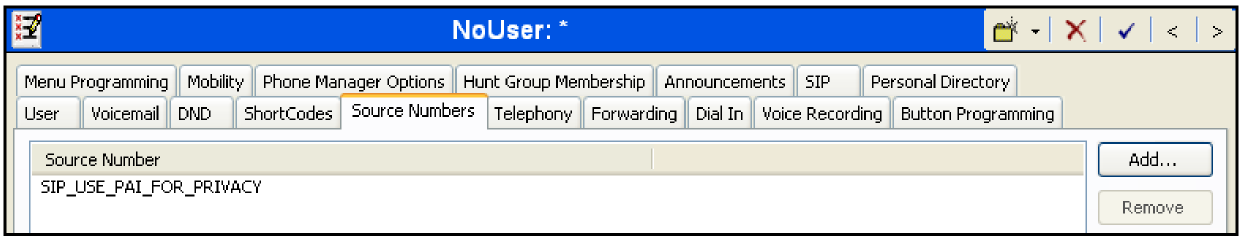
To configure Avaya IP Office to use PAI for privacy calls, navigate to **User** in the Navigation Pane, then **NoUser** in the Group Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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



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

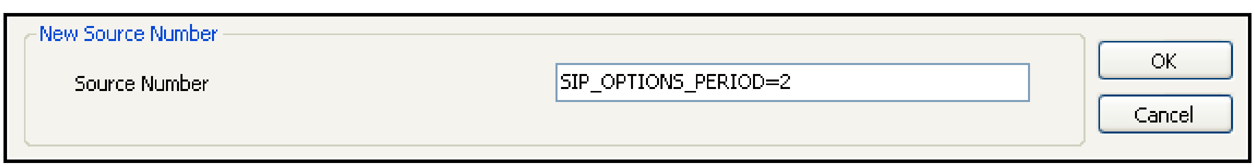


5.12. SIP Options Frequency

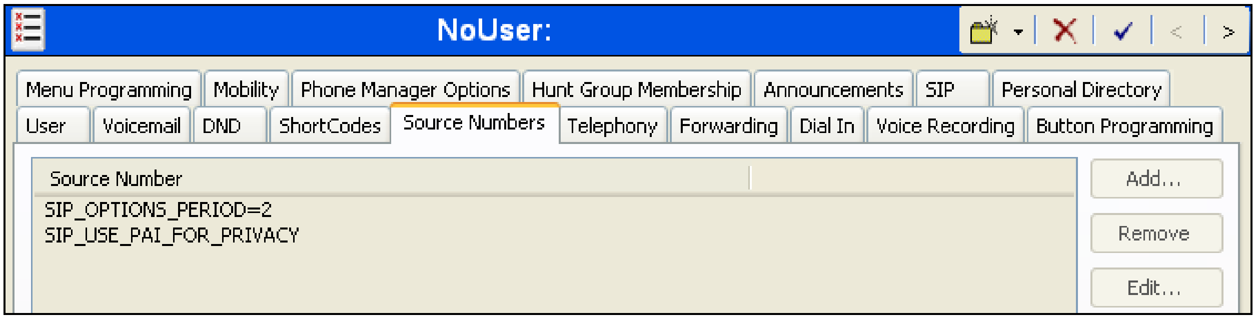
In **Section 5.6.1**, the SIP Line to Level 3 is shown with the **Check OOS** box checked. In the sample configuration, IP Office periodically checks the heath of the SIP Line by sending a SIP OPTIONS message. If there is no response, IP Office can mark the trunk out of service.

If a customer wishes to control how often SIP OPTIONS messages are sent by IP Office, a NoUser Source Number can be configured as follows. This configuration complements the configuration presented in **Section 5.2** and **Section 5.6.1.**

From the Navigation pane, select **User**. From the Group pane, scroll down past the configured users and select the user named **NoUser**. From the NoUser Details pane, select the tab **Source Numbers**.   Press the **Add…** button to the right of the list of any previously configured Source Numbers. In the **Source Number** field shown below, type ***SIP_OPTIONS_PERIOD=X***.   X is a value (in minutes) representing a longer time than the interval configured (in seconds) in the **Binding Refresh Time**.   In the sample configuration, the value used for X was 2 minutes. Click **OK**.



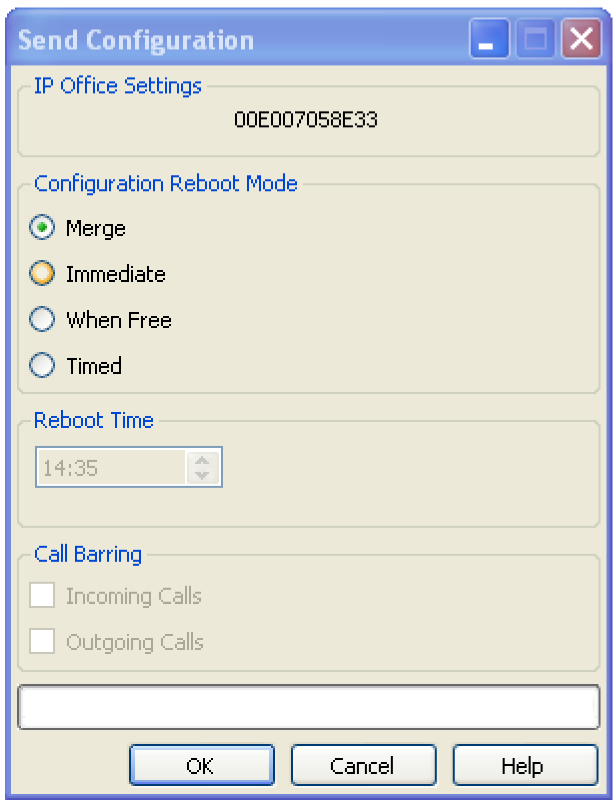
The source number SIP_OPTIONS_PERIOD=2 should now appear in the list of Source Numbers as shown below.



5.13. Save Configuration

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

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



6. Level 3 SIP Trunking Configuration

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

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

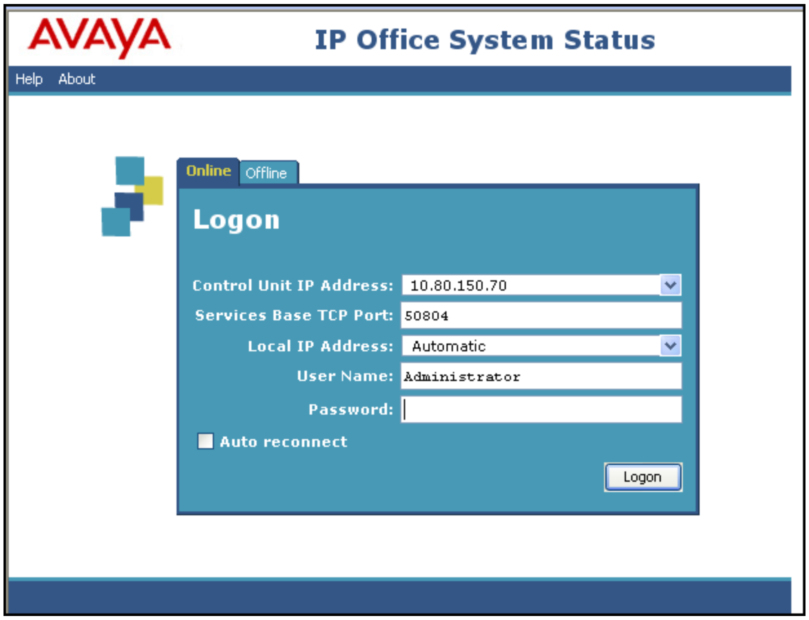
7. Verification Steps

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

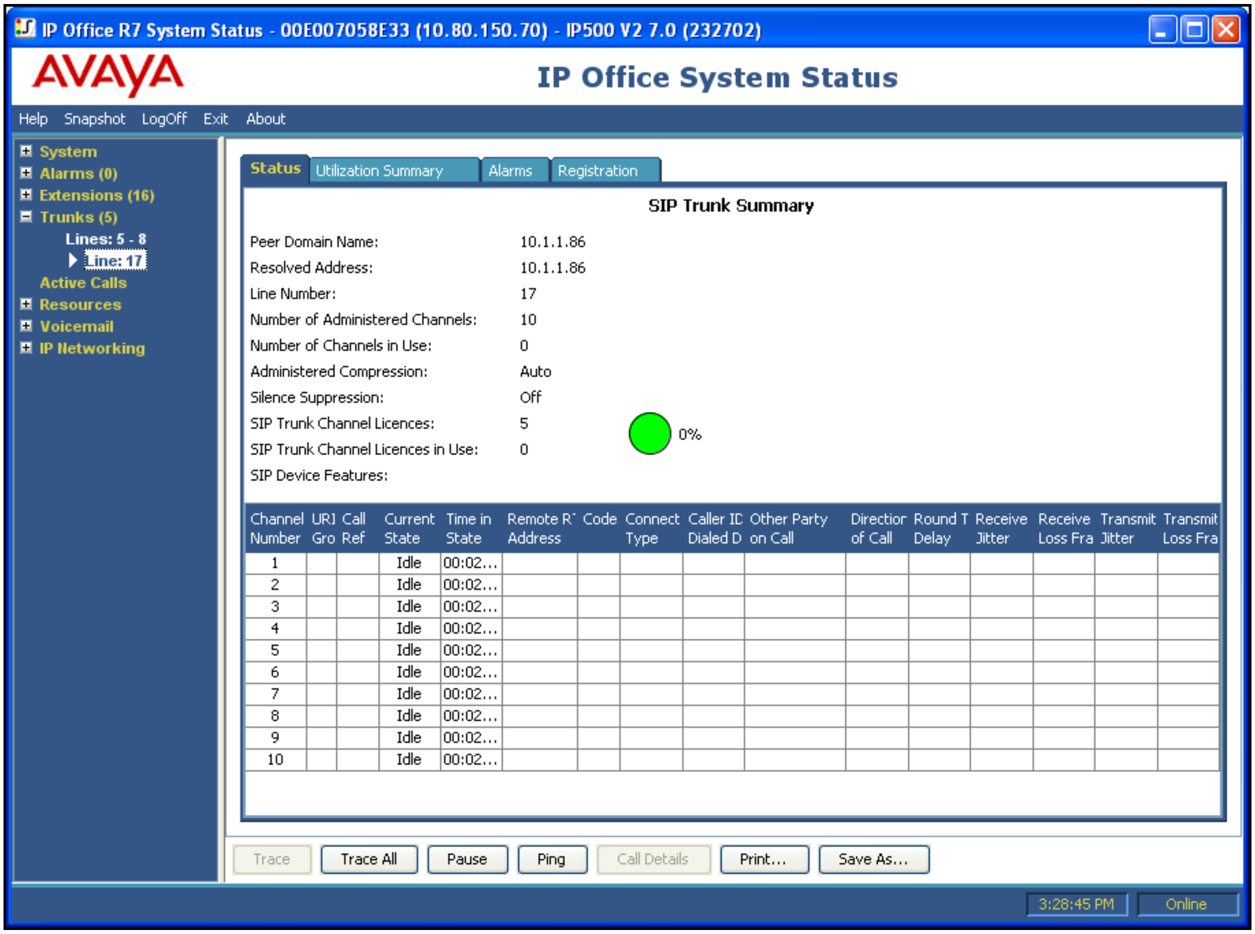
7.1. System Status

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

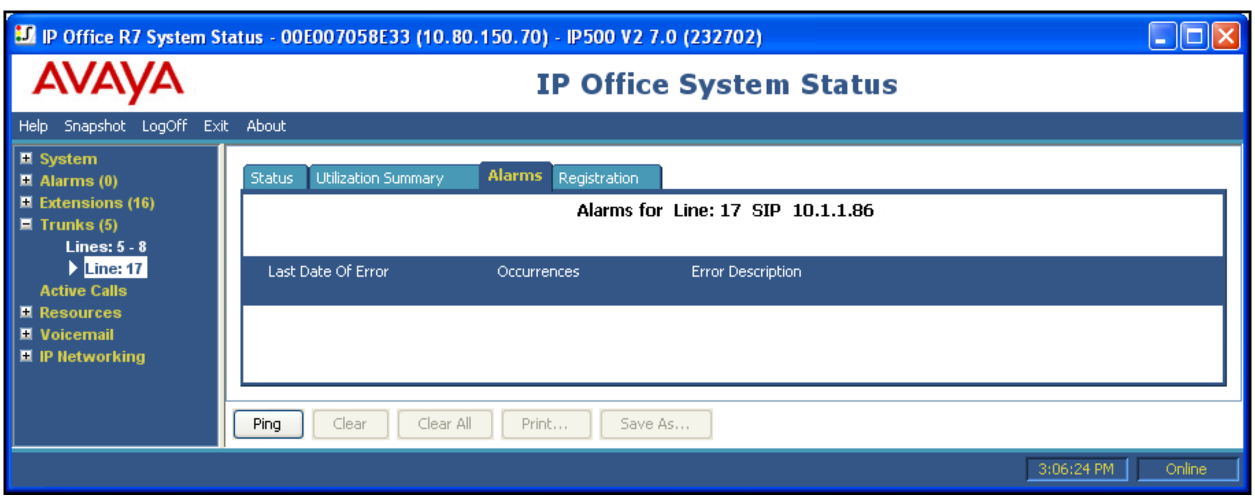
The following screen shows an example **Logon** screen. Enter the IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.



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.



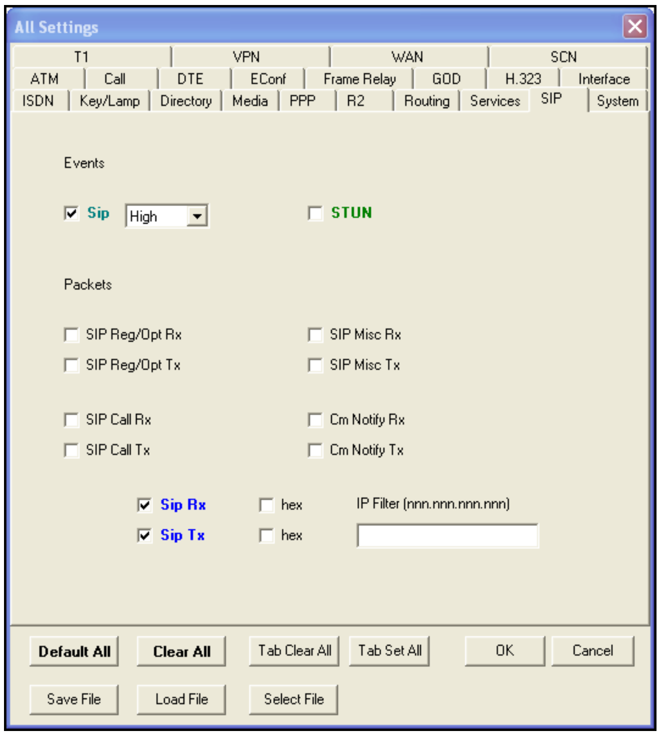
Select the **Alarms** tab and verify that no alarms are active on the SIP line.



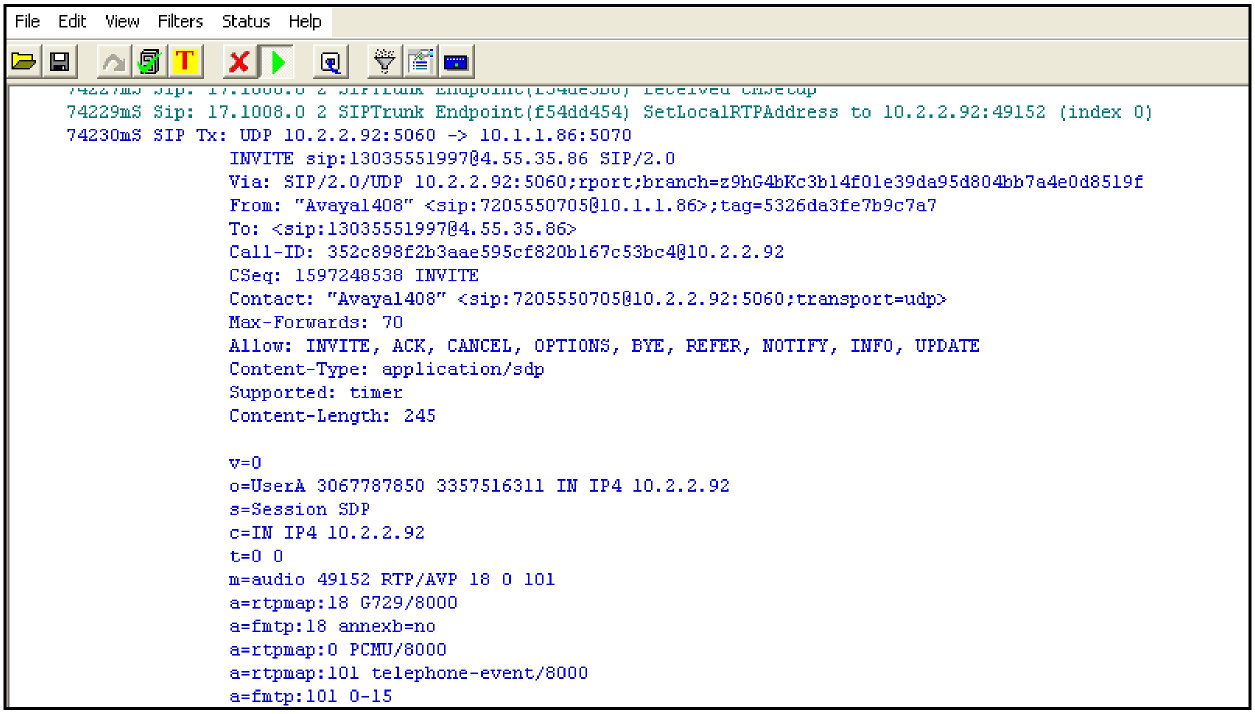
7.2. Monitor

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. All SIP messages will appear in the trace with the color blue. To customize the color, right-click on **SIP Rx** or **SIP Tx** and select the desired color.



As an example, the following shows a portion of the monitoring window for an outbound call from extension 209, whose DID is 720-555-0705, calling out to the PSTN via the Level 3 IP Trunking Service. The telephone user dialed 9-1-303-555-1997.




```
74229mS Sip: 17.1008.0 2 SIPTrunk Endpoint(f54dd454) SetLocalRTPAddress to 10.2.2.92:49152 (index 0)
74230mS SIP Tx: UDP 10.2.2.92:5060 -> 10.1.1.86:5070
                INVITE sip:13035551997@4.55.35.86 SIP/2.0
                Via: SIP/2.0/UDP 10.2.2.92:5060;rport;branch=z9hG4bKc3b14f01e39da95d804bb7a4e0d8519f
                From: "Avaya1408" <sip:7205550705@10.1.1.86>;tag=5326da3fe7b9c7a7
                To: <sip:13035551997@4.55.35.86>
                Call-ID: 352c898f2b3aae595cf820b167c53bc4@10.2.2.92
                CSeq: 1597248538 INVITE
                Contact: "Avaya1408" <sip:7205550705@10.2.2.92:5060;transport=udp>
                Max-Forwards: 70
                Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
                Content-Type: application/sdp
                Supported: timer
                Content-Length: 245

                v=0
                o=UserA 3067787850 3357516311 IN IP4 10.2.2.92
                s=Session SDP
                c=IN IP4 10.2.2.92
                t=0 0
                m=audio 49152 RTP/AVP 18 0 101
                a=rtpmap:18 G729/8000
                a=fmtp:18 annexb=no
                a=rtpmap:0 PCMU/8000
                a=rtpmap:101 telephone-event/8000
                a=fmtp:101 0-15
```

8. Conclusion

These Application Notes describe the configuration necessary to connect Avaya IP Office 7.0 to Level 3 SIP Trunking service. Level 3 SIP Trunking is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. It provides a flexible, cost-saving alternative to traditional hardwired telephony trunks. Level 3 SIP Trunking passed compliance testing. Please refer to **Section 2.2** for any exceptions.

9. Additional References

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

[1] IP Office 7.0 Installation Manual, Document Number 15-601042, Issue 23k, May 22, 2011
http://support.avaya.com/css/P8/documents/100129376

[2] IP Office Release 7.0 Manager 9.0, Document Number 15-601011, Issue 26h, May 22, 2011
http://support.avaya.com/css/P8/documents/100129398

[3] IP Office Release 6.0 System Status Application, Issue 05a, February 12, 2010 Document Number 15-601758
http://support.avaya.com/css/P8/documents/100073300

[4] IP Office Release 7.0 Voicemail Pro Administration, Document Number 15-601063, Issue 26a, May 01, 2011
http://support.avaya.com/css/P8/documents/100129332

[5] IP Office System Monitor, Document Number 15-601019, Issue 02b
http://support.avaya.com/css/P8/documents/100073350

[6] IP Office Release 7.0 1100/1200 Series Phone Installation, Issue 01c, March 2011
http://support.avaya.com/css/P8/documents/100140564

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

10. Appendix A: SIP Line Template

Avaya IP Office Release 7.0 supports a SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

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

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

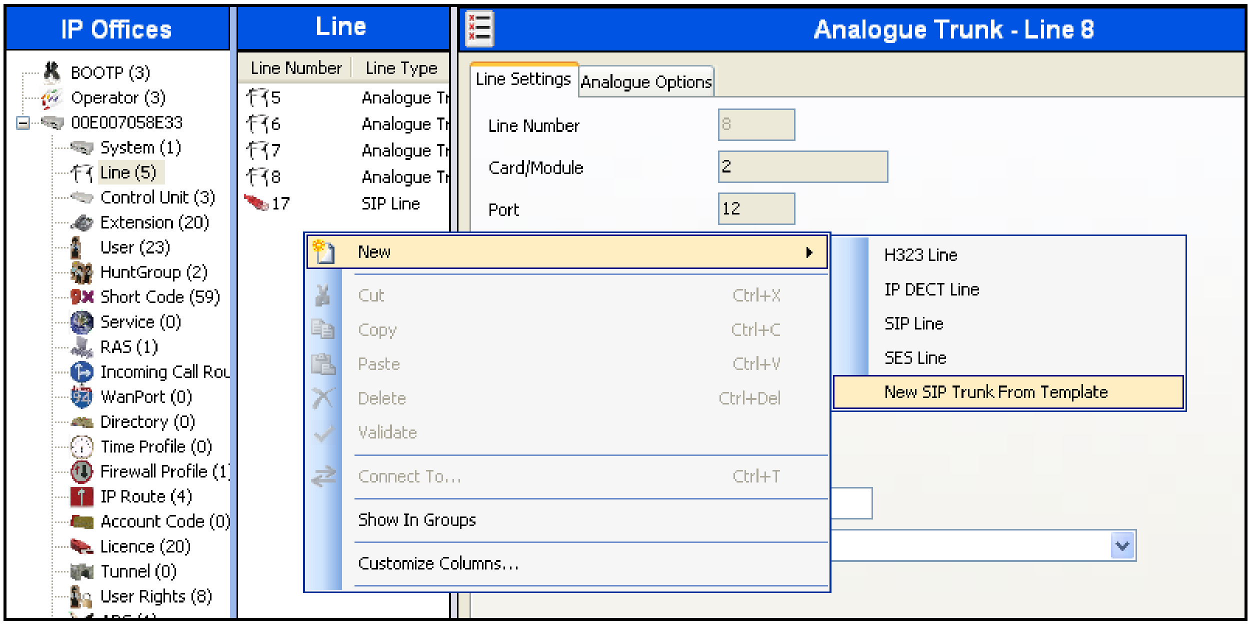
```
&lt;?xml version="1.0" encoding="utf-8"?&gt;
&lt;Template xmlns="urn:SIPTrunk-schema"&gt;
  &lt;TemplateType&gt;SIPTrunk&lt;/TemplateType&gt;
  &lt;Version&gt;20111028&lt;/Version&gt;
  &lt;SystemLocale&gt;enu&lt;/SystemLocale&gt;
  &lt;DescriptiveName&gt;Level 3 SIP Trunk&lt;/DescriptiveName&gt;
  &lt;ITSPDomainName&gt;10.1.1.86&lt;/ITSPDomainName&gt;
  &lt;SendCallerID&gt;CallerIDDIV&lt;/SendCallerID&gt;
  &lt;ReferSupport&gt;true&lt;/ReferSupport&gt;
  &lt;ReferSupportIncoming&gt;1&lt;/ReferSupportIncoming&gt;
  &lt;ReferSupportOutgoing&gt;1&lt;/ReferSupportOutgoing&gt;
  &lt;RegistrationRequired&gt;false&lt;/RegistrationRequired&gt;
  &lt;UseTelURI&gt;false&lt;/UseTelURI&gt;
  &lt;CheckOOS&gt;true&lt;/CheckOOS&gt;
  &lt;CallRoutingMethod&gt;1&lt;/CallRoutingMethod&gt;
  &lt;OriginatorNumber /&gt;
  &lt;AssociationMethod&gt;SourceIP&lt;/AssociationMethod&gt;
  &lt;ITSPProxy&gt;10.1.1.86&lt;/ITSPProxy&gt;
  &lt;LayerFourProtocol&gt;SipUDP&lt;/LayerFourProtocol&gt;
  &lt;SendPort&gt;5070&lt;/SendPort&gt;
  &lt;ListenPort&gt;5060&lt;/ListenPort&gt;
  &lt;DNSServerOne&gt;0.0.0.0&lt;/DNSServerOne&gt;
  &lt;DNSServerTwo&gt;0.0.0.0&lt;/DNSServerTwo&gt;
  &lt;CallsRouteViaRegistrar&gt;true&lt;/CallsRouteViaRegistrar&gt;
  &lt;SeparateRegistrar /&gt;
  &lt;CompressionMode&gt;AUTOSELECT&lt;/CompressionMode&gt;
  &lt;UseAdvVoiceCodecPrefs&gt;true&lt;/UseAdvVoiceCodecPrefs&gt;
```

```
 <AdvCodecPref>G.729(a) 8K CS-ACELP,G.711 ULAW 64K</AdvCodecPref>
 <CallInitiationTimeout>6</CallInitiationTimeout>
 <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
 <VoipSilenceSupression>false</VoipSilenceSupression>
 <ReinviteSupported>true</ReinviteSupported>
 <FaxTransportSupport>FOIP_T38</FaxTransportSupport>
 <UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
 <CodecLockdown>false</CodecLockdown>
 <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>
 <SIPCredentials>
   <Expiry>60</Expiry>
   <RegistrationRequired>false</RegistrationRequired>
 </SIPCredentials>
</Template>
```


To import the above template into a new installation:

1. On the PC where IP Office Manager was installed, copy and paste the above template into a text document named **US_Level3_SIPTrunk.xml**. Move the .xml file to the IP Office Manager template directory (C:\Program Files\Avaya\IP Office\Manager\Templates). It may be necessary to create this directory.

2. Import the template into an IP Office installation by creating a new SIP Line as shown in the screenshot below. In the Navigation Pane on the left, right-click on **Line** then navigate to **New → New SIP Trunk From Template**:



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

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.