Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Cablevision Optimum Voice SIP Trunking with Avaya IP Office Release 8.1 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between Cablevision Optimum Voice and Avaya IP Office Release 8.1.

Optimum Voice SIP Trunking provides PSTN access via a SIP trunk between the business site and the Cablevision cable network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the business.

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

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between Cablevision Optimum Voice and Avaya IP Office Release 8.1.

In the sample configuration, the Avaya IP Office (IPO) solution consists of an Avaya IP Office 500v2 running Release 8.1 software, Avaya Voicemail Pro messaging application, SIP-based Avaya softphones (IP Office Softphone and Flare® Experience for Windows), and Avaya H.323 and SIP hard phones.

Optimum Voice SIP Trunking provides PSTN access via a SIP trunk between the business site and the Cablevision cable network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the business.

2. General Test Approach and Test Results

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.

The general test approach was to configure a simulated business site using Avaya IP Office to connect to Optimum Voice SIP Trunking service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed in **Section 2.1**

2.1. Interoperability Compliance Testing

A simulated business site with Avaya IP Office was connected to Optimum Voice SIP Trunking. To verify SIP trunking interoperability, the following features and functionality were exercised during the interoperability compliance test:

- Response to SIP OPTIONS queries.
- Incoming PSTN calls to H.323 and SIP telephones at the business site. All inbound PSTN calls were routed to the business site across the SIP trunk from the service provider..
- Outgoing PSTN calls from H.323 and SIP telephones at the business site. All outbound PSTN calls were routed from the business site across the SIP trunk to the service provider
- Various call types including: local, long distance, outbound toll-free, international, operator and directory assistance.
- G.711MU codec.
- Caller ID presentation and Caller ID restriction.
- DTMF transmission using in-band tones.
- 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 to PSTN mobile phones on inbound calls.
- Use of SIP REFER and SIP INVITE for call redirection to PSTN.
- Inbound and outbound long-duration calls stability.
- Inbound and outbound long holding time call stability.
- Response to incomplete call attempts and trunk errors.

Items not supported by Optimum Voice SIP Trunking or not tested included the following:

- Inbound toll-free and emergency calls (911) are supported but were not tested as part of the compliance test.
- Operator-assisted calls (0 + 10-digits) are not supported.
- DTMF transmission using RFC 2833 is not supported.
- T.38 Fax is not supported.

2.2. Test Results

Interoperability testing of Optimum Voice SIP Trunking with Avaya IP Office R8.1 was completed with successful results for all test cases with the exception of the observations/limitations described below.

- **OPTIONS**: Cablevision configured its SIP Trunking circuit not to send OPTIONS to the business site. Cablevision responded to OPTIONS from the business site with either “401 Unauthorized” or “200 OK”.
- **Codec Mismatch**: If the codec configured on IPO did not match codec configured for the Cablevision network, Cablevision returned "480 No Routes Found" on outbound INVITE to the premise EdgeMarc router and the router in turn issued "480 Temporarily Unavailable" to IPO. A more appropriate status message, like "488 Not Acceptable Here", could have been used.
- **Idle Appearance FNE (Feature Name Extension)**: When this feature is configured on IPO, the inbound call to this FNE extension will receive a dial tone, and upon input of destination number from the caller by DTMF, the call will be routed to the dialed destination. During compliance testing, the PSTN caller received dial tone which sustained during and after digit input on the calling PSTN phone, and the inbound call was not routed to the expected destination as dialed. IPO support investigated this issue and determined the cause to be inband DTMF used by the Optimum SIP Trunking service. This feature should work properly with outband DTMF per RFC2833.
- **Calling Party Display for Mobile Twinning**: For an inbound call answered at the twinned mobile number, the connected party displayed was the pilot number instead of the original PSTN caller number. IPO passed the original PSTN caller number in the From header of outbound INVITE, but the premise EdgeMarc router changed the PSTN caller number to the pilot number when passing the outbound INVITE to the Cablevision Optimum network.
- **Hold/Resume**: There was no SIP signaling from the network when an active call was placed on hold or resumed from hold at the PSTN phone. User experience was not negatively affected in these cases.
- **Off-net Call Redirection Using REFER**: When REFER was used for forwarding an inbound call to another PSTN party or for blind transfer (by an H.323 phone) of an existing

call off-net to another PSTN party (i.e., the IPO extension completes the transfer without conversing with the transfer destination party first), the outbound REFER message did not pass Digest Authentication by the Cablevison Optimum SIP Trunking service, and the call redirection would fail. The alternative method of off-net call redirection using INVITE was successfully verified by the compliance test and is the recommended configuration (see **Section 5.4**).

2.3. Support

For technical support on the Avaya products described in these Application Notes visit http://support.avaya.com.

For technical support on Cablevision Optimum Voice SIP Trunking, contact Cablevision using the *Local Business Spotlight* links at www.optimumbusiness.com, or call the technical support number at 855-728-2455 for business customers.

3. Reference Configuration

Figure 1 illustrates the test configuration showing a business site connected to Cablevision Optimum Voice SIP Trunking.

Located at the edge of the business site is an Edgewater Networks EdgeMarc 4552 router provided by Cablevision to business customers to serve as a demarcation point for the SIP Trunking service. For testing purposes, the WAN interface of the router was connected to a broadband public Internet connection to access the service. In an actual customer configuration, the router would connect to a cable modem to access the Cablevision network. The LAN port 1 of the router is connected to the LAN at the business site.

Also located at the business site is an Avaya IP Office 500v2 with the COMBO6210/ATM4 expansion card which provides connections for 6 digital stations, 2 analog stations, 4 analog trunks as well as 10-channel VCM (Voice Compression Module) for supporting VoIP codecs. Endpoints include various Avaya IP Telephones (with H.323 and SIP firmware) and SIP-based Avaya softphones (Avaya IP Office Softphone and Avaya Flare® Experience for Windows). The site also has a Windows PC running Avaya Voicemail Pro for providing 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.

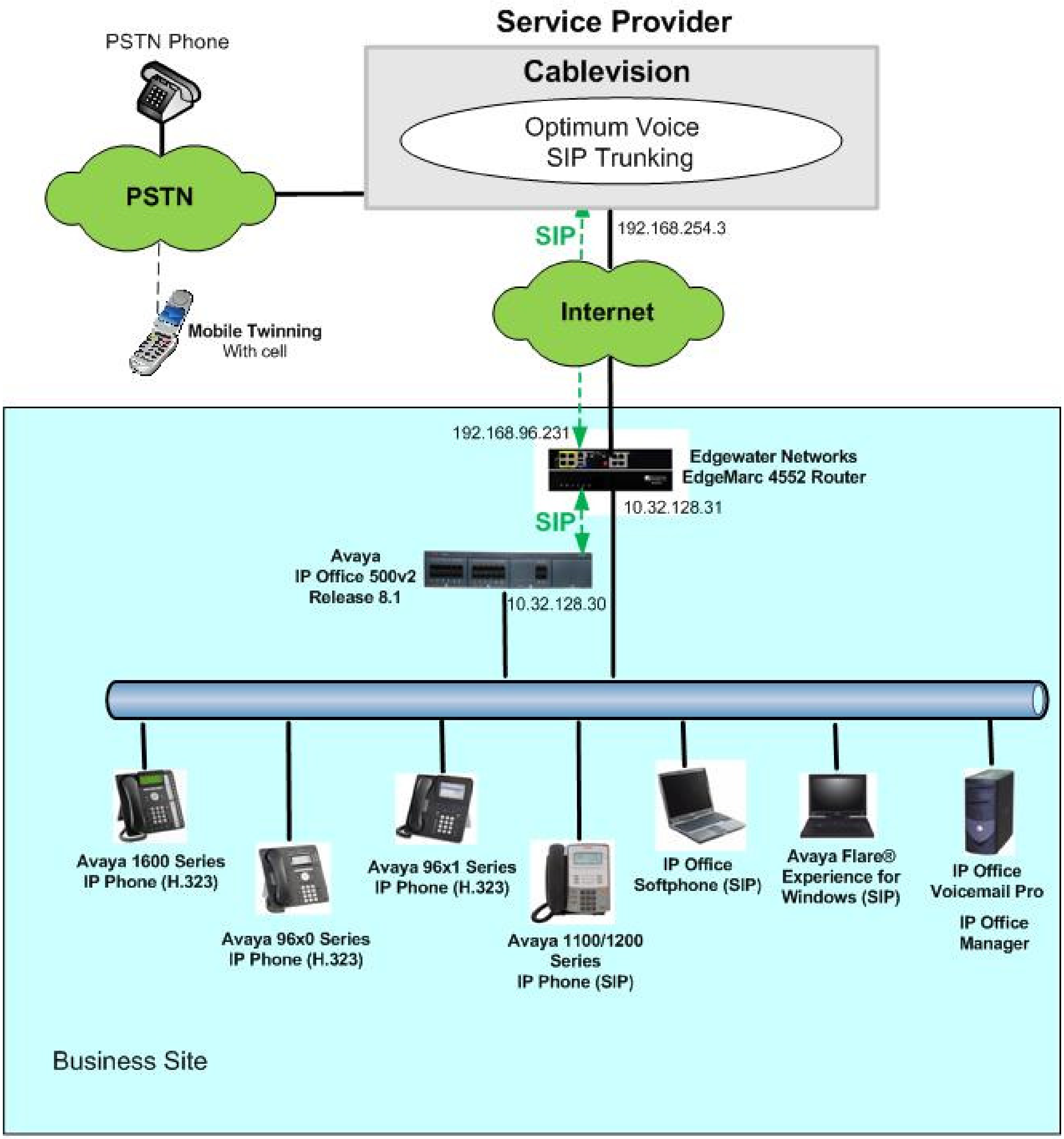


Figure 1: Test Configuration

For security purposes, any actual public IP addresses used in the compliance test were changed to 192.168.x.x throughout these Application Notes where the 3rd and 4th octets were retained from the real addresses.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to Cablevision. The short code of 9 is stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Cablevision. For calls within the North American Numbering Plan (NANP), the user dialed 11 (1 + 10) digits. Thus for these NANP calls, Avaya IP Office sent 11 digits 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, Optimum Voice SIP Trunking sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

In an actual customer configuration, the business site also includes a Cablevison-provided cable modem between the service provider and the premise EdgeMarc router. A complete discussion of the configuration of the cable modem is beyond the scope of these Application Notes.

4. Equipment and Software Validated

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

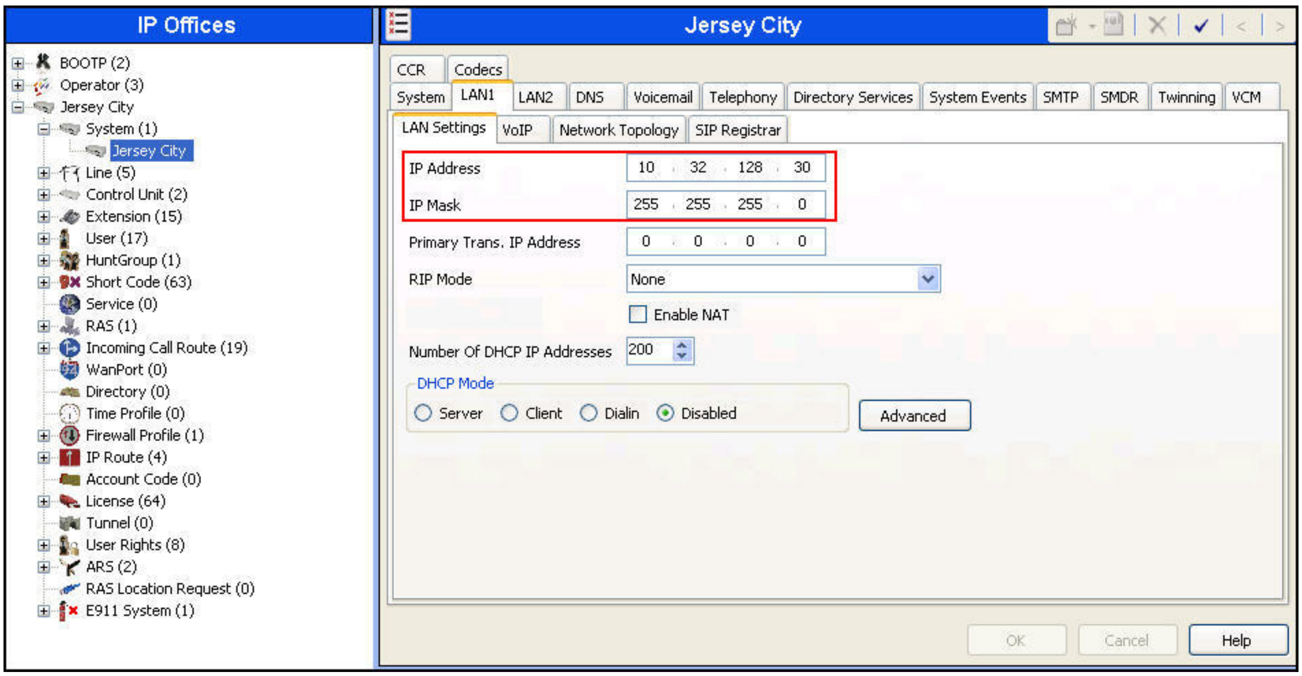
Avaya Telephony Components	
Equipment	**Release**
Avaya IP Office 500v2	8.1 (69)
Avaya IP Office COMBO6210/ATM4 Module	10.1 (69)
Avaya Voicemail Pro	8.1 (9203)
Avaya IP Office Manager	10.1 (69)
Avaya 1120E IP Telephone (SIP)	4.03.12.00
Avaya 1616 IP Telephone (H.323)	Avaya one-X Deskphone 1.3
Avaya 9611G IP Telephone (H.323)	Avaya one-X Deskphone 6.2
Avaya 9630G IP Telephone (H.323)	Avaya one-X Deskphone 3.1
Avaya IP Office Softphone	3.2.3.48 67009
Avaya Flare® Experience for Windows	1.1.1.7
Cablevision Components	
Equipment	**Release**
Edgewater Networks EdgeMarc 4552	11.6.14.1

5. Configure Avaya IP Office

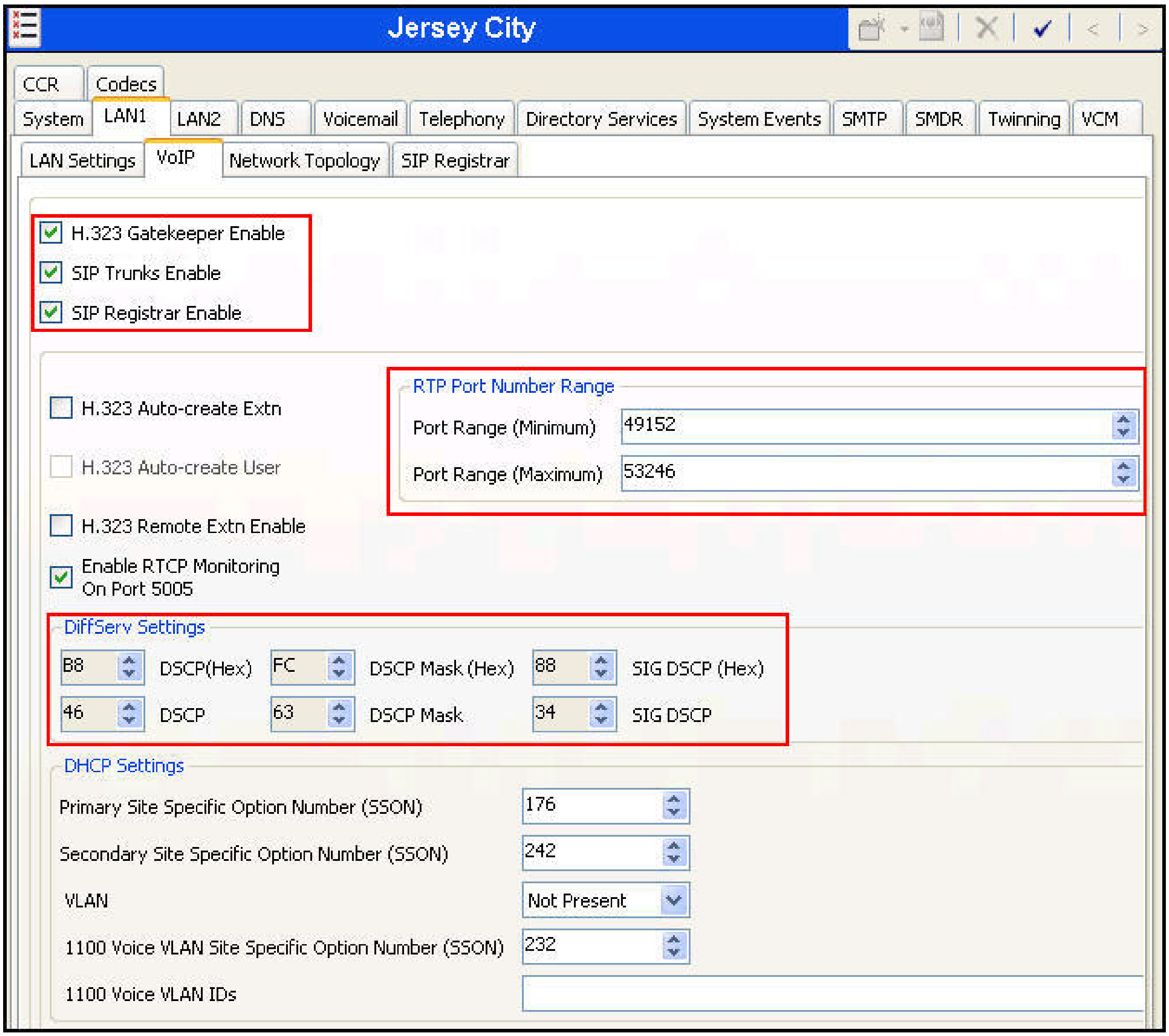
This section describes the Avaya IP Office configuration to support connectivity to Cablevision Optimum Voice SIP Trunking. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window (not shown), and log in with the appropriate credentials. A management window will appear similar to the one in the next section. All the Avaya IP Office configurable components are shown in the left pane known as the Navigation Pane. The pane on the right is the Details Pane. 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 be already in place.

5.1. LAN1 Settings

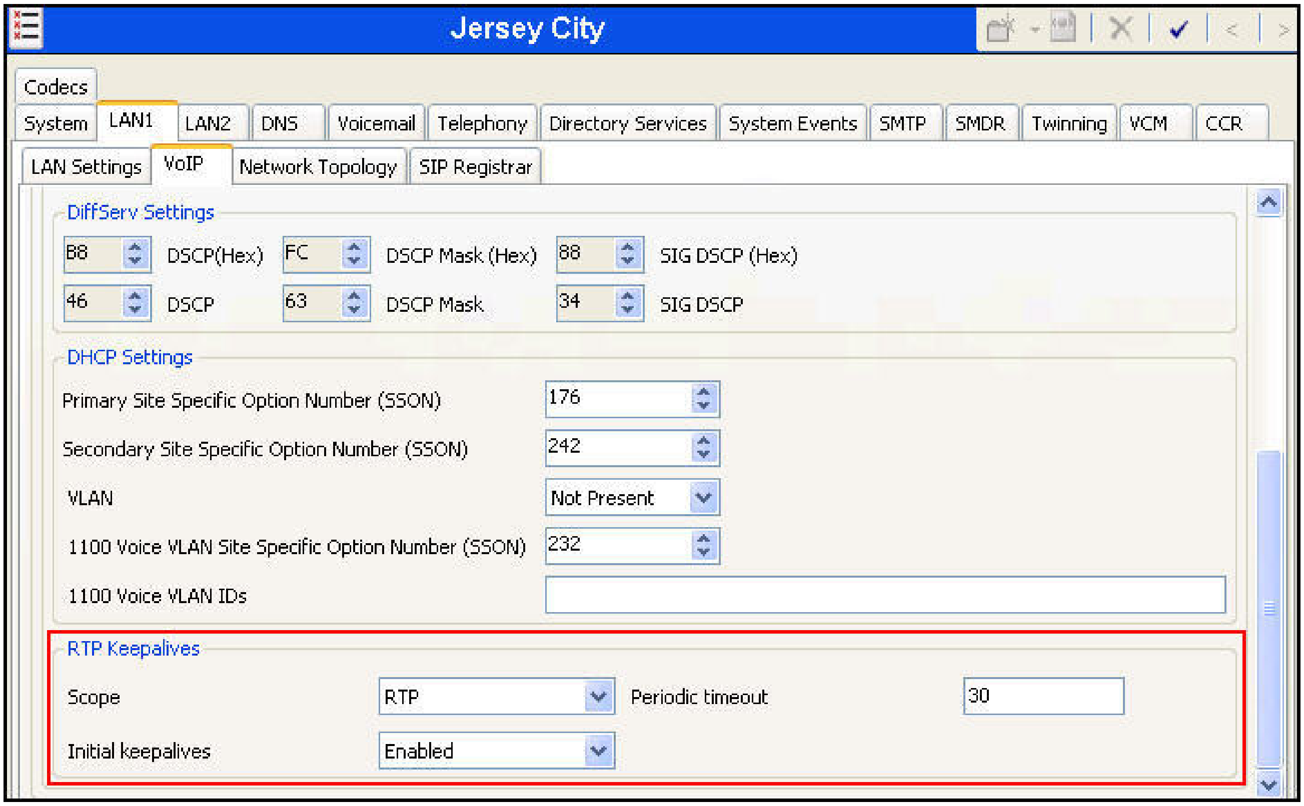
In the sample configuration, ***Jersey City*** was used as the system name and the LAN1 port was used to connect the Avaya IP Office to the business site LAN network. To access the LAN1 settings, first navigate to **System → Jersey City** in the Navigation Pane and then select the **LAN1→ LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are set from values shown in **Figure 1**. All other parameters should be set according to customer requirements.



On the **VoIP** tab under **LAN1** in the Details Pane, check the **SIP Trunks Enable** box to enable SIP trunks. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol. Likewise, the **SIP Registrar Enable** box is checked to allow the use of Avaya SIP endpoints. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. 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.



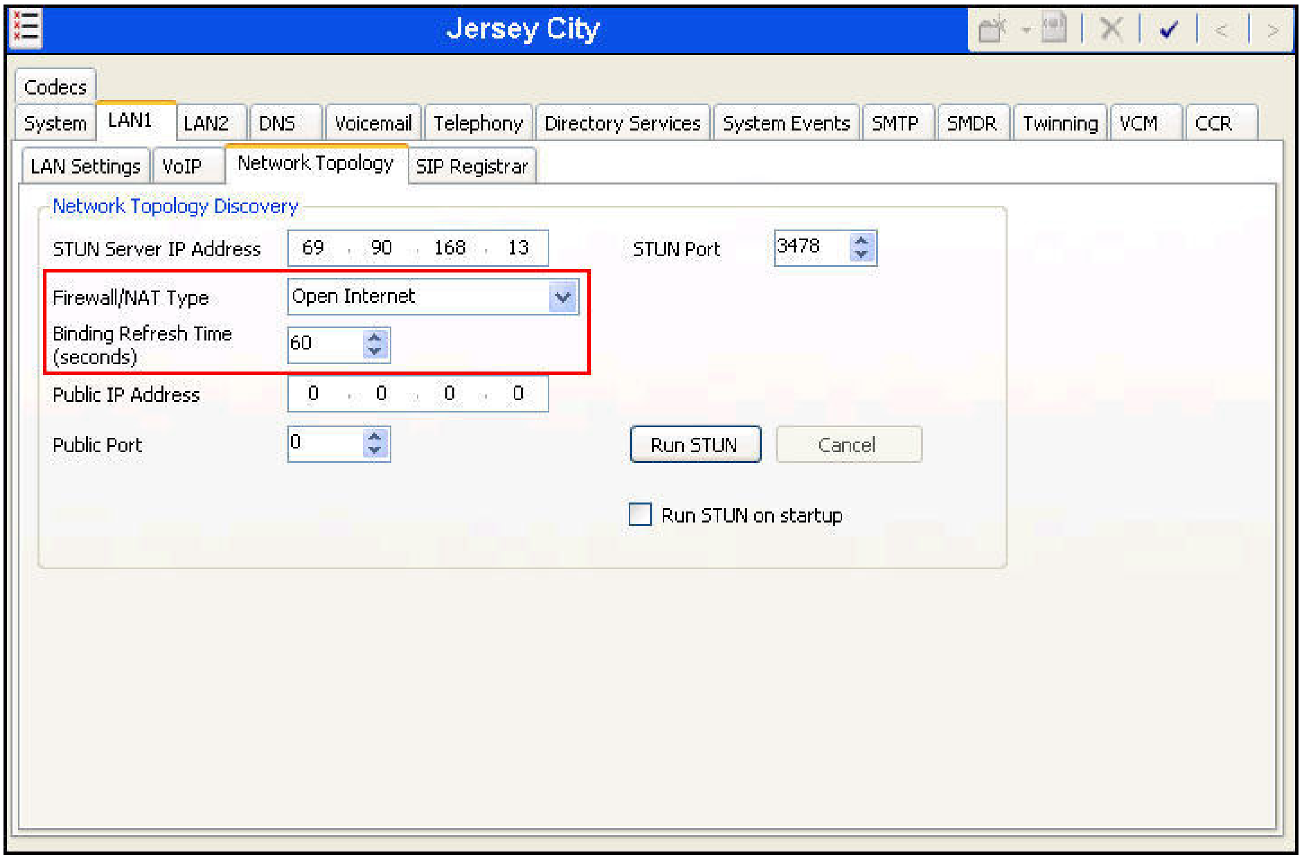
Scrolling down to the **RTP Keepalives** section. Select ***RTP*** for **Scope**; select ***Enabled*** for **Initial keepalives**; enter ***30*** for **Periodic timeout**. These settings direct IP Office to send artificial RTP packets toward the service provider at the start of the call to prevent audio loss in certain offnet call redirection scenarios like twinning inbound call to a mobile phone via the PSTN. This configuration was necessary because the service provider expected IP Office endpoint to send RTP packets first even though there was no IP Office media endpoint involved in this call situation since the call had been re-directed back to PSTN.



All other parameters should be set according to customer requirements.

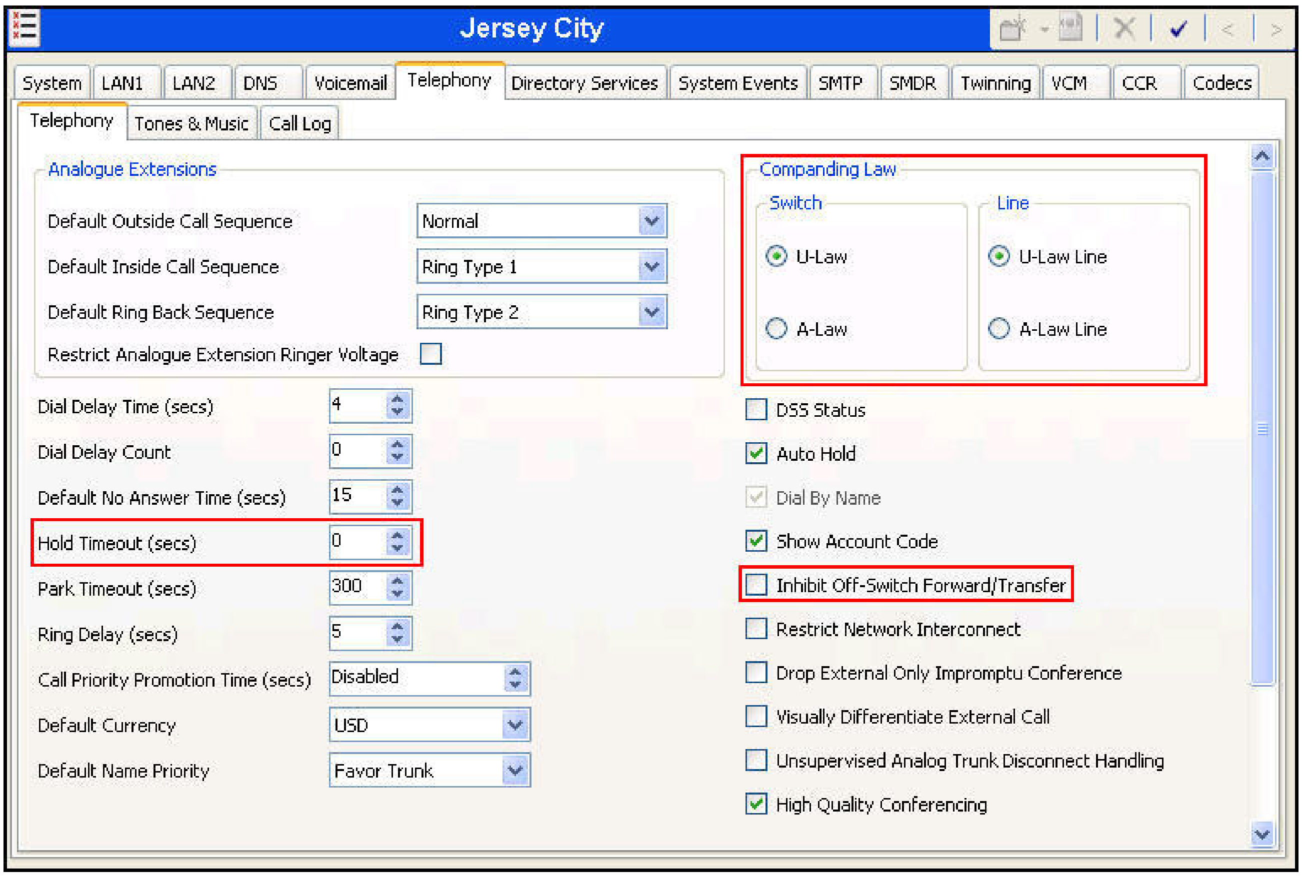
On the **Network Topology** tab in the Details Pane, configure the following parameters:

- 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 ***60***. 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 5.8** for complete details.



5.2. System Telephony Settings

Navigate to the **Telephony → Telephony** Tab on the Details Pane. Choose the **Companding Law** typical for the business location. For North America, ***ULAW*** is used. Check or uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow (option box unchecked) or disallow (option box checked) call forwarding and call transfer to the PSTN via the service provider across the SIP trunk per customer business policies. Enter or select ***0*** for **Hold Timeout (secs)** so that calls on hold will not time out.



5.3. Twinning Calling Party Settings

Navigate to the **Twinning** Tab on the Details Pane. For the compliance test, the **Send original calling party information for Mobile Twinning** box was checked as shown below.

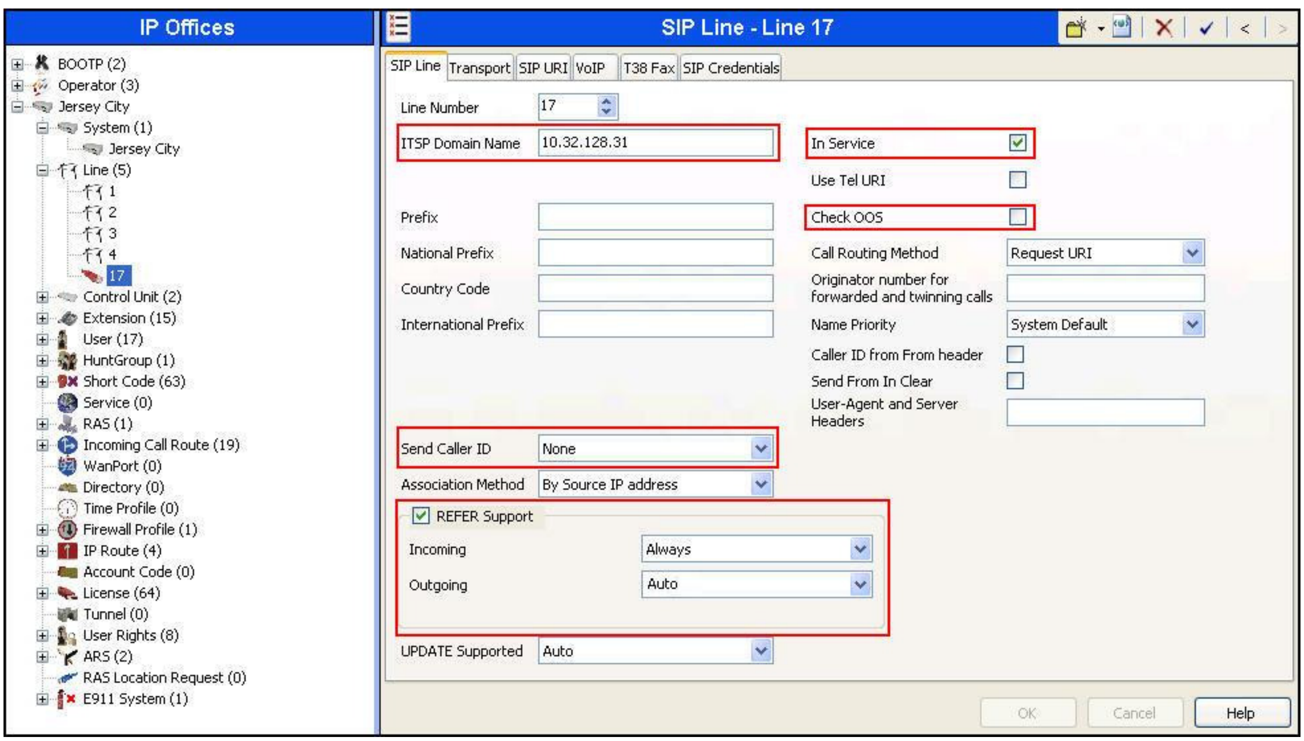
If this box is checked, Avaya IP Office will send the following in the SIP From Header of the INVITE message. The value in the From header determines what gets displayed as the calling party number:

- On calls from an internal extension to another internal phone with twinning enabled, Avaya IP Office will send the calling party number of the originating extension (i.e., DID number assigned to this extension) to the twinned destination.
- On calls from the PSTN to an internal phone with twinning enabled, Avaya IP Office will send the originating PSTN calling party number to the twinned destination.

5.4. Administer SIP Line

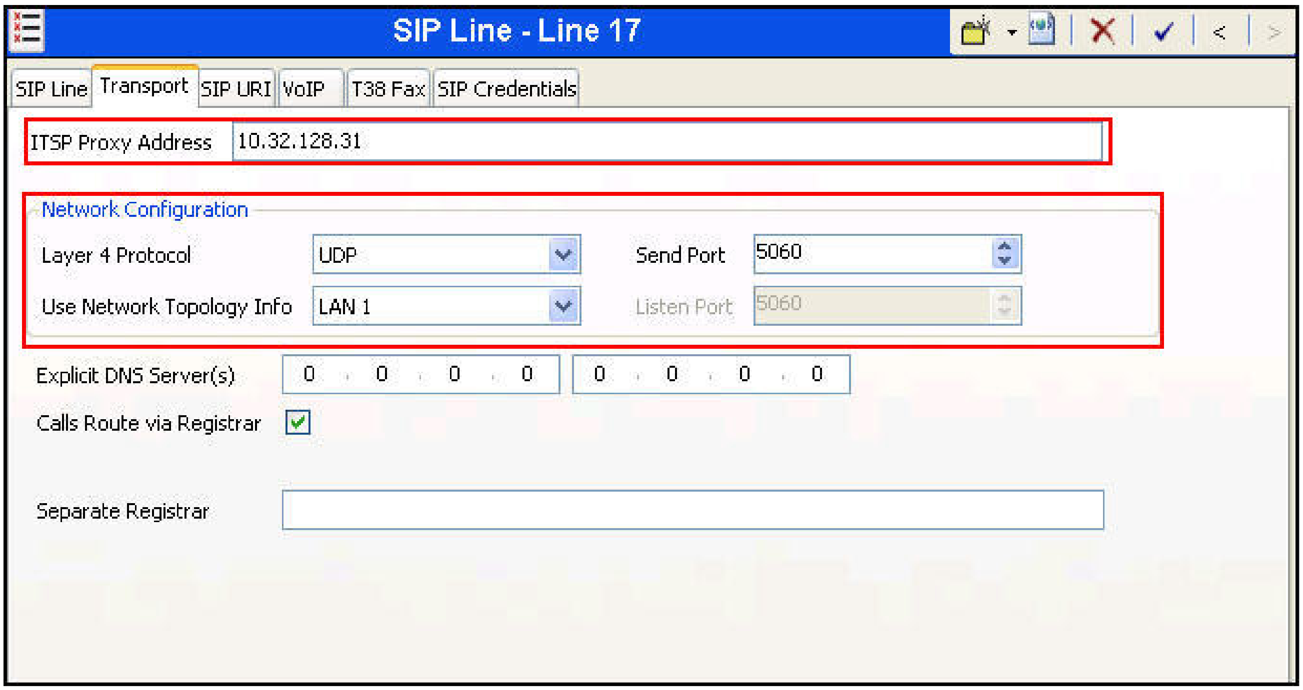
A SIP line is needed to establish the SIP connection between Avaya IP Office and Cablevision Optimum Voice SIP Trunking. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New → SIP Line**. On the **SIP Line** tab in the Details Pane, configure the parameters for the SIP Line. Shown below is a previously administered SIP Line used for the compliance test.

- Set the **ITSP Domain Name** to the IP address of the LAN port 1 interface of the premise EdgeMarc router.
- Set **Send Caller ID** to ***None***. For the compliance test, this parameter was ignored since the **Send original calling party information for Mobile Twinning** box was checked in **Section 4.3**.
- Check the **In Service** box.
- Uncheck the **Check OOS** box. Cablevision Optimum Voice SIP Trunking responds to the SIP OPTIONS messages sent by Avaya IP Office sometimes with "401 Unauthorized" and sometimes with "200 OK". Thus, this setting will prevent Avaya IP Office from taking the SIP trunk out of service.
- Default values may be used for all other parameters.
- If the service provider supports REFER for off-net call redirection (transfer and forward), check **REFER Support** and select ***Always*** for both **Incoming** and **Outgoing**. If the service provider prefers using INVITE for off-net call redirection, select ***Auto*** for **Outgoing** as was the case for the compliance test.



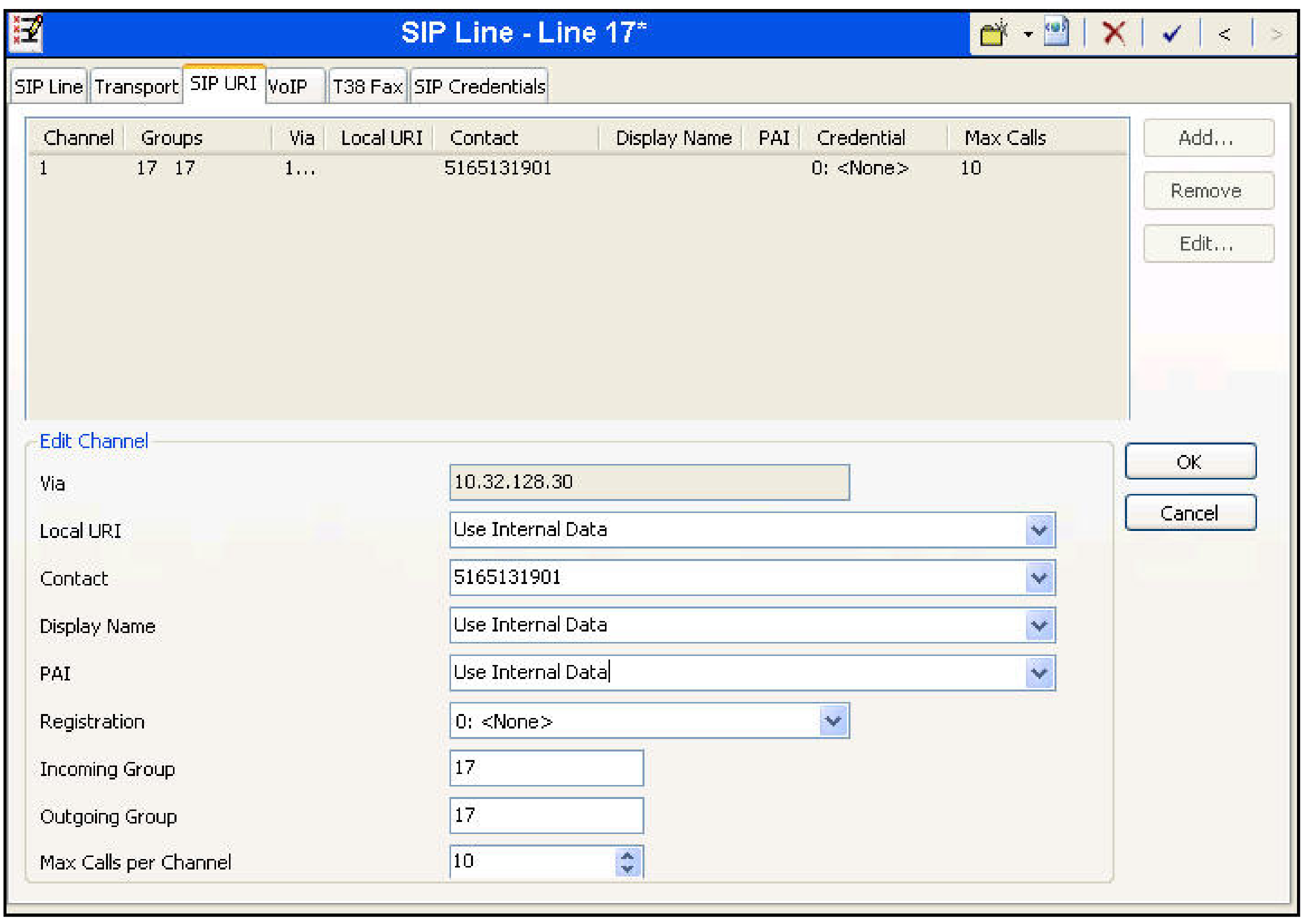
Select the **Transport** tab.

- Set **ITSP Proxy Address** to the IP address of the LAN port 1 interface of the premise EdgeMarc router.
- Set the **Layer 4 Protocol** to ***UDP***.
- Set **Use Network Topology Info** to the network port configured in **Section 5.1**. This associates the SIP Line with the parameters in the **System → LAN1 → Network Topology** tab.
- Set the **Send Port** to ***5060***
- Other parameters may retain default values in the screen below.



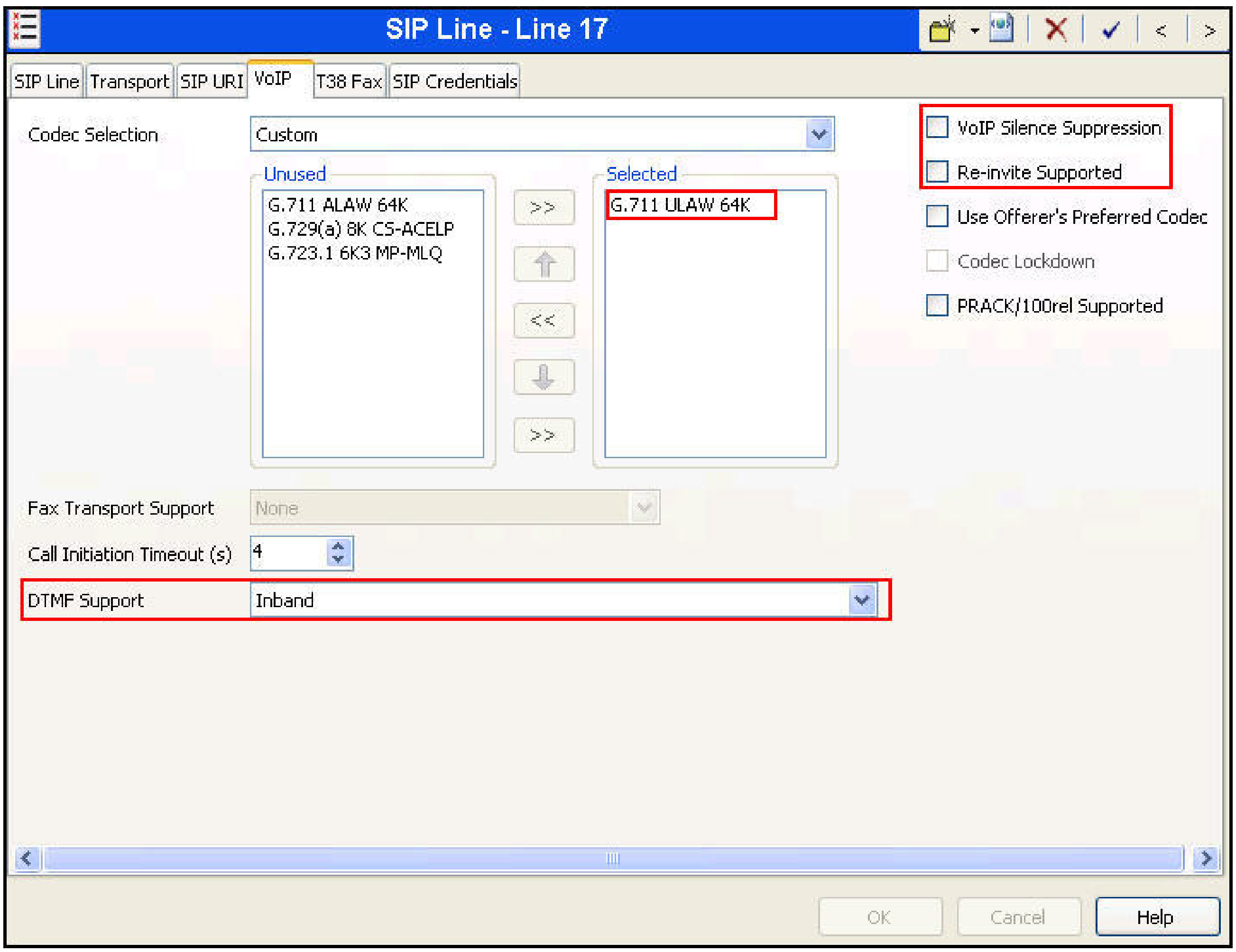
Each SIP URI that Avaya IP Office will accept on this line must be created. To create a SIP URI entry, first select the **SIP URI** tab. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the screen below, a previously configured entry is edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an Avaya IP Office user. The entry was created with the parameters shown below.

- Set **Local URI**, 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 5.6**.
- Set **Contact** to the pilot number provided by Cablevision. This number is used to authenticate all calls from the business site.
- Set **PAI** to ***Internal Data***. With this setting IP Office will populate the SIP P-Asserted-Identity header on outgoing calls with the **SIP Name** set in the **SIP** tab of the call initiating **User** as shown in **Section 5.6**. If ***None*** is selected instead, the P-Preferred-Identity header is used instead of P-Asserted-Identity for compatibility with legacy networks.
- 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.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls allowed.



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

- Set the **Codec Selection** to ***Custom***.
- Choose ***G.711 ULAW 64K*** from the **Unused** box and move this selection to the **Selected** box. This is the only codec supported by Cablevision Optimum Voice SIP Trunking.
- Set the **DTMF Support** field to ***Inband***. This directs Avaya IP Office to send DTMF tones as inband tones in the audio stream instead of RTP events as per RFC2833.
- Uncheck the **VoIP Silence Suppression** option box.
- Uncheck the **Re-invite Supported** option box (this will also automatically set ***None*** for **Fax Transport Support**).
- Default values may be used for all other parameters.

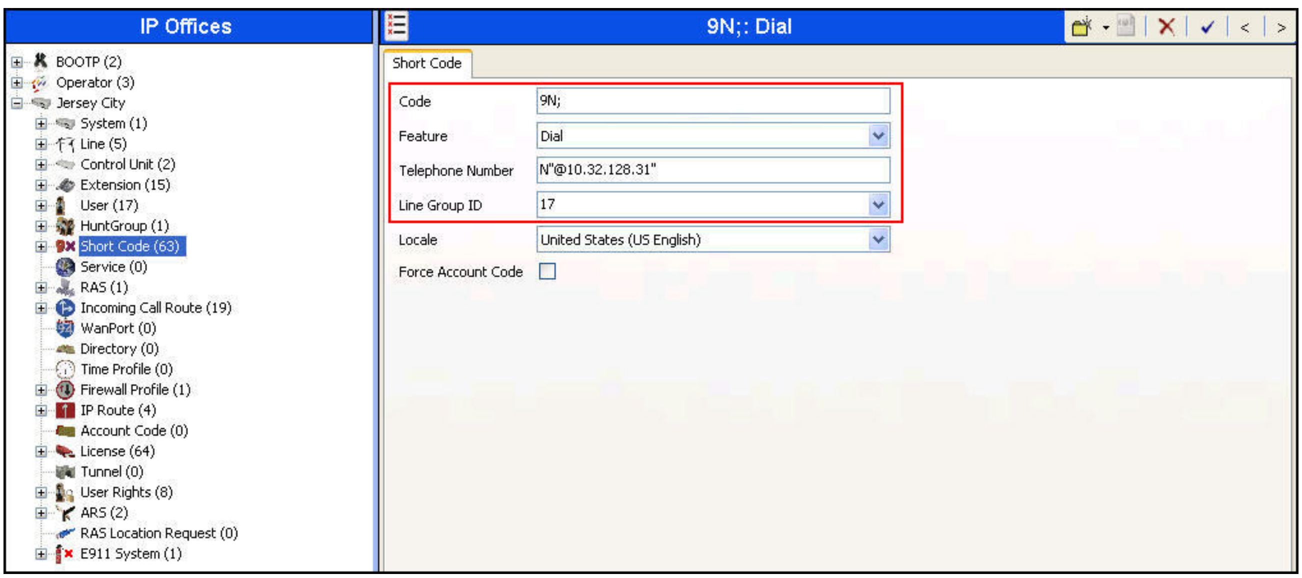


5.5. Short Code

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** (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. 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"@10.32.128.31"***. 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 10.32.128.31 is the IP address of the LAN port 1 interface of the premise EdgeMarc router.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.4**. This short code will use this line group when placing the outbound call.

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

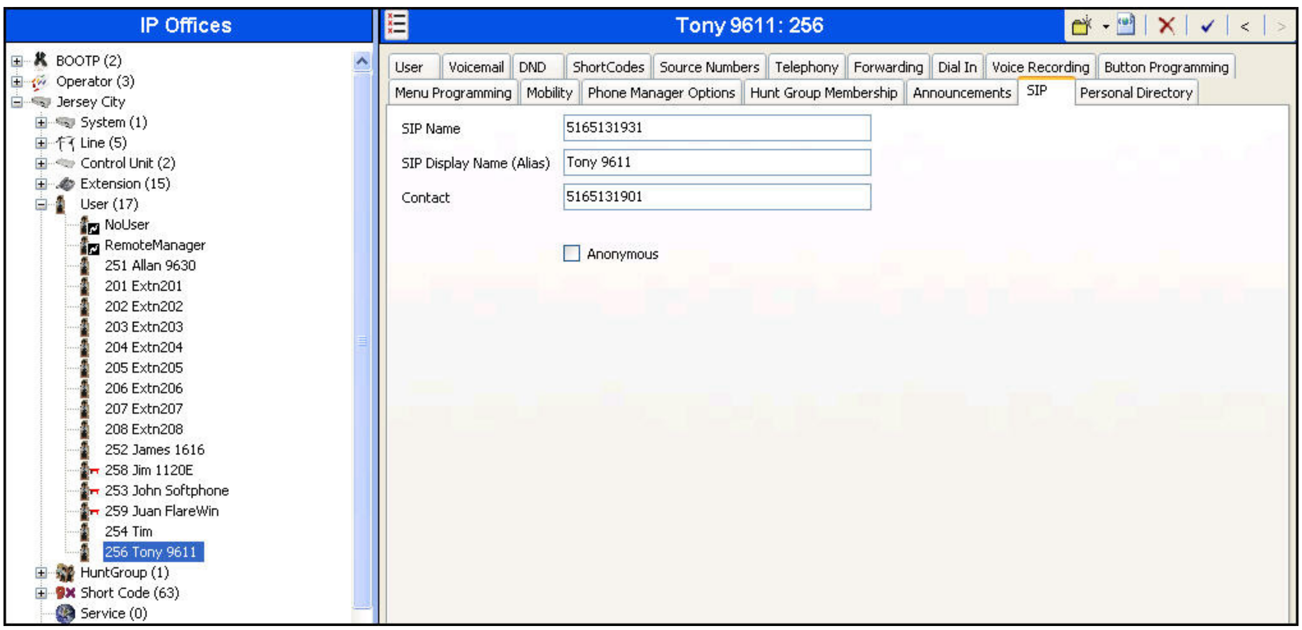


5.6. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.4**. To configure these settings, first navigate to **User**→*Name* in the Navigation Pane where *Name* is the name of the user to be modified. Select the **SIP** tab in the Details Pane. The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From and Contact headers for outgoing SIP trunk calls and allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.4**).

The example below shows the settings for user "Tony 9611" (at Extension 256). The **SIP Name** is set to one of the DID numbers provided by Cablevision. The Contact is set to the pilot number. The Contact should be set to the pilot number for all users. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name.

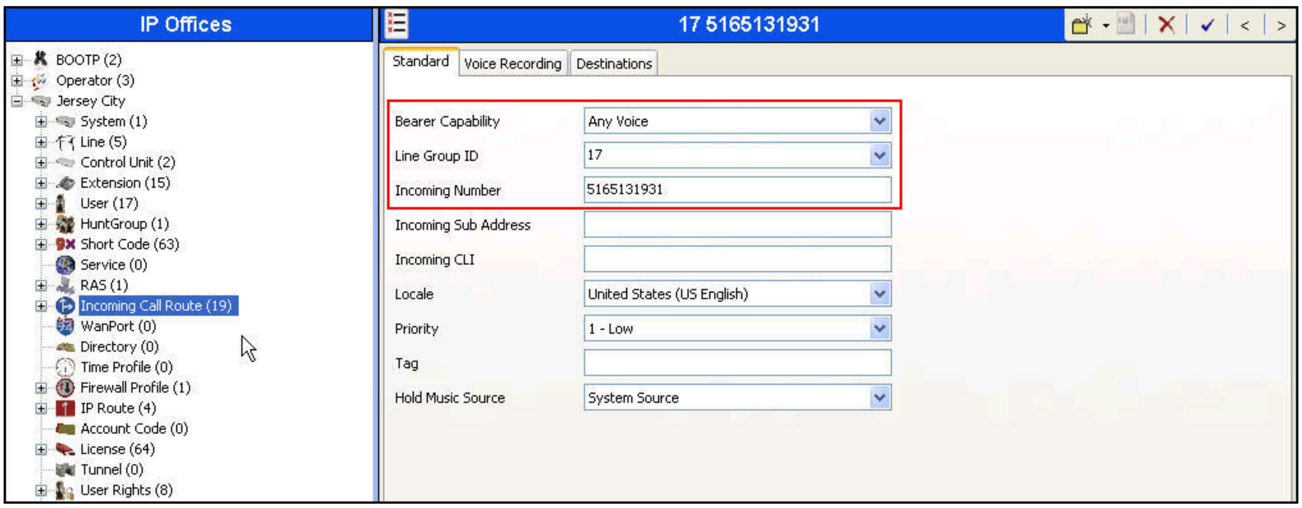
Click the **OK** button (not shown).



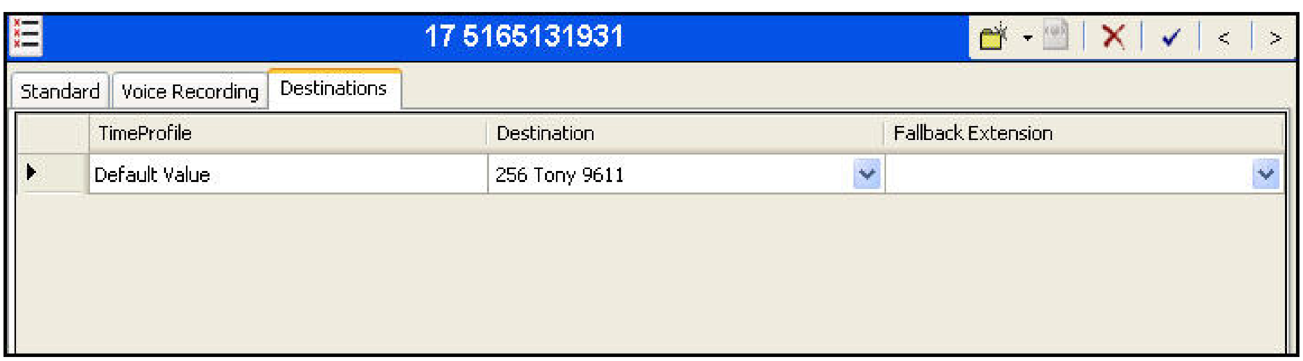
5.7. Incoming Call Route

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 Capability** to ***Any Voice***.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.4**.
- 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 5165131931 on line 17 are routed to user "Tony 9611" at Extension 256.

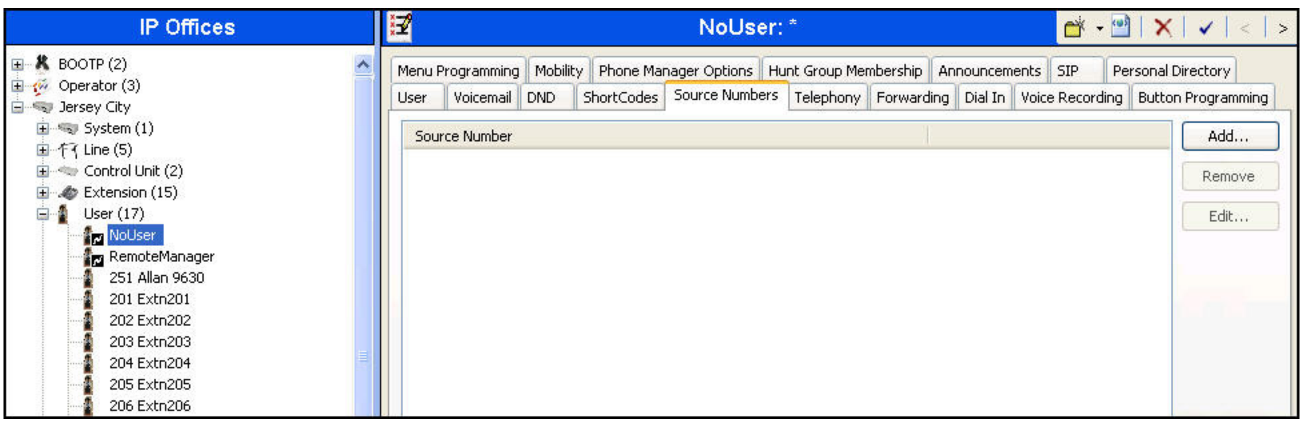


5.8. SIP Options

Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. By default, the IP Office Release 8.1 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 5.1** 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:

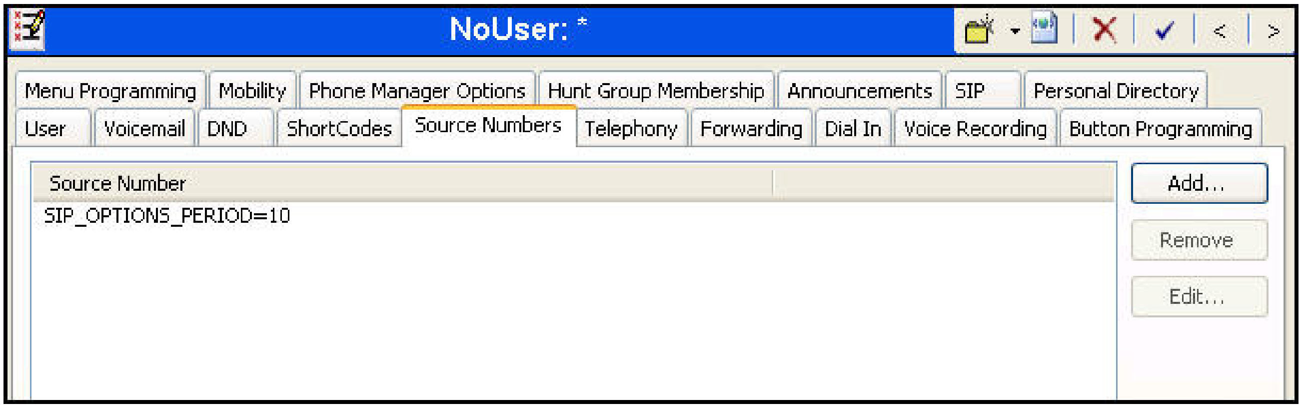
- To use the default value, set Binding Refresh = 0 or 300. OPTIONS will be sent at the 300 second frequency.
- To establish a period of less than 300 seconds, do not define the **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 seconds. The OPTIONS message 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**.

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. For the compliance test, an OPTIONS period of 60 seconds was desired. The **Binding Refresh Time** was set to ***60*** seconds in **Section 5.1**. There was no need to define **SIP_OPTIONS_PERIOD**.



5.9. Save Configuration

Navigate to **File → Save Configuration** (not shown) in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

6. Cablevision Optimum Voice SIP Trunking Configuration

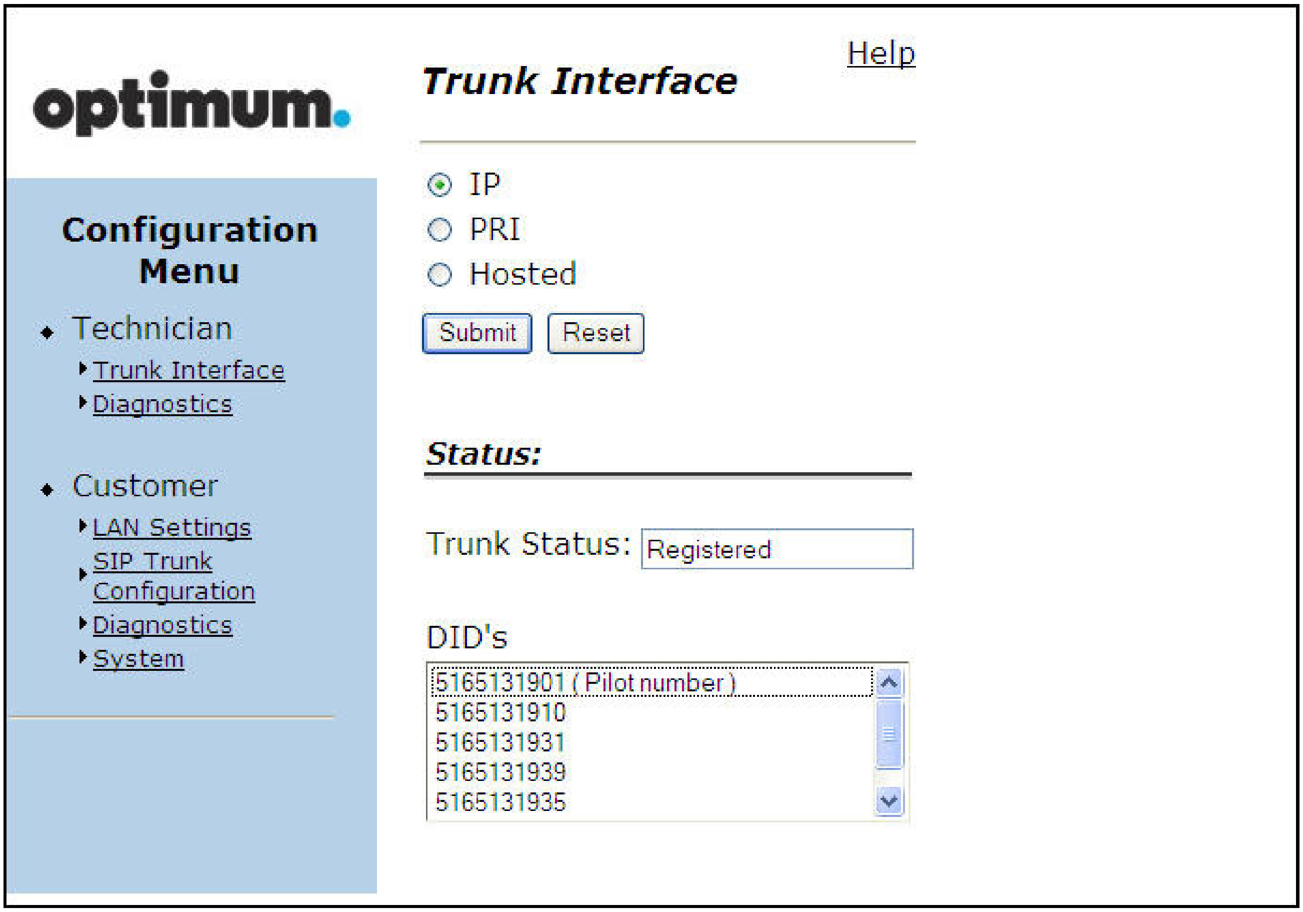
Cablevision is responsible for the configuration of Optimum Voice SIP Trunking including the WAN interface of the premise EdgeMarc router. Cablevision will configure the EdgeMarc router before shipping the device to the customer. Once installed on premise, Cablevision will be able to access the EdgeMarc router remotely via the WAN interface. In addition, Cablevision will provide the customer the necessary information to configure the Avaya IP Office SIP connection including:

- Supported codecs
- Pilot number
- DID numbers

The customer will be responsible for configuring the voice network interface of the EdgeMarc router, along with basic SIP trunk parameters. LAN port 1 of the EdgeMarc router is used for the voice network and will communicate with the Avaya IP Office. LAN port 4 of the EdgeMarc router is used for management and is pre-configured with IP address 10.10.200.1. To perform the customer configuration of the EdgeMarc router, connect a PC to LAN port 4 of the router and assign the PC an IP address on the same subnet as the router. From the PC, launch a Web browser and go to http://10.10.200.1. A login screen will appear. Enter login credentials provided by Cablevision.

Alternatively, from a PC connected to the same subnet to which the LAN port 1 of the EdgeMarc router is connected, launch a Web browser and go to the IP address of the LAN port 1 of the EdgeMarc router. In the configuration for the compliance test as illustrated in Figure 1, this is http://10.32.128.31.

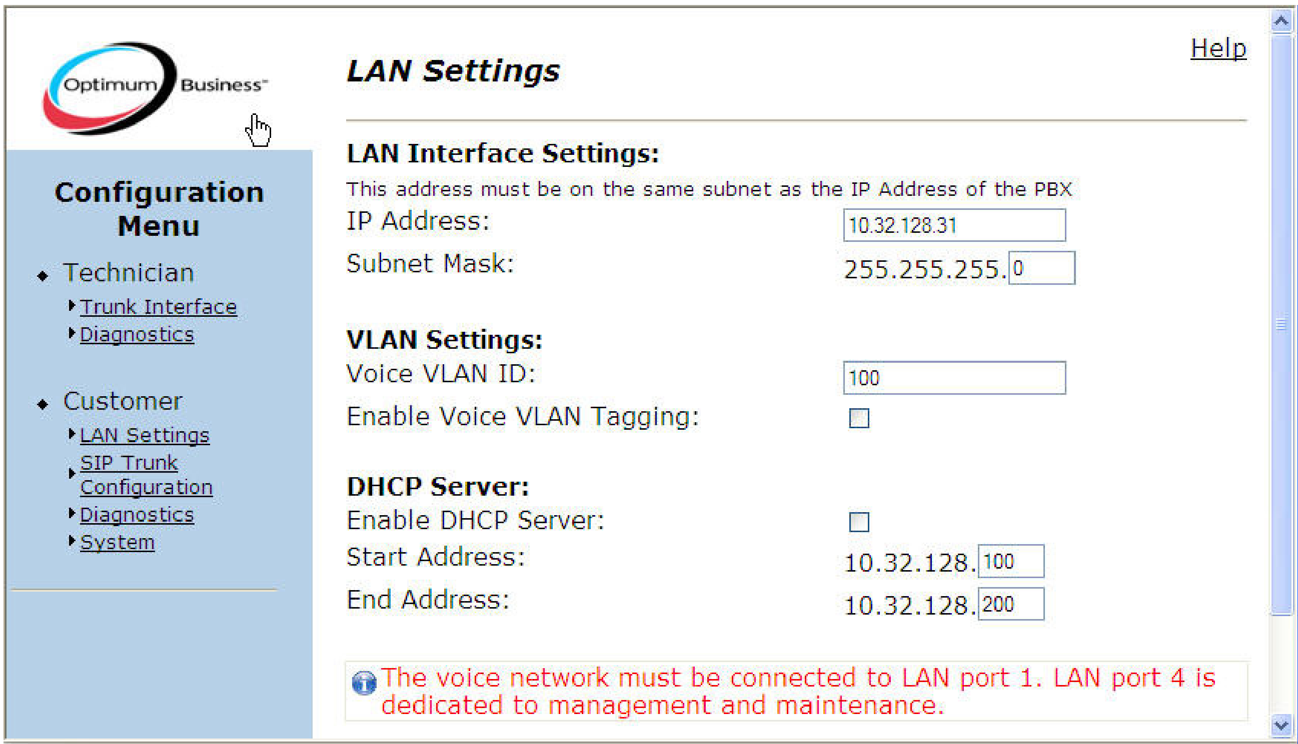
The **Trunk Interface** screen will be first presented. Note that this screen displays the pilot number and DID's provided by Cablevision for the customer account. It also shows the Trunk Interface as IP and Trunk Status as Registered. If the Trunk Status shows anything other than Registered, contact Cablevision support to troubleshoot the SIP Trunking circuit to the business site.



The **LAN Settings** screen will appear as shown below. Enter the following:

- **IP Address**: Enter the IP address assigned to LAN port 1 which will be used for SIP and RTP traffic. This IP address should be on the same subnet as the Avaya IP Office.
- **Subnet Mask**: Enter the subnet mask appropriate for the network where port 1 will be connected.

Default values were used for all other settings. Optionally, the **DHCP Server** functionality can be enabled. However, the compliance test did not use this functionality. Once complete, connect the voice network to LAN port 1.



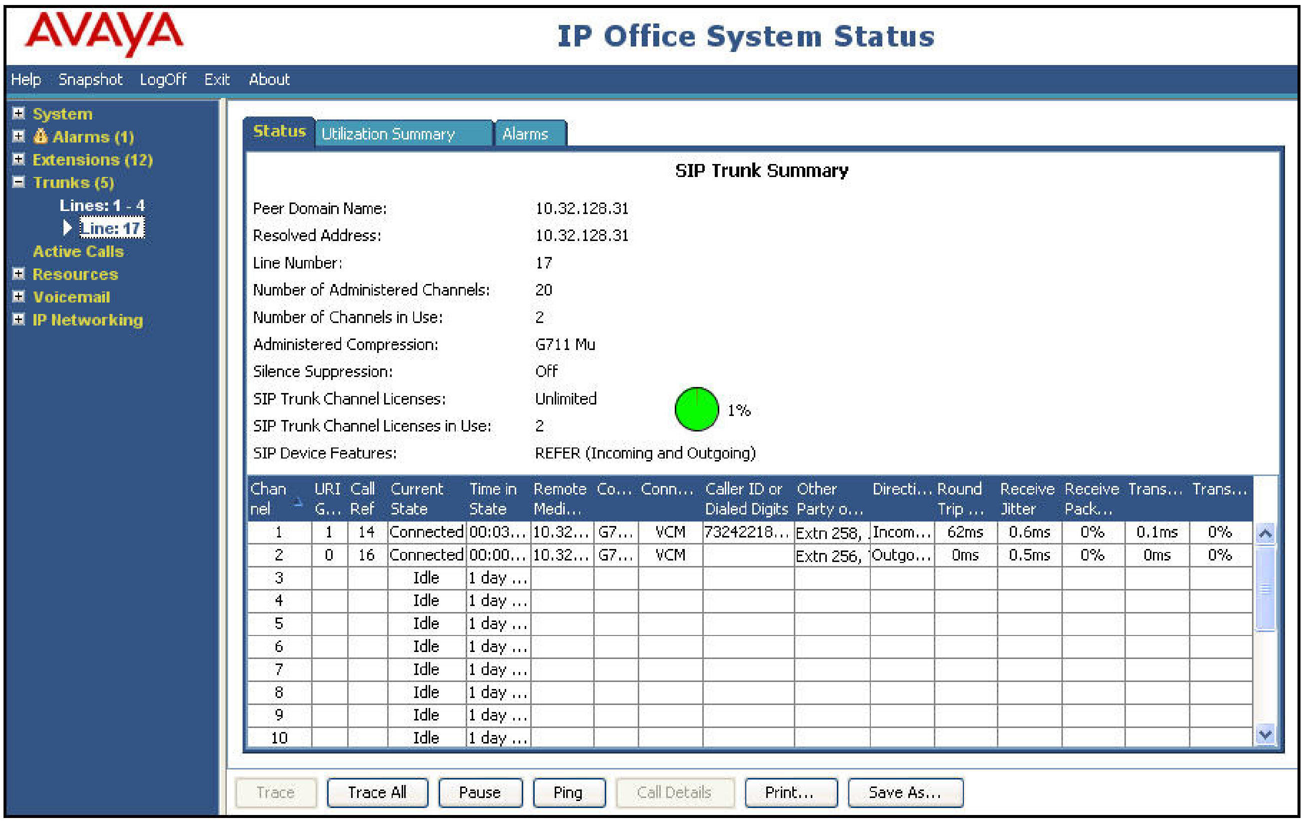
Select the **SIP Trunk Configuration** link from the navigation menu in the left pane. In the right pane, select ***Avaya IP Office*** from the pull-down menu for the **Select your PBX** field. Select the connection option labeled **Passive connection using the local, private IP address of the PBX**. In the **PBX address** field, enter the LAN1 IP address of the Avaya IP Office. Click **Submit**.

Note: the PBX name shown in the **Select your PBX** field could display some other name like "3CX" after the configuration is submitted. This should not impact the operation of the EdgeMarc router.

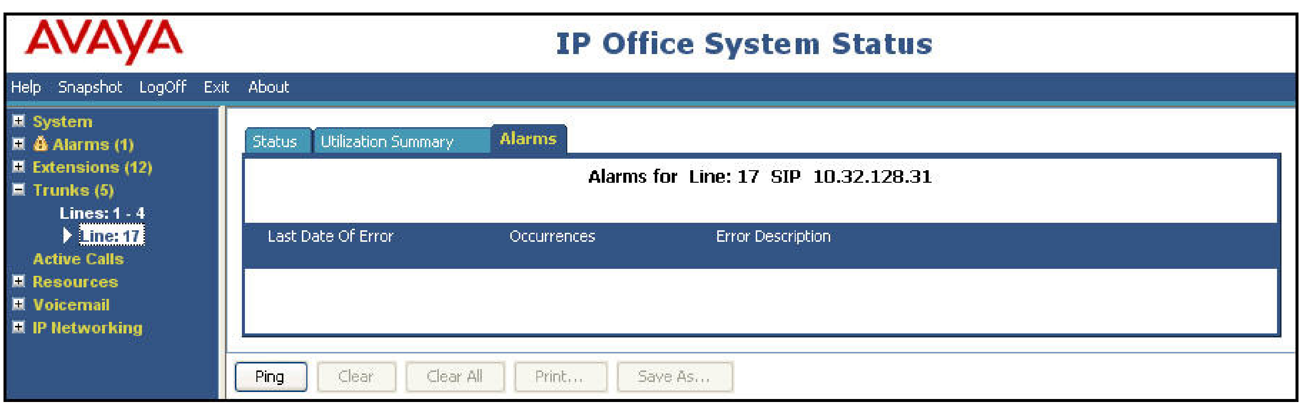
7. Verification Steps

The following steps may be used to verify the configuration:

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application and log in using appropriate login credentials. Select the SIP line of interest from the left pane.
- On the **Status** tab in the right pane, verify that **Current State** for active calls is shown as ***Connected*** and all other SIP Line channels are in ***Idle*** state.



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



- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.
- Verify that a phone connected to PSTN can successfully place a call to the Avaya IP Office with two-way audio.

8. Conclusion

The Optimum Voice SIP Trunking passed compliance testing with the observations/limitations as documented in **Section 2.2**. These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office Release 8.1 and the Cablevision Optimum Voice SIP Trunking as shown in **Figure 1**.

9. Additional References

[1] *IP Office Release 8.1 FP1 Product Description,* Documentation number 15-601041 Issue 26.N, April 2013.
[2] *Avaya IP Office 8.1 Installing IP500/IP500 V2,* Document number15-601042 Issue 27m, July 2013.
[3] *Avaya IP Office 8.1 Implementing Voicemail Pro,* Document number15-601064 Issue 8b, December 2012.
[4] *Avaya IP Office 8.1 FP1 Manager 10.1,* Document number15-601011 Issue 29u, April 2013.
[5] *Avaya IP Office 8.1 Using System Status Application,* Document number15-601758 Issue 07a, May 2013.

Product documentation for Avaya products may be found at http://support.avaya.com.
Product documentation for Optimum Voice SIP Trunking is available from Cablevision.

Appendix: SIP Line Template

Avaya IP Office Release 8.0 and later supports 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 specific installation environment, is included in the SIP Line Template. Therefore it is critical that the SIP Line configuration be verified/updated after a template has been imported, and additional configuration be supplemented using **Section 5.4** in these Application Notes as a reference.

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

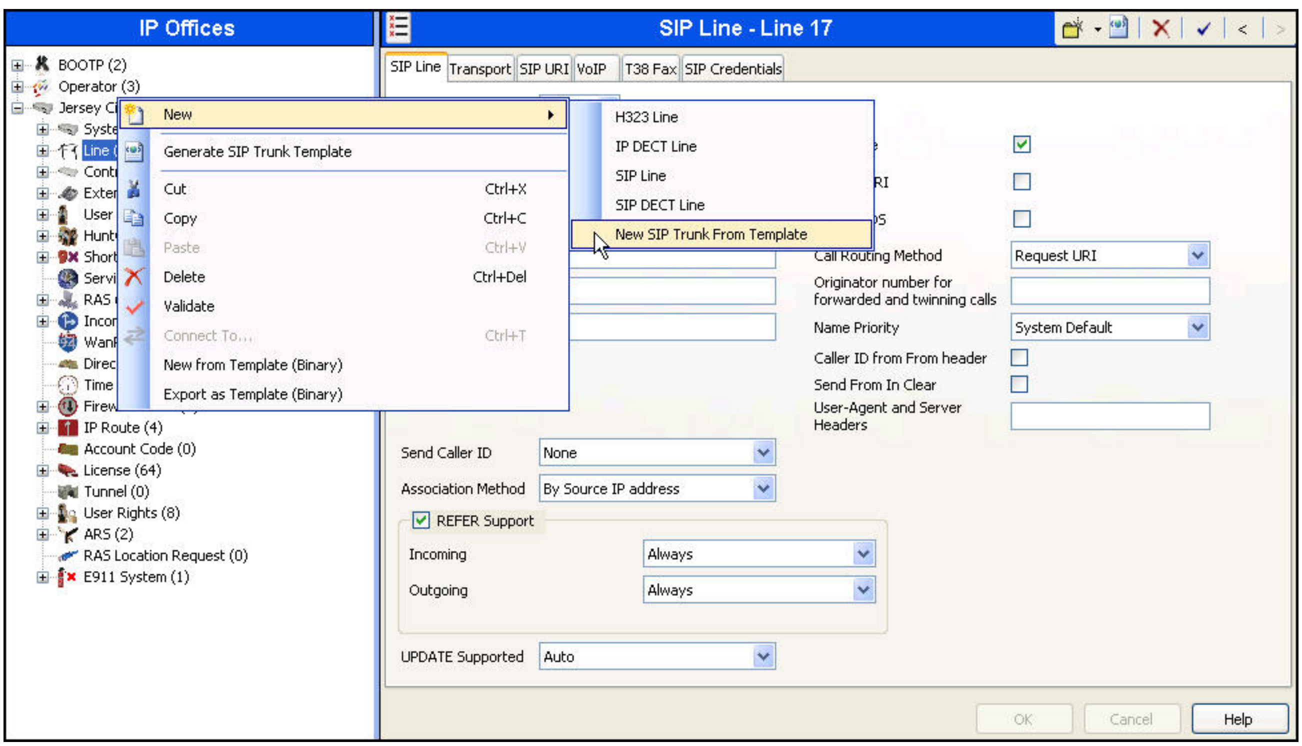

```
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20130816</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>CablevisionOptimumVoice</DescriptiveName>
  <ITSPDomainName>10.32.128.31</ITSPDomainName>
  <SendCallerID>CallerIDNone</SendCallerID>
  <ReferSupport>true</ReferSupport>
  <ReferSupportIncoming>1</ReferSupportIncoming>
  <ReferSupportOutgoing>1</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>false</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <UpdateSupport>UpdateAuto</UpdateSupport>
  <UserAgentServerHeader />
  <CallerIDfromFromheader>false</CallerIDfromFromheader>
  <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
  <ITSPProxy>10.32.128.31</ITSPProxy>
  <LayerFourProtocol>SipUDP</LayerFourProtocol>
  <SendPort>5060</SendPort>
  <ListenPort>5060</ListenPort>
  <DNSServerOne>0.0.0.0</DNSServerOne>
  <DNSServerTwo>0.0.0.0</DNSServerTwo>
  <CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
  <SeparateRegistrar />
  <CompressionMode>AUTOSELECT</CompressionMode>
  <UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs>
  <AdvCodecPref>G.711 ULAW 64K</AdvCodecPref>
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_INBAND</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
  <ReinviteSupported>false</ReinviteSupported>
  <FaxTransportSupport>FOIP_NONE</FaxTransportSupport>
  <UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
  <CodecLockdown>false</CodecLockdown>
  <Rel100Supported>false</Rel100Supported>
```

```
  <T38FaxVersion>0</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>false</UseDefaultValues>
  <ScanLineFixup>true</ScanLineFixup>
  <TFOPEnhancement>true</TFOPEnhancement>
  <DisableT30ECM>true</DisableT30ECM>
  <DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
  <DisableT30MRCompression>false</DisableT30MRCompression>
  <NSFOverride>false</NSFOverride>
</Template>
```


To import the above template into a new installation:

1. Copy and paste the above template into a text document named **`US_CablevisionOptimumSIPTrunk.xml`** on the PC where IP Office Manager was installed. Move the .xml file to the IP Office Manager template directory (C:\Program Files\Avaya\IP Office\Manager\Templates).

2. Import the template into an IP Office installation by creating a new SIP Line as shown in the screenshot below (right clicking **Line** in the left navigation pane):



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