Avaya Solution & Interoperability Test Lab

Application Notes for Configuring TelNet Worldwide SIP Trunking with Avaya IP Office R8.1- Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the TelNet Worldwide SIP Trunking service and Avaya IP Office R8.1.

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

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

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1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the service provider TelNet Worldwide and the Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 8.1, Avaya Voicemail Pro voice messaging application, SIP-based Avaya IP Office Softphone, and Avaya H.323, digital, and analog endpoints.

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2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the TelNet Worldwide 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 with voice messaging application, and various Avaya endpoints.

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 interoperability compliance test.

- Response to SIP OPTIONS queries.
- Incoming PSTN calls to various enterprise phones. Phone types included H.323, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise over the SIP trunk from the service provider.
- Outgoing PSTN calls from various enterprise phones. Phone types included H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise over the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from the SIP-based Avaya IP Office Softphone.
- Inbound and outbound long call and hold time stability.
- Various call types including: local, long distance, international, outbound toll-free, operator services and directory assistance.
- G.729A and G.711MU codecs.
- Caller ID presentation and Caller ID restriction.
- DTMF transmission using RFC 2833.
- Voicemail access and navigation for inbound and outbound calls.
- Response to incomplete call attempts and trunk errors.

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- User features such as hold and resume, transfer, and conference.
- Off-net call forwarding and twinning to mobile phones.
- G.711 pass-through fax.

Items not supported or not tested include the following:

- Inbound toll-free and emergency calls (911) are supported but were not tested as part of the compliance test.
- TelNet Worldwide SIP Trunking does not support T.38 fax.
- TelNet Worldwide SIP Trunking does not support operator-assisted calls (0 + 10-digit dialing).

2.2. Test Results

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

- **OPTIONS –** TelNet Worldwide SIP Trunking was not configured to send SIP OPTIONS messages during the compliance test but would respond to the OPTIONS messages sent by Avaya IP Office.
- **No Matching Codec on Outbound Calls –** When IP Office was configured with a codec unsupported by TelNet Worldwide, outbound INVITE received the response "480 Temporarily unavailable" from the service provider. A more appropriate status message like "488 Not Acceptable Here" could have been returned instead.
- **Outbound Anonymous Call –** When the calling party number was blocked, the outbound call to the PSTN failed. The network returned “604 Does not exist anywhere” to the outbound INVITE; the caller heard intercept tones. TelNet Worldwide was notified about this failure and has been investigating. Inbound anonymous calls worked properly.

2.3. Support

For technical support on the TelNet Worldwide SIP Trunking service, contact TelNet Worldwide Business Customer Care 24.7 by

- Calling 800-508-1254 or
- Sending email to prioritysupport@telnetww.com

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. The test configuration shows an enterprise site connected to TelNet Worldwide SIP Trunking.

Located at the enterprise site is an Avaya IP Office 500v2. 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 1600 Series IP Telephone (with H.323 firmware), an Avaya 9600 Series IP

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Telephone (with H.323 firmware), an Avaya 96x1 Series IP Telephone (with H.323 firmware), a SIP-based Avaya IP Office Softphone, an Avaya 5420 Digital Telephone, an Avaya 6210 Analog Telephone and a fax machine. The site also has a Windows 2003 Server running Avaya Voicemail Pro software for voice messaging application and a Windows XP PC running Avaya IP Office Manager to configure Avaya IP Office.

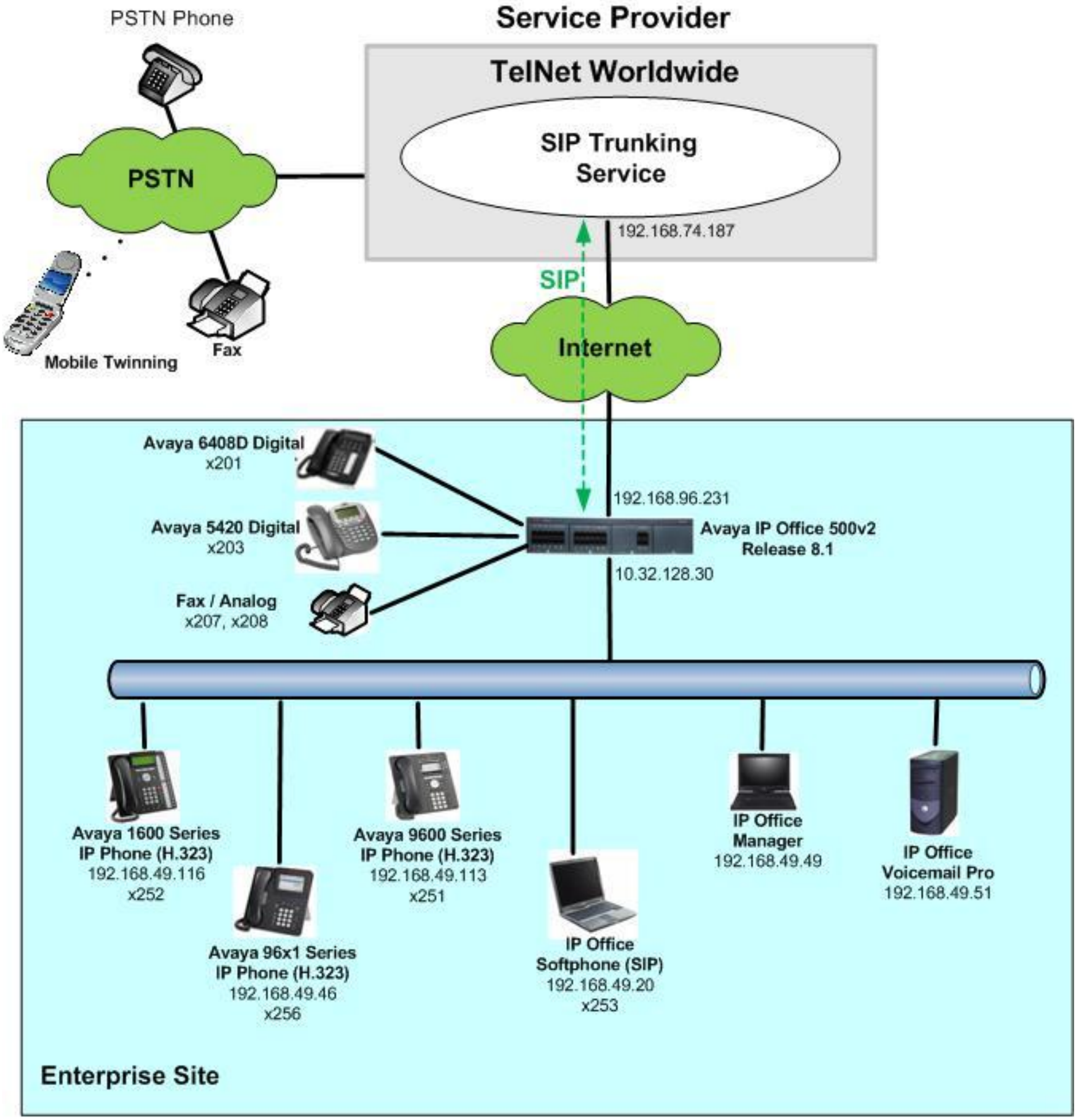

Figure 1: Test Configuration

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For security purposes, any public IP addresses used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been replaced with private 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 TelNet Worldwide. The short code of 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 calls 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. It was configured to send 10 digits in the From header. For inbound calls, TelNet Worldwide 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.

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 500v2	8.1 (63)
Avaya IP Office COMBO6210/ATM4 Module	8.1 (63)
Avaya IP Office Manager running on Windows XP PC	10.1 (63)
Avaya Voicemail Pro running on Windows 2003 Server	8.1.9016.0
Avaya 1600 Series IP Telephones (H.323)	Avaya one-X® Deskphone Value Edition 1.3 SP2
Avaya 9600 Series IP Telephones (H.323)	Avaya one-X® Deskphone Edition 3.1 SP5 (3.1.05S)
Avaya 96x1 Series IP Telephones (H.323)	6.2 SP2 (6.2.2)
Avaya IP Office Softphone	3.2.3.48 67009
Avaya 5420 Digital Telephone	N/A
Avaya 6210 Analog Telephone	N/A
Venta Fax PC Application	6.6.156.385
TelNet Worldwide Components	
Equipment / Software	**Release / Version**
ACME Net-Net 3820	Rev-6.4.0 Patch 1
BroadSoft BroadWorks	Rev 17.SP4

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5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to TelNet Worldwide SIP Trunking. Avaya IP Office is configured through the Avaya IP Office Manager application. From the 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, and log in with the appropriate credentials. A management window will appear similar to the one shown in the next section. The appearance of the Avaya 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 and the Details pane on the right side. 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. LAN2 Settings

In the sample configuration, ***Jersey City*** was used as the system name and the WAN port was used to connect the 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 **Jersey City → 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. All other parameters should be set according to customer requirements.

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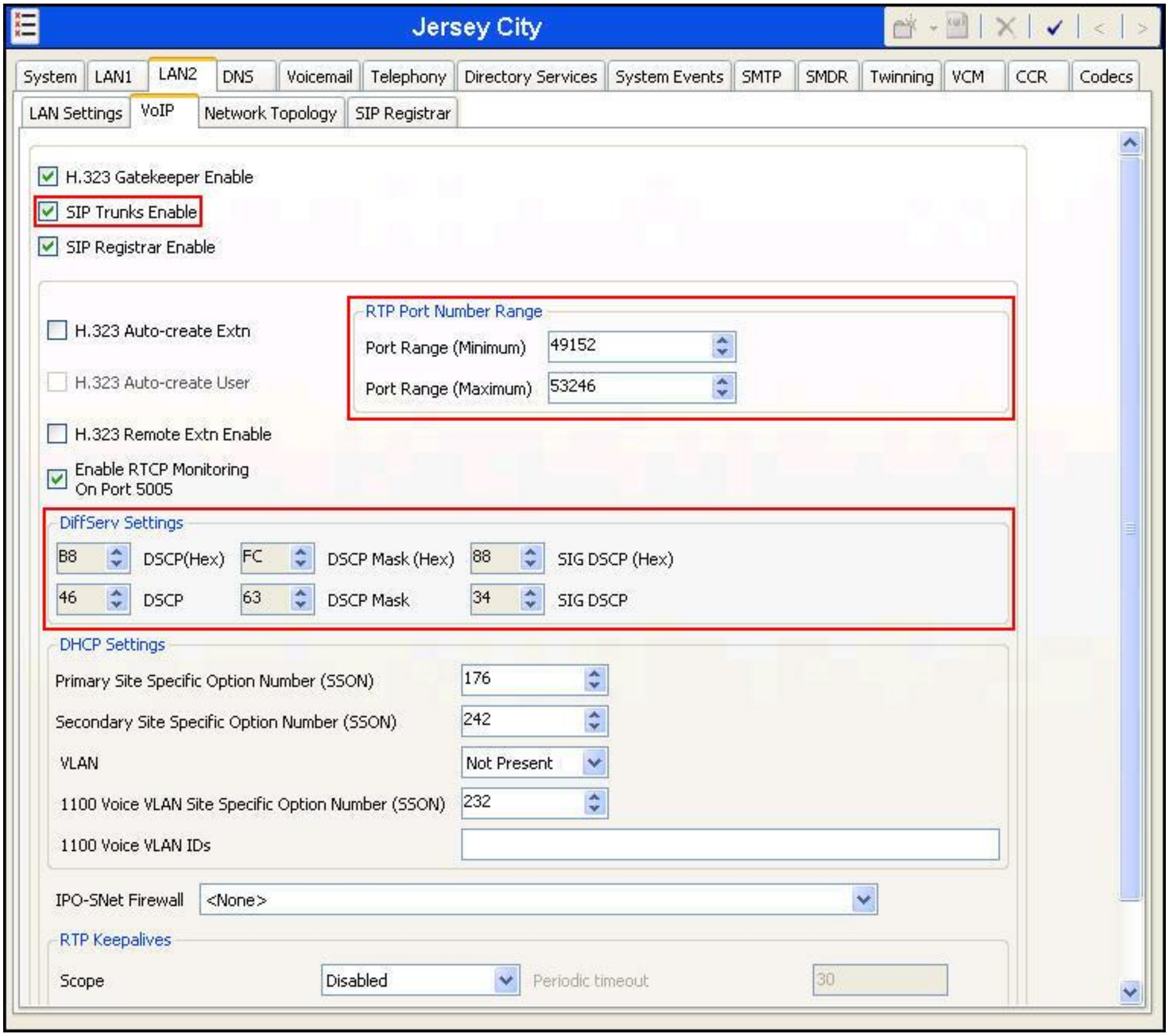
On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2 (the protocol was configured in **Section 5.4** as specified by the service provider). 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. All other parameters should be set according to customer requirements.



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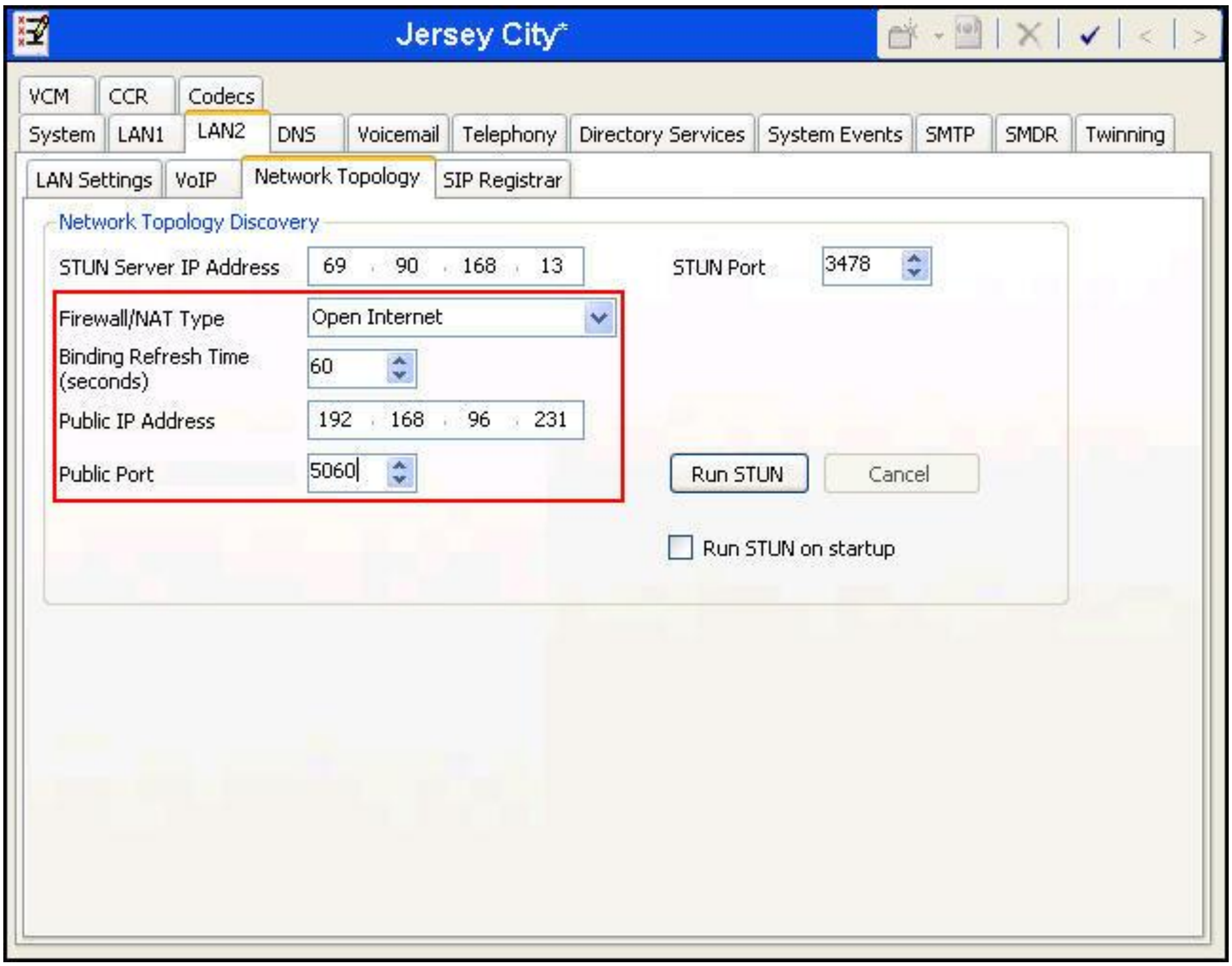
On the **Network Topology** tab of LAN2 in the Details Pane, configure the following parameters:

- 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 IP 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 5.8** for complete details.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- Set **Public Port** to ***5060***.

All other parameters should be set according to customer requirements.



During the 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 TelNet Worldwide, and therefore is not described in these Application Notes.

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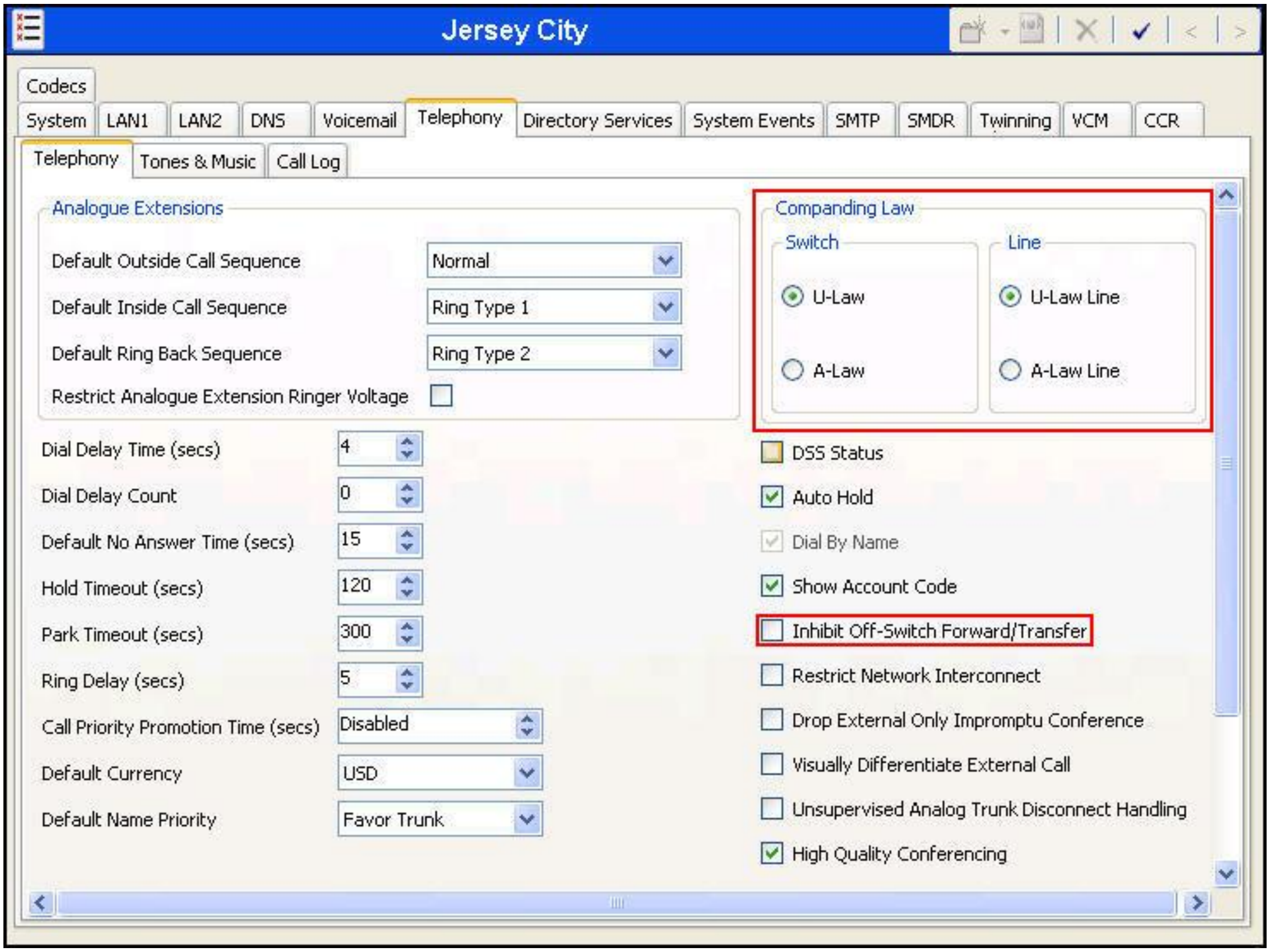
5.2. System Telephony Settings

Navigate to the **Telephony → Telephony** tab on the Details Pane. Choose the **Companding Law** typical for the enterprise location. For the compliance test, ***ULAW*** 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. All other parameters should be set according to customer requirements.



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5.3. Twinning Calling Party Settings

Navigate to the **Twinning** tab on the Details Pane. Uncheck the **Send original calling party information for Mobile Twinning** box. This will allow the Caller ID for Twinning and Call Forwarding to be controlled by the setting on the SIP Line (**Section 5.4**).

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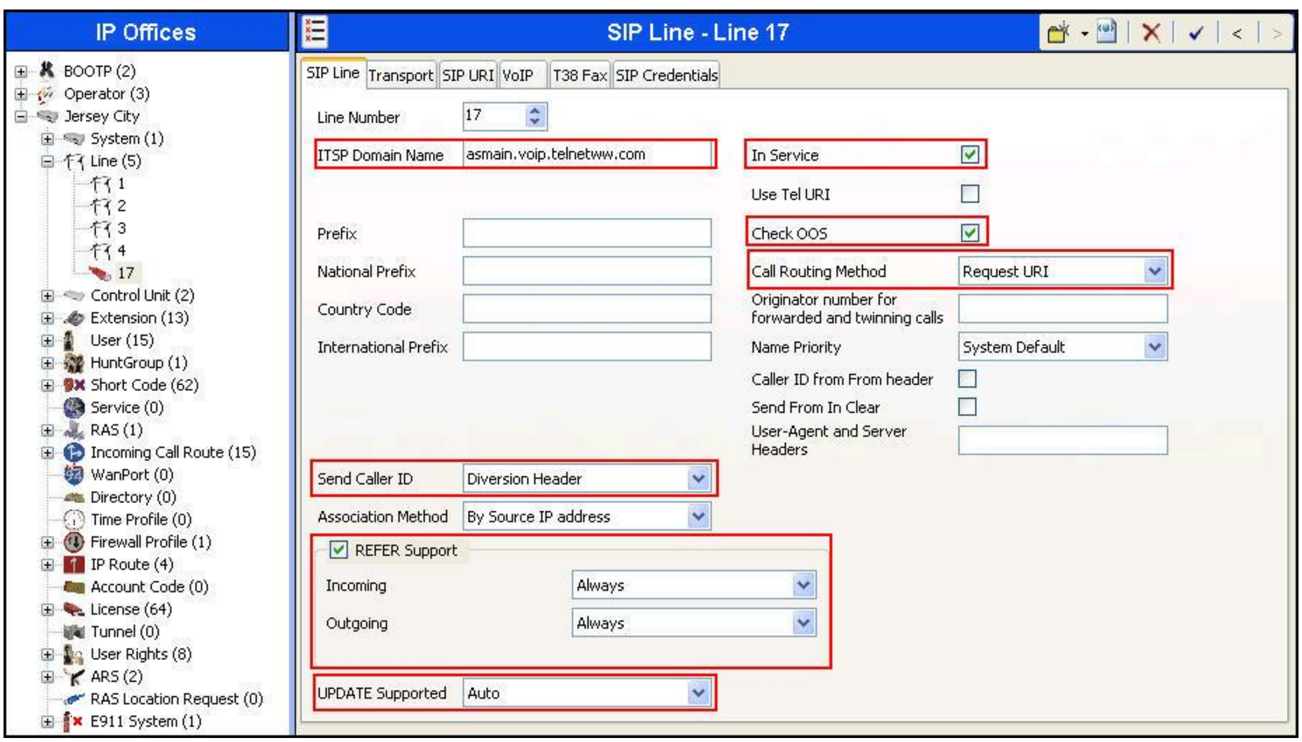
5.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and TelNet Worldwide SIP Trunking. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New → SIP Line** (not shown). On the **SIP Line** tab in the Details Pane, configure the parameters as shown below.

- In the **ITSP Domain Name** field, enter the service provider domain name as provided.
- Check the **In Service** box.
- Check the **Check OOS** box. This option selection directs Avaya IP Office to use the SIP OPTIONS method to periodically check the SIP Line.
- Set **Call Routing Method** to ***Request URI***. Avaya IP Office will route inbound calls based on the number in the Request URI.
- Set **Send Caller ID** to ***Diversion Header***. For forwarded or twinning calls, this setting results in the original calling party appearing in the SIP From header and the forwarding/twinning party in the Diversion header.
- Check the **REFER Support** box and select ***Always*** for **Incoming** and **Outgoing**. These settings direct Avaya IP Office to use SIP REFER message for call redirection to the PSTN as in call transfers.
- Set **UPDATE Supported** to ***Auto***. With this setting Avaya IP Office will send UPDATE messages for session refresh if the other party supports UPDATE.

Default values may be used for all other parameters.



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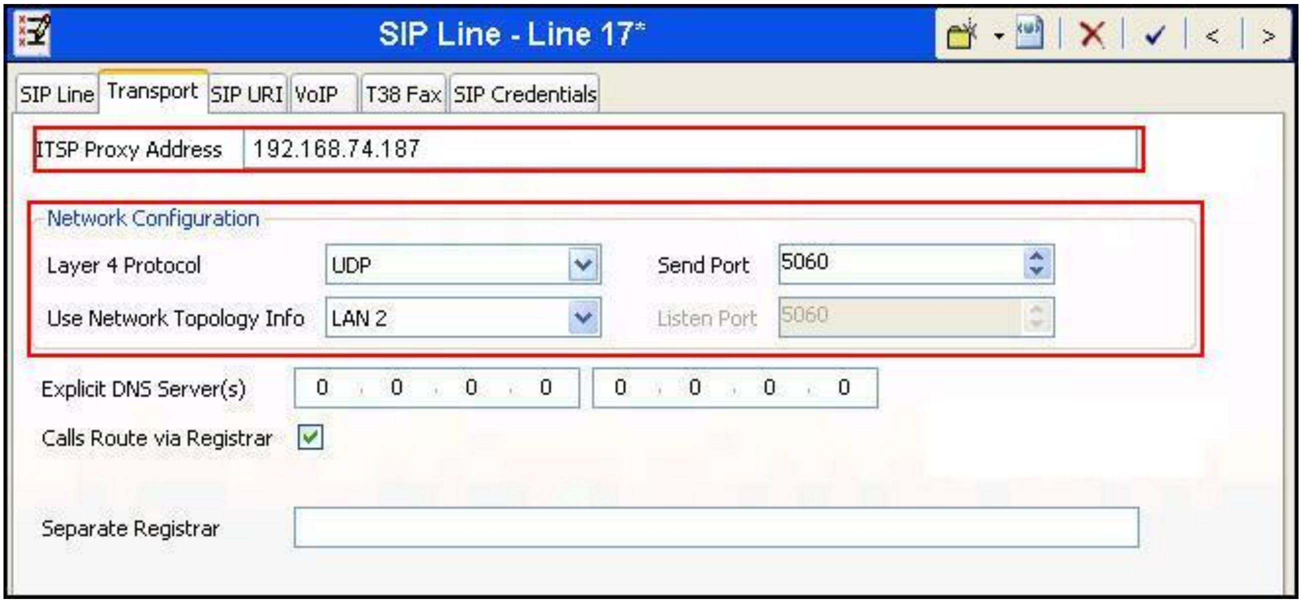
Navigate to the **Transport** tab and set the following:

- Set the **ITSP Proxy Address** to the IP address of the TelNet Worldwide SIP Proxy provided by TelNet Worldwide.
- Set the **Layer 4 Protocol** to ***UDP***.
- Set **Use Network Topology Info** to ***LAN2*** as configured in **Section 5.1**.
- Set the **Send Port** to ***5060***.

Default values may be used for all other parameters.



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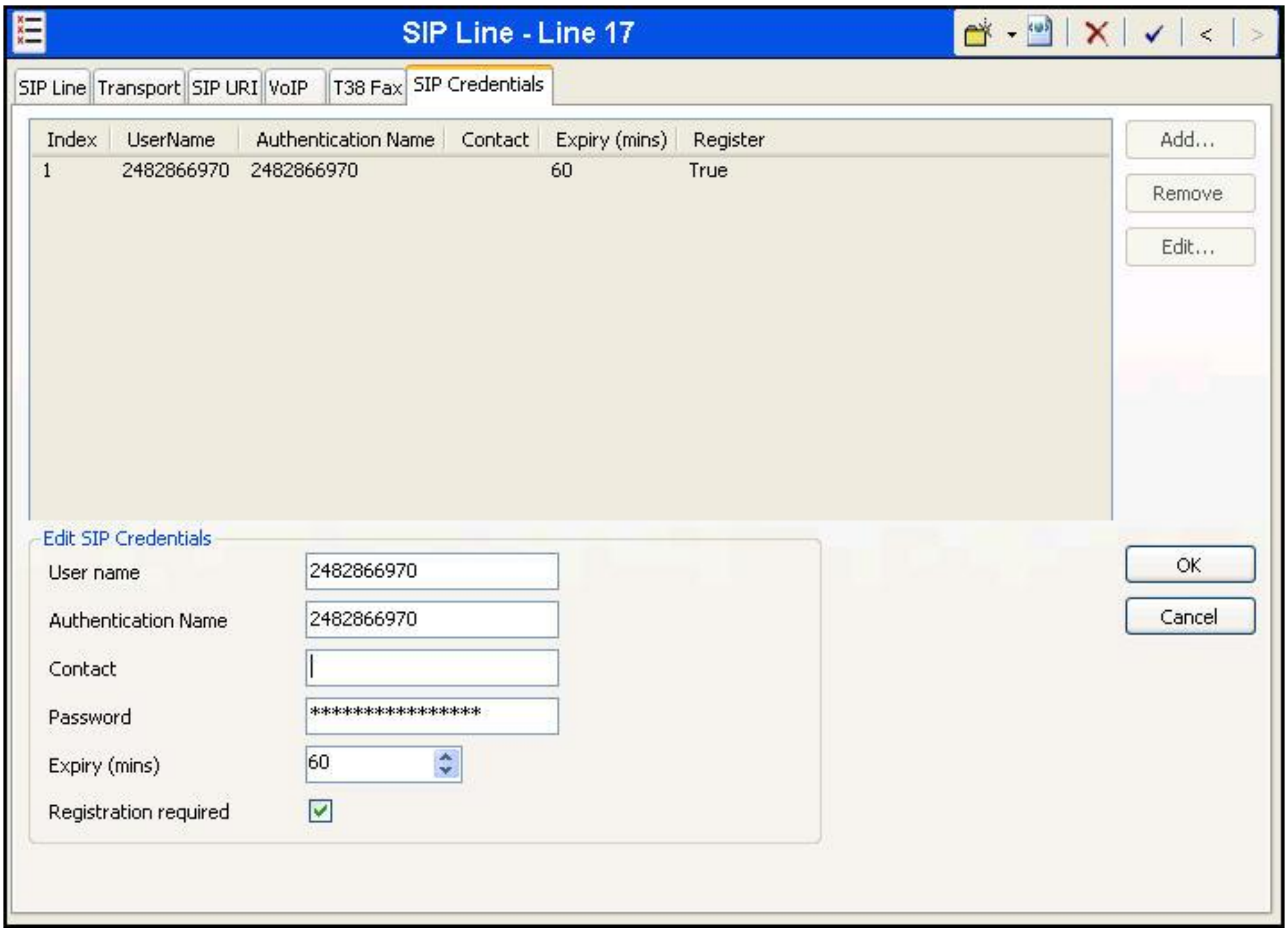
A **SIP Credentials** entry must be created for Digest Authentication used by TelNet Worldwide SIP trunking service to authenticate calls from the enterprise site to the PSTN. To create a **SIP Credentials** entry, first select the **SIP Credentials** tab. Click the **Add** button and the **New Channel** 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 screen below, a previously configured entry is edited. 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.
- Check the **Registration required** option. TelNet Worldwide requires SIP trunk registration.



Select the **SIP URI** tab to create a SIP URI entry or edit an existing 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, a single SIP URI entry was created that matched any number assigned to an Avaya IP Office user. The following screen shows the edit window on a previously configured entry for the compliance test.

- 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**.

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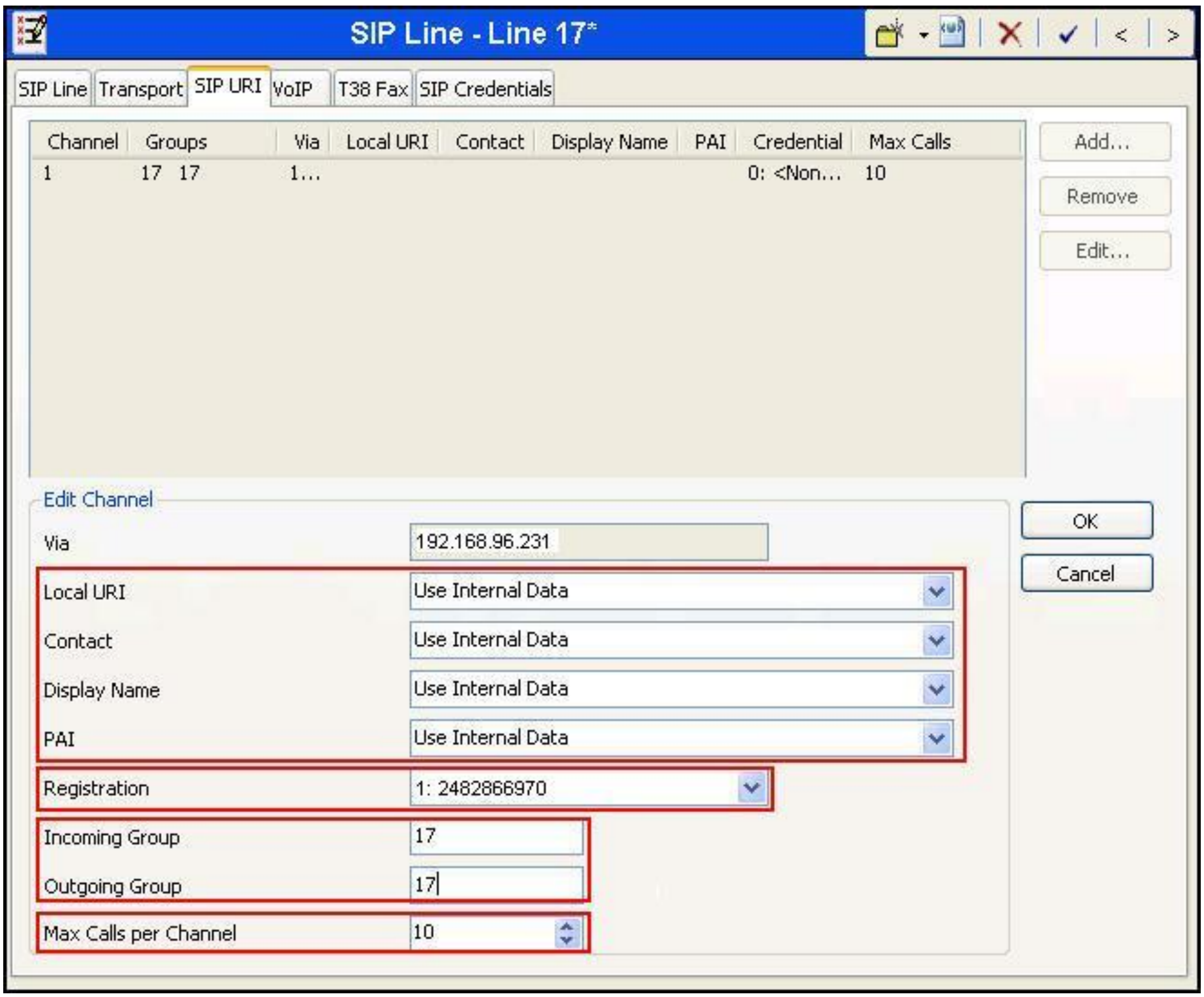
- Set **Contact** and **Display Name** to ***Use Internal Data***. This setting will cause the Contact and Display Name data to be set from the corresponding fields on the **SIP** tab of the individual **User** as shown in **Section 5.6**.
- Set **PAI** to ***Use Internal Data***. This setting directs Avaya IP Office to send the PAI (P-Asserted-Identity) header when appropriate. The PAI header will be populated from the data set in the **SIP** tab of the call initiating **User** as shown in **Section 5.6**.
- For **Registration**, select the account credentials previously configured on the line's **SIP Credentials** tab.
- 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, a new incoming and outgoing group ***17*** was specified that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls allowed using this SIP URI pattern.



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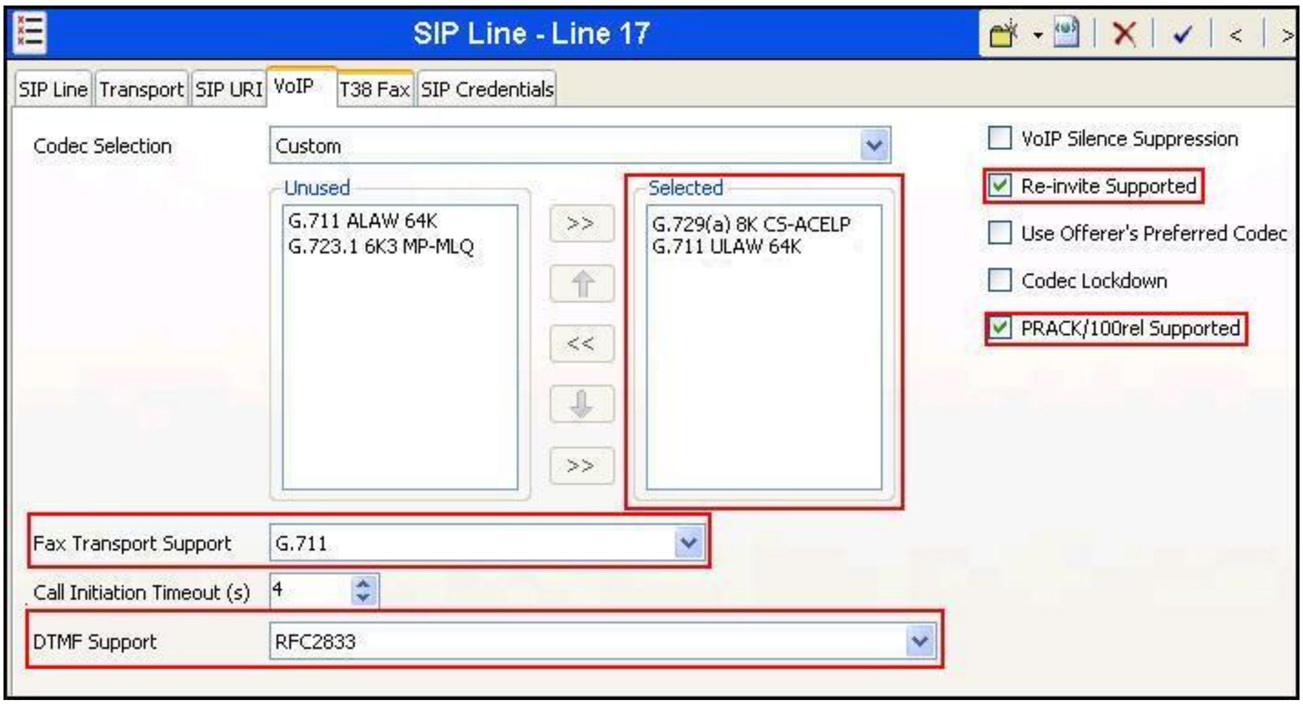
Select the **VoIP** tab, to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below.

- Set the **Codec Selection** to ***Custom***.
- Choose ***G.711 ULAW 64K*** and ***G.729(a) 8K CS-ACELP*** from the **Unused** box and move these 2 selections to the **Selected** box. Use the down/up arrows to order the 2 selected codecs as shown to be consistent with the codec order preference of the TelNet Worldwide SIP Trunking service..
- Check the **Re-invite Supported** box.
- Check the **PRACK/100rel Supported** box. This setting enables support by Avaya IP Office for the PRACK (Provisional Reliable Acknowledgement) message on SIP trunks.
- Set **Fax Transport Support** to ***G.711*** to use G.711 pass-through fax.
- Set the **DTMF Support** field to ***RFC2833***. This directs Avaya IP Office to send DTMF tones using RTP event messages as defined in RFC2833.

Default values may be used for all other parameters.



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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. 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"@asmain.voip.telnetww.com:5060"***. 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 TelNet Worldwide SIP Trunking domain and port number follow the @ sign in the above expression.
- 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.

Default values may be used for all other parameters.

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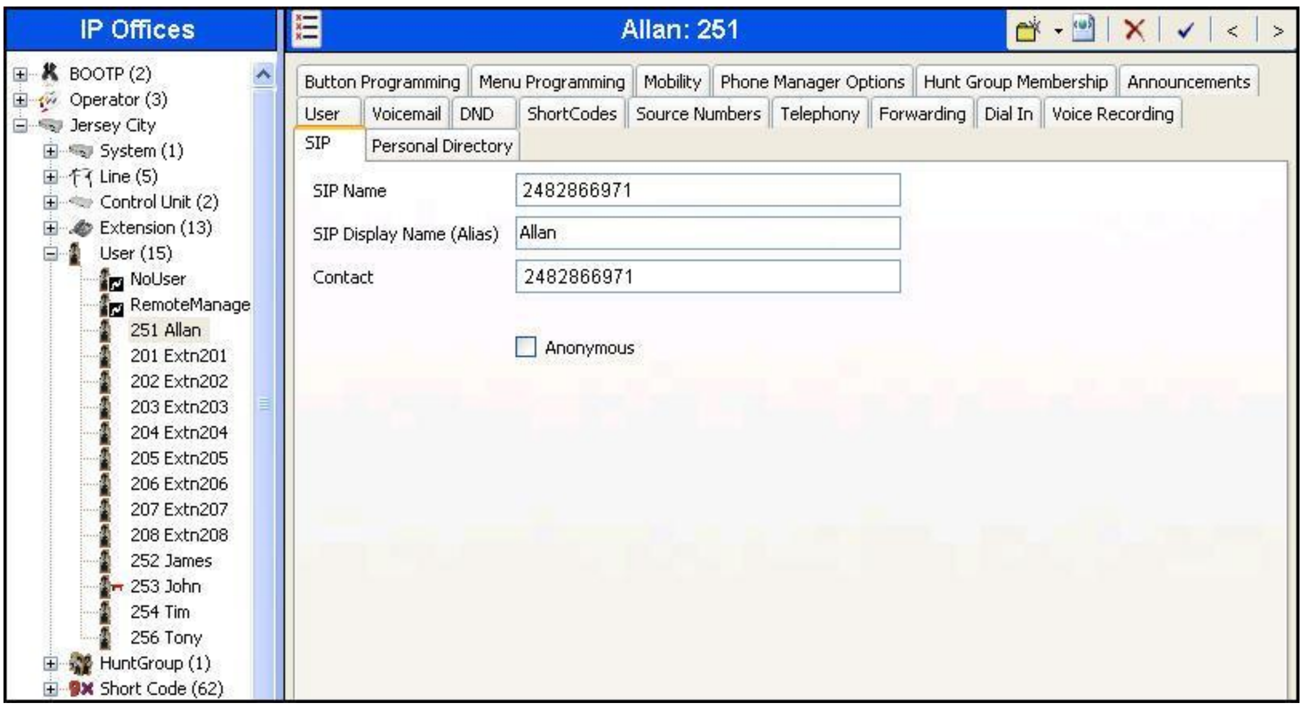
5.6. User

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. In the example below, the name of the user is ***Allan***. Select the **SIP** tab in the Details Pane. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from TelNet Worldwide. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. The value entered for the **Contact** field will be used in the SIP INVITE for outgoing SIP trunk calls. The value entered for the **SIP Name** is used 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.



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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** (not shown). On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capacity** to ***Any Voice***.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.4**.
- 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 from the pull-down list of the **Destination** field. In this example, incoming calls to 2482866971 on line 17 are routed to the user Allan at extension 251.

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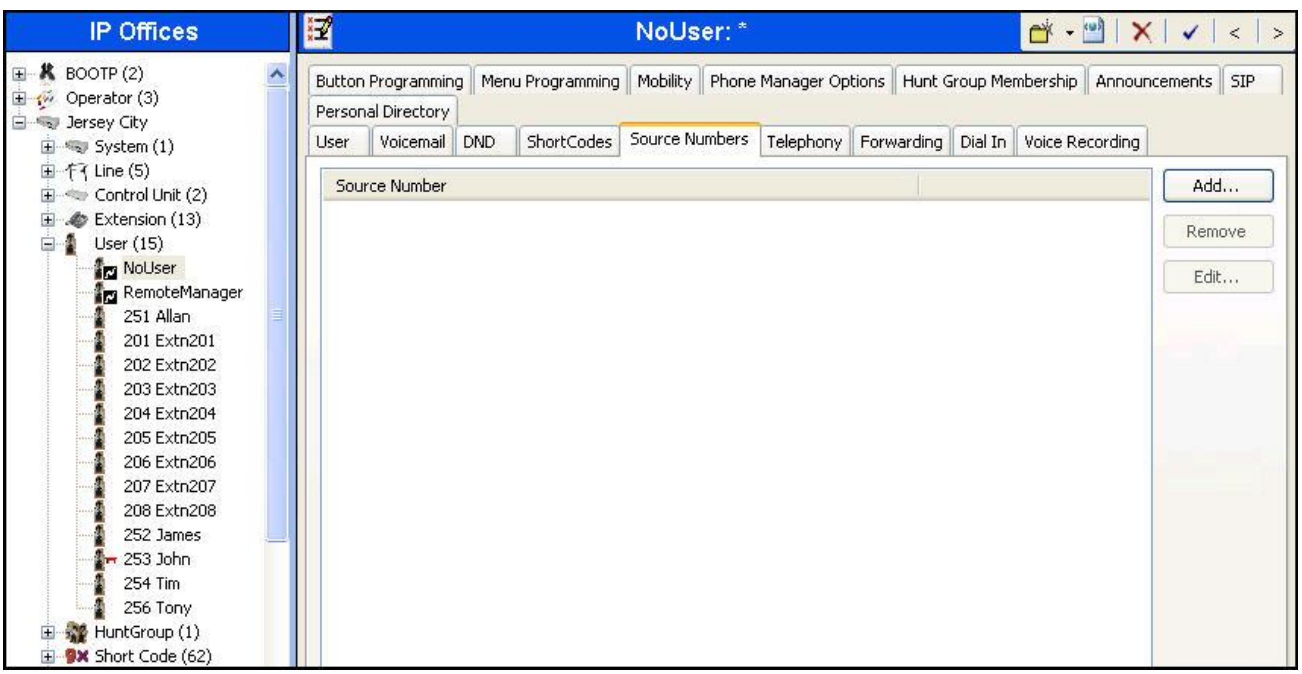
5.8. SIP Options

Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. 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 Numbers** tab of the **NoUser** user. The OPTIONS period is determined in the following manner:

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

For the compliance test, the OPTIONS period of 1 minute was desired. Thus, the **Binding Refresh Time** was set to ***60*** seconds in **Section 5.1** and no **SIP_OPTIONS_PERIOD** parameter was defined.

Alternatively, if an OPTIONS period greater than 300 seconds is desired then define the **SIP_OPTIONS_PERIOD** by doing the following. Navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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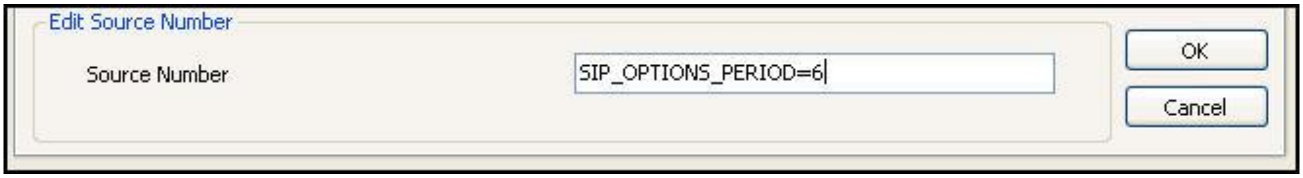
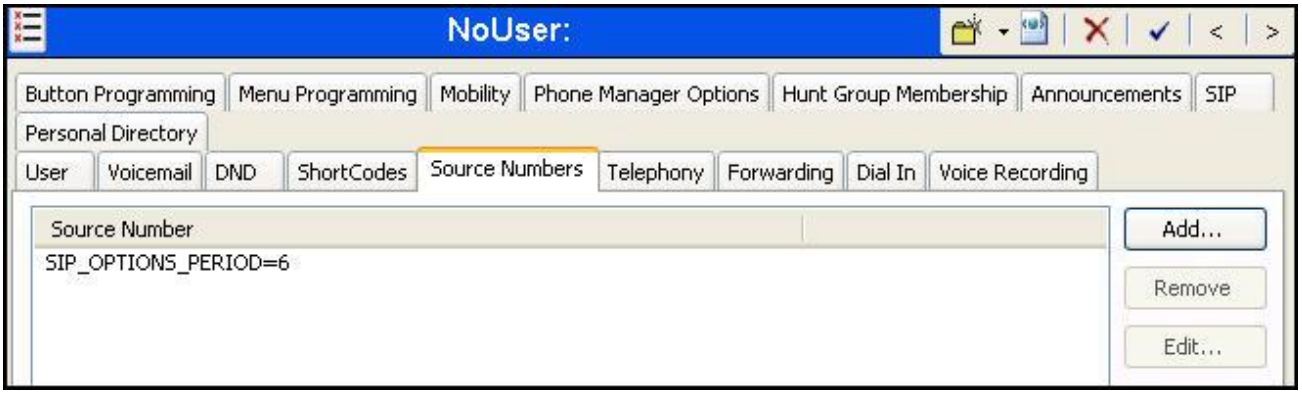
At the bottom of the Details Pane, the **Source Number** field will appear. Enter ***SIP_OPTIONS_PERIOD=X***, where ***X*** is the desired value in minutes. Click **OK**.



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



5.9. Save Configuration

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

6. TelNet Worldwide SIP Trunking Configuration

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

- Network edge IP address and domain of the TelNet Worldwide SIP Trunking service.
- Transport and port for the TelNet Worldwide SIP Trunking service.
- User name and password used for Digest Authentication.
- DID numbers to assign to users.
- Supported codecs and their preference order.
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

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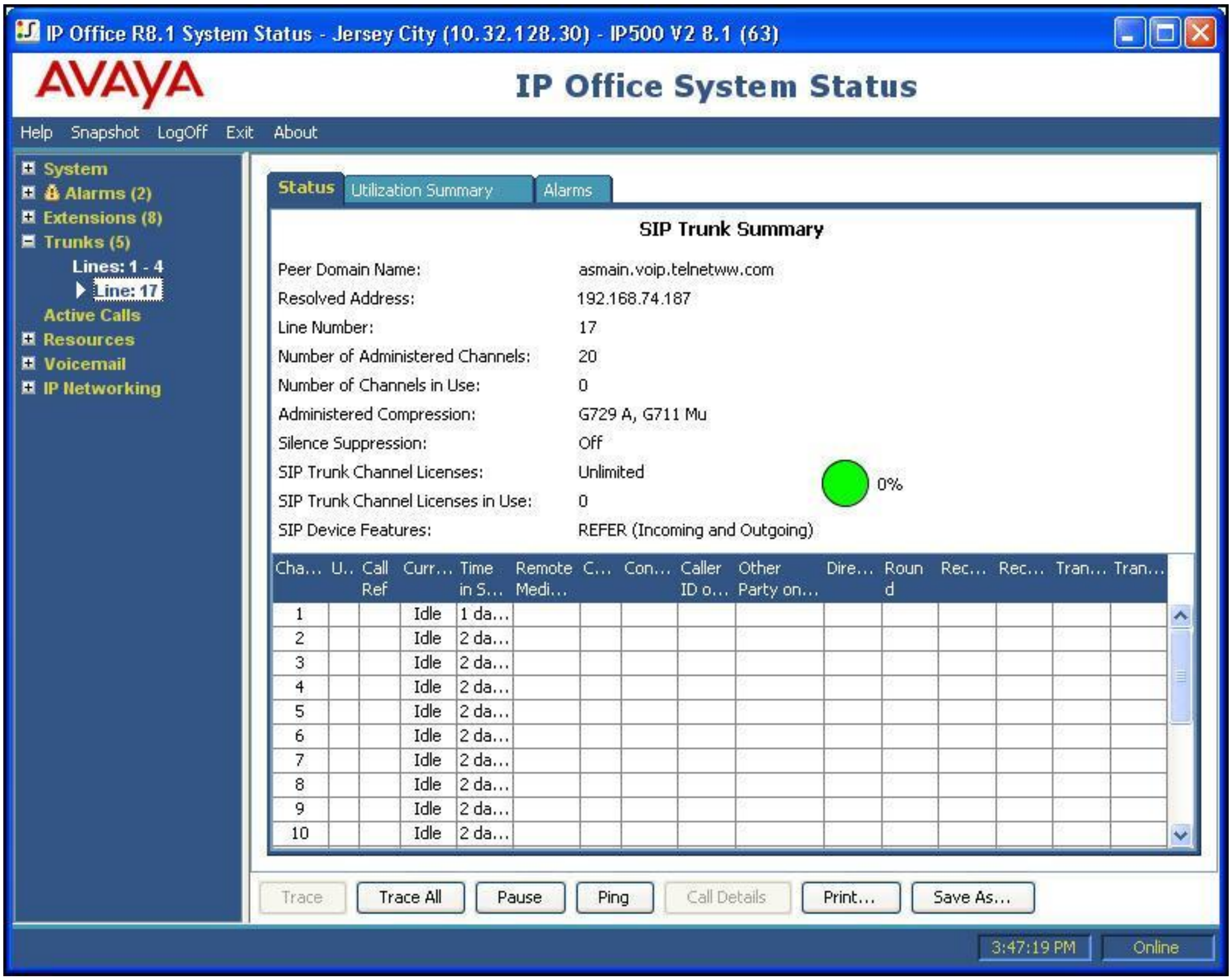
7. Verification Steps

The following steps may be used to verify the configuration:

- Use the Avaya IP Office System Status application to verify the SIP connection state. Launch the application from **Start → Programs → IP Office → System Status** on the Avaya IP Office Manager PC. Select the SIP line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** is ***Idle*** for each channel if no active call is currently in session (as shown below); or some channels are taken by active calls with **Current Sate** shown as ***Connected*** (not shown).



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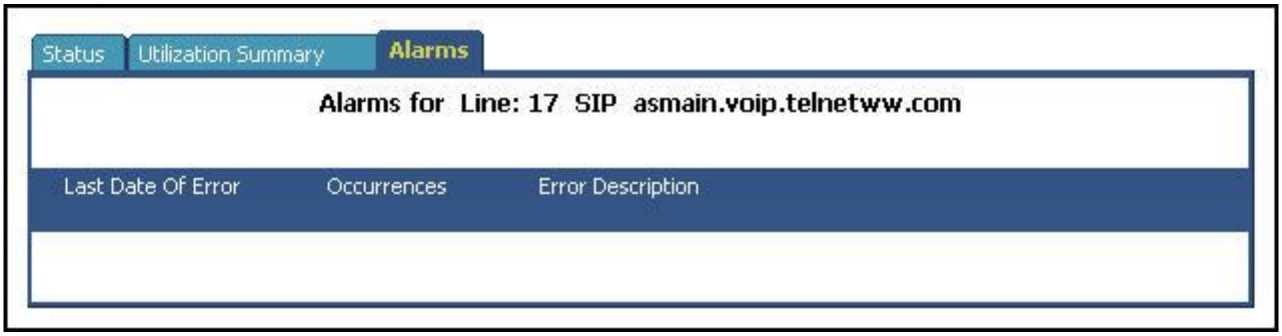
- Select the **Alarms** tab and verify that no alarms are active on the SIP line.



- 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

TelNet Worldwide SIP Trunking passed compliance testing. These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office and TelNet Worldwide SIP Trunking as shown in **Figure 1**. Test results and observations are noted in **Section 2.2**.

9. Additional References

[1] *IP Office 8.1 IP500/IP500 V2 Installation*, Document number 15-601042, Issue 27d, January 2013.
[2] *IP Office R8.1 FP1 Manager 10.1*, Document number 15-601011, Issue 29r, November 2012.
[3] *IP Office Softphone User Guide (Windows)*, Issue 06e, April 2012.
[4] *IP Office System Status Application*, Document number 15-601758, Issue 07a, November 2012.
[5] *IP Office 8.1 Administering Voicemail Pro*, Document number 15-601063, Issue 8b, December 2012.

Product documentation for Avaya products may be found at http://support.avaya.com.
Product documentation for TelNet Worldwide SIP Trunking is available from TelNet Worldwide. See **Section 2.3** on how to contact TelNet Worldwide.

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Appendix: SIP Line Template

From Release 8.0 on, Avaya IP Office 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.

Not all of the configuration information is included in the SIP Line Template. Therefore, it is critical that the SIP Line configuration be verified/updated after a template has been imported and additional configuration be supplemented using **Section 5.4** in these Application Notes as a reference.

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


```xml
<?xml version="1.0" encoding="utf-8" ?>
<Template xmlns="urn:SIPTrunk-schema">
    <TemplateType>SIPTrunk</TemplateType>
    <Version>20130207</Version>
    <SystemLocale>enu</SystemLocale>
    <DescriptiveName>TelNet Worldwide SIP Trunking</DescriptiveName>
    <ITSPDomainName>asmain.voip.telnetww.com</ITSPDomainName>
    <SendCallerID>CallerIDDIV</SendCallerID>
    <ReferSupport>true</ReferSupport>
    <ReferSupportIncoming>1</ReferSupportIncoming>
    <ReferSupportOutgoing>1</ReferSupportOutgoing>
    <RegistrationRequired>false</RegistrationRequired>
    <UseTelURI>false</UseTelURI>
    <CheckOOS>true</CheckOOS>
    <CallRoutingMethod>1</CallRoutingMethod>
    <OriginatorNumber />
    <AssociationMethod>SourceIP</AssociationMethod>
    <LineNamePriority>SystemDefault</LineNamePriority>
    <UpdateSupport>UpdateAuto</UpdateSupport>
    <UserAgentServerHeader />
    <CallerIDfromFromheader>false</CallerIDfromFromheader>
    <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
    <ITSPProxy>192.168.74.187</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.729(a) 8K CS-ACELP,G.711 ULAW 64K</AdvCodecPref>
    <CallInitiationTimeout>4</CallInitiationTimeout>
    <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
```


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```
    <VoipSilenceSupression>false</VoipSilenceSupression>
    <ReinviteSupported>true</ReinviteSupported>
    <FaxTransportSupport>FOIP_G711</FaxTransportSupport>
    <UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
    <CodecLockdown>false</CodecLockdown>
    <Rel100Supported>true</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>
    <SIPCredentials>
        <Expiry>60</Expiry>
        <RegistrationRequired>true</RegistrationRequired>
    </SIPCredentials>
</Template>
```


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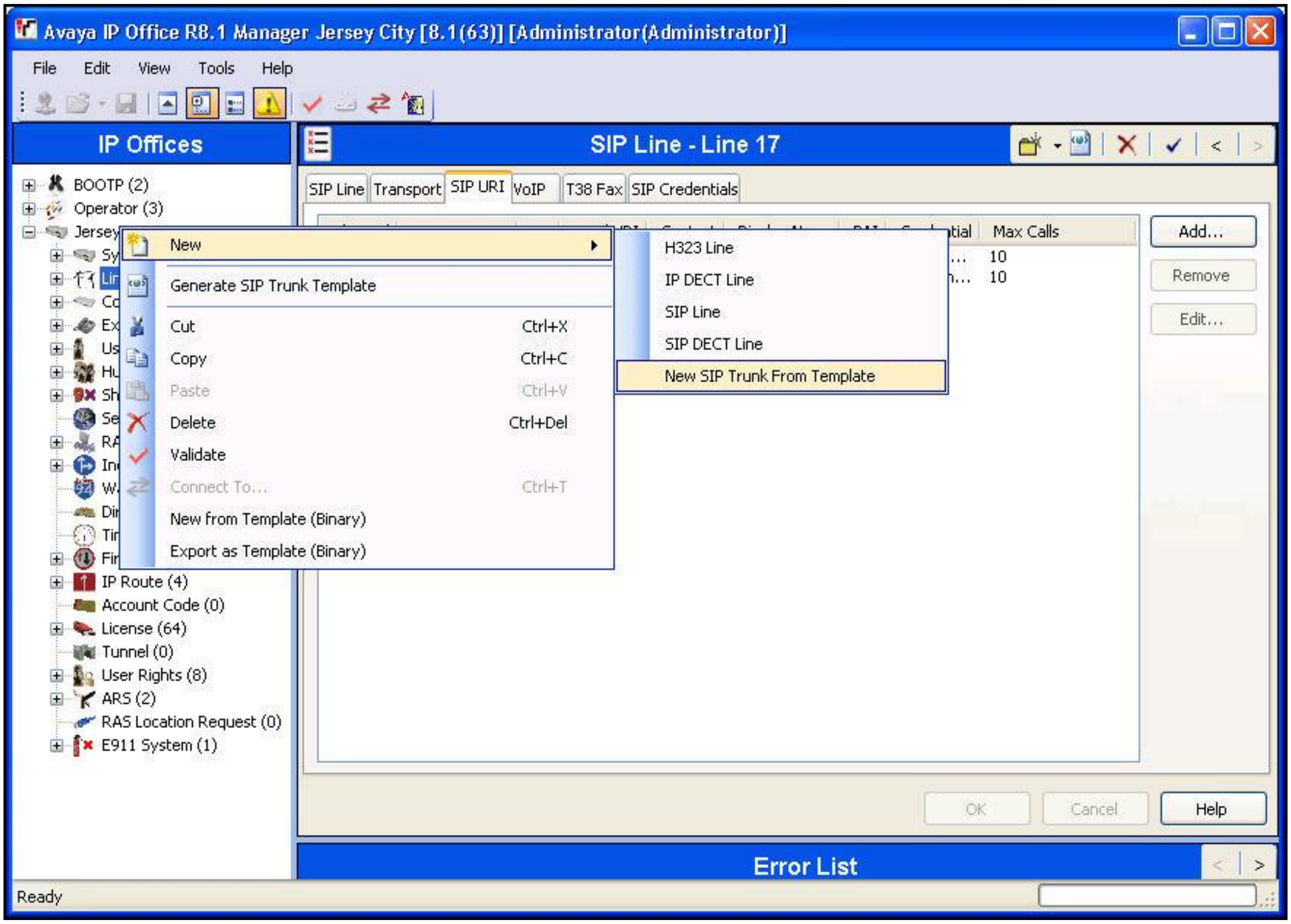
To import the above template into a new installation:

1. On the PC where IP Office Manager is installed, copy and paste the above template into a text document named **US_Worldwide_SIPTrunk.xml**. Move the .xml file to the IP Office Manager template directory (default location is 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. In the Navigation Pane on the left, right-click on **Line** then navigate to **New → New SIP Trunk From Template**:



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

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