



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Configuring DoubleHorn Communications IPOne SIPtrunk Service 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 DoubleHorn Communications IPOne SIPtrunk service and Avaya IP Office R8.1.

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

DoubleHorn Communications 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 the DoubleHorn Communications IPOne SIPtrunk service and the Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 running IP Office Release 8.1 software, Avaya Voicemail Pro voice messaging application, SIP-based Avaya IP Office Softphone, and Avaya H.323, digital, and analog endpoints.

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

## 2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the DoubleHorn Communications 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 the Avaya VoiceMail Pro 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 calls from the PSTN 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 calls to the PSTN 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 duration call and hold time stability.
- Various call types including: local, long distance, international, outbound toll-free, operator assisted calls and directory assistance.
- G.711MU and G.729A codecs.
- Caller ID presentation and Caller ID restriction.
- DTMF transmission using RFC 2833.
- Voicemail access and navigation for inbound and outbound calls.
- Response to incomplete call attempts and trunk errors.

- 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) were not tested as part of the compliance test.
- DoubleHorn Communications SIP Trunking does not support T.38 fax.

## 2.2. Test Results

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

- **OPTIONS** – DoubleHorn Communications 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.
- **Call Session Expiration** – DoubleHorn SIP Trunking has not implemented support for SIP call session expiration and assumes an established call session never expires; therefore DoubleHorn does not issue re-INVITE or UPDATE messages to refresh an active SIP call session.
- **REFER** – When REFER message was used for call transfer to the PSTN, DoubleHorn SIP Trunking accepted the REFER from IP Office with the expected “202 Accepted” message, but would terminate the transferred call about 30 seconds later. The cause of this problem was traced to the carrier that DoubleHorn Communications used for PSTN calls. DoubleHorn Communications has escalated the issue to the carrier’s engineering support. For the compliance test, this problem was worked around by turning off REFER support on Avaya IP Office and using INVITE for call transfers.

## 2.3. Support

For technical support on the DoubleHorn Communications SIP Trunking service, contact DoubleHorn Communications by

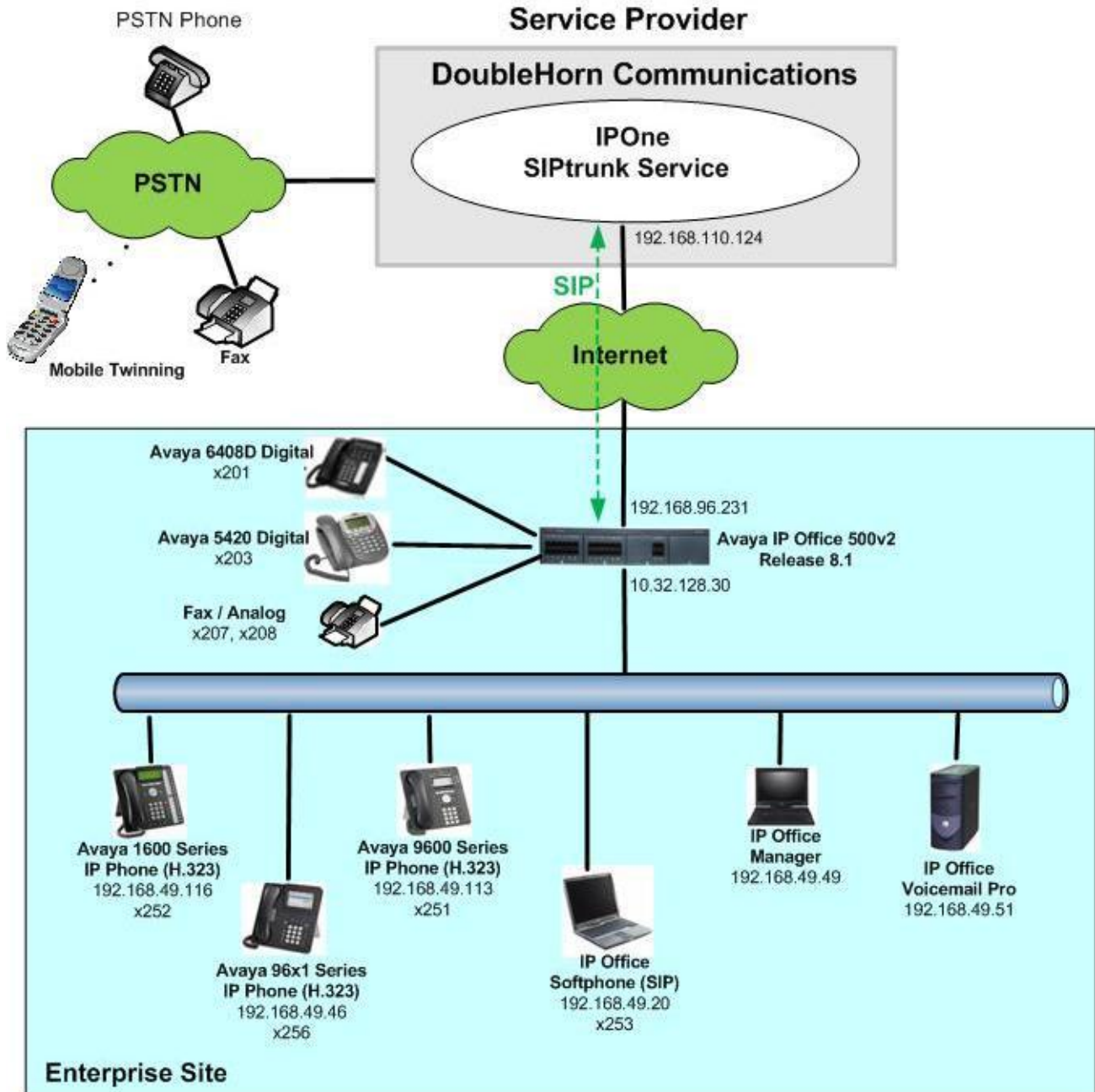
- Calling 855-618-6243
- Accessing the Support Portal at [www.mydoublehorn.com](http://www.mydoublehorn.com)
- Sending email to [dhc.support@doublehorn.com](mailto:dhc.support@doublehorn.com)

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.

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 of various types (H.323, SIP, digital, analog and fax) are connected to Avaya IP Office. 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**

For security purposes, 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 DoubleHorn Communications. 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, DoubleHorn Communications 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
DoubleHorn Communications Components	
Equipment / Software	Release / Version
ACME Net-Net 4500	SCX6.3.9 Patch 7 (Build 330)
Sonus GSX 9000	V09.00.00A203

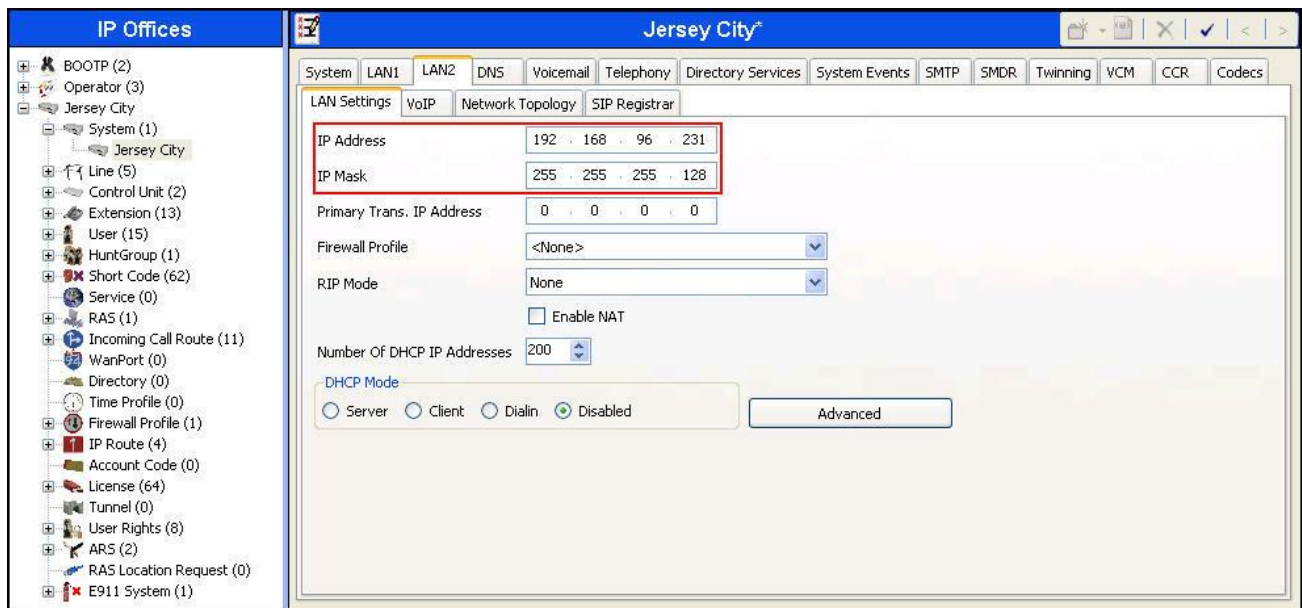
## 5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to DoubleHorn Communications SIP Trunking network. 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 similar to the one shown in the next section will appear. 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 (e.g., twinning and IP Office Softphone support, etc.) is assumed to already be in place.

The configuration screens in this section only show values/settings configured for the compliance test. Defaults were used for other values and may be customized based upon requirements in the field.

### 5.1. 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.



On the **VoIP** tab of LAN2 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 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 screenshot shows the 'Jersey City' configuration window for the 'LAN2' interface. The 'VoIP' tab is selected, and the 'SIP Registrar' sub-tab is active. The 'SIP Trunks Enable' checkbox is checked and highlighted with a red box. Below it, the 'RTP Port Number Range' section is also highlighted with a red box, showing 'Port Range (Minimum)' set to 49152 and 'Port Range (Maximum)' set to 53246. The 'DiffServ Settings' section is highlighted with a red box, showing 'DSCP (Hex)' set to B8, 'DSCP Mask (Hex)' set to FC, 'SIG DSCP (Hex)' set to 88, 'DSCP' set to 46, 'DSCP Mask' set to 63, and 'SIG DSCP' set to 34. The 'DHCP Settings' section shows 'Primary Site Specific Option Number (SSON)' set to 176, 'Secondary Site Specific Option Number (SSON)' set to 242, 'VLAN' set to 'Not Present', '1100 Voice VLAN Site Specific Option Number (SSON)' set to 232, and '1100 Voice VLAN IDs' set to an empty field. The 'IPO-SNet Firewall' is set to '<None>'. The 'RTP Keepalives' section shows 'Scope' set to 'Disabled' and 'Periodic timeout' set to 30.



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**.

The screenshot shows the 'Jersey City\*' Avaya IP Office configuration window. The 'LAN2' tab is selected under the 'System' category. Within the 'LAN2' configuration, the 'Network Topology' sub-tab is active. A 'Network Topology Discovery' dialog box is open, containing the following fields and values:

- STUN Server IP Address: 69 . 90 . 168 . 13
- STUN Port: 3478
- Firewall/NAT Type: Open Internet (selected from a dropdown menu)
- Binding Refresh Time (seconds): 60
- Public IP Address: 192 . 168 . 96 . 231
- Public Port: 5060

At the bottom of the dialog box, there are two buttons: 'Run STUN' and 'Cancel'. Below these buttons is a checkbox labeled 'Run STUN on startup' which is currently unchecked. A red rectangular box highlights the 'Firewall/NAT Type', 'Binding Refresh Time', 'Public IP Address', and 'Public Port' fields.

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 DoubleHorn Communications, and therefore is not described in these Application Notes.



## 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.

Jersey City

Codecs

System LAN1 LAN2 DNS Voicemail **Telephony** Directory Services System Events SMTP SMDR Twinning VCM CCR

Telephony Tones & Music Call Log

Analogue Extensions

Default Outside Call Sequence Normal

Default Inside Call Sequence Ring Type 1

Default Ring Back Sequence Ring Type 2

Restrict Analogue Extension Ringer Voltage ☐

Dial Delay Time (secs) 4

Dial Delay Count 0

Default No Answer Time (secs) 15

Hold Timeout (secs) 120

Park Timeout (secs) 300

Ring Delay (secs) 5

Call Priority Promotion Time (secs) Disabled

Default Currency USD

Default Name Priority Favor Trunk

Companding Law

Switch

☒ U-Law

☐ A-Law

Line

☒ U-Law Line

☐ A-Law Line

DSS Status

☒ Auto Hold

☒ Dial By Name

☒ Show Account Code

☐ Inhibit Off-Switch Forward/Transfer

☐ Restrict Network Interconnect

☐ Drop External Only Impromptu Conference

☐ Visually Differentiate External Call

☐ Unsupervised Analog Trunk Disconnect Handling

☒ High Quality Conferencing

### 5.3. Twinning Calling Party Settings

Navigate to the **Twining** 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**).



The screenshot shows a web-based configuration interface for "Jersey City". The interface has a blue header bar with the title "Jersey City" and a toolbar with icons for help, save, delete, confirm, and navigation. Below the header is a tabbed menu with the following tabs: VCM, CCR, Codecs, System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, and Twining. The "Twining" tab is currently selected and highlighted. The main content area of the "Twining" tab contains a checkbox labeled "Send original calling party information for Mobile Twinning", which is currently unchecked. Below this checkbox is a text input field labeled "Calling party information for Mobile Twinning".

## 5.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and DoubleHorn Communications 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.

- 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.
- Uncheck **REFER Support**. For the compliance test, REFER was not used for call transfer. See the REFER item in the observation/limitation list in **Section 2.2** for details.
- 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.

The screenshot shows the Avaya IP Office configuration interface for SIP Line - Line 17. The interface is divided into a left navigation pane and a main configuration pane. The navigation pane shows a tree view of system components, including IP Offices, Operators, Jersey City, System, Line, Control Unit, Extension, User, HuntGroup, Short Code, Service, RAS, Incoming Call Route, WanPort, Directory, Time Profile, Firewall Profile, IP Route, Account Code, License, Tunnel, User Rights, ARS, RAS Location Request, and E911 System. The main configuration pane displays the configuration for Line 17, with several fields and checkboxes highlighted by red boxes. The highlighted fields include: In Service (checked), Check OOS (checked), Call Routing Method (Request URI), Send Caller ID (Diversion Header), Association Method (By Source IP address), REFER Support (unchecked), Incoming (Always), Outgoing (Always), and UPDATE Supported (Auto).

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

- Set the **ITSP Proxy Address** to the IP address of the service provider SIP Trunking access interface provided by DoubleHorn Communications.
- 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.

The screenshot shows the 'SIP Line - Line 17\*' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '192.168.110.124'. The 'Network Configuration' section is highlighted with a red box, showing 'Layer 4 Protocol' set to 'UDP', 'Send Port' set to '5060', 'Use Network Topology Info' set to 'LAN 2', and 'Listen Port' set to '5060'. Below this, 'Explicit DNS Server(s)' are set to '0.0.0.0' and '0.0.0.0'. 'Calls Route via Registrar' is checked with a green box. The 'Separate Registrar' field is empty.

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 DID 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**.
- 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 **0:<None>** from the pull-down menu. DoubleHorn Communications SIP Trunking does not require trunk registration.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing**

**Group** field. For the compliance test, the 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.

**SIP Line - Line 17\***

SIP Line Transport **SIP URI** VoIP T38 Fax SIP Credentials

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	17 17	1...					0: <Non...	10

Add...  
Remove  
Edit...

**Edit Channel**

Via: 192.168.96.231

Local URI: Use Internal Data

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: Use Internal Data

Registration: 0: <None>

Incoming Group: 17

Outgoing Group: 17

Max Calls per Channel: 10

OK  
Cancel

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 codec selection to the **Selected** box. Though the G.729A codec was also tested during the compliance test, the finalized configuration used G.711MU since it is the only codec supported by DoubleHorn Communications SIP Trunking.
- 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.

The screenshot shows the 'SIP Line - Line 17' configuration window with the 'VoIP' tab selected. The 'Codec Selection' is set to 'Custom'. The 'Unused' list contains G.711 ALAW 64K, G.723.1 6K3 MP-MLQ, and G.729(a) 8K CS-ACELP. The 'Selected' list contains G.711 ULAW 64K. The 'Fax Transport Support' is set to G.711. The 'Call Initiation Timeout (s)' is set to 4. The 'DTMF Support' is set to RFC2833. The 'Re-invite Supported' and 'PRACK/100rel Supported' checkboxes are checked. The 'VoIP Silence Suppression', 'Use Offerer's Preferred Codec', and 'Codec Lockdown' checkboxes are unchecked.

Unused	Selected
G.711 ALAW 64K	G.711 ULAW 64K
G.723.1 6K3 MP-MLQ	
G.729(a) 8K CS-ACELP	

Fax Transport Support: G.711

Call Initiation Timeout (s): 4

DTMF Support: RFC2833

VoIP Silence Suppression: ☐

Re-invite Supported: ☒

Use Offerer's Preferred Codec: ☐

Codec Lockdown: ☐

PRACK/100rel Supported: ☒



## 5.5. Short Code

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

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. The **9N;** short code, used for the compliance test, will be invoked when the user dials 9 followed by any number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N"@192.168.110.124"**. 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 of the DoubleHorn Communications SIP Trunking access interface follows 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.

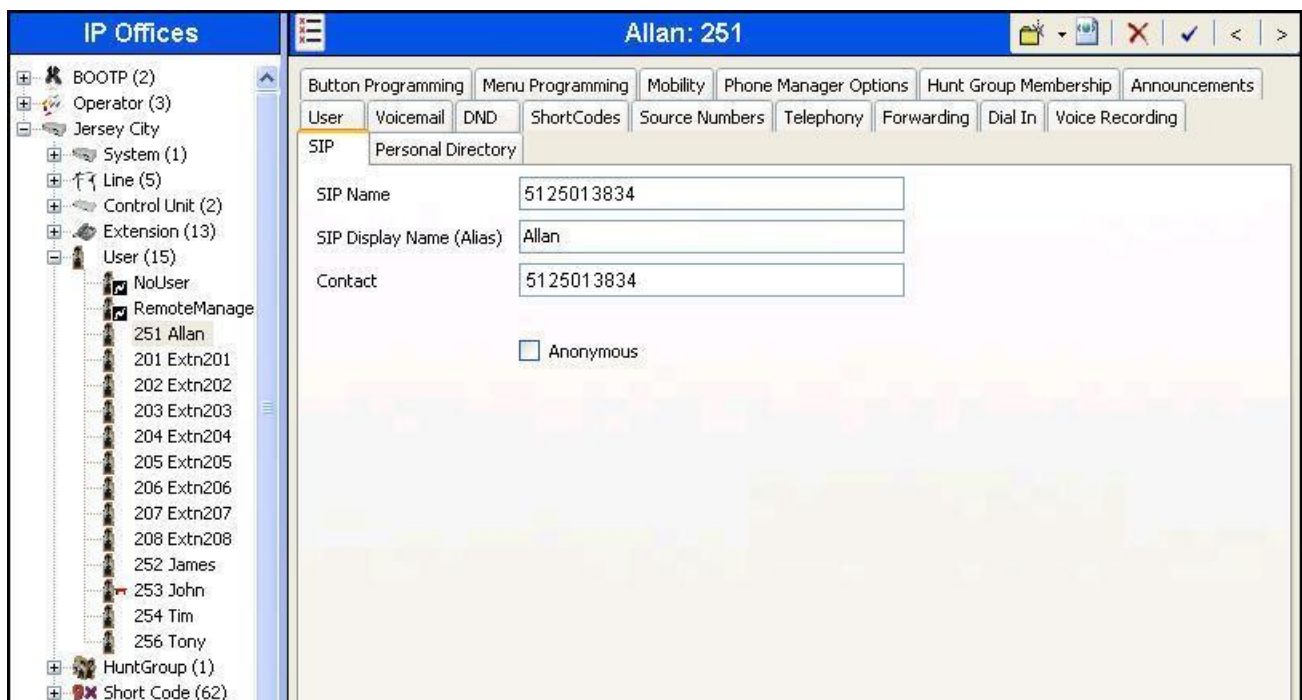
Default values may be used for all other parameters.

The screenshot displays the Avaya SIP Office configuration interface. On the left is the 'IP Offices' navigation pane, which includes a tree structure with items like BOOTP (2), Operator (3), Jersey City, System (1), Jersey City, Line (5), Control Unit (2), Extension (13), User (15), HuntGroup (1), Short Code (62), Service (0), RAS (1), Incoming Call Route (1), WanPort (0), and Directory (0). The 'Short Code (62)' item is selected. The main area on the right is titled '9N;; Dial\*' and contains a 'Short Code' configuration form. The form fields are: 'Code' with the value '9N;;', 'Feature' with a dropdown menu set to 'Dial', 'Telephone Number' with the value 'N"@192.168.110.124"', 'Line Group ID' with a dropdown menu set to '17', 'Locale' with a dropdown menu set to 'United States (US English)', and 'Force Account Code' with an empty checkbox. A red rectangular box highlights the 'Code', 'Feature', 'Telephone Number', and 'Line Group ID' fields.

## 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 by DoubleHorn Communications. 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 Contact header for outgoing SIP INVITE to the service provider. 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.



## 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 DID number on which this route should match. Matching is right to left.

Default values can be used for all other fields.

The screenshot shows the 'Incoming Call Route' configuration window with the 'Standard' tab selected. The left navigation pane shows a tree structure with 'Incoming Call Route (19)' selected. The main area contains the following fields:

Bearer Capacity	Any Voice
Line Group ID	17
Incoming Number	5125013834
Incoming Sub Address	
Incoming CLI	
Locale	United States (US English)
Priority	1 - Low
Tag	
Hold Music Source	System Source

On the **Destinations** tab, select the destination from the pull-down list of the **Destination** field. In this example, incoming calls to 5125013834 on line 17 are to be routed to the user Allan at extension 251.

The screenshot shows the 'Incoming Call Route' configuration window with the 'Destinations' tab selected. The main area contains a table with the following data:

TimeProfile	Destination	Fallback Extension
Default Value	251 Allan	

## 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:

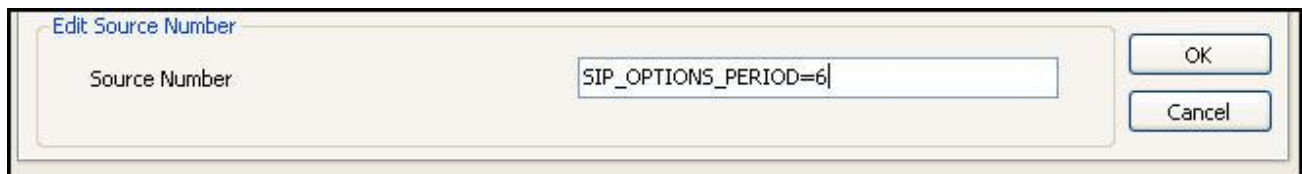
- If no **SIP\_OPTIONS\_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 300 seconds is used.
- To establish a period less than 300 seconds, do not define a **SIP\_OPTIONS\_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 seconds. 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.

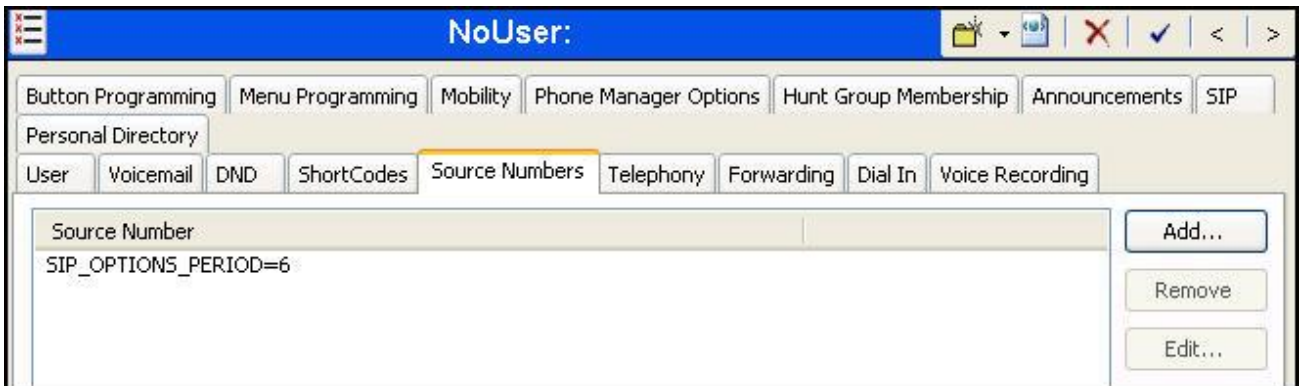
Alternatively, if an OPTIONS period greater than 300 seconds is desired then define the **SIP\_OPTIONS\_PERIOD** by doing the following. Navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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. DoubleHorn Communications SIP Trunking Configuration

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

- Network edge IP address of the DoubleHorn Communications SIP Trunking service
- Transport and port for the DoubleHorn Communications SIP Trunking connection to the Avaya IP Office at the enterprise site.
- 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.



## 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 State** shown as *Connected* (not shown).

The screenshot displays the Avaya IP Office System Status application. The left sidebar contains a navigation menu with options: System, Alarms (2), Extensions (11), Trunks (5), Lines: 1 - 4, Line: 17 (selected), Active Calls, Resources, Voicemail, and IP Networking. The main window has tabs for Status, Utilization Summary, and Alarms. The Status tab is active, showing a SIP Trunk Summary with the following details:

- Peer Domain Name: sip://192.168.110.124
- Resolved Address: 192.168.110.124
- Line Number: 17
- Number of Administered Channels: 20
- Number of Channels in Use: 0
- Administered Compression: G711 Mu
- Silence Suppression: Off
- SIP Trunk Channel Licenses: Unlimited
- SIP Trunk Channel Licenses in Use: 0
- SIP Device Features: UPDATE (Incoming and Outgoing)

A green circle indicates 0% utilization. Below the summary is a table showing the status of 10 channels. All channels are currently Idle.

Cha...	U...	Call	Curr...	Time	Remote C...	Con...	Caller	Other	Dire...	Roun	Rec...	Rec...	Tran...	Tran...
Ref			in S...	Medi...			ID o...	Party on...		d				
1			Idle	00:2...										
2			Idle	1 da...										
3			Idle	1 da...										
4			Idle	1 da...										
5			Idle	1 da...										
6			Idle	1 da...										
7			Idle	1 da...										
8			Idle	1 da...										
9			Idle	1 da...										
10			Idle	1 da...										

At the bottom of the window, there are buttons for Trace, Trace All, Pause, Ping, Call Details, Print..., and Save As...



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

Status	Utilization Summary	Alarms
Alarms for Line: 17 SIP sip://192.168.110.124		
Last Date Of Error	Occurrences	Error Description

- 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 DoubleHorn Communications IPhone SIPtrunk service passed compliance testing. These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office and DoubleHorn Communications 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 FPI 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 the DoubleHorn Communications IPhone SIPtrunk service is available from DoubleHorn Communications. See **Section 2.3** on how to contact DoubleHorn Communications.

## 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 version="1.0" encoding="utf-8" ?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20130222</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>DoubleHorn SIP Trunking</DescriptiveName>
  <ITSPDomainName />
  <SendCallerID>CallerIDDIV</SendCallerID>
  <ReferSupport>false</ReferSupport>
  <ReferSupportIncoming>0</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>192.168.110.124</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>
```

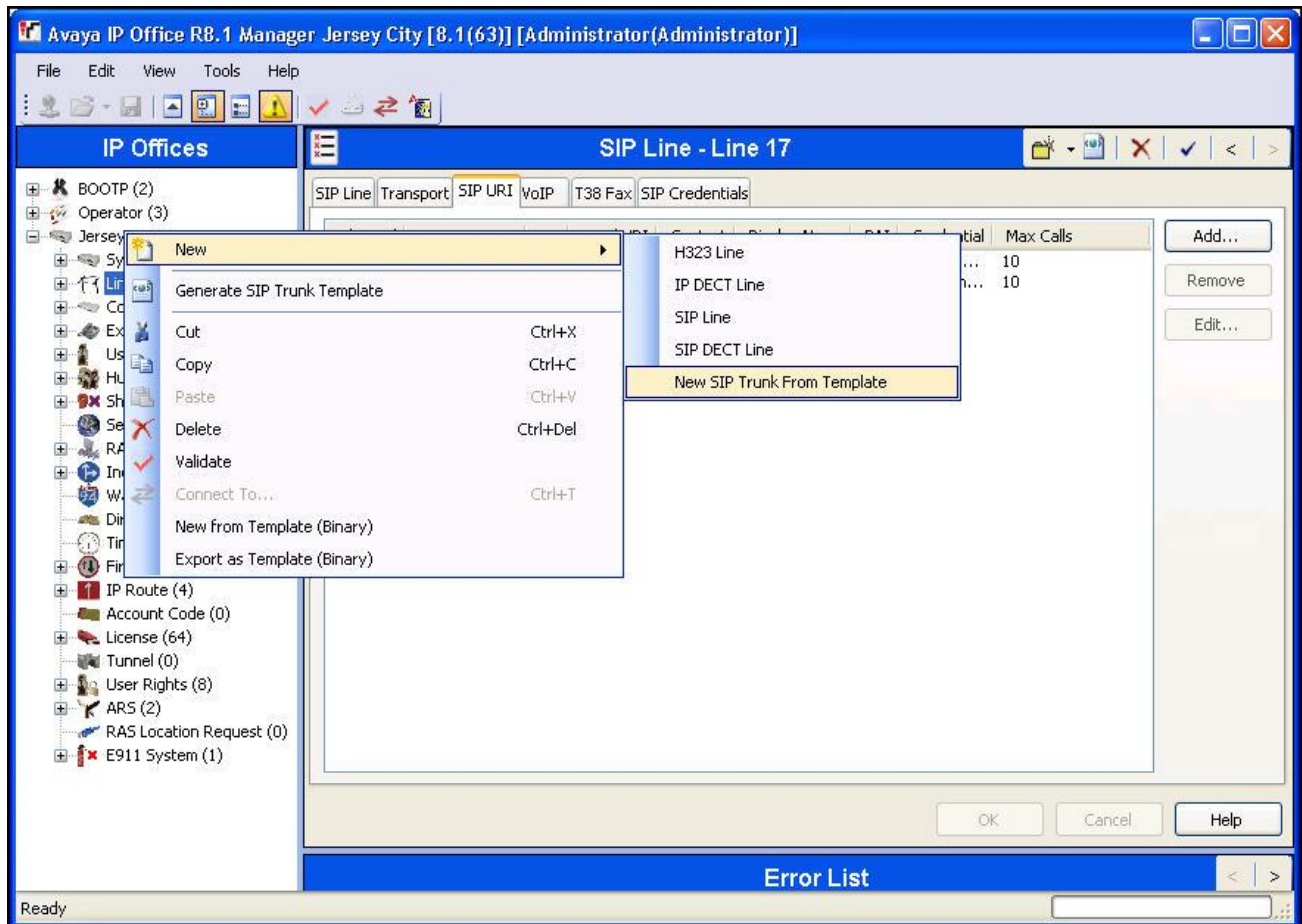
```

<DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
<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>
</Template>

```

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\_DoubleHorn\_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**:



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



The screenshot shows a Windows-style dialog box titled "Template Type Selection". It contains three dropdown menus: "Locale" set to "United States (US English)", "Country" set to "United States", and "Service Provider" set to "DoubleHorn". To the right of the "Service Provider" dropdown is an unchecked checkbox labeled "Display All". At the bottom right are two buttons: "Create new SIP Trunk" and "Cancel".

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