Avaya Solution & Interoperability Test Lab

Application Notes for IntelePeer CoreCloud SIP Trunking Service with Avaya IP Office Release 8.1 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring IntelePeer CoreCloud Session Initiation Protocol (SIP) Trunking Service with Avaya IP Office Release 8.1.

IntelePeer CoreCloud SIP Trunking Service provides PSTN access via a SIP Trunk between the enterprise and IntelePeer CoreCloud networks as an alternative to legacy analog or ISDN-PRI trunks. This approach generally results in lower cost for the enterprise.

IntelePeer 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 service provider IntelePeer CoreCloud and Avaya IP Office solution. In the sample configuration, Avaya IP Office solution consists of Avaya IP Office (IP Office) Release 8.1 and various Avaya endpoints.

IntelePeer CoreCloud (ICC) SIP Trunking Service referenced within these Application Notes is designed for business customers. The service enables PSTN calling via a broadband WAN connection using SIP protocol. This converged network solution is a cost effective alternative to traditional PSTN trunks such as analog and/or ISDN-PRI.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using IP Office to connect to ICC. This configuration (shown in **Figure 1**) was used to exercise the feature and functionality tests listed in **Section 2.1**.

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 ICC SIP Trunking interoperability, following features and functionalities were exercised during the compliance testing:

- Incoming PSTN calls to various phone types including SIP, H.323, digital and analog telephones at the enterprise. All incoming calls from PSTN are routed to the enterprise across the SIP Trunk from the service provider networks.
- Outgoing PSTN calls from various phone types including SIP, H.323, digital and analog telephone at the enterprise. All outgoing calls to PSTN are routed from the enterprise across the SIP trunk to the service provider networks.
- Incoming and outgoing PSTN calls to/from Avaya IP Office Softphone using both SIP and H.323 protocols.
- Dialing plans including local, long distance, outgoing toll-free calls, local directory assistance (411), etc.
- Calling Party Name presentation and Calling Party Name restriction.
- Proper codec negotiation of G.711MU and G.729 codecs.
- Proper early media transmissions G.711MU and G.729 codecs.
- Proper media transmission using G.711MU and G.729 codecs.
- DTMF tone transmissions as out-of-band RTP event per RFC 2833.
- Voicemail navigation for incoming and outgoing calls.
- Telephony features such as hold and resume, call transfer, call forward and conferencing.
- Off-net call transfer with re-INVITE method.

- Off-net call forward with Diversion method.
- Mobility Twinning incoming calls to mobile phones with Diversion method.
- Response to OPTIONS heartbeat.
- Response to incomplete call attempts and trunk errors.
- Proper fax over IP with T.38 codes.

The following features are supported by ICC SIP trunk service but were not tested as part of this testing.

- Operator Assisted Call 0 and 0 +10 digits.
- Local directory assistance (411).
- Emergency service (911).

ICC SIP trunk service does not support the following:

- ICC does not support SIP REFER method for call redirection.

2.2 Test Results

Interoperability testing of ICC with Avaya IP Office solution was successfully completed with the exception of the observations/limitations described below.

1. **ICC does not send OPTIONS but response to IP Office OPTIONS with 200OK**.

2. **Calling Party Name and Number are not updated if IP Office off-net redirects (by transferring or forwarding) an incoming or outgoing call back to PSTN**. Before (and after) completing the off-net redirection, IP Office did not send UPDATE or re-INVITE signaling to update the call display on PSTN parties. This is a known behavior of IP Office with no available resolution at this time. This issue has low user impact, it is listed here simply as an observation.

3. **Calling Party Name and Number are not updated if IP Office off-net redirects (by transferring or forwarding) an incoming or outgoing call to internal station**. Before (or after) completing the local redirection to internal station, IP Office did not send UPDATE or re-INVITE signaling to update the call display on PSTN party. This is a known behavior of IP Office with no available resolution at this time. This issue has low user impact, it is listed here simply as an observation.

4. **Outbound Fax T.38 does not work –** The ICC intermittently negotiates the T.38 fax call. Once the fax call does negotiate successfully, there is no signal being sent from the ICC network to IPO for fax transmission.

2.3 Support

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

For technical support on IntelePeer CoreCloud SIP Trunking Service, contact IntelePeer technical support at http://www.intelepeer.com.

3. Reference Configuration

Figure 1 below illustrates the test configuration. It shows an enterprise site connected to the ICC networks through the Internet.

For confidentiality and privacy purposes, actual public IP addresses and PSTN routable phone numbers used in the certification testing have been replaced with fictitious parameters throughout the Application Notes.

The Avaya components used to create the simulated customer site including:

- Avaya IP Office 500v2
- Avaya Voicemail Pro for IP Office
- Avaya 9600 Series H.323 IP Telephones
- Avaya 11x0 Series SIP IP Telephones
- Avaya IP Office soft-phones (SIP and H.323 modes)
- Avaya 9508 Digital Telephones
- Avaya Analog 8809 Telephones

Located at the enterprise site is Avaya IP Office 500v2 with the MOD DGTL STA16 expansion to provide connection for 16 digital stations, the PHONE 8 module to provide connection for 8 analog stations and the 64-channel Voice Compression Module (VCM) for supporting VoIP codec. IP Office has the LAN port that connects to ICC networks via the Internet.

Mobility Twinning is configured for some IP Office users so that incoming calls to these user phones can also be delivered to the configured mobile phones.

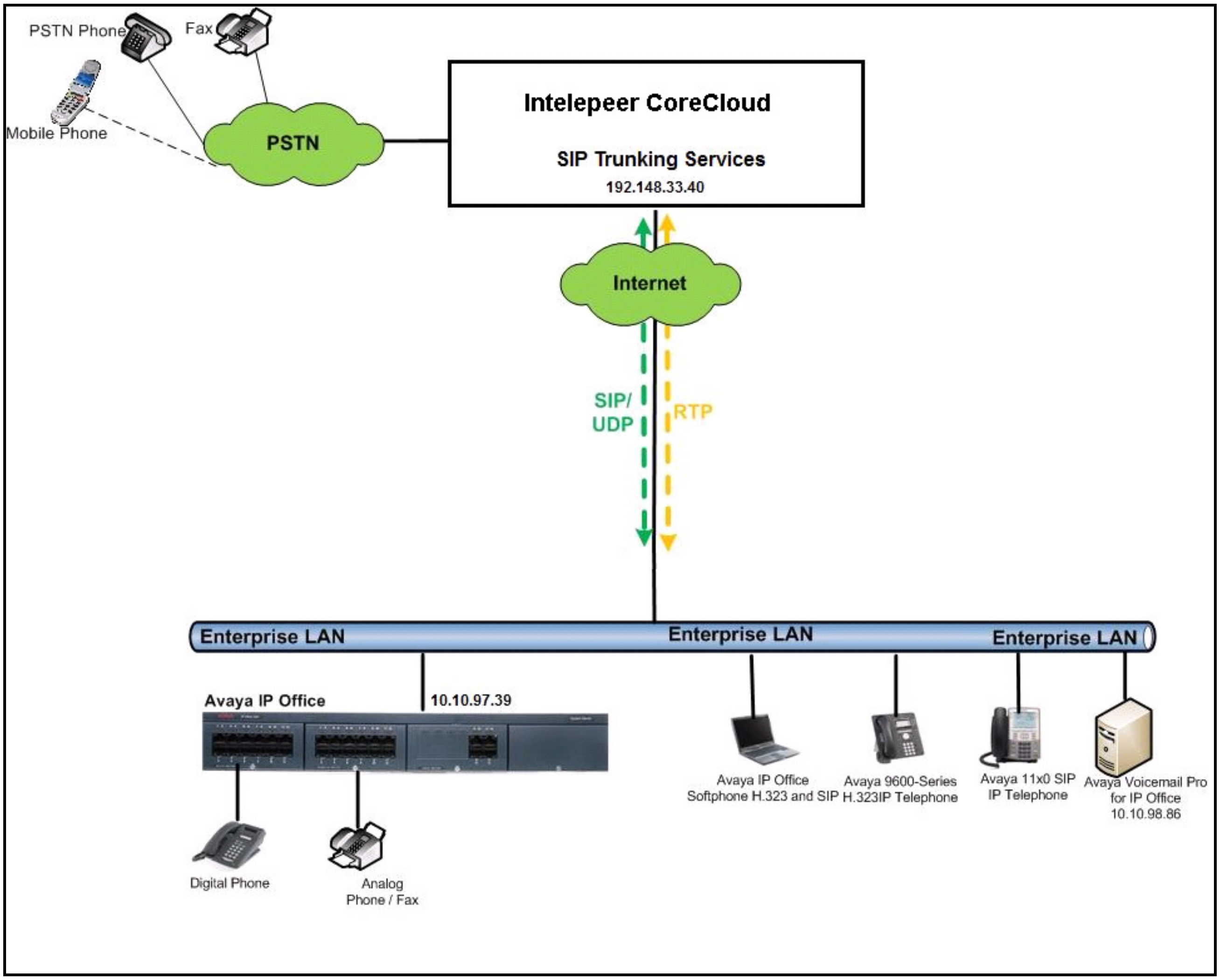


Figure 1: Avaya IP Telephony Network Connecting to ICC SIP Trunking Service

For the compliance testing, ICC provided the service provider public SIP domain as its Central Office (CO) IP address **192.148.33.40** and the enterprise public SIP domain as the Avaya IP Office IP address **10.10.97.39**. These public SIP domains will be used for public SIP and RTP traffics between ICC and the Avaya IP Office using transport protocol UDP.

For outgoing calls, IP Office sent 11 digits in destination headers, e.g. "Request-URI" and "To", and sent 10 digits in source headers, e.g. "From", "Contact", and "P-Asserted-Identity". For incoming calls, ICC sent 10 digits in destination headers and sent 11 digits in source headers.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and the enterprise such as a Firewall. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes. However, it should be noted that SIP and RTP traffic between the service provider and the enterprise 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.

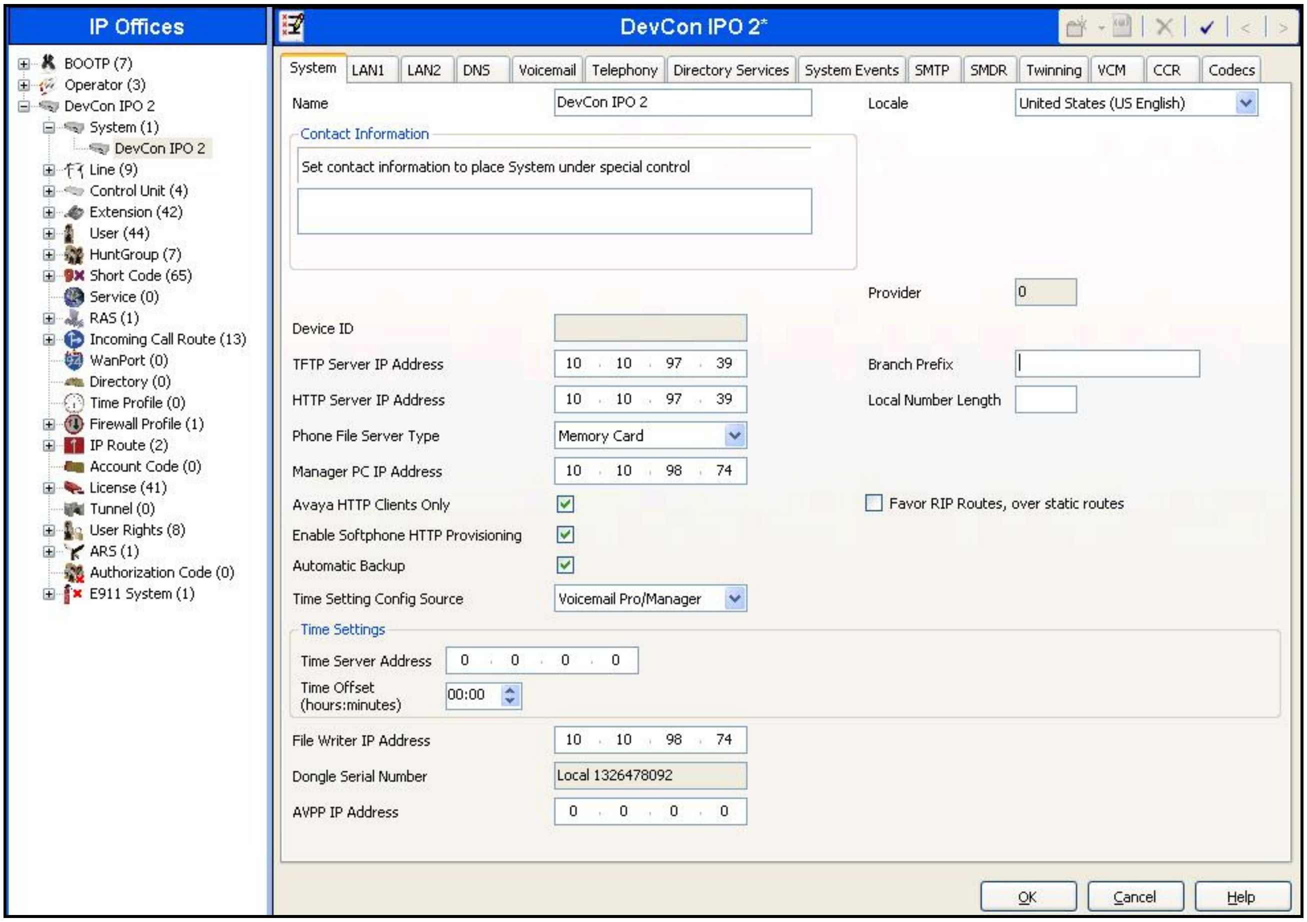
Avaya Telephony Components	
Equipment/Software	**Release/Version**
Avaya IP Office 500v2	8.1 (67)
Avaya IP Office DIG DCP*16 V2	8.1 (67)
Avaya IP Office Ext Card Phone 8	8.1
Avaya IP Office Manager	10.1 (67)
Avaya Session Border Controller for Enterprise (running on Portwell CAD-0208 platform)	6.2 (6.2.0 Q30)
Avaya Voicemail Pro for IP Office	8.1.1003.0
Avaya 9630G IP Telephone (H.323)	Avaya one-X® Deskphone Edition S3.2
Avaya 11x0 IP Telephone (SIP)	SIP11x0e04.03.12.00
Avaya IP Office Softphone	3.2.3.20 64770
Avaya Digital Telephone (9508)	N/A
Avaya Analog 8809 Telephone	N/A

IntelePeer CoreCloud SIP Trunking Service Components	
Equipment/Software	**Release/Version**
Sonus GSX9000	V07.03.07 R006

Testing was performed with IP Office 500v2 R8.1, but it also applies to IP Office Server Edition R8.1. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R8.1 to support analog or digital endpoints or trunks.

5. Configure IP Office

This section describes IP Office configuration required to interwork with ICC. It is configured through Avaya IP Office Manager (IP Office Manager) which is a PC application. On the PC, select **Start → Programs → IP Office → Manager** to launch IP Office Manager. Navigate to **File → Open Configuration**, select proper IP Office from pop-up window, and log in with the appropriate credentials. A management window will appear as shown below. The appearance of IP Office Manager can be customized using the **View** menu (not shown). In the screenshots presented in this section, the **View** menu was configured to show the **Navigation Pane** on the left side and the **Details Pane** on the right side. These panes will be referenced throughout these Application Notes.

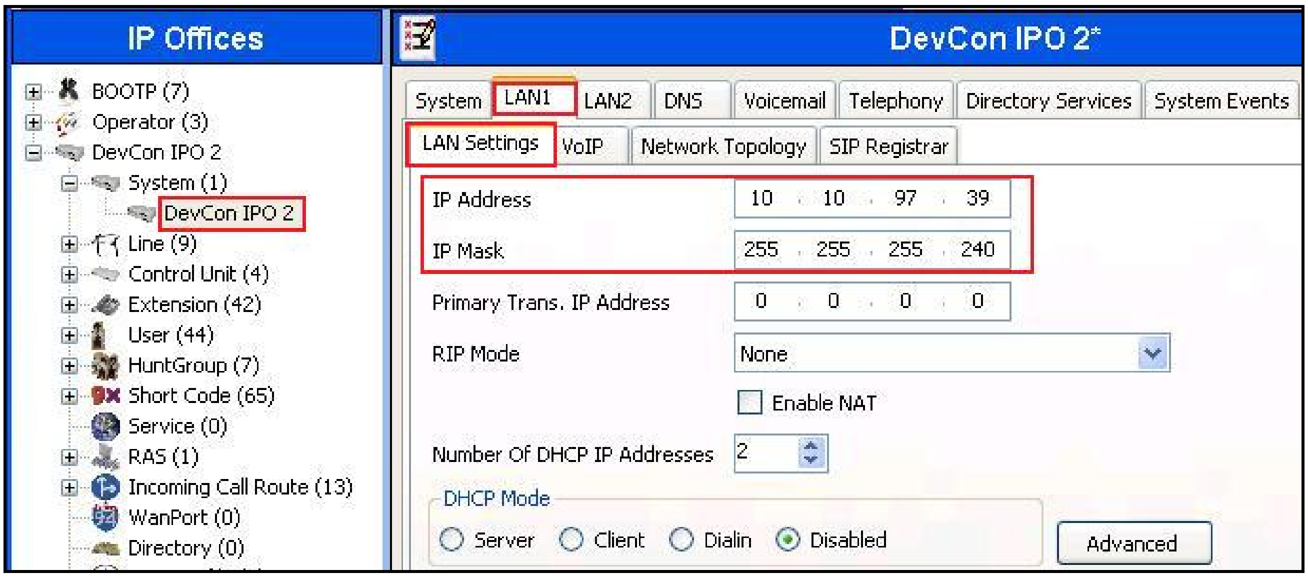


These Application Notes assume the basic installation and configuration have already been completed and are not discussed here. For further information on IP Office, please consult **References** in **Section 9**.

5.1 LAN

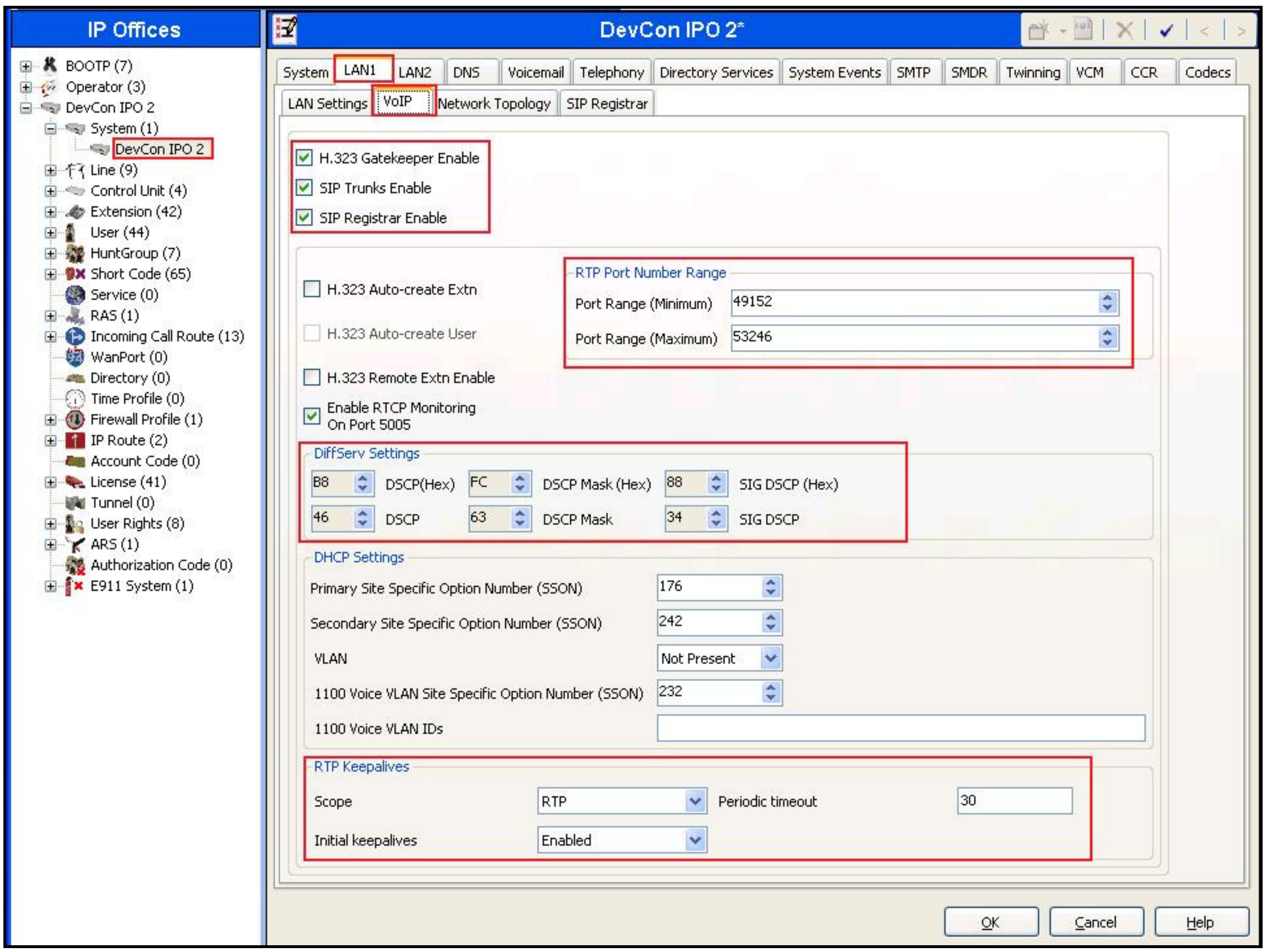
In the sample configuration, IP Office was configured with the system name **DevCon IPO 2** and LAN port was used to connect to ICC networks via the Internet. **LAN1** settings correspond to the LAN port on IP Office. To access **LAN1** settings, navigate to **System (1) → DevCon IPO 2** in Navigation Pane then in Details Pane navigate to the **LAN1→ LAN Settings** tab. The **LAN1** settings for the compliance testing were configured with following parameters.

- Set the **IP Address** field to the LAN IP address, e.g. **10.10.97.39**.
- Set the **IP Mask** field to the subnet mask of the public network, e.g. **255.255.255.240**.
- All other parameters should be set according to customer requirements.
- Click OK to commit (not shown) then press Ctrl + S to save.



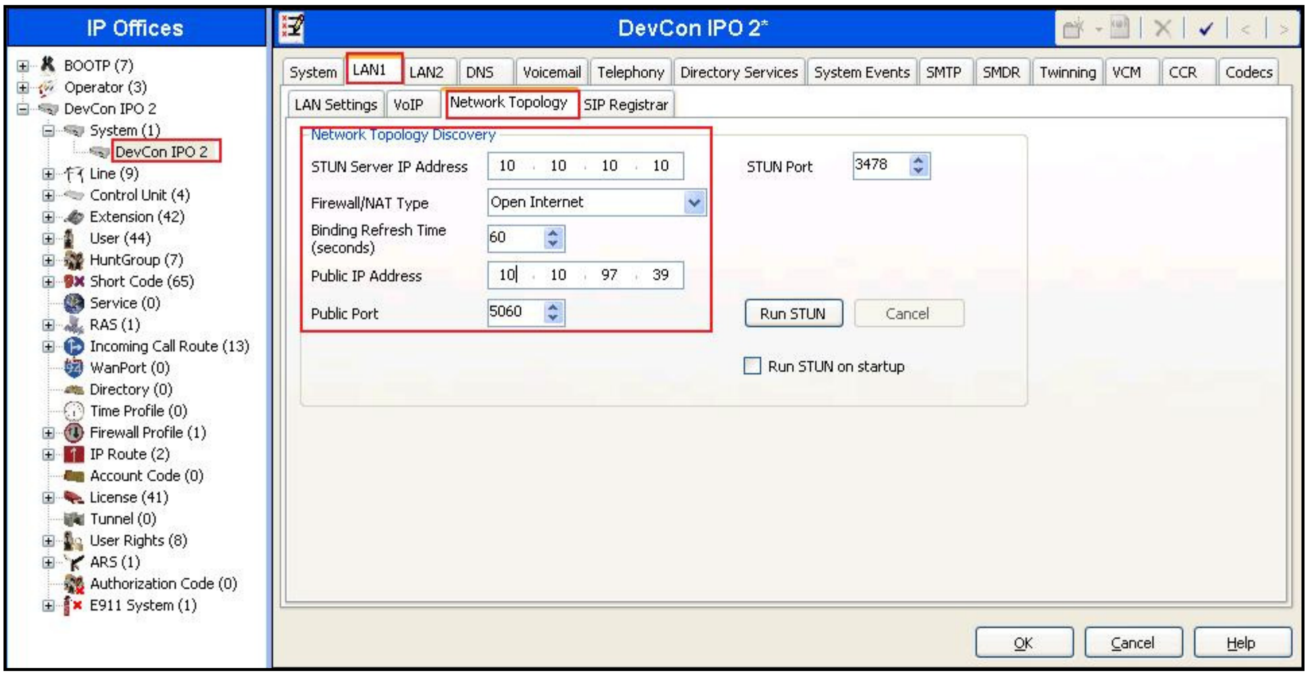
The **VoIP** tab as shown in the screenshot below was configured with following settings.

- Check the **H323 Gatekeeper Enable** to allow Avaya IP Telephones/Softphones using the H.323 protocol to register.
- Check the **SIP Trunks Enable** to enable the configuration of SIP Trunk connecting to ICC.
- Check the **SIP Registrar Enable** to allow Avaya IP Telephones/Softphones to register using the SIP protocol.
- Verify the **RTP Port Number Range** settings for a specific range for the RTP traffic. The **Port Range (Minimum)** and **Port Range (Maximum)** values were kept as default.
- Verify the **DiffServ Settings** were kept as default for the Differentiated Services Code Point (DSCP) parameters in the IP packet headers 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.
- Verify **RTP Keepalives** settings were enabled with **Scope** as **RTP**, **Periodic timeout** in **30** seconds, and **Initial keepalives** as **Enabled**. This allows IP Office to send IP packets to keep the active RTP session alive in every 30 seconds if there is no audio detected on the SIP Trunk.
- All other parameters should be set according to customer requirements.
- Click OK to commit, then press Ctrl + S to save.



In the **Network Topology** tab, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. In the compliance testing, it was set to **Open Internet**. With this configuration, even the default STUN settings are populated but they will not be used.
- Set the **Binding Refresh Time (seconds)** to **60**. This value is used to determine the frequency that IP Office will send OPTIONS heartbeat to the service provider.
- Set the **Public IP Address** to IP Office LAN IP address, e.g. **10.10.97.39**.
- Set the **Public Port** is set to **5060**.
- All other parameters should be set according to customer requirements.
- Click OK to commit then press Ctrl + S to save.



5.2 IP Route

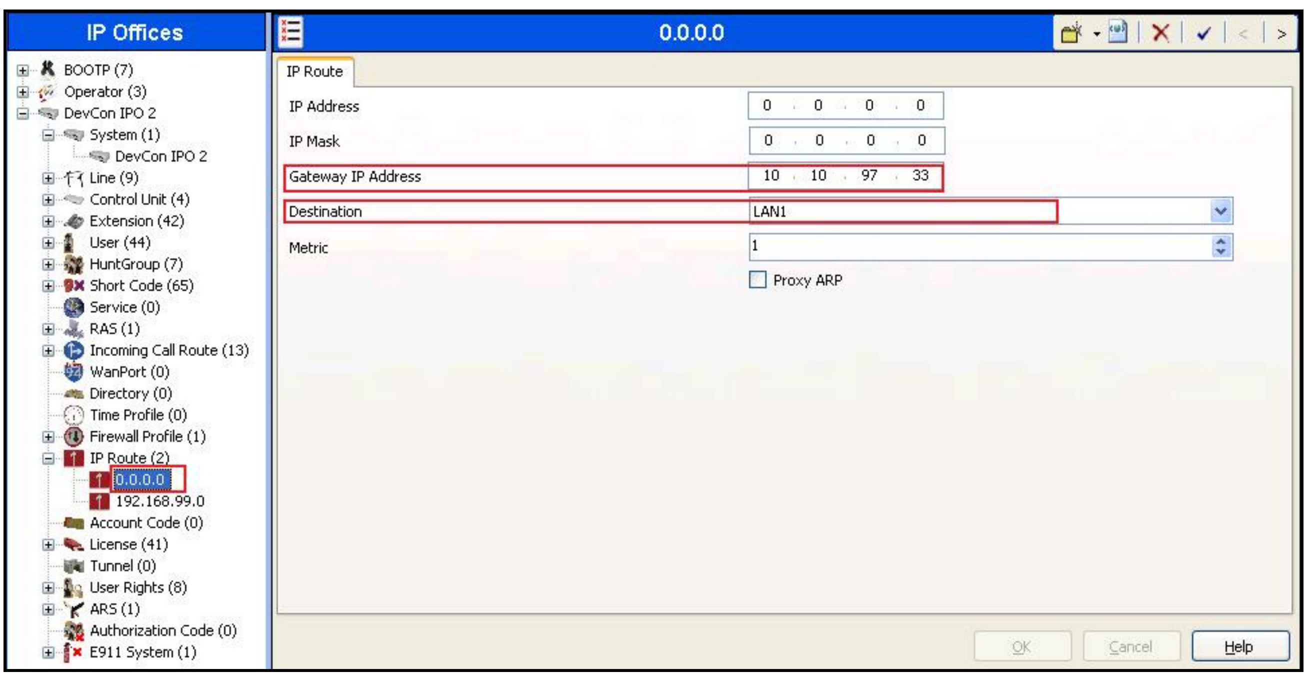
IP Route settings include an IP Route **0.0.0.0** on LAN1 connecting to ICC for SIP, RTP traffics and connecting to the private enterprise networks.

To create an IP Route, select **IP Route** in the Navigation Pane, then click "**Create a New Record**" icon (not shown).

The IP Routes were configured using the following settings.

- Set the **IP Address** to the address of the destination network.
- Set the **IP Mask** to the subnet mask of the destination network.
- Set the **Gateway IP Address** to the IP address of the enterprise gateway that routes traffic to the destination network.
- Set the **Destination** to the interface **LAN1**.
- All other parameters should be set according to customer requirements.
- Click OK to commit then press Ctrl + S to save.

The following screenshot shows the IP Route **0.0.0.0** that was created on **LAN1** for SIP and RTP traffics to ICC. **LAN1** was assigned to the network address **0.0.0.0** and default subnet mask **0.0.0.0**. The default gateway was set to IP address **10.10.97.33** which is an internal gateway on the enterprise network that connects to **LAN1**.

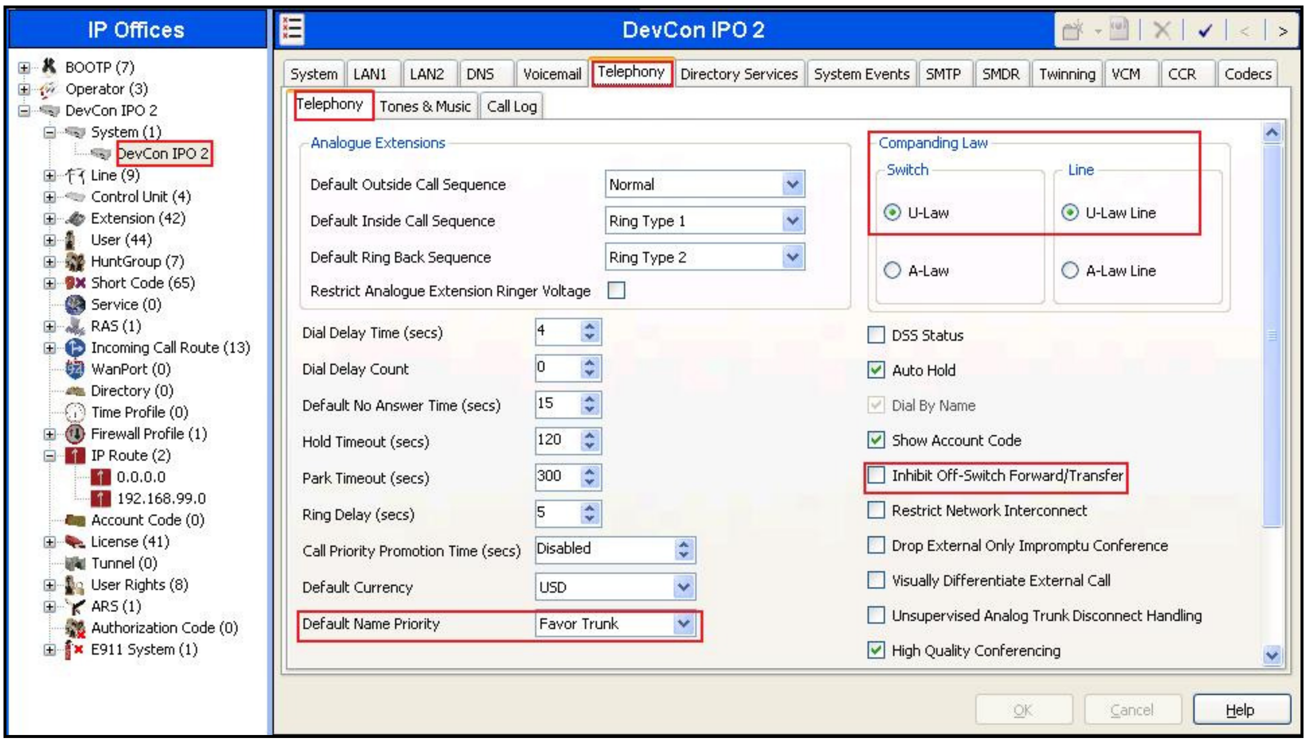


5.3 System Telephony and Codecs

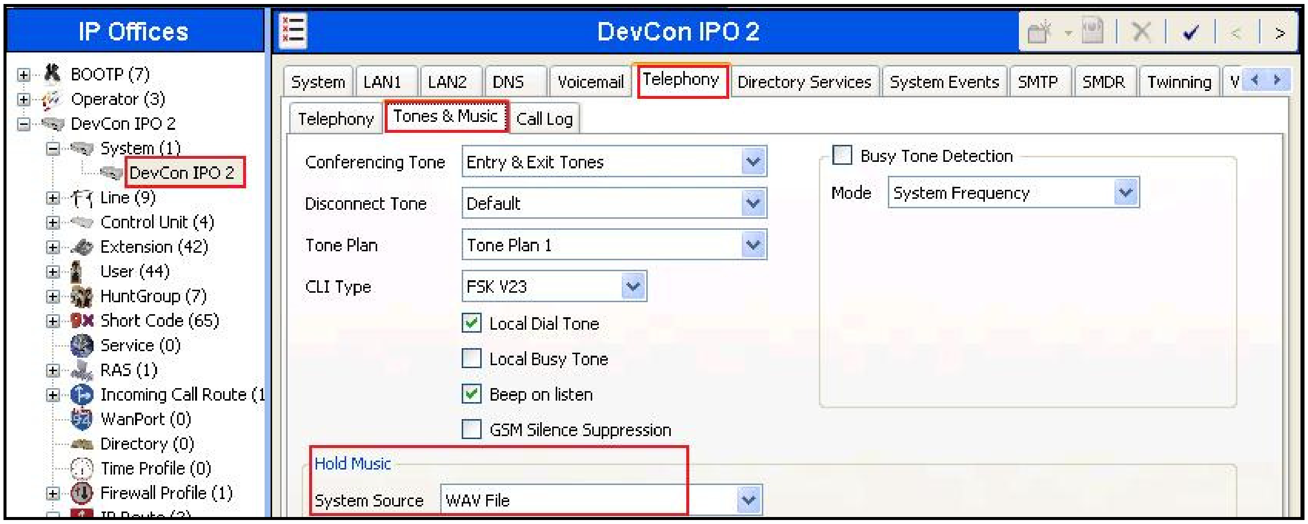
Navigate to the **System (1) → DevCon IPO 2** in the Navigation Pane then select **Telephony → Telephony** tab in the Details Pane.

The **Telephony** settings were configured with following parameters.

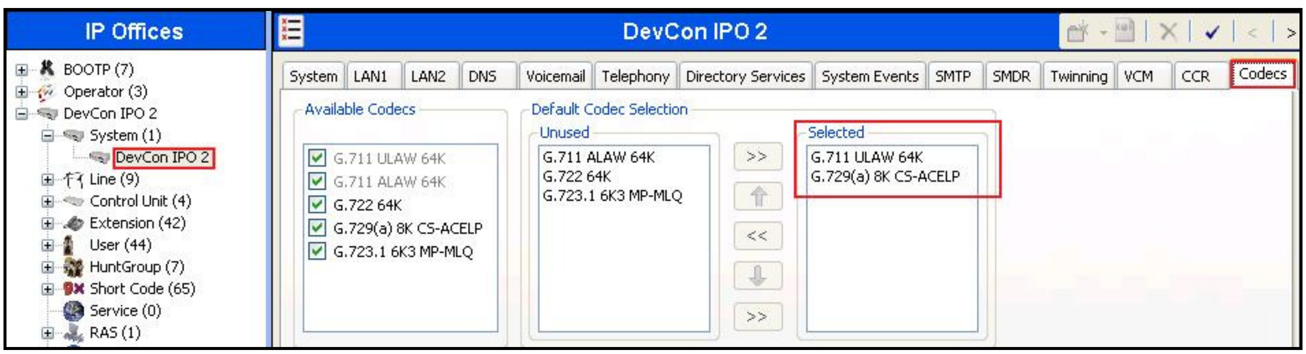
- Choose the **Companding Law** typical for the enterprise location. For North America, **U-LAW** was used for both **Switch** and **Line**.
- Set **Default Name Priority** to **Favor Trunk**. This allows IP Office to use information received from SIP Trunk for call display purpose rather than overriding it with pre-defined internal settings.
- Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to PSTN via the service provider SIP Trunk.
- Click OK to commit then press Ctrl + S to save.



Under **Tones & Music** tab as shown below, **Hold Music** was configured with **System Source** to use **WAV File** which is an uploaded medium to provide Music on Hold on the SIP Trunk.



For **Codecs** settings, navigate to the **System (1) → DevCon IPO 2** in the Navigation Pane, and then select **Codecs**. The **Codecs** settings are shown in the screenshot below with G.729 and G.711MU were selected. In the compliance testing, ICC supported both G.729 and G.711MU.



Click OK to commit (not shown) then press Ctrl + S to save.

5.4 Twinning Calling Party Information

When using Twinning, Calling Party Number displayed on the twinned phone is controlled by two parameters. The first parameter is the **Send original calling party information for Mobile Twinning** box on the **System→Twinning** tab. The second parameter is the **Send Caller ID** parameter on the **SIP Line** form shown in **Section 5.5.1**.

For the compliance testing, the **Send original calling party information for Mobile Twinning** as shown below was unchecked. This setting allows **Send Caller ID** parameter that was set to **Diversion Header** in **Section 5.5** to be used. IP Office will send the following in the "From" header:

- On calls from an internal extension to a twinned phone, IP Office sends Calling Party Number of the originating extension.
- On calls from the PSTN to a twinned phone, IP Office sends Calling Party Number of the originating PSTN party.

5.5 Administer SIP Line

A SIP Line was needed to establish the SIP Trunk between IP Office and ICC.

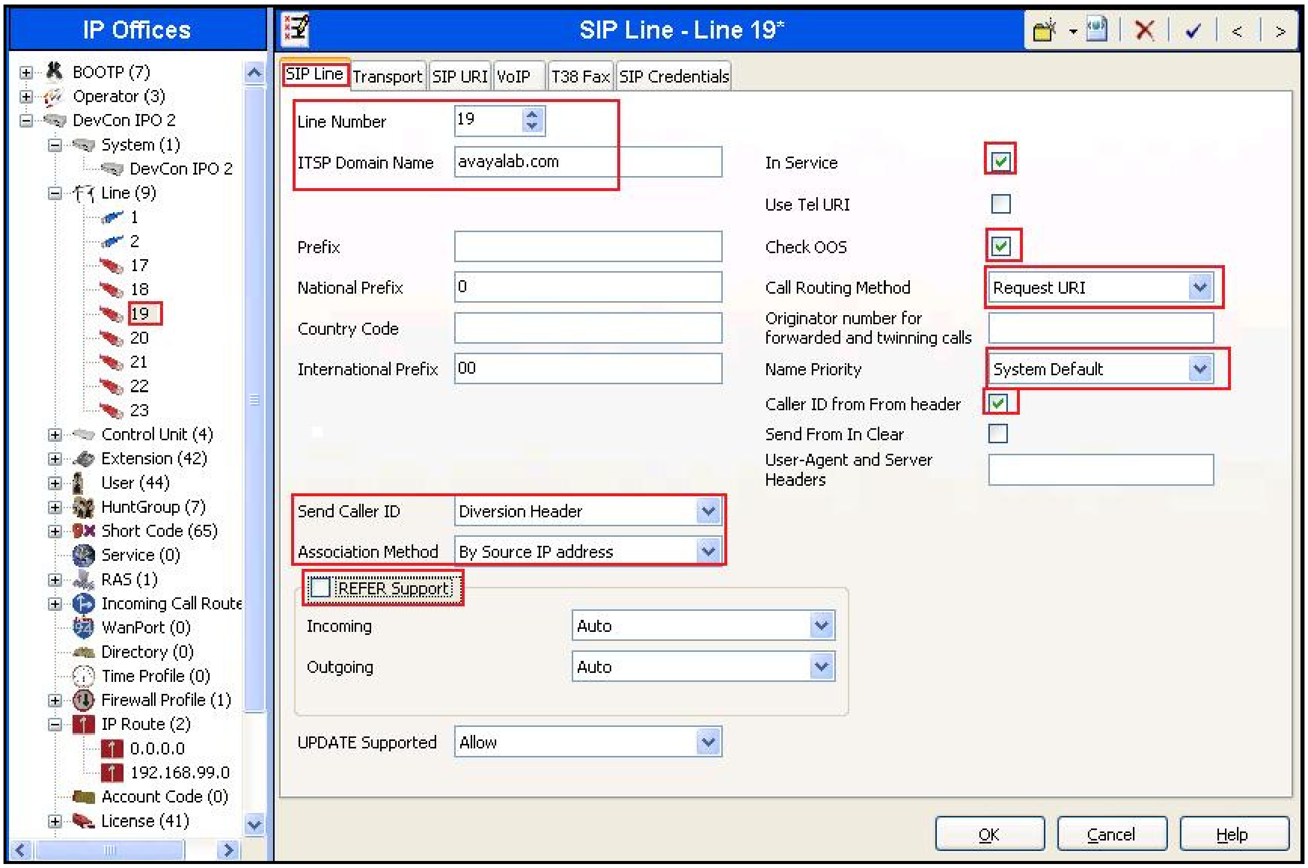
To create a SIP Line, navigate to **Line** in the left Navigation Pane then select **New → SIP Line** (not shown).

5.5.1 Administer SIP Line Settings

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

- Set the **Line Number** to an unassigned number, e.g. **19**.
- Set the **ITSP Domain Name** to the FQDN or IP address that will be used as the enterprise SIP domain so that IP Office uses this domain as the URI-Host of the "From", "P-Asserted-Identity" and "Diversion" headers. In the compliance testing, the enterprise SIP domain was defined as **avayalab.com** for the traffic between IP Office and the ICC.
- Set the **Send Caller ID** to **Diversion Header**. For the compliance testing, this parameter was used for Caller ID since **Send original calling party information for Mobile Twinning** was unchecked in **Section 5.4**.
- Set the **Association Method** to **By Source IP address**. This setting allows IP Office to apply the configuration for the public SIP Trunk to incoming and outgoing calls from/ to ICC if the traffics were originated from/ to the pre-defined IP address of the far end proxy server.
- Uncheck the **REFER Support**, since ICC does not supported REFER method.
- Set the **UPDATE Supported** field to **Allow** as ICC supported the UPDATE method in this certification testing.
- Check the **In Service** box.

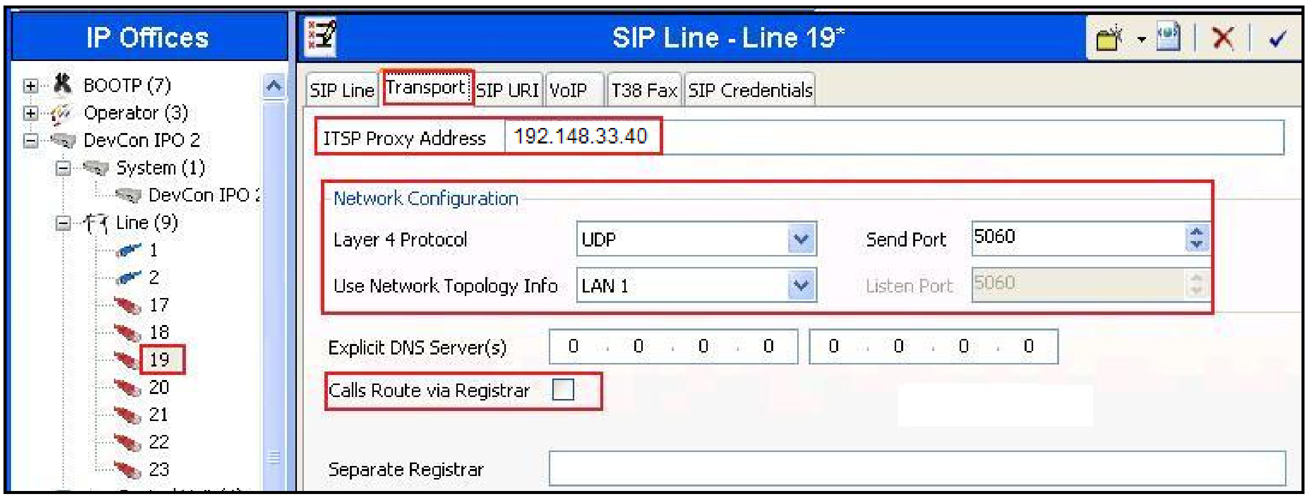
- Check the **Check OOS** box. With this option selected, IP Office will send the OPTIONS heartbeat to check status of the SIP Trunk.
- Set the **Call Routing Method** field to **Request URI**.
- Set the **Name Priority** field to **System Default**.
- Check the **Call ID from From header** box.
- Default values may be used for all other parameters.
- Click OK to commit then press Ctrl + S to save.



5.5.2 Administer Transport Settings

Select the **Transport** tab then configure the parameters as shown below.

- The **ITSP Proxy Address** was set to the IP Address of the ICC **192.148.33.40** as shown in **Figure 1**.
- In the **Network Configuration** area, **UDP** was selected as the **Layer 4 Protocol** and the **Send Port** was set to **5060**.
- The **Use Network Topology Info** parameter was set to **LAN 1**. This associates the SIP Line 19 with the parameters in the **System → LAN1 → Network Topology** tab.
- The **Calls Route via Registrar** was unchecked. In this certification testing, ICC did not support the dynamic Registration on the SIP Trunk.
- Other parameters retain default values.
- Click OK to commit (not shown) then press Ctrl + S to save.

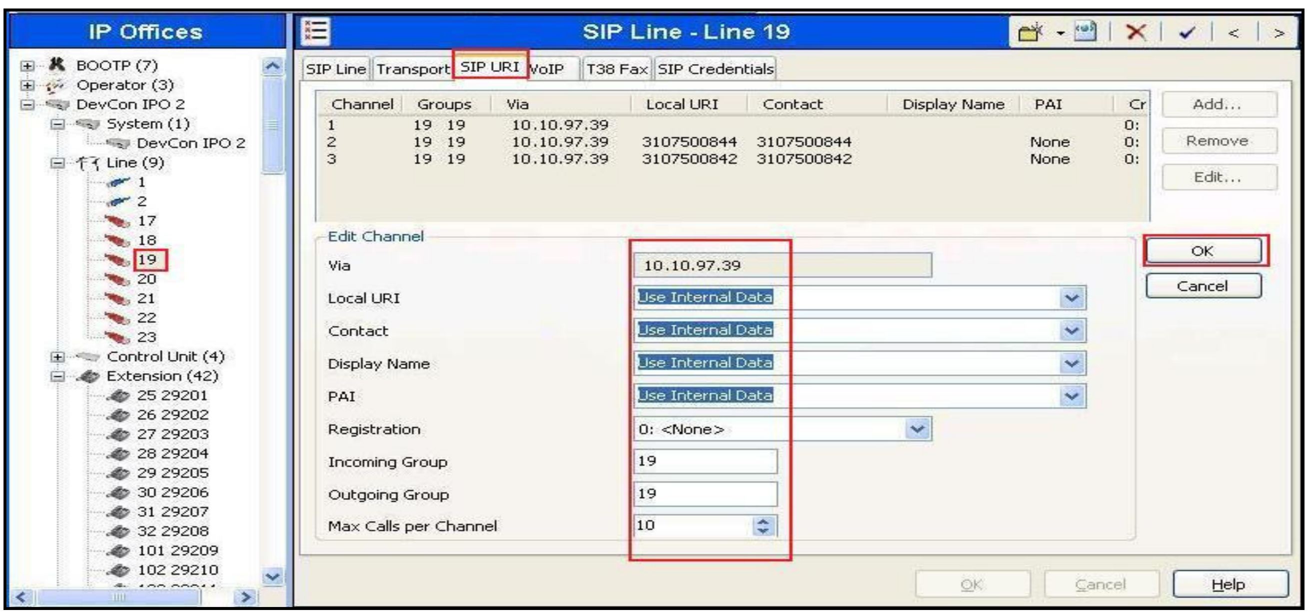


5.5.3 Administer SIP URI Settings

SIP URIs entry must be created to match Calling Party Number for incoming calls or to present Calling Party Number for outgoing calls on the SIP Line. Select the **SIP URI** tab then click the **Add** button and the **New Channel** area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the example screenshot below, previously configured entries were edited.

For the compliance testing, SIP URI entry with **Channel 1** was created for incoming and outgoing calls. Its parameters were shown below:

- Set the **Local URI**, **Contact**, **Display Name** and **PAI** to **Internal Data**. This setting will use Calling Party Number defined under the **SIP** tab of **User** as shown in **Section 5.7** for the public SIP calls.
- For the **Registration** field, select **<None>** to disable the Registration.
- Associate SIP Line 19 to the **Incoming Group** and **Outgoing Group**. The line group number will be used in defining incoming or outgoing call routes for this SIP Line.
- Set the **Max Calls per Channel** to **10** which is the number of simultaneous SIP calls that are allowed using this SIP URI pattern.



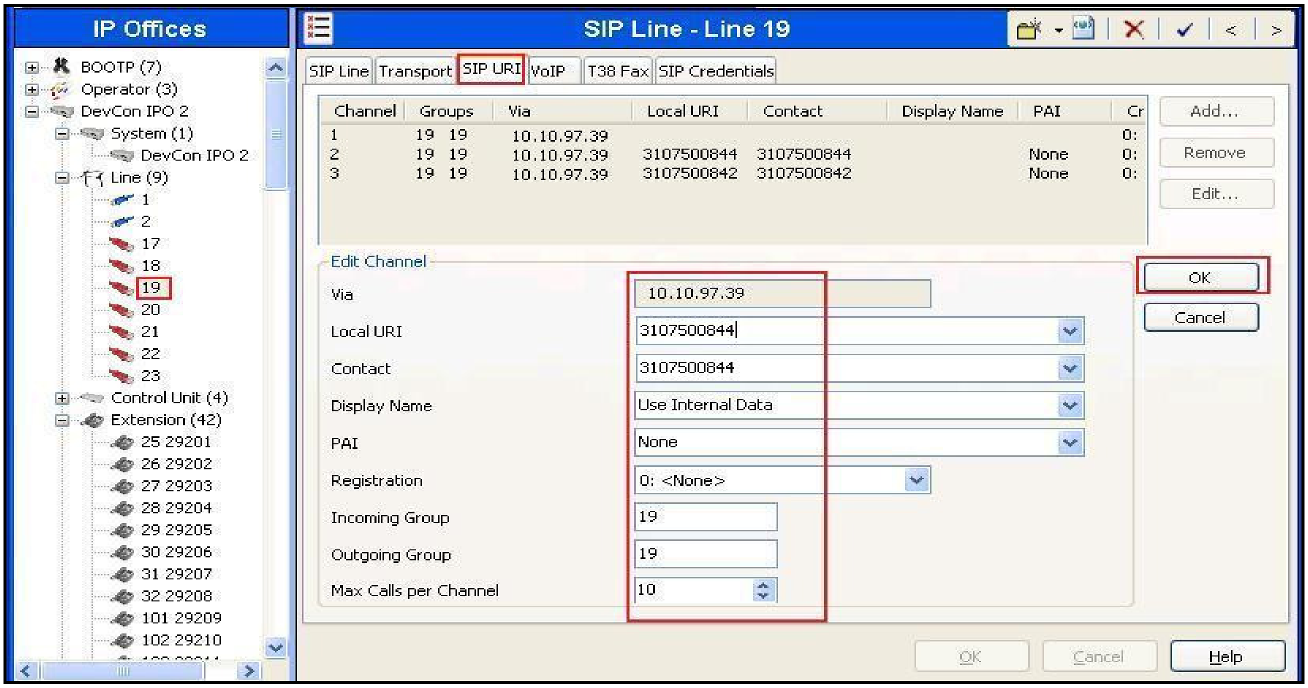
SIP URI entry **Channel 2** and **Channel 3** were similarly created for incoming calls appropriately to pre-define DID numbers **3107500844** and **3107500842** to access to Feature Name Extension 00 (FNE00) and Voicemail respectively. The Short Codes for FNE00 was defined in **Section 5.6** to provide Dial Tone and Mobile Callback for mobility extension.

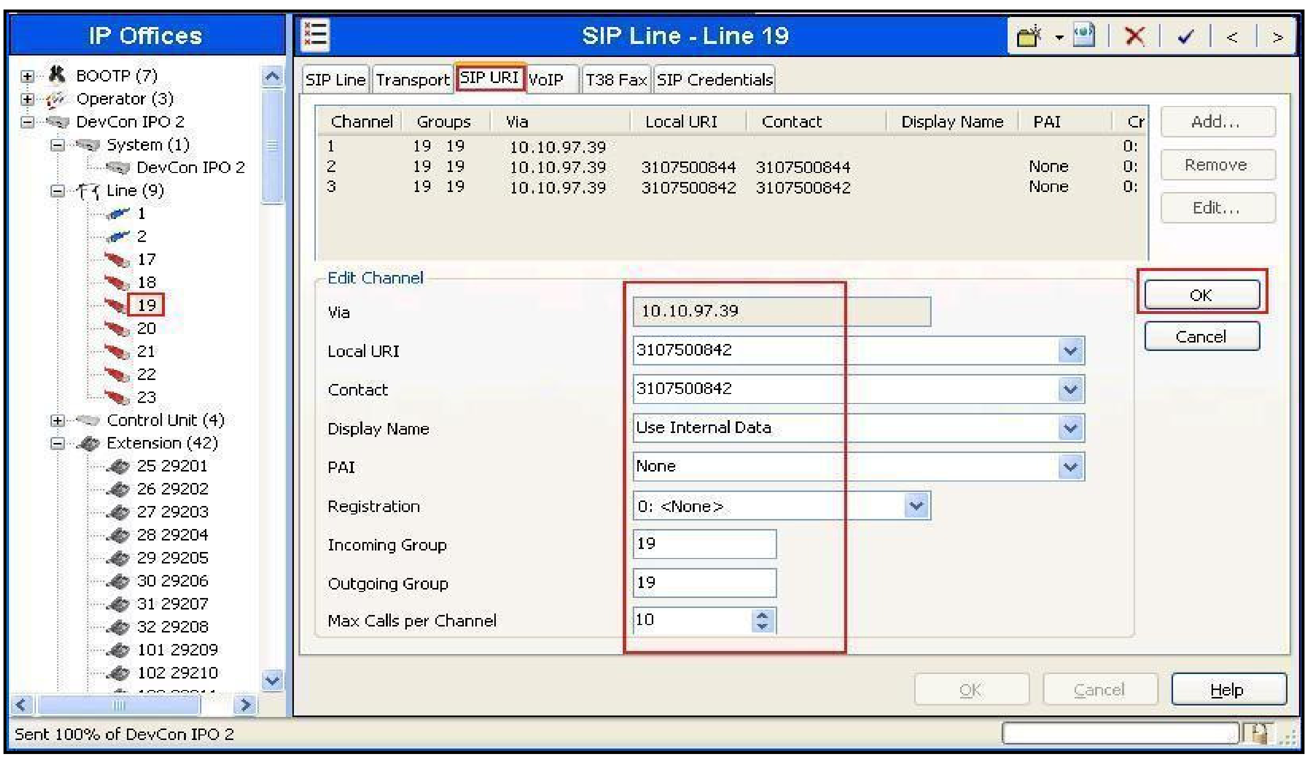
The **Channel 2,** as shown in the screenshot below, was configured with following parameters.

- Set the **Local URI** and **Contact** fields to pre-define DID number **3107500844** appropriately for **Channel 2** and **3107500842** for **Channel 3**.
- Associate **Incoming Group** and **Outgoing Group** to SIP Line 19.
- Set the **Max Calls per Channel** field to **10**.

- Other parameters retain default values.
- Click OK to commit.

SIP URI entry for **Channel 2 and Channel 3** are shown below respectively:



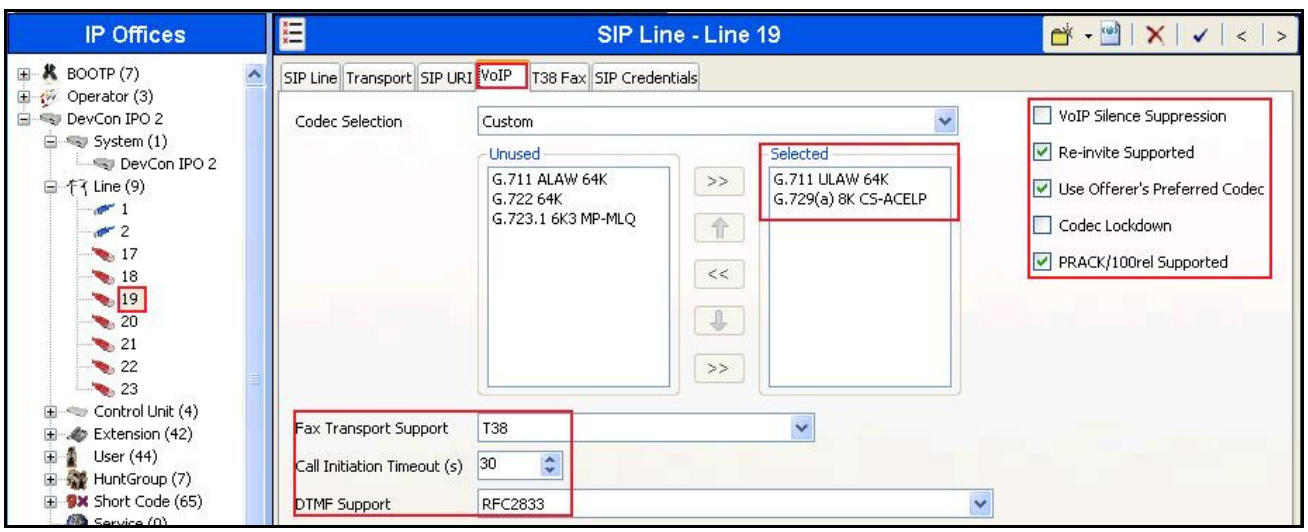


Click OK to commit then press Ctrl + S to save.

5.5.4 Administer VoIP Settings

Select the **VoIP** tab then set the Voice over Internet Protocol parameters of the SIP Line as following:

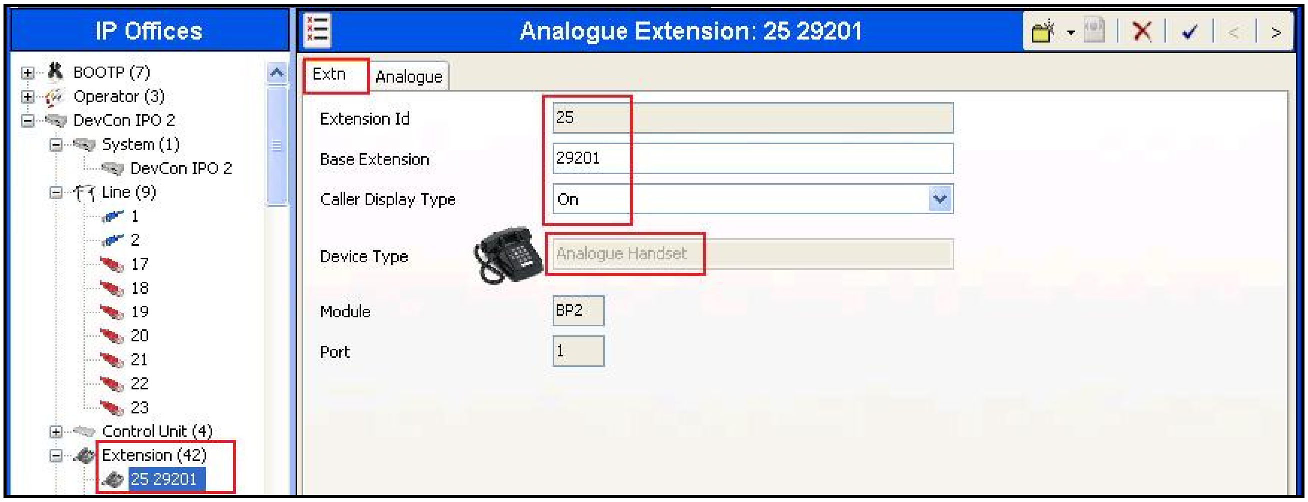
- The **Codec Selection** can be selected by choosing **System Default** from the pull-down menu to use the System Codecs as defined in **Section 5.3**. The codec order was configured as **G.711 ULAW 64K** and **G.729 (a) 8K CS-ACELP** which are supported by ICC. IP Office includes these codes in the right prioritized order in the Session Description Protocol (SDP) offer or answer defined for the RTP traffic.
- Set the **Fax Transport Support** to **T.38** from the pull-down menu.
- Set the **Call Initiation Timeout (s)** to **30** seconds to allow a long enough duration for a public call to be established over the SIP Trunk.
- Set the **DTMF Support** to **RFC2833** from the pull-down menu. This directs IP Office to send out-of-band DTMF tones using RTP events per RFC 2833.
- Uncheck the **VoIP Silence Suppression** box. By un-checking the **VoIP Silence Suppression** box, calls can be established with the G.729 codec but without silence suppression.
- Check the **Re-invite Supported** box.
- Check **Use Offerer's Preferred Codec** box.
- Uncheck **Codec Lockdown** box.
- Check the **PRACK/100rel** because ICC supported the "100rel" signaling as described in RFC 3262.
- Default values may be used for all other parameters.
- Click OK to commit (not shown) then press Ctrl + S to save.



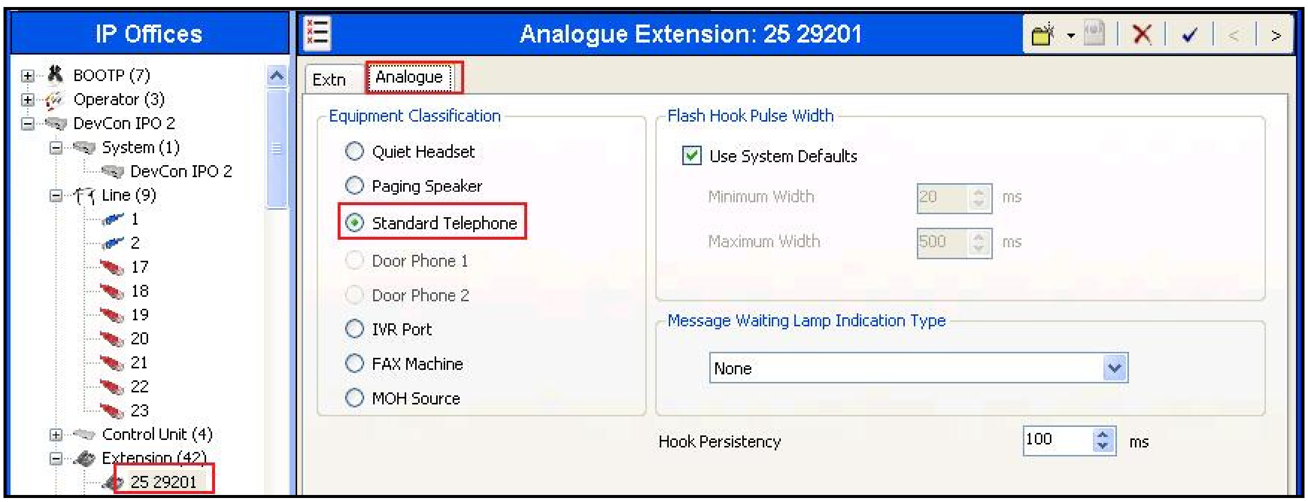
5.5.5 Administer T38 Fax Settings

Select the **T38 Fax** tab then check the **Use Default Values** to retain the default settings for T38 Fax (not shown).

Note: In this testing, the configuration for IP Office to work with ICC SIP trunk service in Fax T.38 mode, the analog extension (where the fax equipment is connected to) is being configured as follow: Navigate to **Extension** in the left Navigation Pane then select an Analog Extension, **25 29201.** On the **Extn** tab on the Details Pane confirm the settings are as shown.



Select **Analogue** tab on the Details Pane, confirm the settings are as shown.



5.6 Short Code

Short Codes were defined to route general outgoing calls and private outgoing calls to PSTN over the SIP Line, incoming calls from mobility extension to access FNE hosted on IP Office or incoming calls to retrieve voice message on IP Office VoiceMail Pro.

To create a short code, select **Short Code** in the left Navigation Pane then right-click and select **New** (not shown). On the **Short Code** tab in the Details Pane, configure the parameters for the new short code to be created.

The screenshot below shows the details of the Short Code **9N;** that was created for outgoing calls in the test configuration. The digit **9** was used as a prefix that IP Office user will dial to access to SIP Trunk for outgoing calls to PSTN.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, it is **9N;**. This short code will be invoked when the user dials **9** followed by any number.
- Set the **Feature** to **Dial**. This is the feature that the short code will invoke.
- Set the **Telephone Number** to **9N"@avaylab.com"**. This field is used to construct the "Request URI" and "To" headers of outgoing calls. The value **N** represents the number dialed by the user. The host part following the "@" is the enterprise SIP domain.
- Set the **Line Group ID** field to **19** which is the outgoing line group number defined on the **SIP URI** tab of the **SIP Line** in **Section 5.5.1**. This short code will use this line group when placing outgoing calls.
- Set **Locale** to **United State (US English)**.

For incoming calls from mobility extension to FNE features hosted by IP Office to provide **Dial Tone** functionality, Short Code **FNE00** was created. The **FNE00** was configured with the following parameters.

- In the **Code** field, enter the FNE feature code as **FNE00** for **Dial Tone**.
- Set the **Feature** field to **FNE Service**.
- Set the **Telephone Number** field to **00** for **FNE00**.
- Set the **Line Group ID** field to **0**.
- Retain default values for other fields.

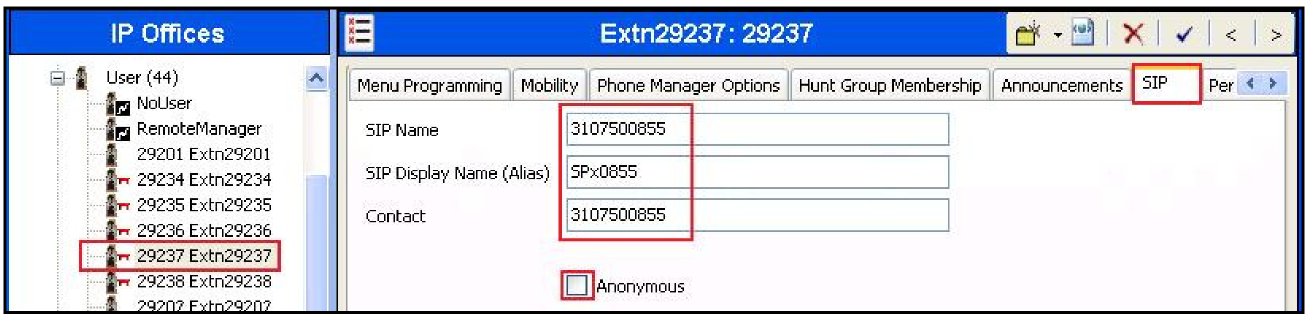
Following screenshots illustrate **FNE00** configuration.

When complete, click OK to commit (not shown) then press Ctrl + S to save.

5.7 User

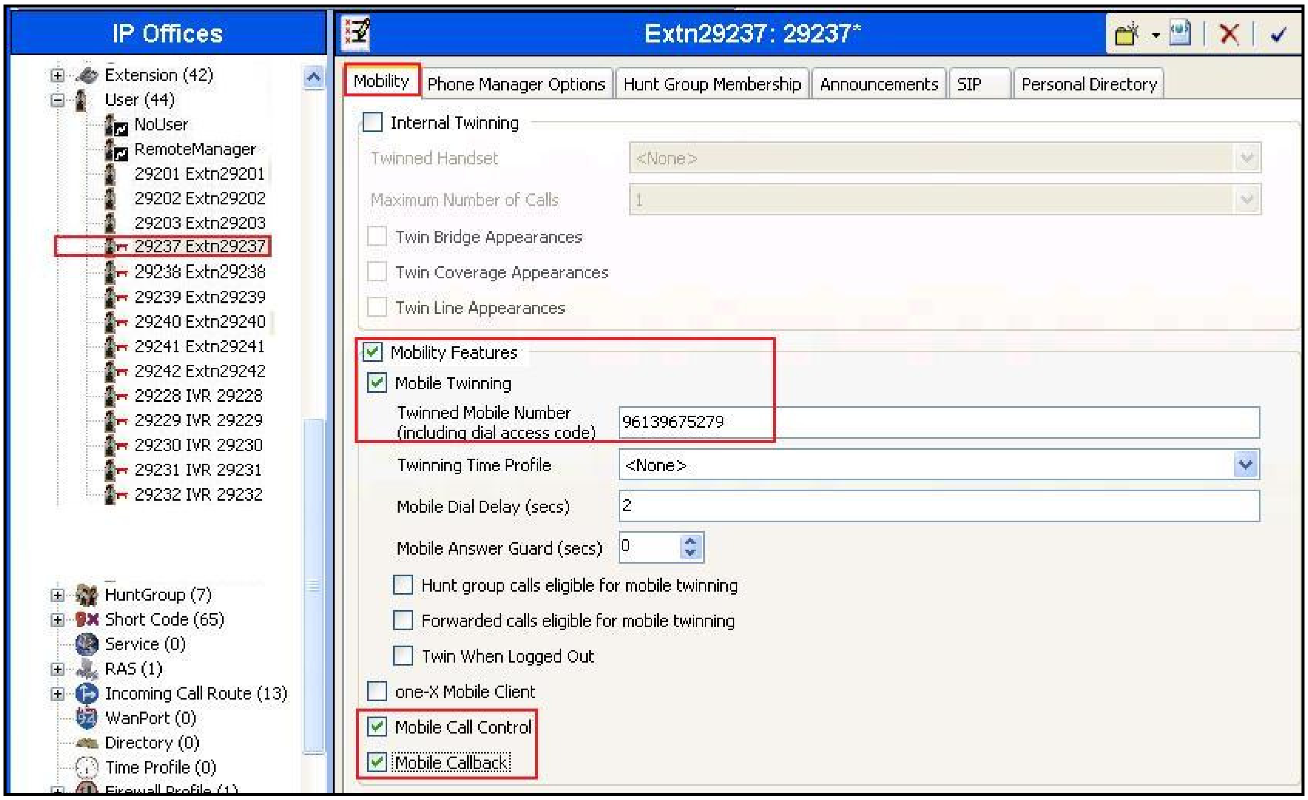
Configure the SIP parameters for each user that will be placing and receiving calls via the SIP Line as defined in **Section 5.5**. To configure these settings, first select **User** in the left Navigation Pane, and then select the name of the user to be modified. In the example below, with the user **Extn29237** selected, select the **SIP** tab in the Details Pane.

- The values entered for the **SIP Name** and **Contact** fields are used as the URI-User in the "From" header for outgoing calls. They also allow matching of URI-User for incoming calls without having to enter this number as an explicit SIP URI for the SIP Line (see **Section 5.5**). The **SIP Name** and **Contact** fields were set to one of the DID numbers assigned to the enterprise by ICC, e.g. **3107500855**.
- The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name, e.g. **SPx0855**. If all calls involving this user and a SIP Line should be considered private
- The **Anonymous** box may be checked to withhold the user information from the networks.



Mobile Twinning feature may be enabled on the user to allow incoming calls to simultaneously alert the desk phone and the mobile phone. The following screenshot shows the **Mobility** tab.

- The **Mobility Features** and **Mobile Twinning** boxes were checked.
- The **Twinned Mobile Number** was configured with the number to reach the twinned mobile telephone, in this case it was **91613XXX5279** including digit 9 as the dial access code and 1613XXX5279 as the mobility extension.
- Check **Mobile Call Control** to allow incoming call from mobility extension to access FNE00 (see **Section 5.6**).
- Other options can be set according to customer requirements.



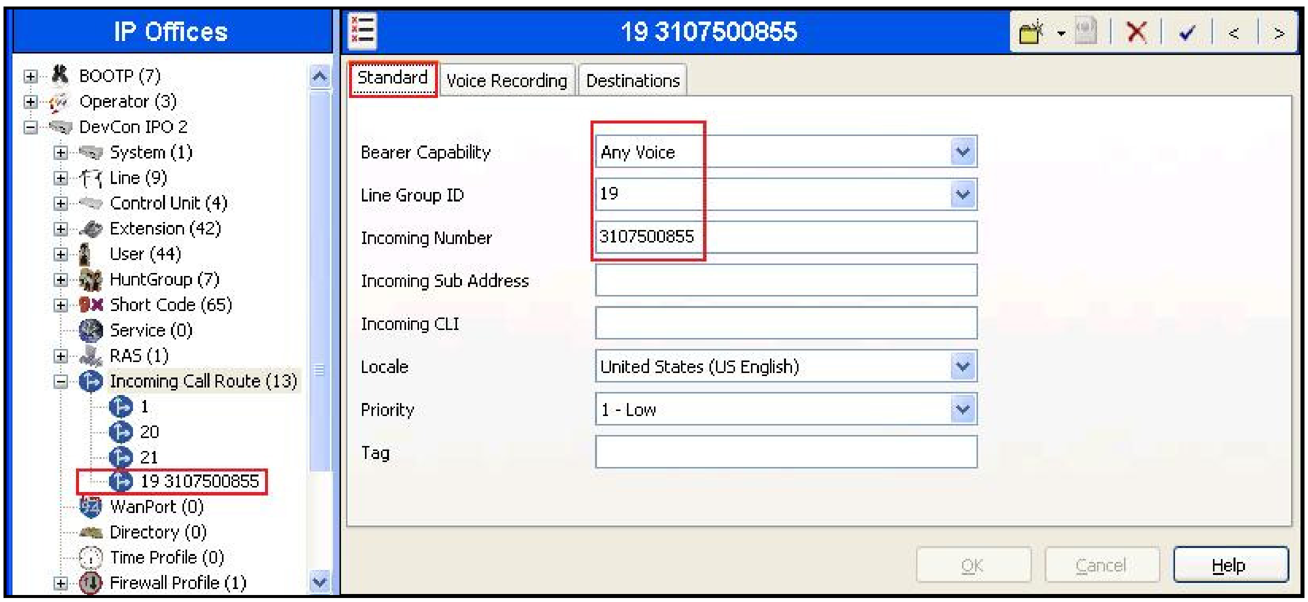
When complete, click OK to commit (not shown) then press Ctrl + S to save.

5.8 Incoming Call Route

An Incoming Call Route maps an incoming call on a specific SIP 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 on the **Incoming Call Route** in the left Navigation Pane and select **New** (not shown). On the **Standard** tab of the Details Pane, enter the following parameters.

- Set the **Bearer Capability** to **Any Voice**.
- Set the **Line Group ID** to SIP Line **19** as defined in **Section 5.5**.
- Set the **Incoming Number** to the DID number that associate to the internal extension.
- Set **Locale** to **United State (US English)**
- Default values can be used for all other fields.

The screenshot below shows Incoming Call Route **19 3107500855** configured to receive incoming call to DID number **3107500855** then alert local station **29237**.



On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to **3107500855** on SIP Line 19 are routed to extension **29237 Extn29237**.

Following screenshots show Incoming Call Routes to receive incoming calls on DID number **4169169608** that similarly configured to access **FNE00** and **Voicemail**. The **Destinations** were appropriately defined as FNE00 and VoiceMail. **Note**: FNE00 was entered manually by selecting **Destination** as **DialIn** (not shown) then input the appropriate FNE feature code.

When complete, click OK to commit (not shown) then press Ctrl + S to save.

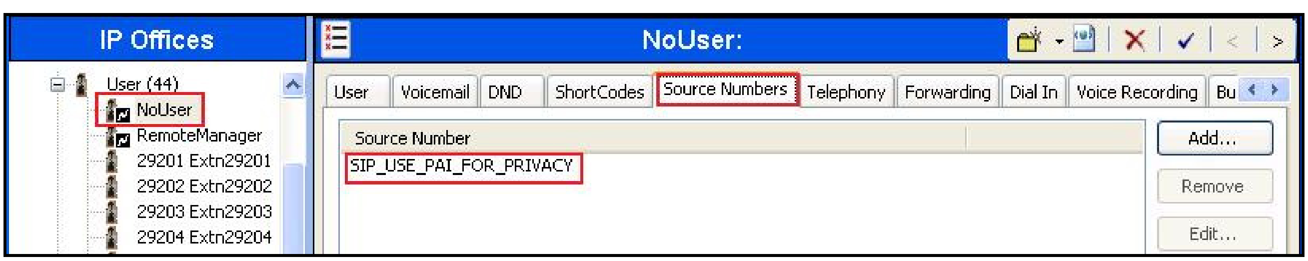
5.9 Privacy/Anonymous Calls

For outgoing calls with privacy (anonymous) enabled, IP Office will replace Calling Party Number in the "From" and "Contact" headers with "restricted" and "anonymous" respectively. IP Office can be configured to use the "P-Preferred-Identity" or "P-Asserted-Identity" header to pass the actual Calling Party information for authentication and billing purposes. For the compliance testing, the "P-Asserted-Identity" header was used.

To configure IP Office to use the "P-Asserted-Identity" header for private calls, navigate to **User → noUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button (not shown).

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

The **SIP_USE_PAI_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below.



When complete, click OK to commit (not shown) then press Ctrl + S to save.

5.10 Save Configuration

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

6. ICC SIP Trunking Service Configuration

ICC is responsible for the configuration of ICC SIP Trunking Service. ICC will provide the customer with necessary information to configure SIP Trunk for the Avaya IP Office solution. Thus, the configuration of ICC SIP trunking server will not be discussed in these Application Notes. The provided information from ICC includes:

- IP address of the ICC SIP proxy.
- DID numbers.
- Supported codecs.
- A customer specific SIP signaling reference.

The sample configuration between the enterprise and ICC for the compliance testing was a static configuration. There was no registration on the SIP Trunk implemented on either ICC or enterprise side.

7. Verification and Troubleshooting

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of useful troubleshooting tips that can be used to troubleshoot the solution.

7.1 Verification Steps

The following activities are made to each test scenario:

- Verify that endpoints at the enterprise site can place calls to PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
- Verify that endpoints at the enterprise site can receive calls from PSTN and that the call can remain active for more than 35 seconds.
- Verify that the user on the PSTN can end an active call by hanging up.
- Verify that an endpoint at the enterprise site can end an active call by hanging up.

7.2 Protocol Traces

The following SIP message headers are inspected using sniffer trace analysis tool:

- Request-URI: Verify the request number and SIP domain.
- From: Verify the display name and display number.
- To: Verify the display name and display number.
- P-Asserted-Identity: Verify the display name and display number.
- Privacy: Verify privacy masking with “user, id”.
- Diversion: Verify the display name and display number.

The following attributes in SIP message body are inspected using sniffer trace analysis tool:

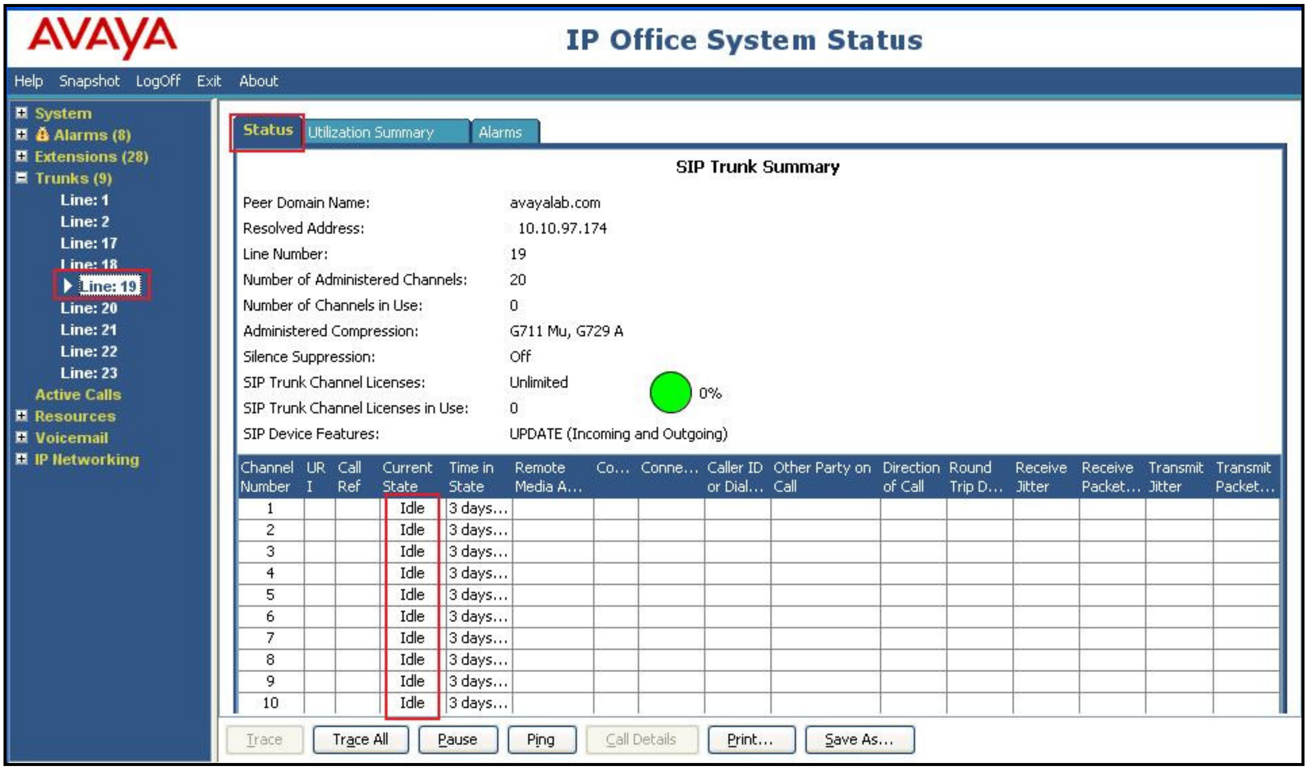
- Connection Information (c line): Verify IP addresses of near end and far end endpoints.
- Time Description (t line): Verify session timeout value of near end and far end endpoints.
- Media Description (m line): Verify audio port, codec, DTMF event description.
- Media Attribute (a line): Verify specific audio port, codec, ptime, send/ receive ability, DTMF event and fax attributes.

7.3 Troubleshooting

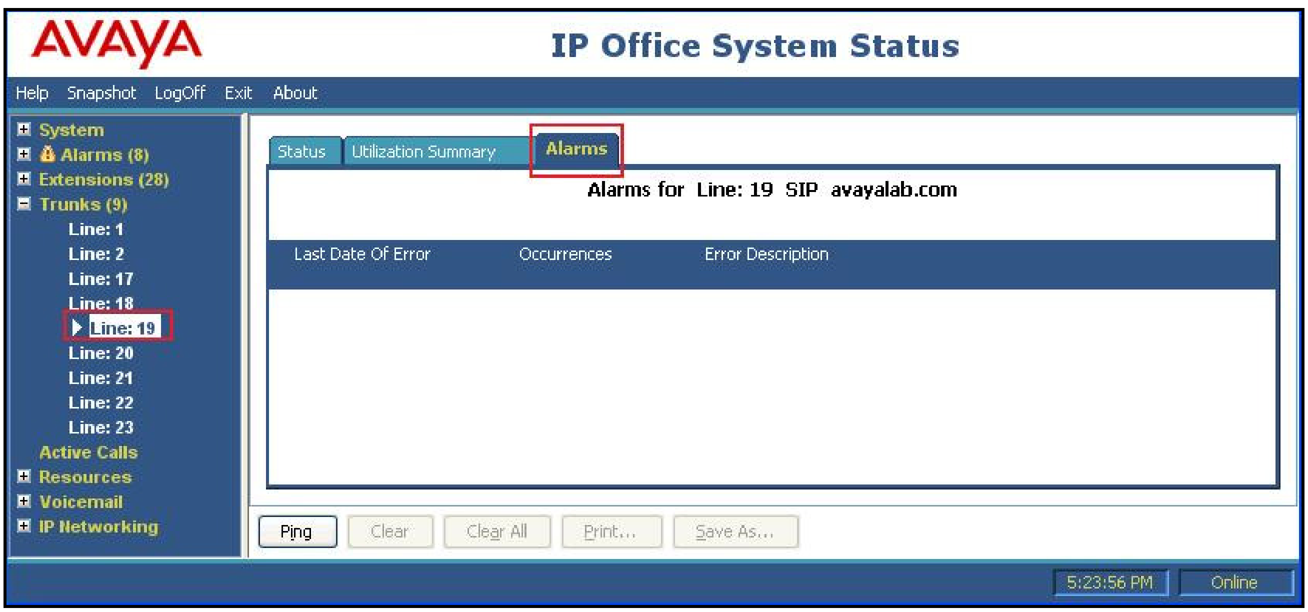
7.3.1 IP Office System Status

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 from **Start → Programs → IP Office → System Status** on the PC where IP Office Manager is installed. 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 (assuming no active calls at present time).



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



7.3.2 Sniffer Traces Analysis

Using a network sniffing tool e.g. Wireshark to monitor the SIP signaling between the enterprise and ICC. The sniffer traces are captured on the public internet.

Following screenshots show an example incoming call from ICC to the enterprise.

- Incoming INVITE request from ICC.

```
INVITE sip:3107500844@10.10.97.39:5060 SIP/2.0
Via: SIP/2.0/UDP 192.148.33.40:5060;branch=z9hG4bK02B982c5fc82193792c
From: <sip:6139675258@192.148.33.40>;tag=gK024a516b
To: <sip:3107500844@10.10.97.39>
Call-ID: 218295744_18373145@192.148.33.40
CSeq: 32238 INVITE
Max-Forwards: 16
Allow: INVITE,ACK,CANCEL,BYE,PRACK,UPDATE,OPTIONS,MESSAGE,PUBLISH
Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay,
multipart/mixed
Contact: <sip:6139675258@192.148.33.40:5060>
P-Preferred-Identity: <sip:6139675258@192.148.33.40:5060>
Supported: timer,100rel
Session-Expires: 1800
Min-SE: 90
Content-Length:  315
Content-Disposition: session; handling=required
Content-Type: application/sdp

v=0
o=Sonus_UAC 22055 30882 IN IP4 192.148.33.40
s=SIP Media Capabilities
c=IN IP4 192.148.122.53
t=0 0
m=audio 8092 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=ptime:20
a=rtcp:8093
```

- 200OK response from the enterprise.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.148.33.40:5060;branch=z9hG4bK02B982c5fc82193792c
From: <sip:6139675258@192.148.33.40>;tag=gK024a516b
To: <sip:3107500844@10.10.97.39>;tag=77983a77bde50628
Call-ID: 218295744_18373145@192.148.33.40
CSeq: 32238 INVITE
Contact: "SPx0842" <sip:3107500844@10.10.97.39:5060;transport=udp>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
Supported: timer,100rel
Server: IP Office 8.1 (67)
Min-SE: 1800
Require: timer
Session-Expires: 1800;refresher=uac
Content-Type: application/sdp
Content-Length: 203

v=0
o=UserA 4271051162 786533294 IN IP4 10.10.97.39
s=Session SDP
c=IN IP4 10.10.97.39
t=0 0
m=audio 49156 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Following screenshots show an example outgoing call from the enterprise to ICC.

- Outgoing INVITE request from the enterprise.

```
INVITE sip:6139675258@avayalab.com SIP/2.0
Via: SIP/2.0/TCP 10.10.97.39:5060;rport;branch=z9hG4bKbc1415bce61aa296cfe3e189f6a94795
From: "SPx0842" <sip:3107500844@avayalab.com>;tag=0280b655e6b71d0f
To: <sip:6139675258@avayalab.com>
Call-ID: 95fffcfab694e0328c4591a084dc4097
CSeq: 1430514418 INVITE
Contact: "SPx0842" <sip:3107500844@10.10.97.39:5060;transport=tcp>
Max-Forwards: 70
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
Content-Type: application/sdp
Supported: timer,100rel
User-Agent: IP Office 8.1 (67)
P-Asserted-Identity: "SPx0842" <sip:3107500844@10.10.97.39:5060>
Content-Length: 250

v=0
o=UserA 667427473 3811411377 IN IP4 10.10.97.39
s=Session SDP
c=IN IP4 10.10.97.39
t=0 0
m=audio 49154 RTP/AVP 18 0 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

- Incoming 200OK response from ICC.

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.97.39:5060;branch=z9hG4bKbc1415bce61aa296cfe3e189f6a94795;rport=5060
From: "SPx0842" <sip:3107500844@avayalab.com>;tag=0280b655e6b71d0f
To: <sip:6139675258@avayalab.com>;tag=gK049bd1dc
Call-ID: 95fffcfab694e0328c4591a084dc4097
CSeq: 1430514418 INVITE
Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay,
multipart/mixed
Contact: <sip:6139675258@192.148.33.40:5060;transport=tcp>
Allow: INVITE,ACK,CANCEL,BYE,PRACK,UPDATE,OPTIONS,MESSAGE,PUBLISH
Require: timer
Supported: timer
Session-Expires: 1800;refresher=uac
Content-Length:  267
Content-Disposition: session; handling=required
Content-Type: application/sdp

v=0
o=Sonus_UAC 4105 8855 IN IP4 192.148.33.40
s=SIP Media Capabilities
c=IN IP4 192.148.122.51
t=0 0
m=audio 16284 RTP/AVP 18 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=ptime:20
a=rtcp:16285
```

8. Conclusion

These Application Notes describe the configuration necessary to connect Avaya IP Office Release 8.1 and IntelePeer CoreCloud SIP Trunking Service.

All of the test cases have been executed. Despite the number of observations seen during testing as noted in **Section 2.2**, the test results met the objectives outlined in **Section 2.1**. The IntelePeer CoreCloud SIP Trunking Service is considered **compliant** with Avaya IP Office Release 8.1.

9. References

[1] *IP Office 8.1 IP500/IP500 V2 Installation*, Document Number 15-601042, Issue 27f, 04 March 2013.
[2] *IP Office 8.1 Manager FP1 10.1*, Document Number 15-601011, Issue 29t, 20 February 2013.
[3] *IP Office 8.1 Administering Voicemail Pro*, Document Number 15-601063, Issue 8b, 11 December 2012.
[4] *Administering Avaya Session Border Controller for Enterprise,* Release 6.2, Issue 2, March 2013.
[5] *Installing Avaya Session Border Controller for Enterprise,* Release 6.2, Issue 2, March 2013.
[6] *Upgrading Avaya Session Border Controller for Enterprise,* Release 6.2, Issue 2, March 2013.

Documentation for Avaya products may be found at http://support.avaya.com.

Product documentation for IntelePeer CoreCloud SIP Trunking Service is available from IntelePeer.

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.