Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Vitelity SIP Trunk Service with Avaya IP Office 9.1 Server Edition Solution - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider Vitelity and Avaya IP Office 9.1 Server Edition.

Vitelity SIP Trunk Service (Vitelity) provides PSTN access via a SIP trunk between the enterprise and the Vitelity network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results

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 Vitelity and Avaya IP Office Server Edition solution. In the sample configuration, the Avaya IP Office Server Edition solution consists of the Primary Server running the Avaya IP Office Server Edition Linux software Release 9.1, Avaya IP Office Server Edition Expansion System (500V2), and Avaya Communicator for Windows (SIP mode), Avaya H.323, Avaya SIP, digital and analog endpoints.

Vitelity referenced within these Application Notes is designed for business customers. The service enables local and long distance PSTN calling via standards-based SIP trunks as an alternative to legacy analog or digital trunks, without the need for additional TDM enterprise gateways and the associated maintenance costs.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office Server Edition and connecting to Vitelity.

This configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed in **Section 2.1**. Note: NAT devices added between Avaya IP Office Server Edition and the Vitelity network should be transparent to SIP signaling.

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

A simulated enterprise site with Avaya IP Office Server Edition was connected to Vitelity. To verify SIP trunking interoperability, following features and functionality were exercised during the interoperability compliance test:

- Incoming PSTN calls to various phone types. Phone types included Avaya H.323, Avaya SIP, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types. Phone types included Avaya H.323, Avaya SIP, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls from/to the Avaya Communicator for Windows (SIP mode).
- Inbound and outbound long hold time call stability.
- Various call types including: local, long distance, international call, inbound toll-free, outbound toll-free, 411 Directory Assistant and 911 services.

- Codec G.729A and G.711MU.
- Caller number/ID presentation.
- Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls.
- DTMF transmission using RFC 2833.
- Voicemail navigation for inbound and outbound calls.
- Telephony features such as hold and resume, transfer, and conference.
- Fax G.711 pass-through.
- Off-net call forwarding.
- Twinning to mobile phones on inbound calls.

Item not supported include the following:

- SIP REFER in Call Transfer (transferring calls with the PSTN back to the PSTN).
- SIP OPTIONS sent by Vitelity.
- Outbound calls to Operator Assisted Call.
- Faxing using T.38.
- The setting of G.729A as the first priority codec on Vitelity system. (Vitelity always uses codec G.711MU as the first priority codec).

2.2. Test Results

Interoperability testing of Vitelity was completed with successful results for all test cases with the exception of following limitations:

- **Blind Call Transfer to PSTN using Avaya 1140E SIP phone did not complete until transferee picked up the call** - PSTN phone called to Avaya 1140E SIP phone, then Avaya 1140E SIP phone answered the call and performed blind transfer to another PSTN endpoint. The expected behavior of Avaya 1140E SIP phone is after the transfer is completed, the phone should display “transfer completed”. But in this case, after user pressed “transfer” button and answered question of “Consultative transfer with party ?” with “No”, which implied the blind transfer, the transferee PSTN phone rang and the Avaya 1140E SIP phone was still displayed “transferring” until the transferee PSTN phone answered the call. The work around was to hang up the Avaya 1140E SIP phone. This is a known limitation on Avaya 1140E SIP phone. There is no user impact. Transfer is still completed with two-way audio.

- **The call was dropped between PSTN phones if the Avaya IP phone disconnected the conference** – Call scenario was when Avaya IP phone hosted the conference among PSTN phones. Then, the Avaya IP phone disconnected the conference. The call was dropped between PSTN phones. The issue is under investigation.

2.3. Support

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

For technical support on the Vitelity, please contact customer service at 1-888-898-4835 or visit: http://www.vitelity.com/

3. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to Vitelity through the public IP network. For confidentiality and privacy purposes, actual public IP addresses used in this testing have been masked out and replaced with fictitious IP addresses throughout the document.

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

- IP Office Server Edition Primary Server
- IP Office Server Edition Expansion System (500V2)
- Avaya 9600 Series IP Deskphones (H.323)
- Avaya 11x0 Series IP Deskphones (SIP)
- Avaya 1408 Digital phones
- Avaya Analog phones
- Avaya Communicator for Windows (SIP)

The Primary Server consists of a HP ProLiant DL360 server, running the Avaya IP Office Server Edition Linux software Release 9.1. The server is the only mandatory component required to support SIP trunking and IP endpoints. Avaya Voicemail Pro runs as a service on the Primary Server. The LAN1 port of the Primary Server (Eth0) is connected to the enterprise LAN. The LAN2 port (Eth1) was not used during the compliance test.

The optional Expansion System (500V2) is used for the support of digital, analog and additional IP stations. It consists of an Avaya IP Office 500v2 with the MOD DGTL STA16 expansion module which provides connections for 16 digital stations to the PSTN, and the extension PHONE 8 card which provides connections for 8 analog stations to the PSTN as well as 64-channel VCM (Voice Compression Module) for supporting VoIP codecs. The LAN1 port of the Avaya IP Office IP500V2 is connected to the enterprise LAN. The LAN2 port was not used.

A separate Windows 7 Enterprise PC runs Avaya IP Office Server Edition Manager to configure and administer Avaya IP Office system.

Mobility Twinning is configured for some of the Avaya IP Office Server Edition users so that calls to these user’s phones will also ring and can be answered at configured mobile phones.

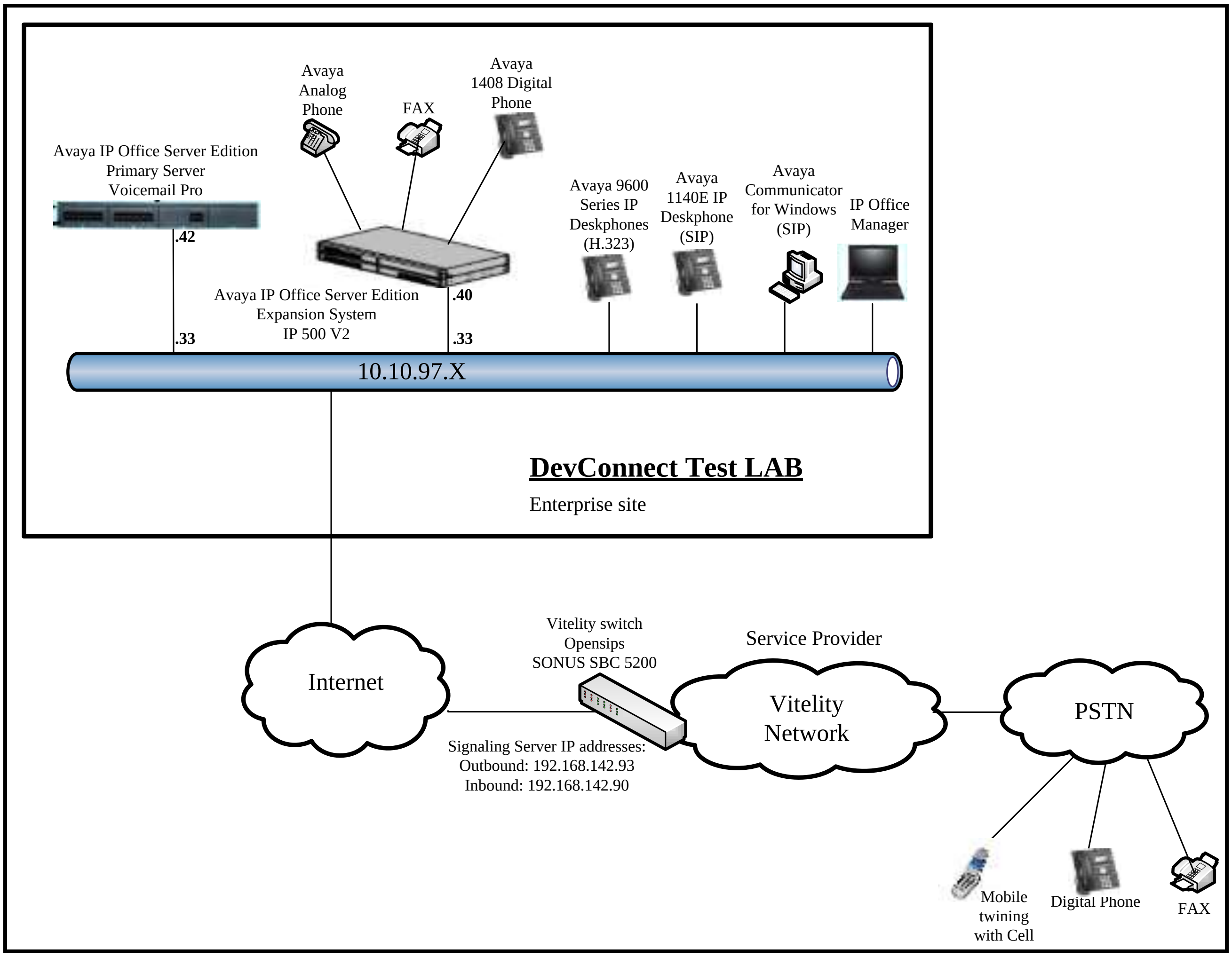


Figure 1: Test Configuration for Avaya IP Office Server Edition with Vitelity SIP Trunk Service

Inbound calls from the service provider via SIP trunk arrive to Server Edition Primary Server, where Incoming Call Routes are checked to determine the call destination. In the event that the destination of the incoming call is an endpoint in the Expansion System (500V2), the call is sent via the Small Community Network (SCN) H.323 trunk to the expansion IP500V2 for routing to the final endpoint. This SCN H.323 trunk is automatically created during the initial process of addition of the Expansion System to the IP Office Server Edition solution.

Similarly, outbound calls from the enterprise to the PSTN are routed via the SIP trunk to the Vitelity network. Calls originated from extensions registered to the Primary Server are routed directly to Vitelity, while calls originated from extensions on the Expansion System are sent to the Primary Server via SCN H.323 trunk, before being routed to Vitelity via the SIP trunk.

For the purposes of the compliance test, Avaya IP Office Server Edition users dialed a short code of 6 + N digits to send digits across the SIP trunk to Vitelity. The short code of 6 was stripped off by

Avaya IP Office Server Edition but the remaining N digits were sent unaltered to Vitelity. For calls within the North American Numbering Plan (NANP), the user would dial 11 (1 + 10) digits. Thus, for these NANP calls, Avaya IP Office Server Edition would send 11 digits in the Request URI and the To header of an outbound SIP INVITE request. It was configured to send 10 digits in the From field. For inbound calls, Vitelity sent 10 digits in the Request URI and the To header of an inbound SIP INVITE request.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and Avaya IP Office Server Edition, such as, a session border controller or a 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 Server Edition 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:

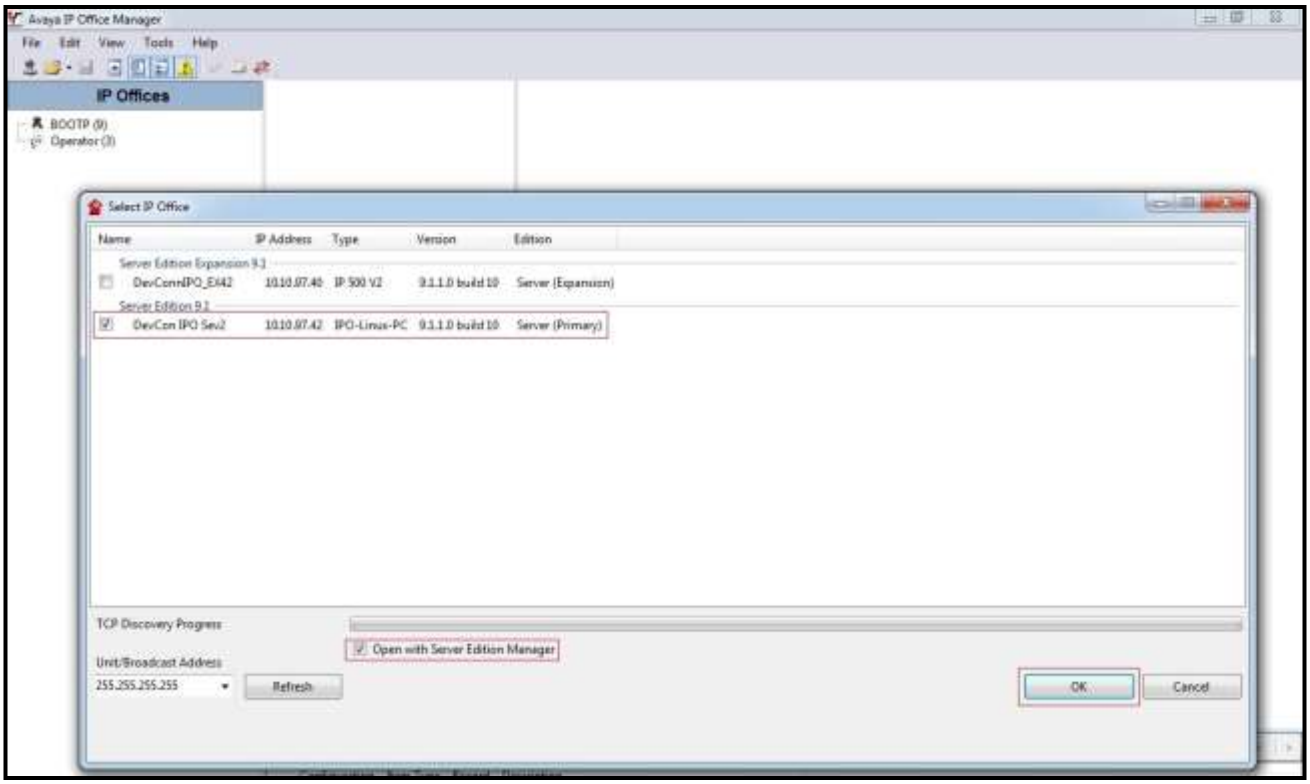
Avaya Telephony Components	
Equipment	**Release**
Avaya IP Office Server Edition solution • Primary Server HP ProLiant DL360 G7 – IPO-Linux-PC • Voicemail Pro • Expansion System (V2) – IP 500 V2 • Avaya IP Office Analogue PHONE 8 • Avaya IP Office VCM64/PRID U • Avaya IP Office DIG DCPx16 V2	9.1.100.10 9.1.100.10 9.1.100.3 9.1.100.10 9.1.100.10 9.1.100.10 9.1.100.10
Avaya IP Office Server Edition Manager	9.1.100.10
Avaya 1140E SIP	04.04.18.00
Avaya IP 9640G	S3.2
Avaya IP 9630	S3.2
Avaya Communicator for Windows (SIP)	2.0.3.30
Avaya Digital Telephone (1408D)	R40
Avaya Symphony 2000 Analog Telephone	N/A
HP Officejet 4500 (fax)	N/A
Vitelity Components	
Equipment	**Release**
OpenSIPS	1.11.4

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition without T.38 Fax Service.

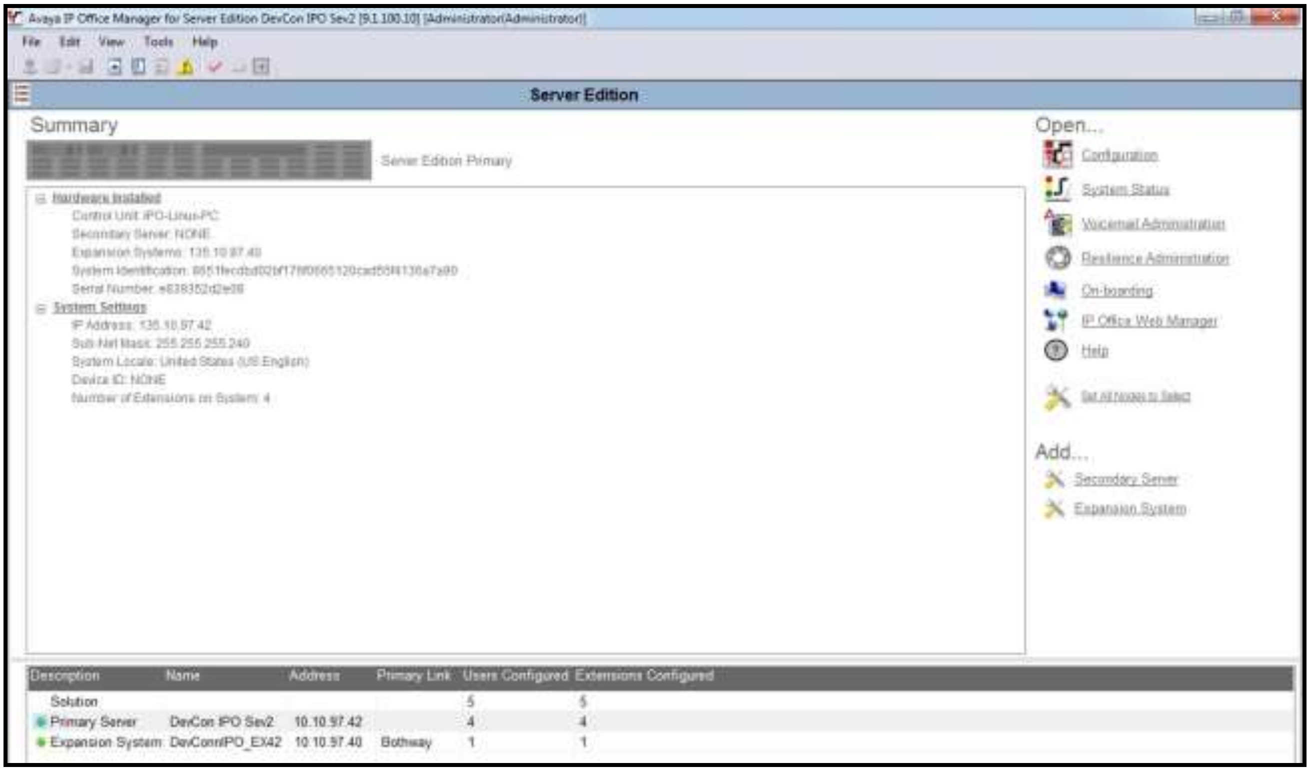
5. Configure Avaya IP Office Server Edition Solution

This section describes the Avaya IP Office Server Edition solution configuration necessary to support connectivity to the Vitelity SIP Trunk Service. It is assumed that the initial installation and provisioning of the Server Edition Primary Server and Expansion System has been previously completed and therefore is not covered in these Application Notes. For information on these installation tasks refer to **[3]** in the Additional References **section 9**.

This section describes the Avaya IP Office Server Edition configuration to support connectivity to Vitelity. Avaya IP Office Server Edition is configured through the Avaya IP Office Server Edition Manager PC application. From a PC running the Avaya IP Office Server Edition Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office Server Edition system from the pop-up window, making sure the box for **Open with Server Edition Manager** is checked. Log in using appropriate credentials.



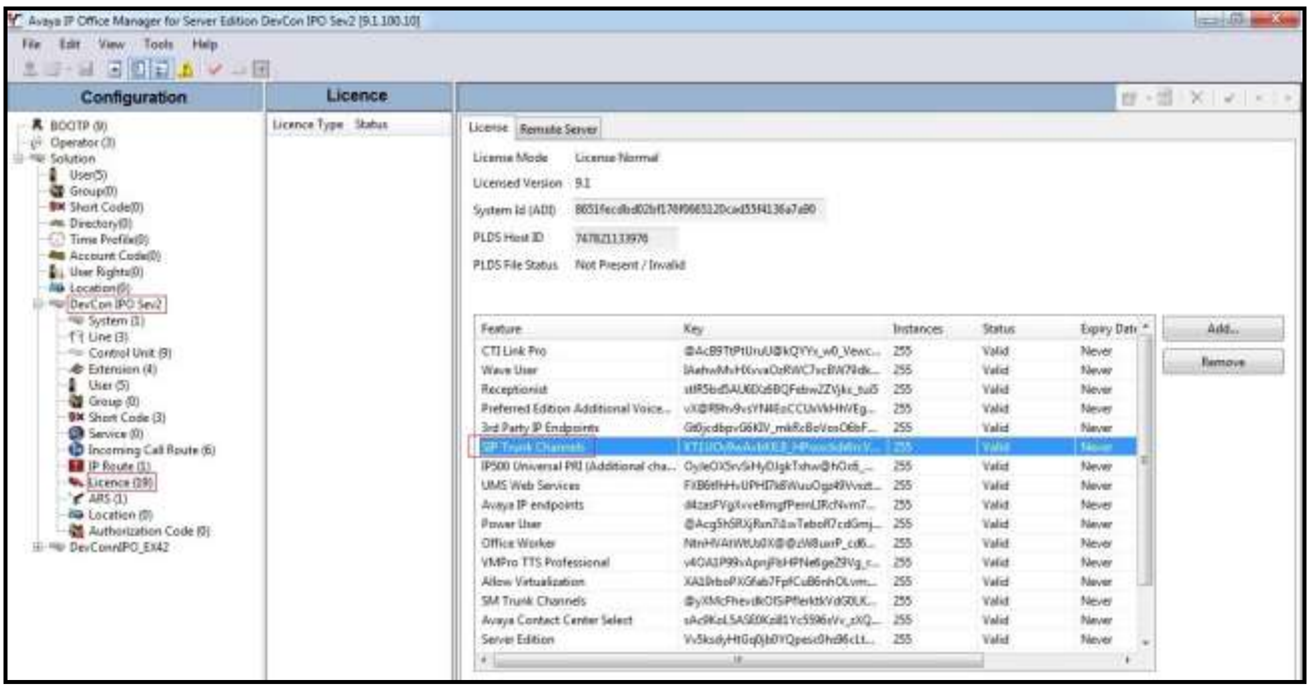
The appearance of the IP Office Server Edition Manager can be customized using the **View** menu. In the screens presented in this section, it includes the system inventory of the servers and links for administration and configuration tasks.



5.1. Licensing

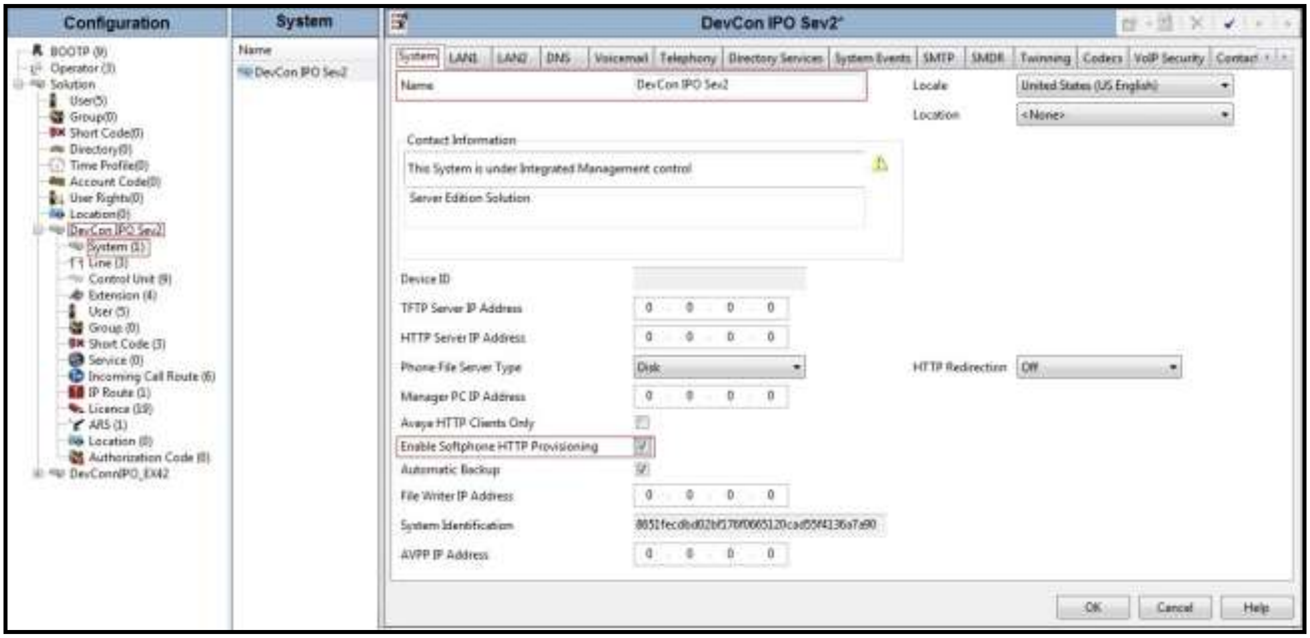
The configuration and features described in these Application Notes require the IP Office Server Edition system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

Licenses for an IP Office Server Edition solution are based on a combination of centralized licensing done through the IP Office Server Edition Primary Server, and server specific licenses that are entered into the configuration of the system requiring the feature. SIP Trunk Channels are centralized licenses, and they are entered into the configuration of the Primary Server. Note that when centralized licenses are used to enable features on other systems, such as SIP trunk channels, the Primary Server allocates those licenses to the other systems only after it has met its own license needs. To verify that there is a SIP Trunk Channels license with sufficient capacity, select License under Primary on the Navigation pane and SIP Trunk Channels in the Group pane. Confirm that there is a valid license with sufficient "Instances" (trunk channels) in the Details pane. Note that the actual License Key in the screen below was edited for security purposes.



5.2. System Tab

Navigate to **System (1)** under the **DevCon IPO Sev2** on the left pane and select the **System** tab in the Details pane. The Name field can be used to enter a descriptive name for the system. In the reference configuration, **DevCon IPO Sev2** was used as the name in the Primary Server. Make sure to check the **Enable Softphone HTTP Provisioning** box to enable the support of Avaya IP Office Softphone.



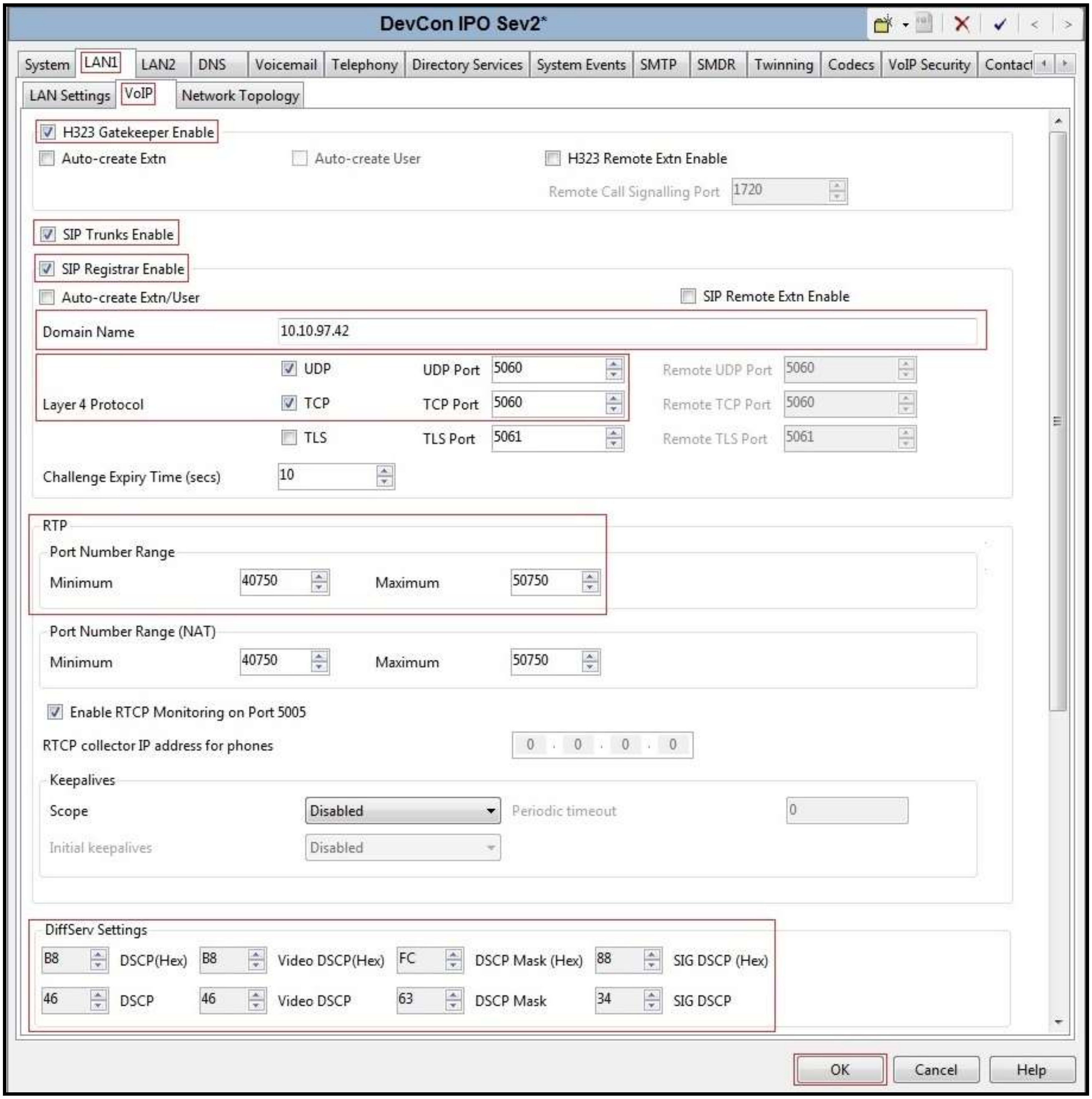
5.3. LAN1 Settings

In the sample configuration, LAN1 is used to connect both the Primary Server and the Expansion System to the enterprise network. The **DevCon IPO Sev2** was used as the Primary Server name and **DevConnIPO_EX42** was used as the Expansion System name.

To configure the LAN1 settings on the Primary Server, complete the following steps. Navigate to **DevCon IPO Sev2 → System (1)** in the Navigation and Group Panes and then navigate to the **LAN1 → LAN Settings** tab in the Details Pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office Server Edition LAN1 port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements. Click **OK** to submit the change.

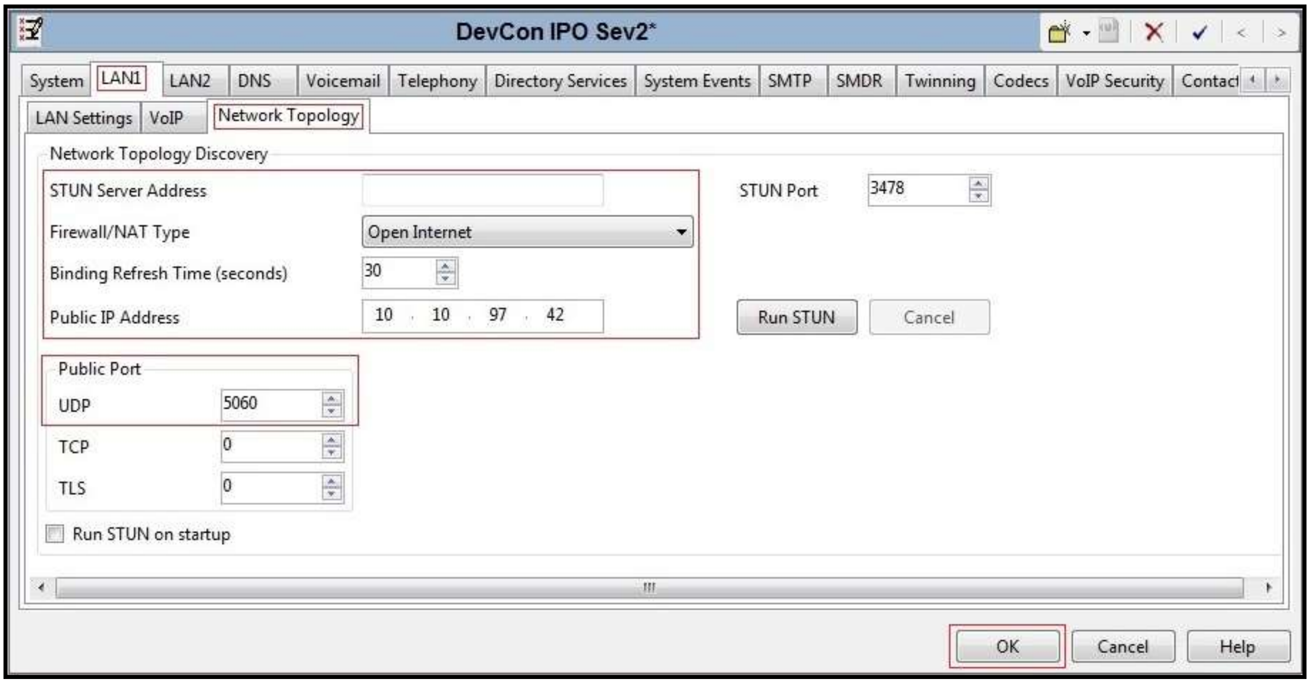
The **VoIP** tab as shown in the screenshot below was configured with following settings.

- Check the **H323 Gatekeeper Enable** to allow Avaya IP Deskphones/Softphones using the H.323 protocol to register.
- Check the **SIP Trunks Enable** to enable the configuration of SIP Trunk connecting to Vitelity.
- Check the **SIP Registrar Enable** to allow Avaya IP Deskphones/Softphones to register using the SIP protocol.
- Input **Domain Name** as **10.10.97.42**.
- The **Layer 4 Protocol** use **UDP** with **UDP Port** as **5060**, and **TCP** with **TCP Port** as **5060**.
- 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.
- All other parameters should be set according to customer requirements.
- Click **OK** to submit the changes.

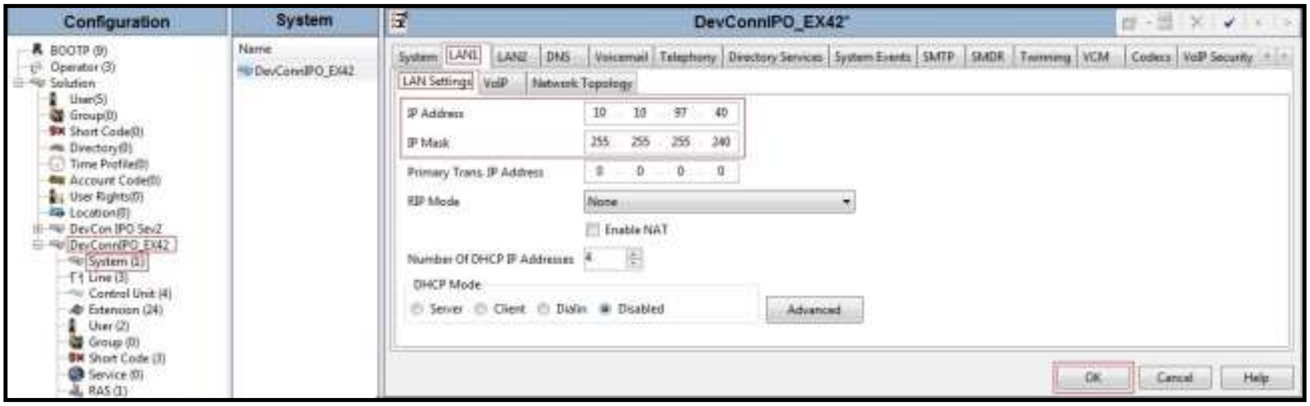

DevCon IPO Sev2*
System
LAN1
LAN2
DNS
Voicemail
Telephony
Directory Services
System Events
SMTP
SMDR
Twinning
Codecs
VoIP Security
Contact
LAN Settings
VoIP
Network Topology
H323 Gatekeeper Enable
Auto-create Extn
Auto-create User
H323 Remote Extn Enable
Remote Call Signalling Port
1720
SIP Trunks Enable
SIP Registrar Enable
Auto-create Extn/User
SIP Remote Extn Enable
Domain Name
10.10.97.42
Layer 4 Protocol
UDP
UDP Port
5060
Remote UDP Port
5060
TCP
TCP Port
5060
Remote TCP Port
5060
TLS
TLS Port
5061
Remote TLS Port
5061
Challenge Expiry Time (secs)
10
RTP
Port Number Range
Minimum
40750
Maximum
50750
Port Number Range (NAT)
Minimum
40750
Maximum
50750
Enable RTCP Monitoring on Port 5005
RTCP collector IP address for phones
0 . 0 . 0 . 0
Keepalives
Scope
Disabled
Periodic timeout
0
Initial keepalives
Disabled
DiffServ Settings
B8
DSCP(Hex)
B8
Video DSCP(Hex)
FC
DSCP Mask (Hex)
88
SIG DSCP (Hex)
46
DSCP
46
Video DSCP
63
DSCP Mask
34
SIG DSCP
OK
Cancel
Help

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

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to **Open Internet**. With this configuration, STUN Server will not be used.
- Set the **Binding Refresh Time (seconds)** to **30**. This value is used as one input to determine the frequency at which Avaya IP Office Server Edition will send SIP OPTIONS messages to the service provider.
- Set **Public IP Address** to the IP address of the Avaya IP Office Server Edition LAN1 port.
- Set **Public Port** for **UDP** as **5060**.
- All other parameters should be set according to customer requirements.
- Click **OK** to submit the changes.



To configure the LAN1 settings tab for the Expansion System, navigate to **DevConnIPO_EX42 → System (1)** in the Navigation and Group Panes and then navigate to the **LAN1 → LAN Settings** tab in the Details Pane. The IP Address and IP Mask fields should be populated with the values assigned during the Expansion System initial installation process. Verify the configuration or modify the values if needed. While DHCP was disabled during the compliance test, this parameter should be set according to customer requirements. Other settings were left at their default values. Click **OK** to submit the change.

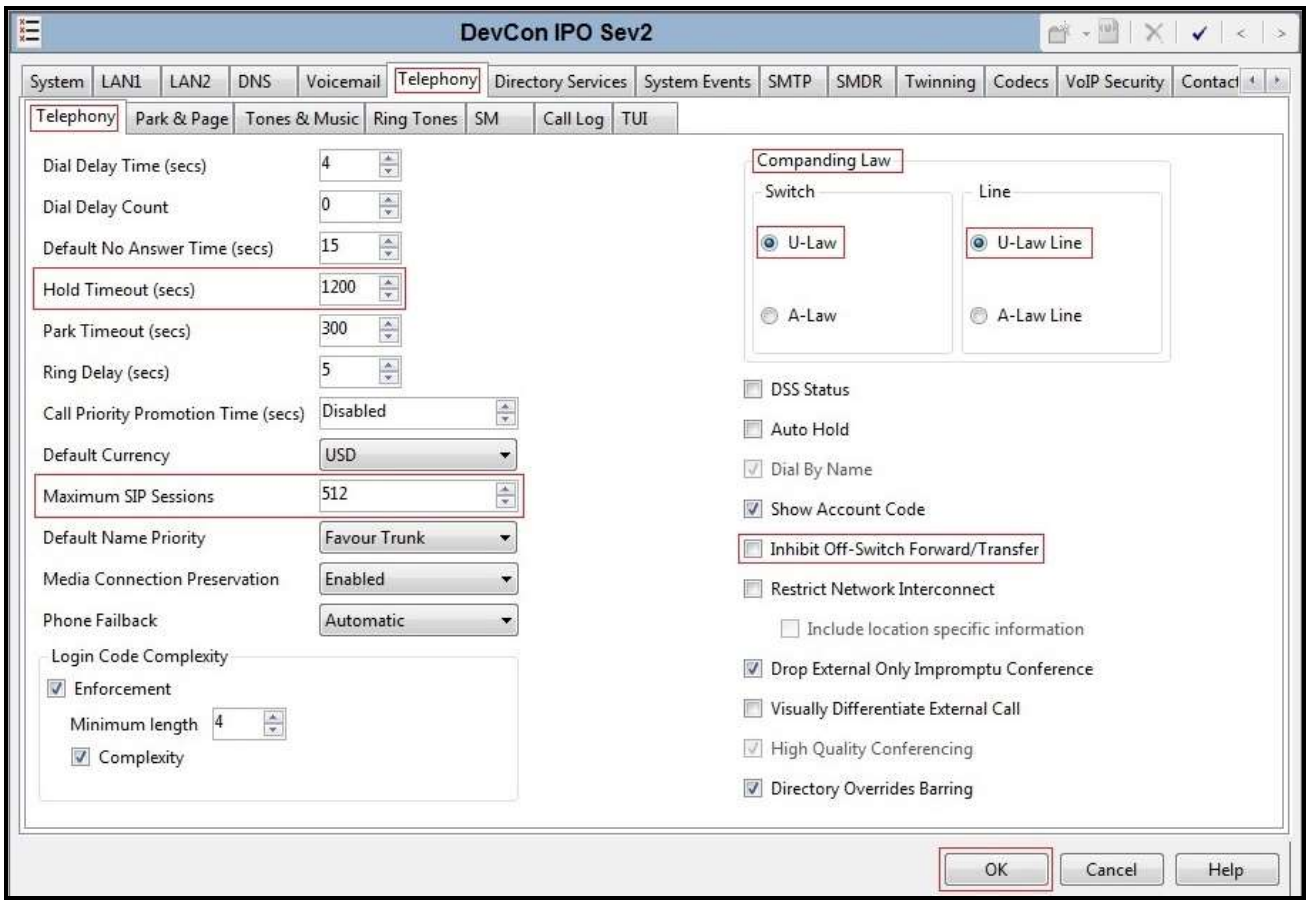


The **VoIP** tab for LAN1 in the Expansion System can be configured using the same values previously described for the **VoIP** tab in the Primary Server. The parameters in **Network Topology** tab can be left in the default settings for the Expansion System.

5.4. System Telephony Settings

Navigate to **DevCon IPO Sev2 → System (1)** in the Navigation and Group Panes (not shown) and then navigate to the **Telephony → Telephony** tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location. For North America, **U-Law** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfers to the PSTN via the service provider across the SIP trunk. Set **Hold Timeout (secs)** to **1200**.

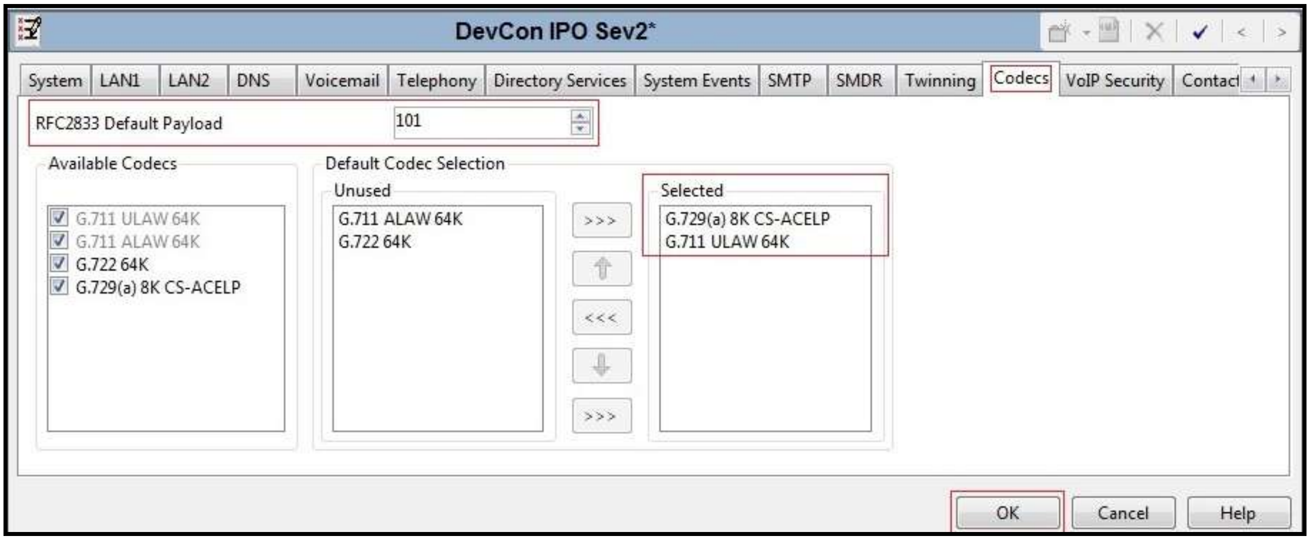
The **Maximum SIP Sessions** field appears only in Server Edition systems. On Server Edition systems, the Maximum SIP Sessions value must match the total number of SIP set and trunks calls that can occur at the same time. The Maximum SIP Sessions setting determines the number of SIP Trunk Channel licenses reserved for concurrent sessions on any SIP trunks provided by the server. Those licenses are reserved from the pool of SIP Trunk Channel licenses in the configuration of the Primary Server shown on **Section 5.1**. In the compliance test, 512 sessions were reserved on the Primary Server. Defaults were used for all other settings. Click **OK** to submit the changes.



Navigate to **DevConnIPO_EX42 → System (1)** and repeat the steps above to configure the **Telephony** settings for the Expansion System. Since the SIP trunk will be terminated on the Primary Server, it was not necessary to enter a value in the Maximum SIP Sessions field in this case, and the default value of 0 was used (not shown).

5.5. System Codec Settings

Navigate to **DevCon IPO Sev2 → System (1)** in the Navigation and Group Panes (not shown) and then navigate to the **Codecs** tab in the Details Pane. Choose the **RFC2833 Default Payload** as IP Office Server Edition default of **101**. Select codecs **G.729(a) 8K CS-ACELP**, and **G.711 ULAW 64K** that Vitelity supports. Click **OK** to submit the changes.



5.6. Twinning Calling Party Settings

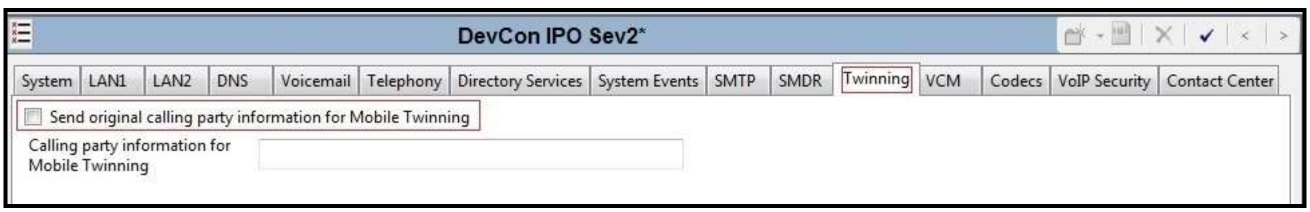
When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. These parameters only affect twinning and do not impact the messaging or operation of other redirected calls such as forwarded calls. The first parameter is the **Send original calling party information for Mobile Twinning** box on the **Twinning** tab, as shown below. The second parameter is the **Send Caller ID** parameter on the **SIP Line** form (shown in **Section 5.7.2**).

If **Send original calling party information for Mobile Twinning** on the **Twinning** tab is checked, the setting of the second parameter is ignored and Avaya IP Office Server Edition will send the following in the SIP From Header:

- On calls from an internal extension to a twinned phone, Avaya IP Office Server Edition will send the calling party number of the originating extension.
- On calls from the PSTN to a twinned phone, Avaya IP Office Server Edition will send the calling party number of the host phone associated with the twinned destination (instead of the number of the originating caller).

If this option is unchecked, the value sent in the SIP From header is determined by the setting of the second parameter mentioned above.

- For the compliance test, the **Send original calling party information for Mobile Twinning** box in the **DevConn IPO Sev2 → System (1) →Twinning** tab was unchecked. The value sent in the SIP From header is determined by the setting of the **Send Caller ID** parameter on the **SIP Line** form.



5.7. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office Server Edition and Vitelity SIP Trunk service. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Server Edition Manager to create a SIP Line. Follow the steps in **Section 5.7.1** to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses.
- SIP Credentials (if applicable).
- SIP URI entries.
- Setting of the Use Network Topology Info field on the Transport tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Section 5.7.2.**

Also, the following SIP Line settings are not supported on Basic Edition:

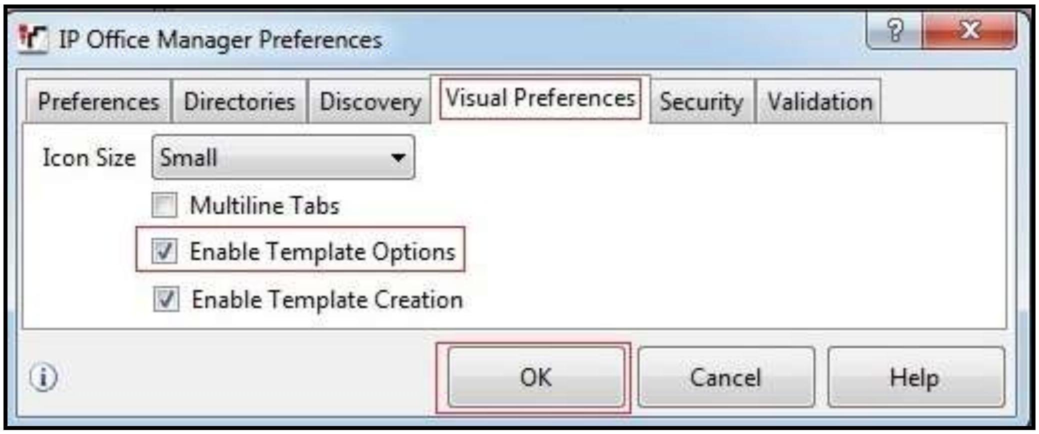
- SIP Line – Originator number for forwarded and twinning calls.
- Transport – Second Explicit DNS Server.
- SIP Credentials – Registration Required.
- SIP Advanced Engineering.

Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Section 5.7.2.**
For the compliance test, SIP Line 17 was used as trunks for outgoing calls and SIP Line 18 was used as trunks for incoming calls.

5.7.1. Create SIP Line from Template.

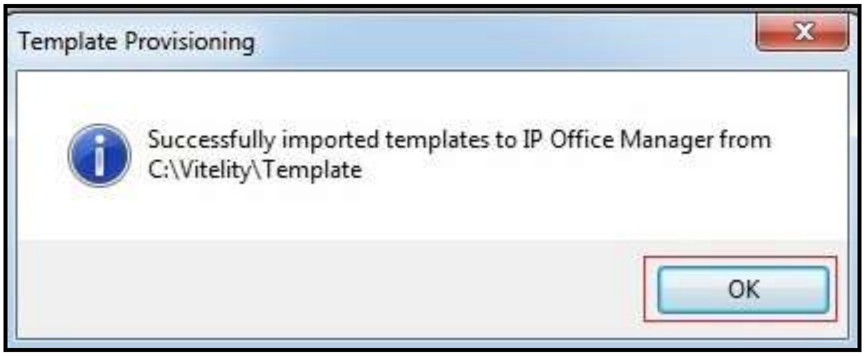
1. Copy the template files to the computer where IP Office Server Edition Manager is installed. Rename the template files to **AF_Vitelity_17_SIPTrunk.xml** (for SIP Line 17) and **AF_Vitelity_18_SIPTrunk.xml** (for SIP Line 18). The file names are important in locating the proper template file in **Step 5**.

2. Verify that template options are enabled in IP Office Server Edition Manager. In IP Office Server Edition Manager, navigate to **File → Preferences**. In the IP Office Server Edition Manager Preferences window that appears, select the **Visual Preferences** tab. Verify that the box is checked next to **Enable Template Options**. Click **OK** to submit the changes.



3. Import the template into IP Office Server Edition Manager. From IP Office Server Edition Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Server Edition Manager pull-down menus in **Step 5**.

In the pop-up window (not shown) that appears, select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window below will appear stating success (or failure). Then click **OK** to continue. If preferred, this step may be skipped if the template file is copied directly to the IP Office template directory.



4. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New SIP Trunk from Template**.

5. In the subsequent Template Type Selection pop-up window, check **Display All** and select **AF_Vitelity_17_SIPTrunk** from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**AF_Vitelity_17_SIPTrunk.xml**) created in **Step 1**. Click **Create new SIP Trunk** to finish creating the SIP trunk.

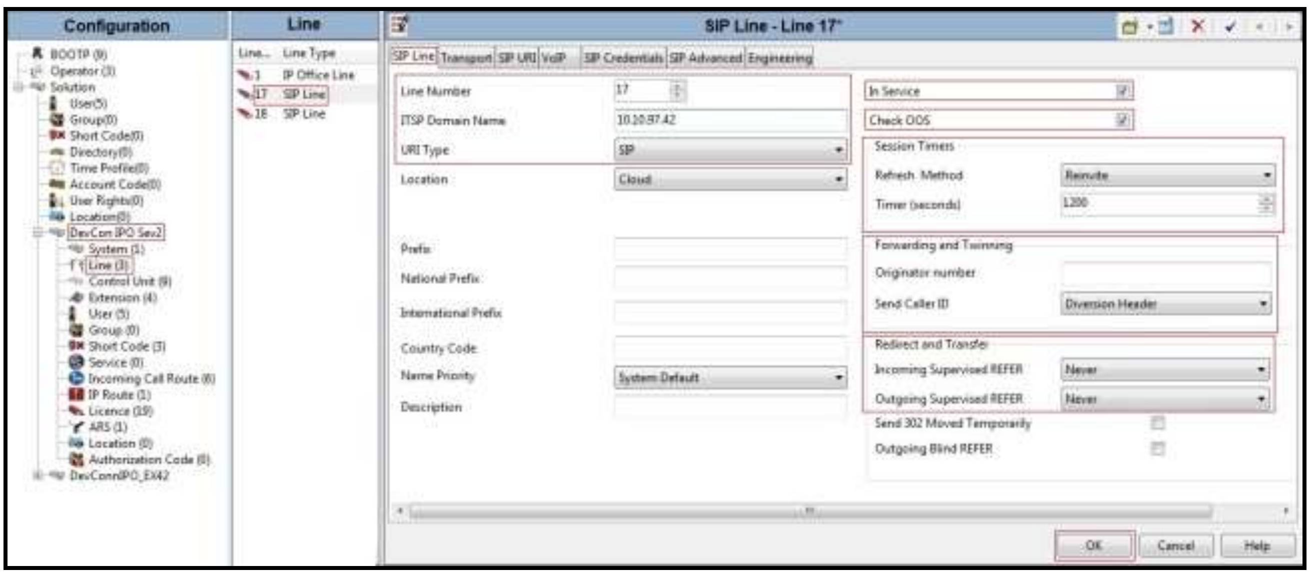
Select **AF_Vitelity_18_SIPTrunk** from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**AF_Vitelity_18_SIPTrunk.xml**) created in **Step 1**. Click **Create new SIP Trunk** to finish creating the SIP trunk.

6. Once the SIP Lines are created, verify the configuration of the SIP Lines with the configuration shown in **Section 5.7.2** and **5.7.3**.

5.7.2. Create SIP Line Manually for Outgoing Calls

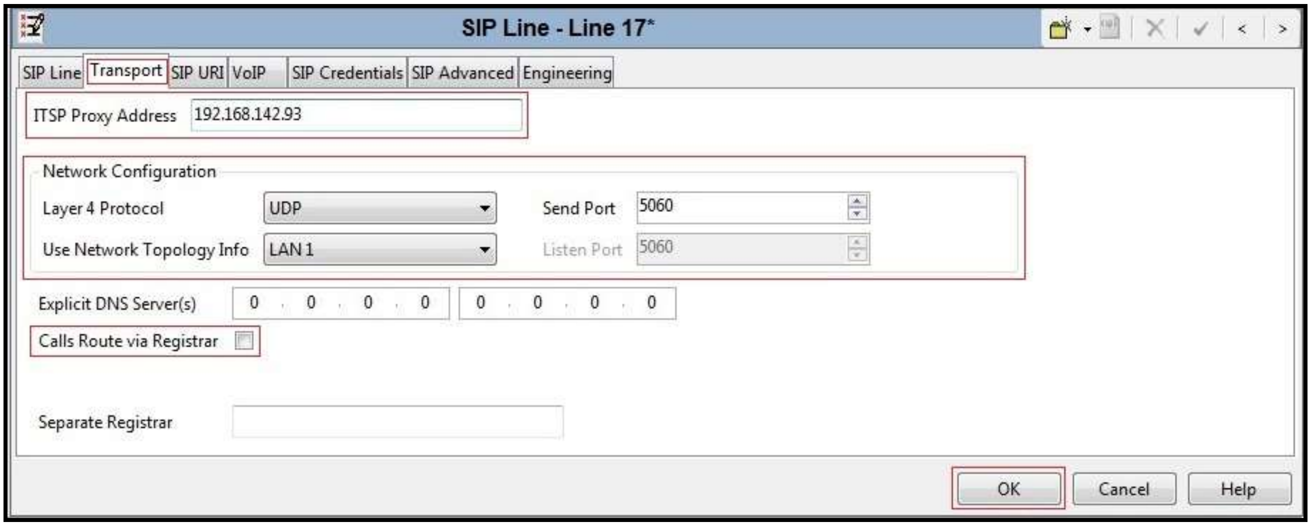
To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New → SIP Line** (not shown).
On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Set **ITSP Domain Name** to the IP address of Avaya IP Office Server Edition LAN1 port. This field is used to specify the default host part of the SIP URI in the From, To, R-URI fields for outgoing calls. For the user making the call, the user part of the From SIP URI is determined by the settings of the SIP URI channel record being used to route the call (see SIP URI → Local URI). For the destination of the call, the user part of the To and R-URI fields are determined by dial short codes of the form 6N;/N"@192.168.142.93" where N is the user part of the SIP URI and "@192.168.142.93" is optional and can be used to override the host part of the To and R-URI.
- Check the **In Service** and **Check OOS** boxes.
- Set **URI Type** to **SIP**.
- For **Session Timers**, set **Refresh Method** to **Reinvite** with **Timer (seconds)** to **1200**.
- For **Forwarding and Twinning**, set **Send Caller ID** to **Diversion Header**.
- For **Redirect and Transfer**, set **Incoming Supervised REFER** and **Outgoing Supervised REFER** to **Never**. Note: Vitelity did not support the REFER for compliance testing.
- Default values may be used for all other parameters.
- Click **OK** to commit then press Ctrl + S to save.



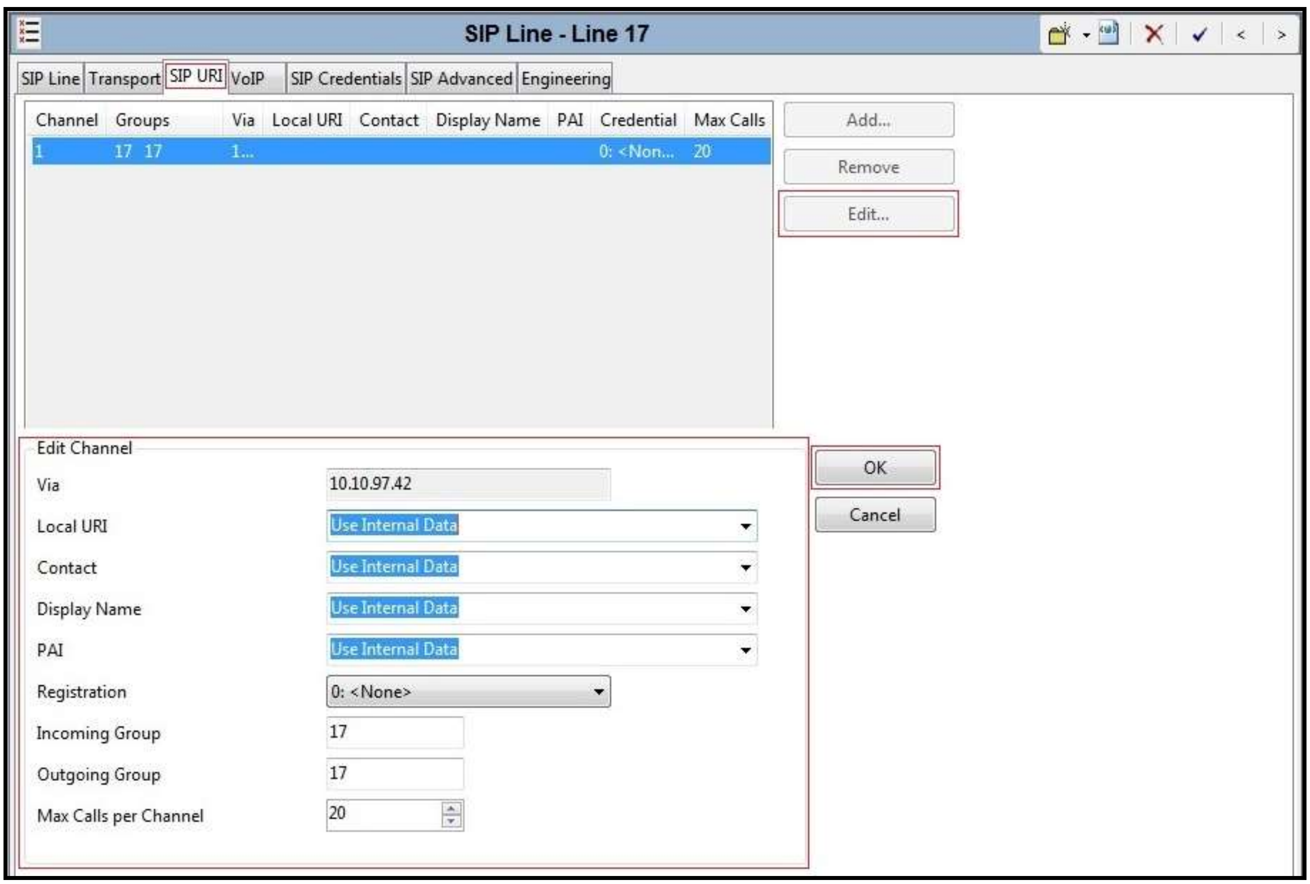
On the **Transport** tab in the Details Pane, configure the parameters as shown below:

- The **ITSP Proxy Address** was set to the IP Address of Vitelity signaling server **192.168.142.93** as shown in **Figure 1**. This is the SIP Proxy address used for outgoing SIP calls.
- In the **Network Configuration** area, **UDP** was selected as the **Layer 4 Protocol** and the **Send Port** was set to **5060** which is the port number supported by Vitelity.
- The **Use Network Topology Info** parameter was set to **LAN 1**. This associates the SIP Line 17 with the parameters in the **DevCon IPO Sev2 → System (1) → LAN1 → Network Topology** tab. The **Listen Port** was set to **5060** which is the port number supported by Vitelity.
- The **Calls Route via Registrar** was unchecked. In this certification testing, Vitelity did not support the dynamic Registration on the SIP Trunk.
- Other parameters retain default values.
- Click **OK** to commit then press Ctrl + S to save.



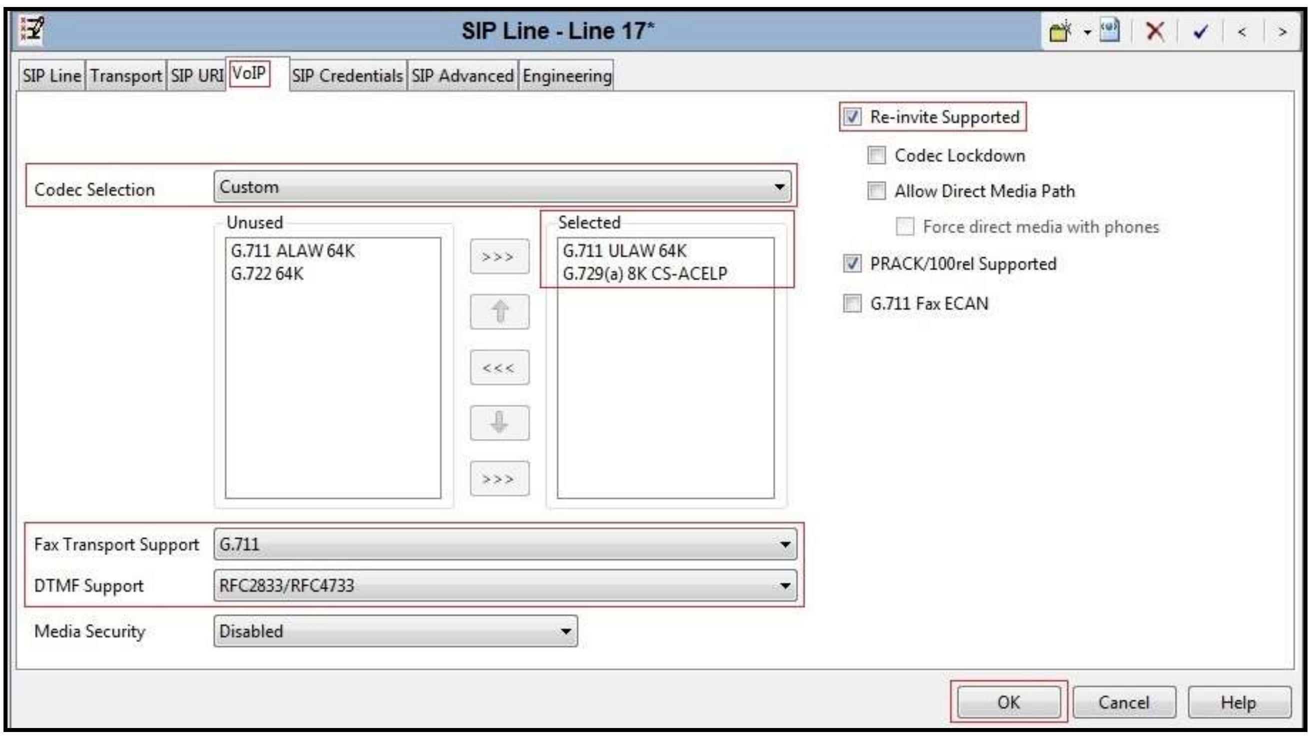
A SIP URI entry must be created to match for each outgoing number that Avaya IP Office Server Edition will route on this line. Select the **SIP URI** tab; click the **Add** button and the **New Channel** area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top, and click **Edit…** button. In the example screen below, a previously configured entry is edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an Avaya IP Office Server Edition user. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact**, **Display Name**, and **PAI** to **Use Internal Data**. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.9**.
- Set **Registration** to **0: <None>** as Vitelity did not require registration.
- Associate this line with an outgoing line group in the **Outgoing Group** field. This line group number will be used in defining outgoing call routes for this line. For the compliance test, a new line group **17** was defined that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Click **OK** to submit the changes.



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

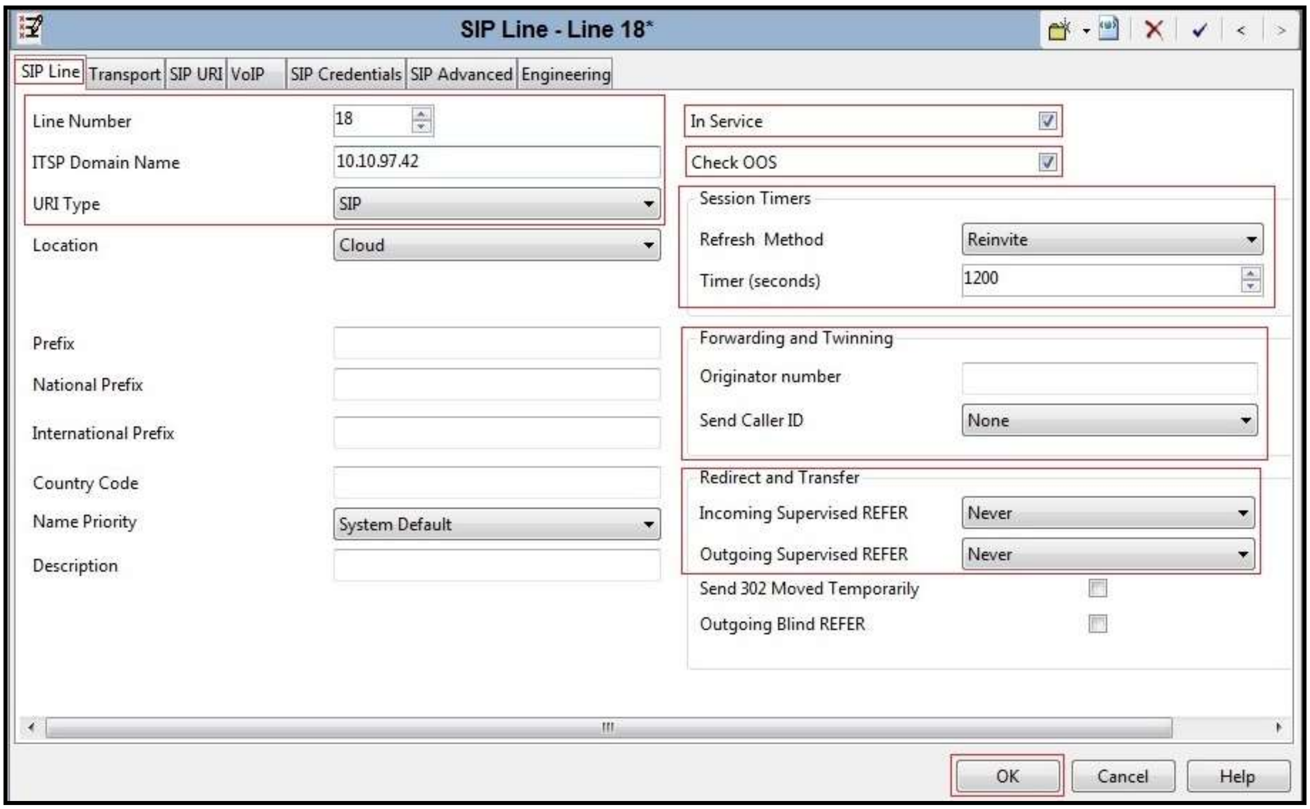
- The **Codec Selection** can be selected by choosing **Custom** from the pull-down menu, allowing an explicit ordered list of codecs to be specified. The **G.711 ULAW 64K**, and **G.729(a) 8K CS –ACELP** codecs are selected. Avaya IP Office Server Edition supports these codecs, which are sent to the Vitelity, in the Session Description Protocol (SDP) offer, in that order.
- Check the **Re-invite Supported** box.
- Set **Fax Transport Support** to **G.711** from the pull-down menu. Vitelity supports Fax G.711 pass-through mode.
  Note: Fax T.38 is not supported because Avaya IP Office Server Edition R9.1.1.0 Build 10 is a Linux based system.
- Set the **DTMF Support** to **RFC2833/RFC4733** from the pull-down menu. This directs Avaya IP Office Server Edition to send DTMF tones using RTP events messages as defined in RFC2833.
- Default values may be used for all other parameters**.**
- Click **OK** to submit the changes.



5.7.3. Create SIP Line Manually for Incoming Calls

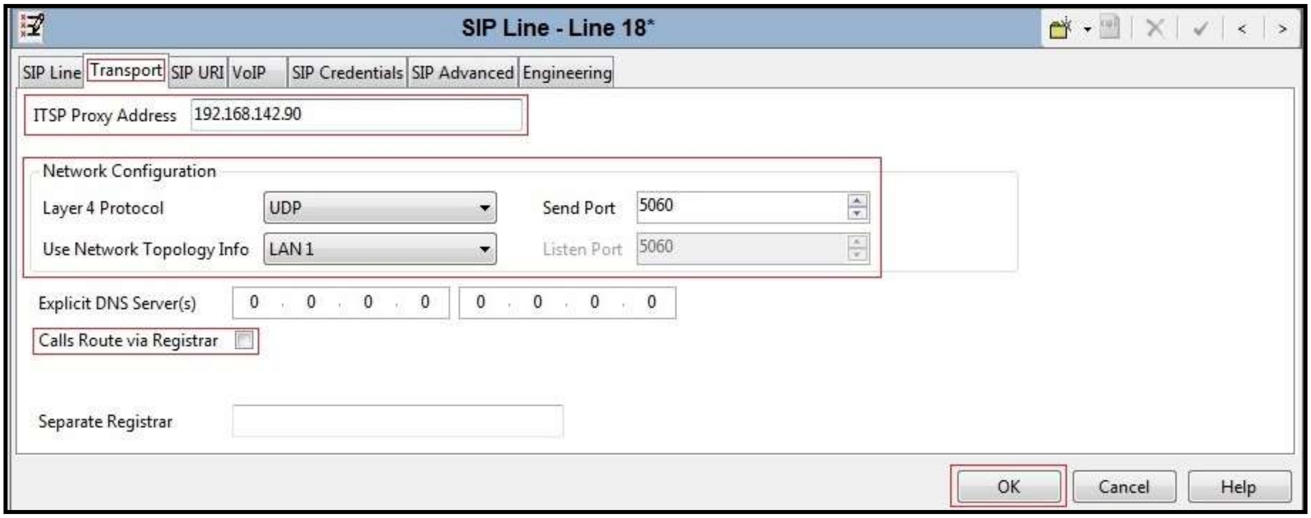
To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New → SIP Line** (not shown).
On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Set **ITSP Domain Name** to the IP address of Avaya IP Office Server Edition LAN1 port.
- Check the **In Service** and **Check OOS** boxes.
- Set **URI Type** to **SIP**.
- For **Session Timers**, set **Refresh Method** to **Reinvite** with **Timer (seconds)**: **1200**.
- For **Forwarding and Twinning**, set **Send Caller ID** to **None**.
- For **Redirect and Transfer**, set **Incoming Supervised REFER** and **Outgoing Supervised REFER** to **Never**.
- Default values may be used for all other parameters.
- Click **OK** to commit then press Ctrl + S to save.



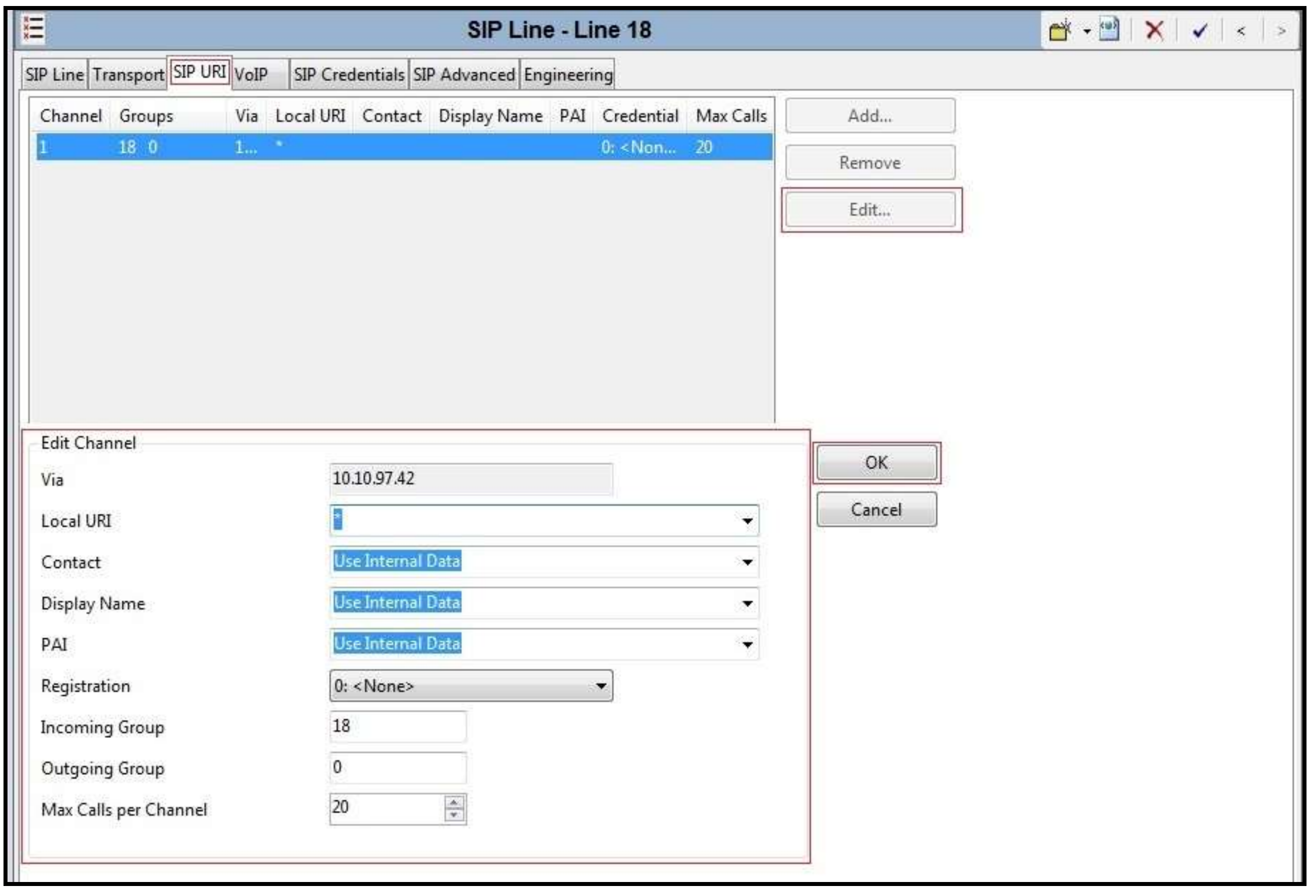
On the **Transport** tab in the Details Pane, configure the parameters as shown below:

- The **ITSP Proxy Address** was set to the IP Address of Vitelity signaling server **192.168.142.90** as shown in **Figure 1**. This is the SIP Proxy address used for incoming SIP calls.
- In the **Network Configuration** area, **UDP** was selected as the **Layer 4 Protocol** and the **Send Port** was set to **5060** which is the port number supported by Vitelity.
- The **Use Network Topology Info** parameter was set to **LAN 1**. This associates the SIP Line 18 with the parameters in the **DevCon IPO Sev2 → System (1) → LAN1 → Network Topology** tab. The **Listen Port** was set to **5060** which is the port number supported by Vitelity.
- The **Calls Route via Registrar** was unchecked.
- Other parameters retain default values.
- Click **OK** to commit then press Ctrl + S to save.



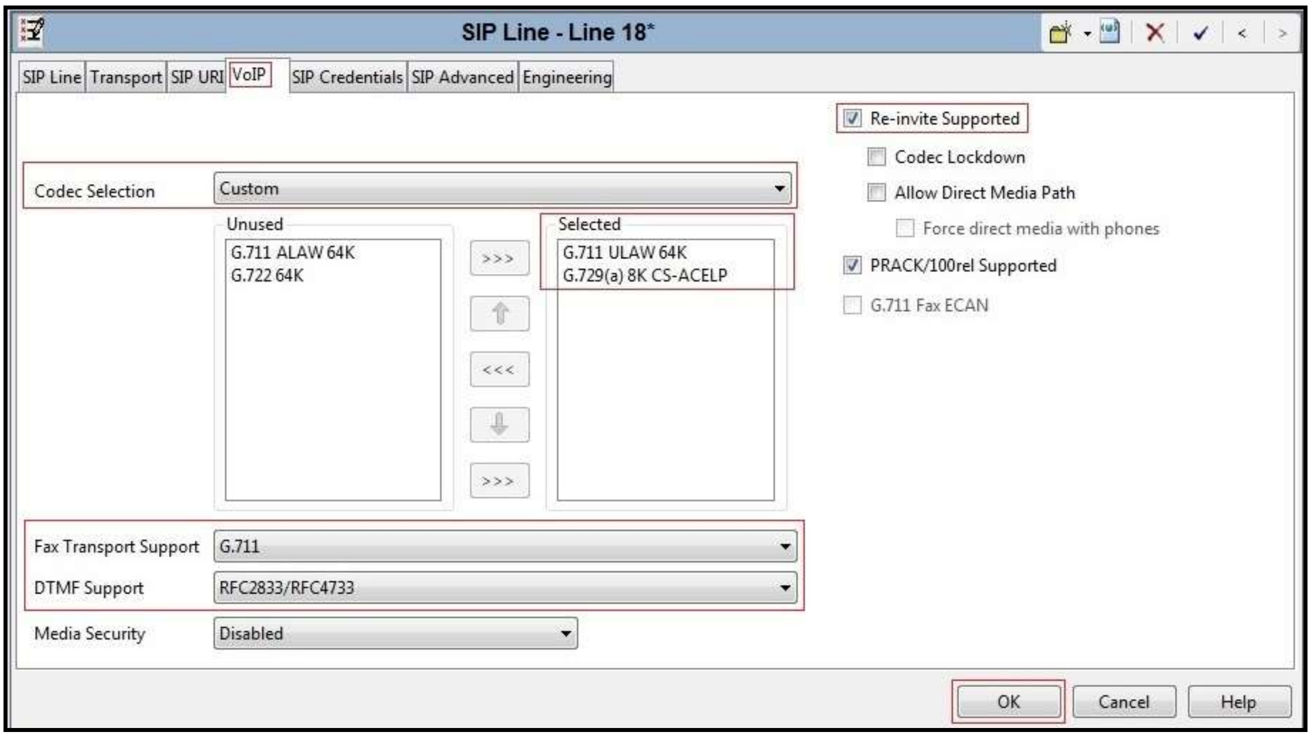
A SIP URI entry must be created to match each incoming number that Avaya IP Office Server Edition will accept on this line. Select the **SIP URI** tab; click the **Add** button and the **New Channel** area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top, and click **Edit…** button. In the example screen below, a previously configured entry is edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an Avaya IP Office Server Edition user. The entry was created with the parameters shown below:

- Set **Local URI** to ***** to accept any incoming calls. Set **Contact**, **Display Name**, and **PAI** to **Use Internal Data**. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.9**.
- Set **Registration** to **0: <None>** as Vitelity did not require registration.
- Associate this line with an incoming line group in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. For the compliance test, a new line group **18** was defined that only contains this line (line 18).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Click **OK** to submit the changes.



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

- The **Codec Selection** can be selected by choosing **Custom** from the pull-down menu, allowing an explicit ordered list of codecs to be specified. The **G.711 ULAW 64K**, and **G.729(a) 8K CS –ACELP** codecs are selected. Avaya IP Office Server Edition supports these codecs, which are sent by the Vitelity.
- Check the **Re-invite Supported** box.
- Set **Fax Transport Support** to **G.711** from the pull-down menu. Vitelity supports Fax G.711 pass-through mode.
  Note: Fax T.38 is not supported because Avaya IP Office Server Edition R9.1.1.0 Build 10 is a Linux based system.
- Set the **DTMF Support** to **RFC2833/RFC4733** from the pull-down menu. This directs Avaya IP Office Server Edition to send DTMF tones using RTP events messages as defined in RFC2833.
- Default values may be used for all other parameters**.**
- Click **OK** to submit the changes.



5.8. Outgoing Call Routing

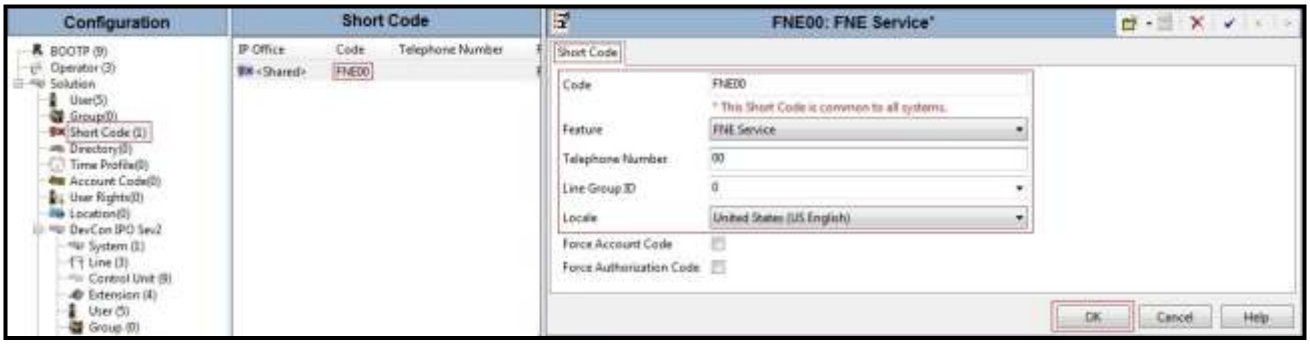
5.8.1. Short Code in Primary Server

Define a short code to route outbound traffic on the SIP line to Vitelity. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane 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 screen below shows the details of the previously administered "6N;" short code used in the test configuration.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **6N;**, this short code will be invoked when the user dials 6 followed by any number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N"@192.168.142.93"**. 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 host part following the "@" is the IP address of Vitelity signaling server.
- Set the **Line Group ID** to the **Outgoing Group 17** defined on the **SIP URI** tab on the **SIP Line** in **Section 5.7.2**. This short code will use this line group when placing the outbound call.
- Set the **Locale** to **United States (US English)**.
- Default values may be used for all other parameters**.**
- Click **OK** to submit the changes.

The feature of incoming calls from mobility extension to idle-appearance FNE (Feature Name Extension) is hosted by Avaya IP Office Server Edition. The Short Code **FNE00** was configured with following parameters:

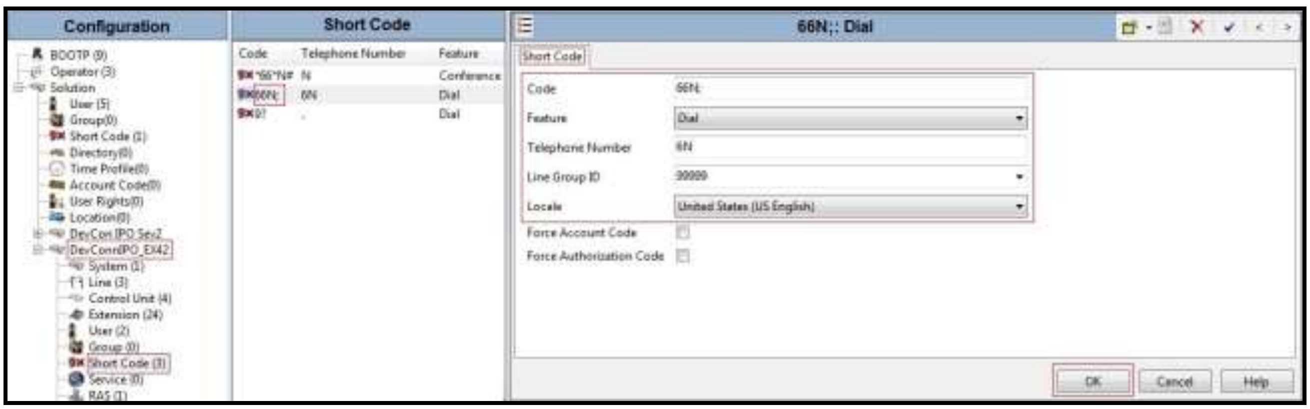
- For **Code** field, enter FNE feature code as **FNE00** for dial tone.
- Set **Feature** to **FNE Service**.
- Set **Telephone Number** to **00**.
- Set **Line Group ID** to **0**.
- Set the **Locale** to **United States (US English)**.
- Default values may be used for other parameters.
- Click **OK** to submit the changes.



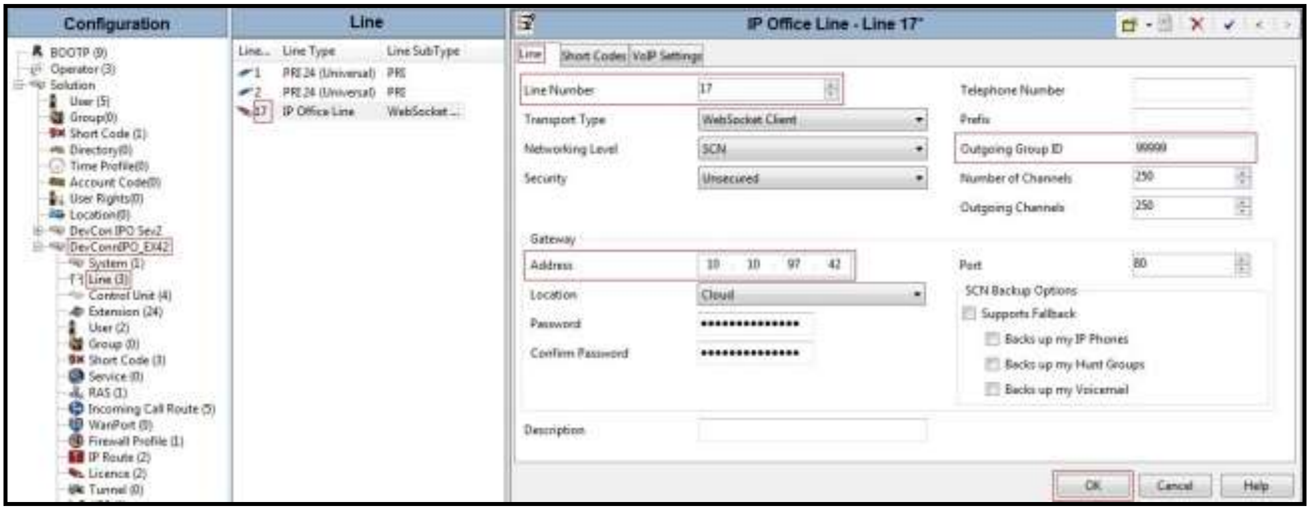
5.8.2. Short Code and SIP Line in Expansion System

Define a short code to route outbound traffic on the SIP line to Vitelity. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane 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 screen below shows the details of the previously administered "66N;" short code used in the test configuration.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **66N;**, this short code will be invoked when the user (using Avaya analog or digital phones) dials 66 followed by any number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **6N**.
- Set the **Line Group ID** to the **99999** defined on the **Outgoing Group ID** of the H.323 line connecting the Expansion System to the Primary Server. This short code will use this line group when placing the outbound call via Avaya IP Office Server Edition Primary Server.
- Set the **Locale** to **United States (US English)**.
- Default values may be used for all other parameters**.**
- Click **OK** to submit the changes.



Verify the ID of the H.323 line connecting the Expansion System to the Primary Server. To do this, select Expansion Line on the navigation pane and select the H.323 line on the Group pane (line 17 on the screen below). Make note of the **Outgoing Group ID 99999** on the Details pane. The **Address** is Avaya IP Office Server Edition LAN1 IP Address **10.10.97.42**.

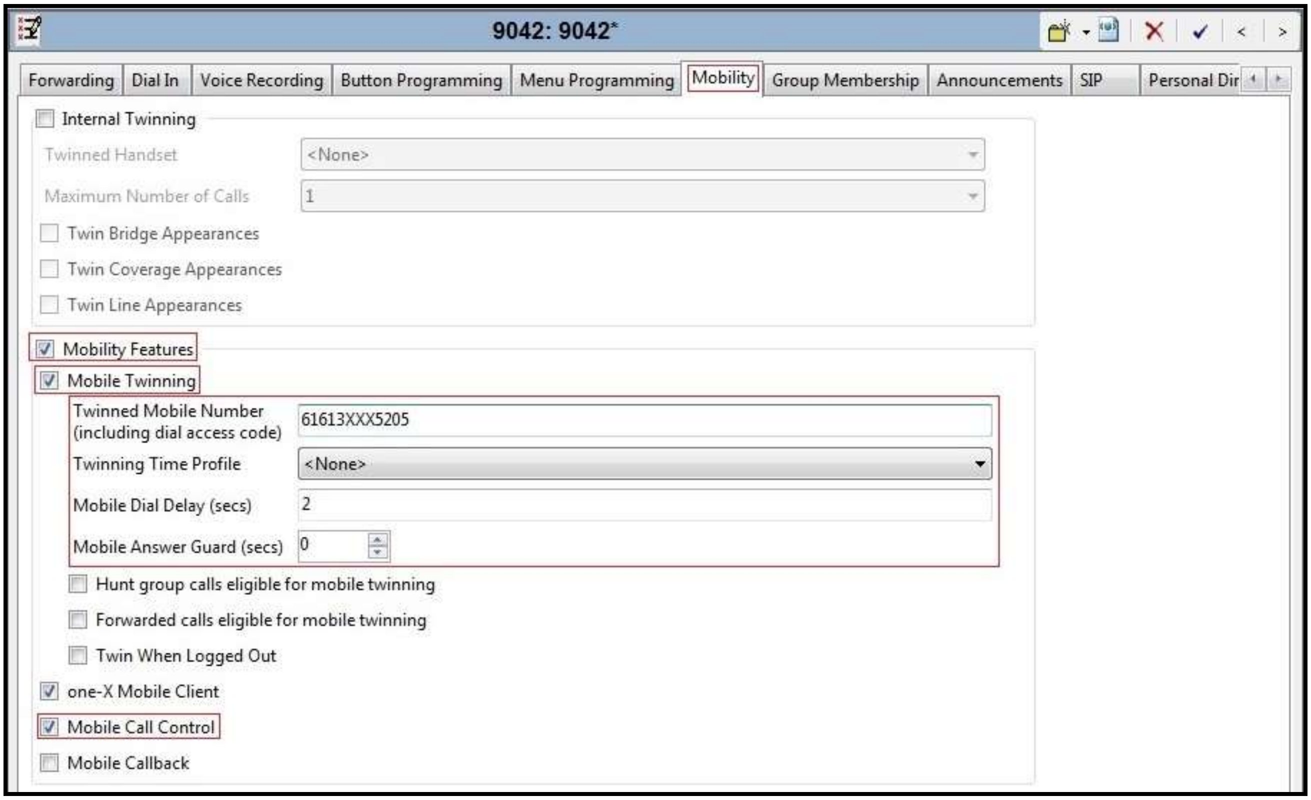


5.9. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.7**. To configure these settings, first select **User** in the left Navigation Pane, then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is **9042**. Select the **SIP** tab in the Details Pane.

The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls. They also allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line. The example below shows the settings for user **9042**. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise provided by Vitelity. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network.

One of the H.323 IP Phones at the enterprise site uses the Mobile Twinning feature. The following screen shows the **Mobility** tab for **User 9042**. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case **61613XXX5205**. Check **Mobile Call Control** to allow incoming calls from mobility extension to access FNE00 (see **Section 5.8.1**). Other options can be set according to customer requirements.

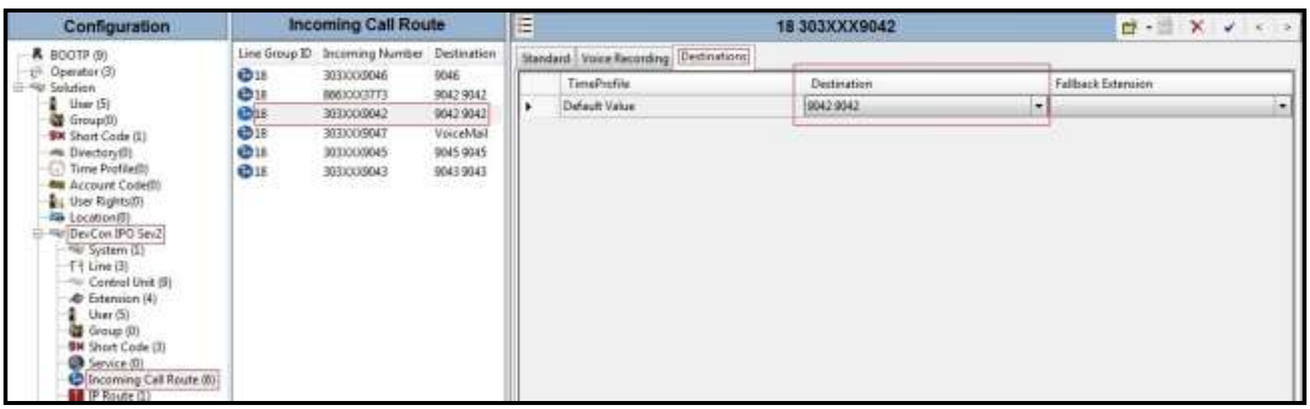


5.10. Incoming Call Route

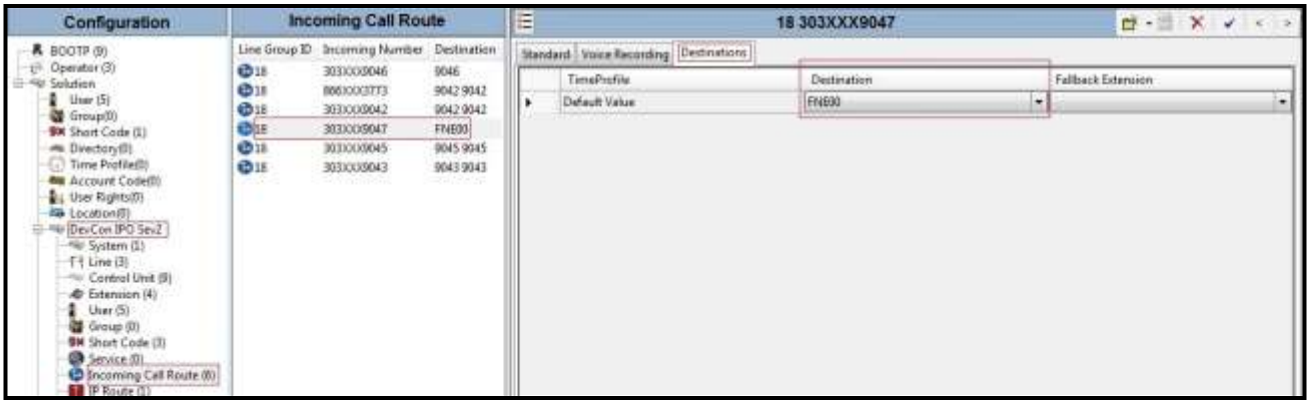
An Incoming Call Route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by service provider. To create an incoming call route, select **Incoming Call Route** in the left Navigation Pane, then right-click in the center Group Pane and select **New** (Not shown). On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capability** to **Any Voice**.
- Set the **Line Group ID** to the **Incoming Group 18** defined on the **SIP URI** tab on the **SIP Line** in **Section 5.7.3**.
- Set the **Incoming Number** to the incoming DID number on which this route should match.
- Default values can be used for all other fields.

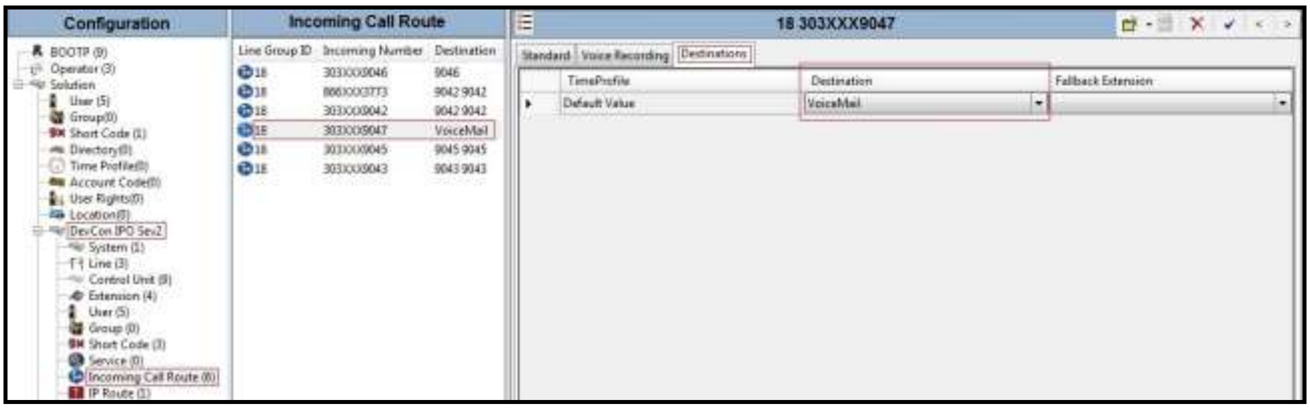
On the **Destination** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to **303XXX9042** on line 18 are routed to extension **9042**.



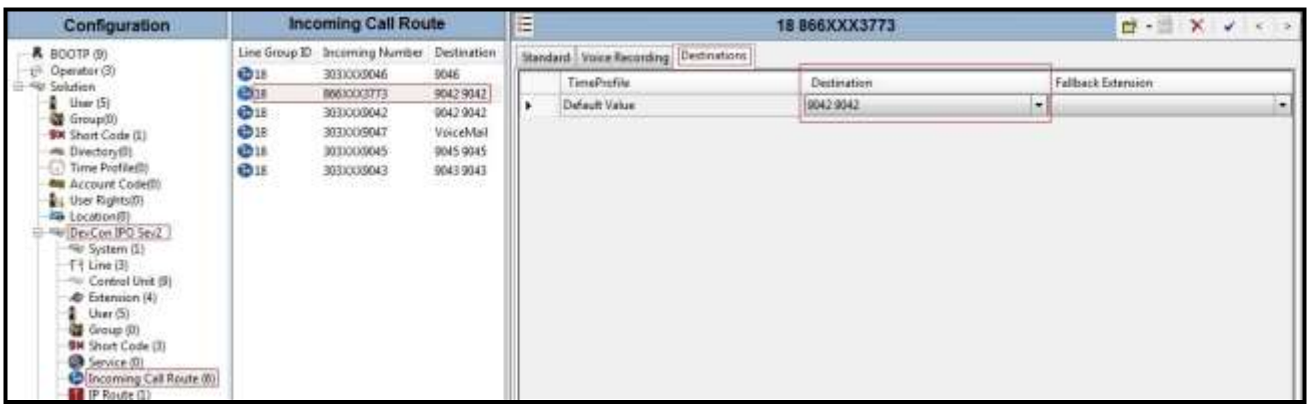
For testing purpose, the incoming calls to DID number **303XXX9047** were configured to access **FNE00**. The **Destination** was appropriately defined as **FNE00** as below screenshot:



For testing purpose, the incoming calls to DID number **303XXX9047** were also configured to access **VoiceMail**. The **Destination** was appropriately defined as **VoiceMail** as below screenshot:



For Inbound Toll-Free testing purpose, any incoming calls to DID number **866XXX3773** were also configured to route to extension **9042**. The **Destination** was appropriately defined as **9042** as below screenshot:



5.11. 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. Vitelity SIP Trunk Configuration

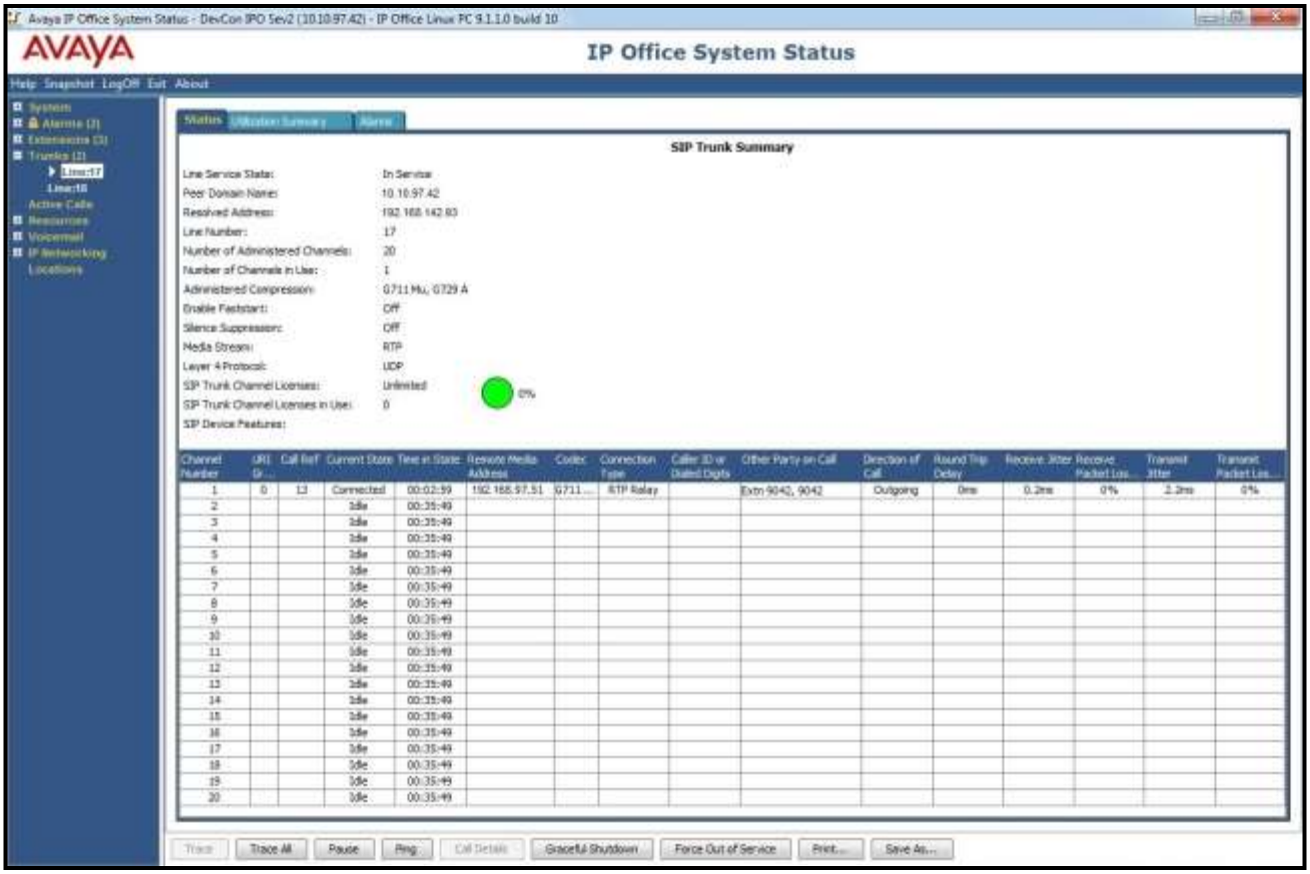
Vitelity is responsible for the configuration of Vitelity SIP Trunk Service. The customer must provide the IP address used to reach the Avaya IP Office Server Edition at the enterprise. Vitelity will provide the customer necessary information to configure the SIP connection between Avaya IP Office Server Edition and Vitelity. The provided information from Vitelity includes:

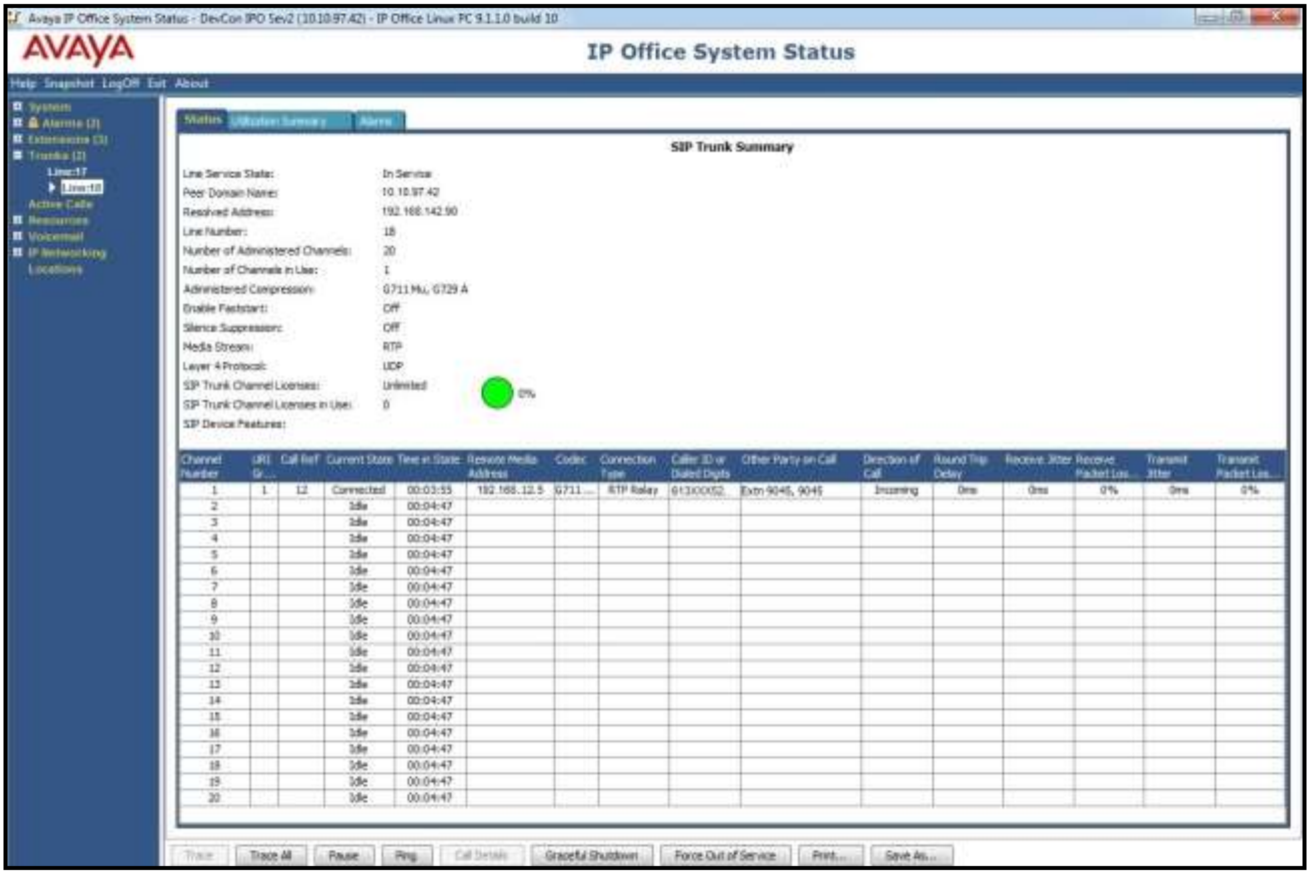
- IP address and port number used for signaling or media through any security.
- DID numbers.
- Vitelity SIP Trunk Specification.

7. Verification Steps

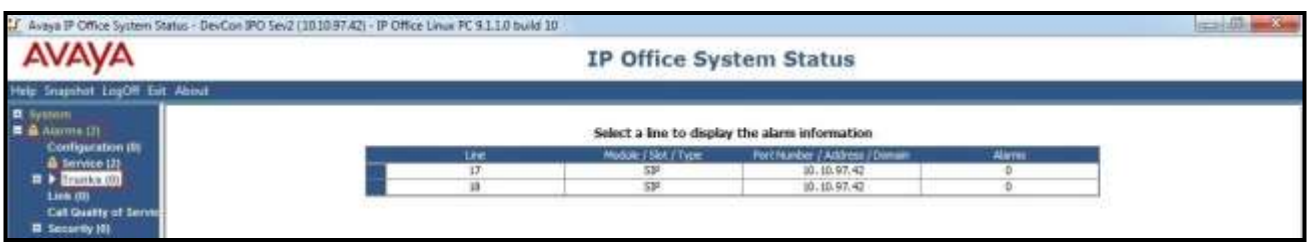
The following steps may be used to verify the configuration:

- Use the Avaya IP Office Server Edition 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 Avaya IP Office Server Edition Manager was installed. Select the SIP line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** for each channel (The below screen-shots showed 2 active calls at present time).





Use the Avaya IP Office Server Edition System Status application to verify that no alarms are active on the SIP line. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Server Edition Manager was installed. Select **Alarm → Trunks** to verify that no alarms are active on the SIP lines.



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

8. Conclusion

Vitelity passed compliance testing excepting the limitation in **Section 2.2**. These Application Notes describe the procedures required to configure the SIP connection between Avaya IP Office Server Edition and the Vitelity as shown in **Figure 1**.

9. Additional References

[1] IP Office 9.1 Administering Avaya IP Office Platform with Manager, Release 9.1, Issue 10.03, February 2015.

[2] Avaya IP Office™ Platform Server Edition Reference Configuration Release 9.1, Issue 02.06, April 2015.

[3] Deploying Avaya IP Office™ Platform Server Edition Solution Release 9.1, Issue 02.07, April 2015.

Product documentation for Avaya products may be found at: http://support.avaya.com. Additional IP Office documentation can be found at: http://marketingtools.avaya.com/knowledgebase/ipoffice/general/rss2html.php?XMLFILE=manuals.xml&TEMPLATE=pdf_feed_template.html.

Product documentation for Vitelity SIP Trunking may be found at: http://www.vitelity.com

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.