Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 10.0 with VTX Connect PABX-IP SIP Trunking – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between VTX Connect PABX-IP and Avaya IP Office.

VTX Connect PABX-IP provides PSTN access via a SIP Trunk connected to the VTX Voice over Internet Protocol (VoIP) network as an alternative to legacy analogue or digital trunks. VTX Services SA is a member of the Avaya DevConnect Service Provider program.

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 VTX Connect PABX-IP and Avaya IP Office. Customers using this Avaya SIP-enabled enterprise solution with VTX's SIP Trunk are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks. This approach generally results in lower cost for the enterprise customer.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to VTX Connect PABX-IP. This configuration (shown in **Figure 1**) was used to exercise the features and functionality 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

Avaya IP Office was connected to VTX Connect PABX-IP. To verify SIP trunking interoperability the following features and functionality were exercised during the interoperability compliance test:

- Incoming PSTN calls to various phone types including H.323, SIP and analogue telephones at the enterprise. Calls were routed to the enterprise across the SIP trunk from VTX.
- Outgoing PSTN calls from various phone types including H.323, SIP and analogue telephones at the enterprise. Calls were routed from the enterprise across the SIP trunk to VTX.
- Inbound and outbound PSTN calls to/from an Avaya Communicator for Windows client.
- Various call types including: local, international, toll free (outbound) and directory assistance.
- Calls using G.711A, G.711MU and G.729A codecs.
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using T.38.
- Caller ID presentation and Caller ID restriction.
- DTMF transmission using RFC 2833.
- Voicemail navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, and conference.
- Off-net call forwarding and mobile twinning.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for VTX Connect PABX-IP with the following observations:

- During testing, occasional intermittent issues were found with calls to and from the test PSTN phones. These included lack of audio, no indication of a busy PSTN phone and failure of incoming fax calls. These issues were due to the test environment and were not interoperability issues with VTX Connect.
- No inbound toll-free access was available for testing
- No test call was booked with Emergency Services Operator.
- When forwarding to a PSTN phone using REFER, the network responds to the REFER message on leg 2 with NOTIFY (200 OK) and the IP office clears the call. IP Office then sends a "405 Method Not Allowed" message. This does not interfere with the Call Forwarding.
- When doing a blind transfer of an outbound PSTN call to a PSTN phone, the IP Office sends a REFER on leg 1. The network successfully transfers the call, but then appears to use 407 Proxy Authentication Required to clear leg 2 before the dialogue is complete. IP Office sends a duplicate INVITE in response which the network rejects with a "481 Unknown Dialogue". This is not a particularly graceful way to clear the call on IP Office, but it is successfully established in the network.
- When testing consultative transfer of an inbound PSTN call to a PSTN number from an H.323 endpoint, IP Office sent 405 Method Not Allowed after the call was cleared. The call was successfully established in the network.
- On completion of a consultative transfer to a PSTN test phone from an 1140 SIP endpoint, the phone indicated "Transfer Failed" even though the transfer was successful. This did not occur when transferring to a DDI number and is assumed to be a limitation of the test environment and not an interoperability issue.
- Incoming fax calls failed when made from a fax machine in the Avaya Galway Lab. The calls were successful when made from VTX premises. This is assumed to be an issue with the test environment and not an interoperability issue.
- During testing of incoming fax, it was observed that incorrect RTP port numbers were used for the media when the SDP in the re-INVITE from IP Office had the audio media line specified before T.38. This was resolved using a SIP Line Custom String in the Engineering tab of the SIP Line as described in **Section 5.5.2**.

2.3. Support

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

For technical support on VTX Services products please contact the VTX Services support team at:
https://www.vtx.ch/fr/support

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to VTX Connect PABX-IP. Located at the enterprise site are an Avaya IP Office Server Edition and an Avaya IP Office 500 V2 as an Expansion. Endpoints include an Avaya 1600 Series IP Telephone (with H.323 firmware), Avaya 9600 Series IP Telephones (with H.323 firmware), an Avaya 1140e SIP Telephone, an Avaya Analogue Telephone and a fax machine. The site also has a Windows 7 PC running Avaya IP Office Manager to configure the Avaya IP Office as well as Avaya Communicator for Windows for mobility testing. For security purposes, public IP addresses and PSTN routable phone numbers used in the compliance test are not shown in full.

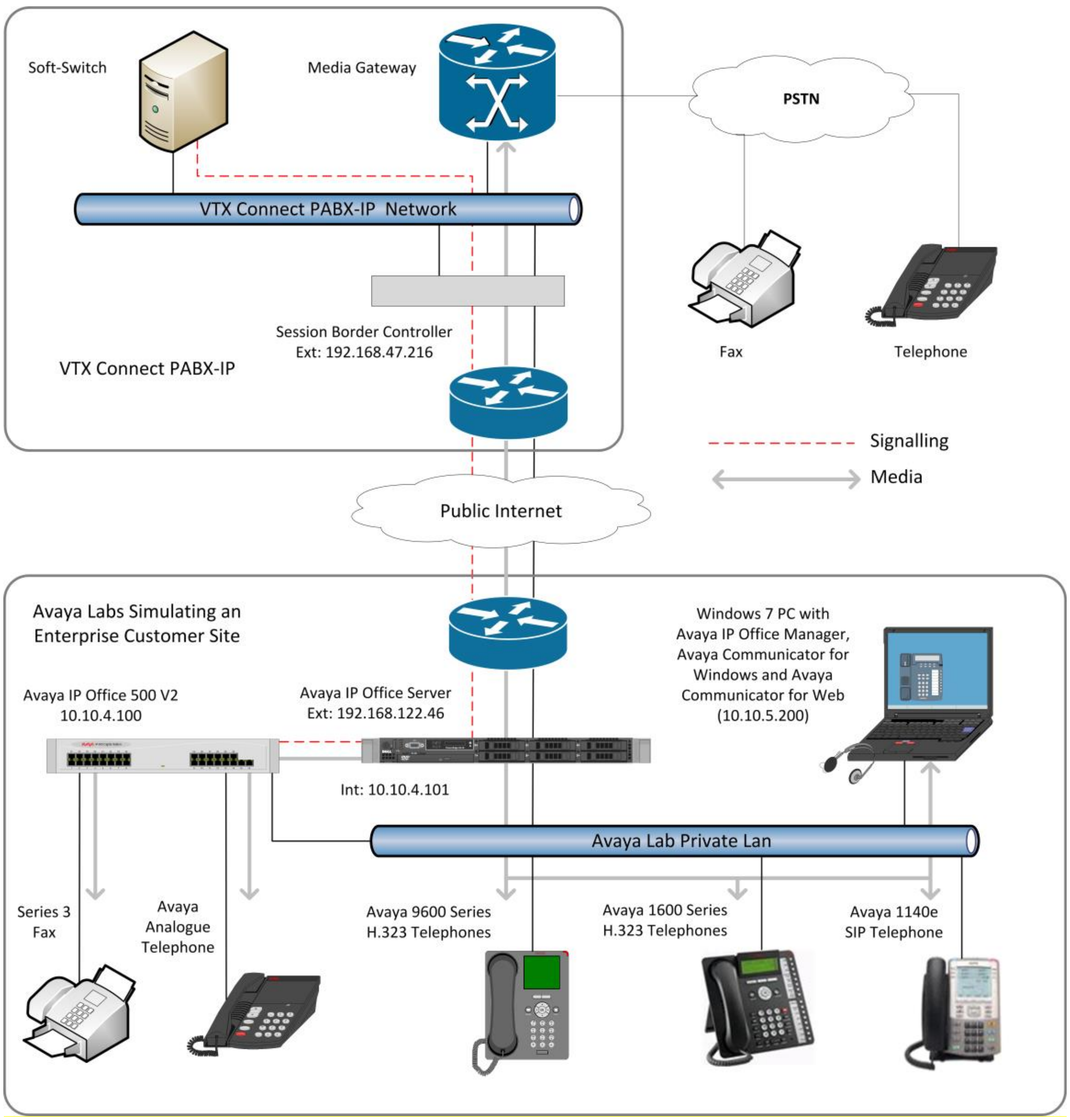


Figure 1: VTX Connect PABX-IP to Avaya IP Office Topology

4. Equipment and Software Validated

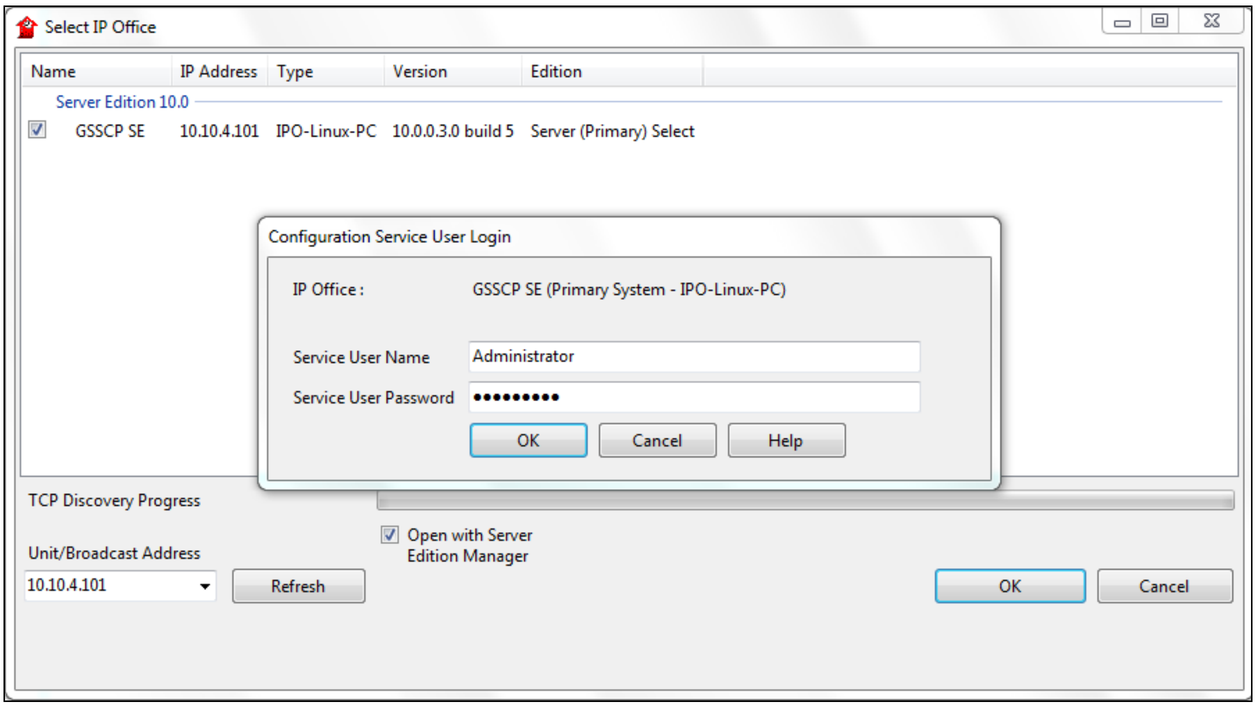
The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya	
Avaya IP Office Server Edition	10.0.0.3.0 build 5
Avaya IP Office 500 V2 Expansion	10.0.0.3.0 build 5
Avaya 1140e IP SIP Telephone	04.04.23.00
Avaya 1608 IP Phone (H.323)	1.350B
Avaya 9608 IP Phone (H.323)	6.6.4.01 V474
Avaya 98390 Analogue Phone	N/A
Avaya Communicator for Windows	2.1.3.0-NGUE-FLAREWINIPOREGRESSION10-JOB1.237
Avaya IP Office Server Edition Manager	Version 10.0.0.3.0 build 5
VTX	
VTX Connect-PBX	VTX Connect-PBX-4
Platform: TELES C5 IP-Centrex	version 5.0
SBC: Acme	version SCX 6.3.0 build 606

Testing was performed with IP Office Server Edition with 500 V2 Expansion R10.0. 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. Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks, this includes T.38 fax.

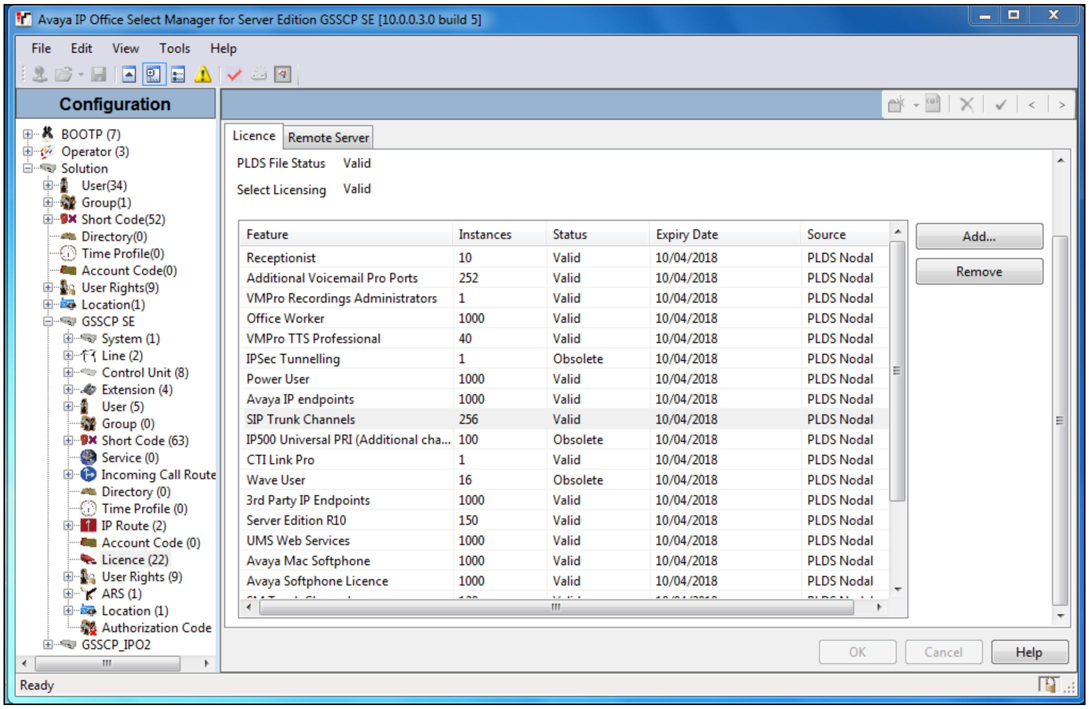
5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to VTX Connect PABX-IP. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration** (not shown), select the appropriate Avaya IP Office system from the pop-up window and log in with the appropriate credentials. A management window will appear similar to the one in the next section. All the Avaya IP Office configurable components are shown in the left pane known as the Navigation pane. The pane on the right is the Details pane. These panes will be referenced throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the Service Provider (such as mobile twinning) is assumed to already be in place.



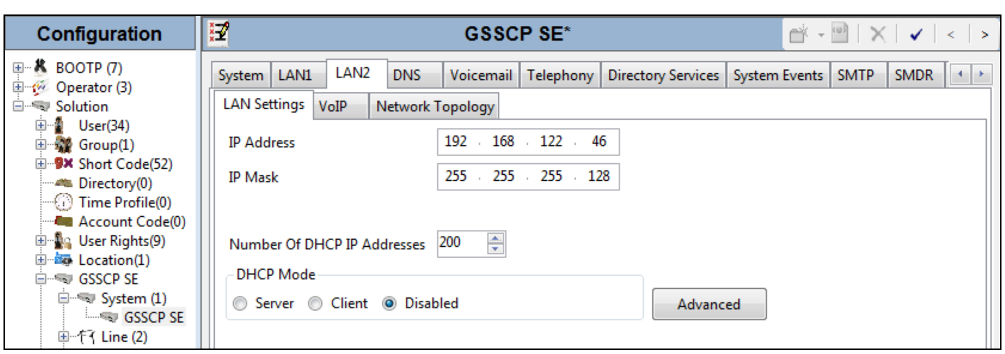
5.1. Verify System Capacity

Navigate to **License** in the Navigation pane. In the Details pane verify that the **License Status** for **SIP Trunk Channels** is Valid and that the number of **Instances** is sufficient to support the number of SIP trunk channels provisioned by VTX.



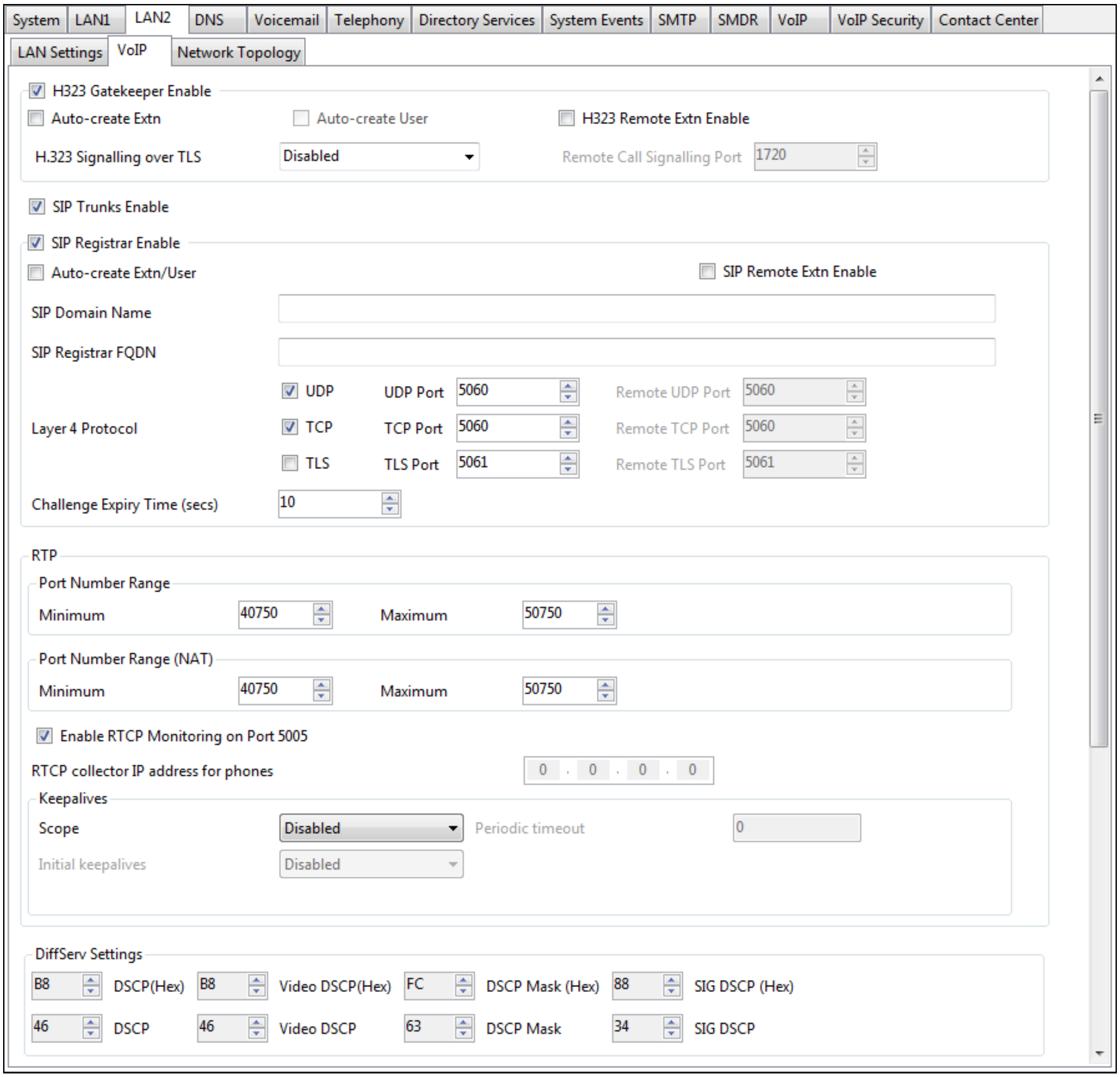
5.2. LAN2

In the sample configuration, the LAN2 port was used to connect the Avaya IP Office to the VPN server. To access the LAN2 settings, first navigate to **System → <IP Office Name>** in the Navigation pane where IP Office Name is the name of the IP Office. This is **GSSCP_IPO2** for the 500 V2 in the GSSCP test environment. Navigate to the **LAN2 → LAN Settings** tab in the Details pane. The **IP Address** and **IP Mask** fields are the external interface of the IP Office. All other parameters should be set according to customer requirements. On completion, click the OK button (not shown).



On the **VoIP** tab in the Details pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. If SIP Endpoints are to be used such as the Avaya Communicator for Windows and the Avaya 1140e, the **SIP Registrar Enable** box must also be checked. Define the port to be used for the signalling transport, in the test environment **TCP** was used and the port number was left at the default value of **5060**.

Scroll down for further configuration. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office requests RTP media to be sent to a UDP port in the configurable range for calls using LAN2. The range used for testing was the default IP Office Server setting of **40750** to **50750**.

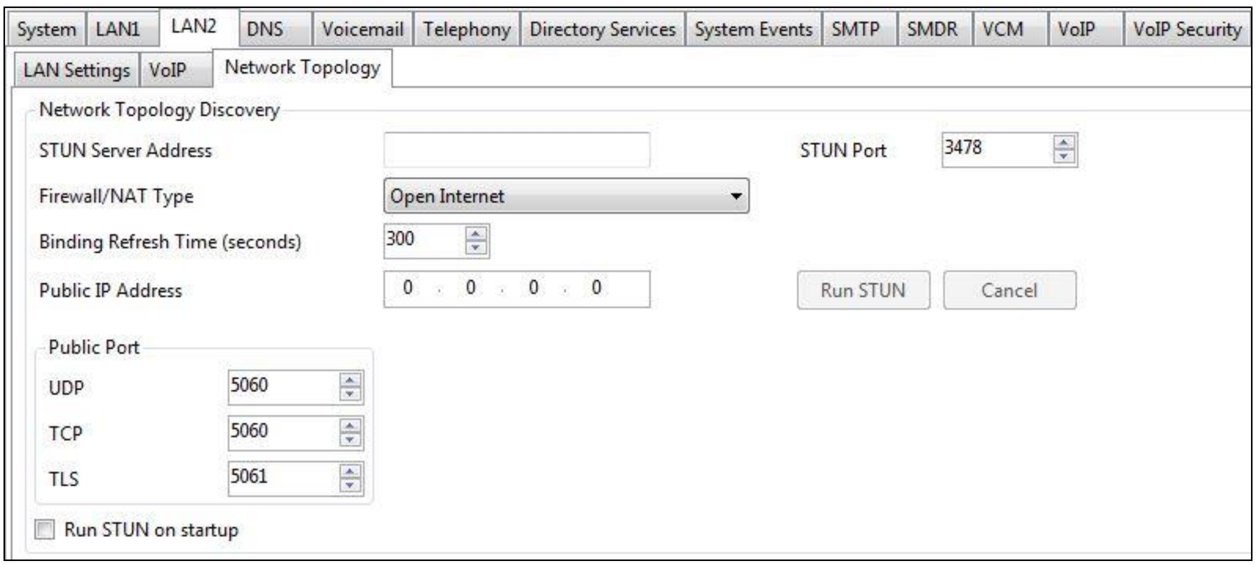


Note: Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP header with specific values to support Quality of Services policies for both signalling and media. DSCP for media can be set for both voice and video. The **DSCP** field is the value used for voice and the **SIG DSCP** is the value used for signalling. For the compliance test, the DSCP values were left at their default values.

All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).

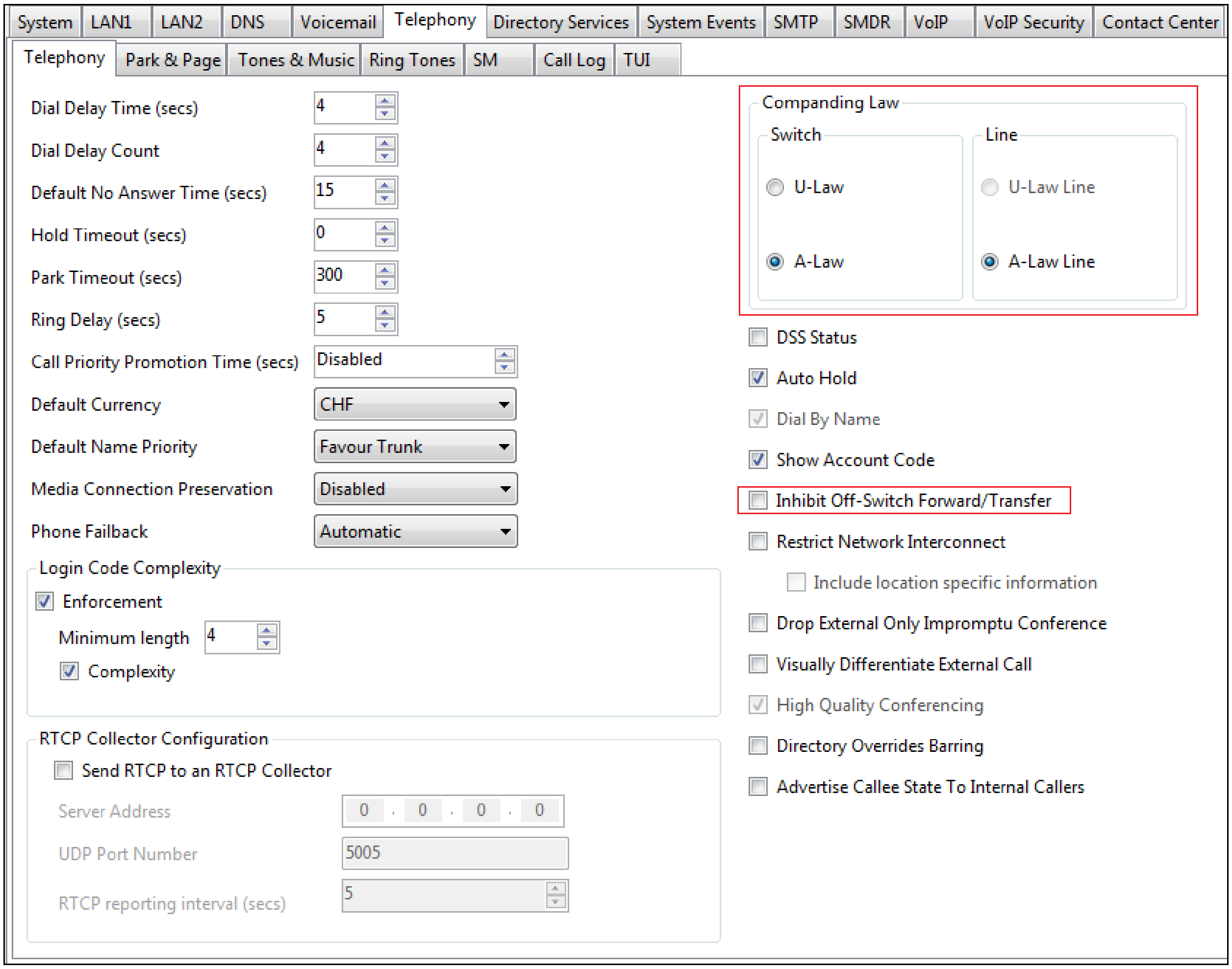
On the **Network Topology** tab in the Details pane, leave the **STUN Server Address** blank and the Firewall/NAT Type at **Open Internet** as NAT is not required in this configuration.

The Network Topology tab can be used to set the **Binding Refresh Time** for the periodic sending of OPTIONS. This is intended for use where OPTIONS are required at intervals of less than 300 seconds. As SIP registration was used for testing, the sending of OPTIONS was not required.



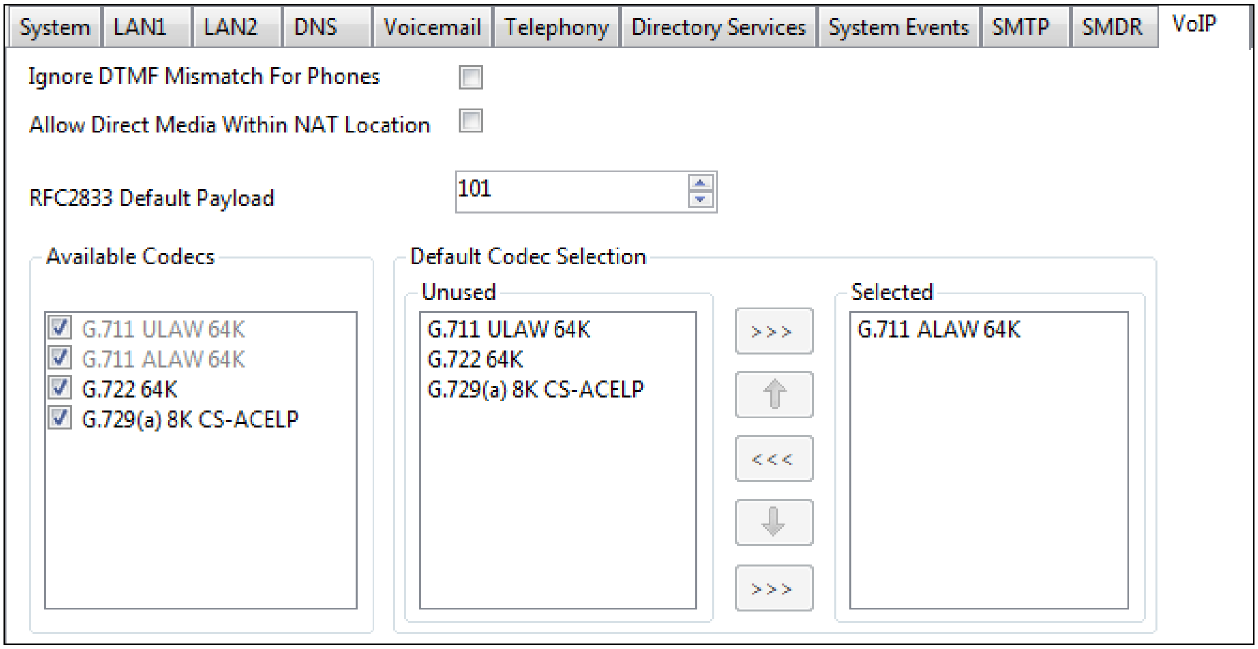
5.3. System Telephony Settings

Navigate to the **Telephony → Telephony** tab on the Details pane. Choose the **Companding Law** typical for the enterprise location. For Europe, **A-Law** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the Service Provider across the SIP trunk. On completion, click the **OK** button (not shown).



5.4. Codec Settings

Navigate to the **VoIP** tab on the Details pane. Check the Available Codecs boxes as required for the IP endpoints. Note that **G.711 ULAW 64K** and **G.711 ALAW 64K** are greyed out and always available. Once available codecs are selected, they can be used or unused by using the horizontal arrows as required. Note that in test **G.711 ALAW 64K** was used as the default codec. The order of priority can be changed using the vertical arrows. On completion, click the **OK** button (not shown).



Note: The codec settings for IP endpoints can also be used for the SIP Trunk by selecting **System Default** in the **Codec Selection** as shown in **Section 5.5.2**.

5.5. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and VTX Connect PABX-IP. 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 Manager to create a SIP Line. Follow the steps in **Section 5.5.1** 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 5.5.2**.

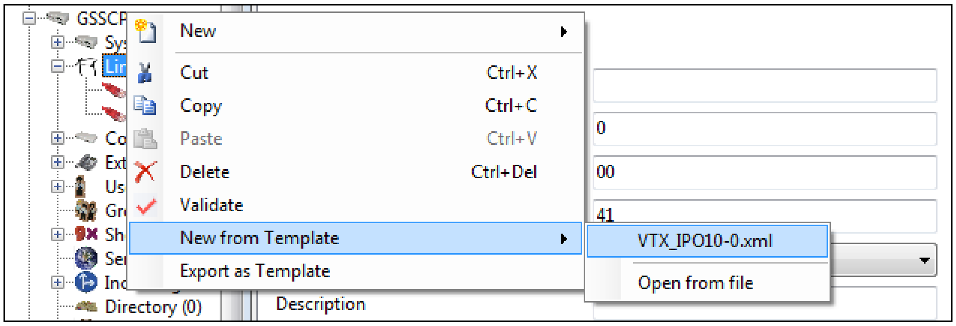
Also, the following SIP Line settings are not supported on Basic Edition:

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

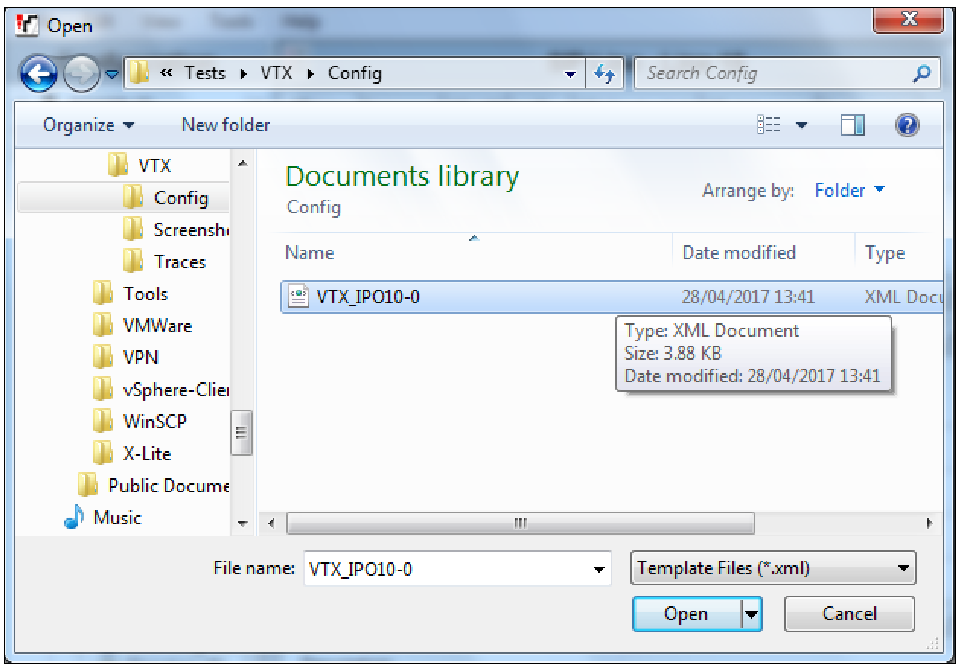
Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation pane and select **New→SIP Line** (not shown). Then, follow the steps outlined in **Section 5.5.2**.

5.5.1. SIP Line From Template

Copy the template file to the computer where IP Office Manager is installed. To create the SIP Trunk from the template, right-click on **Line** in the Navigation pane, then navigate to **New → New from Template**.



Note: If the template file was imported into the IP Office it will appear as an option in the menu for creating a SIP Line from a template. If the template file was not imported but is present on the local machine, click on **Open from file** and navigate to the directory where the template was copied and select it.

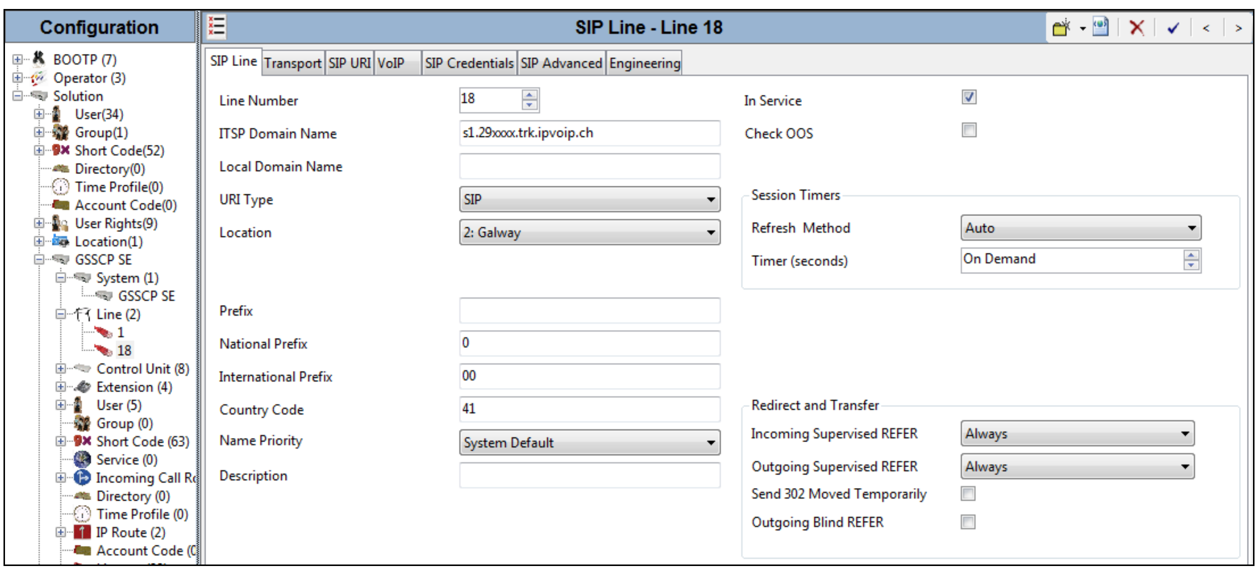


The SIP Line is automatically created and can be verified and edited as required using the configuration described in **Section 5.5.2**.

5.5.2. Manual SIP Line Configuration

On the **SIP Line** tab in the Details pane, configure the parameters below to connect to VTX.

- Set the **ITSP Domain Name** to that provided by VTX Services. The domain name specified here is used in SIP messages instead of the IP address of the IP Office.
- Leave **Prefix** blank as it is not used
- Set **National Prefix** to 0 and **International Prefix** to 00 so that national and international numbers can be correctly identified.
- Set **Country Code** to 41 for Switzerland so that the calling party number of inbound numbers is converted to national format for display on IP Office extensions.
- Leave the **Check OOS** box unchecked so that the SIP Trunk is not taken out of service when there is no response to OPTIONS. As SIP registration is used, OPTIONS are not required.
- Ensure the **In Service** box is checked.
- Leave the **Session Timers** parameters at default values. As SIP registration is used, session timers are not required.
- Select **Always** in the **Incoming Supervised REFER** and **Outgoing Supervise REFER** drop down menus. VTX Connect PABX-IP supports REFER and it is not used automatically where OPTIONS messages are not received.
- Leave all other fields at default settings.

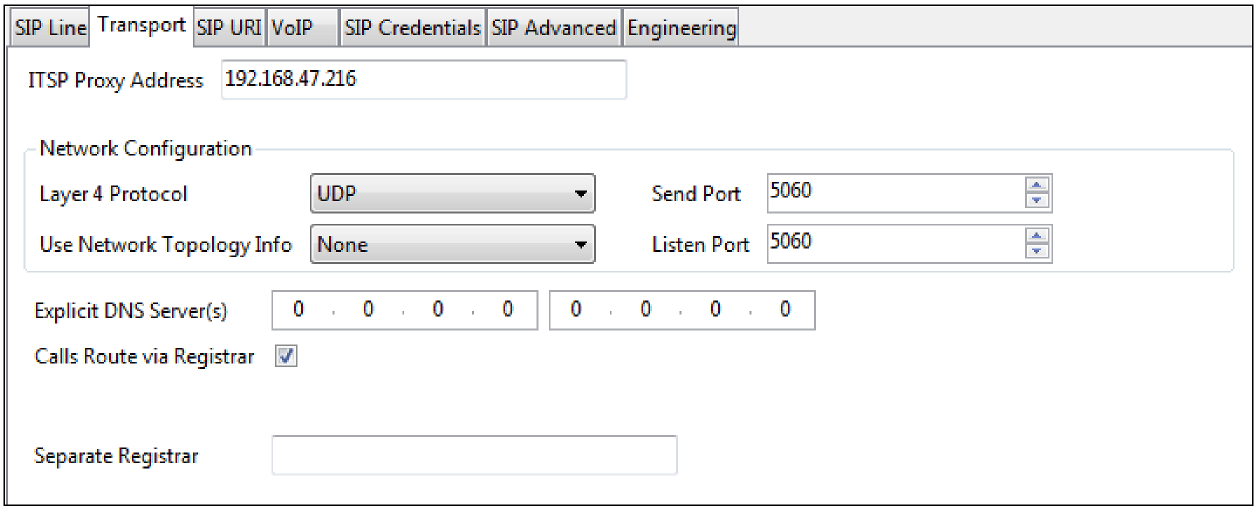


On completion, click the **OK** button (not shown).

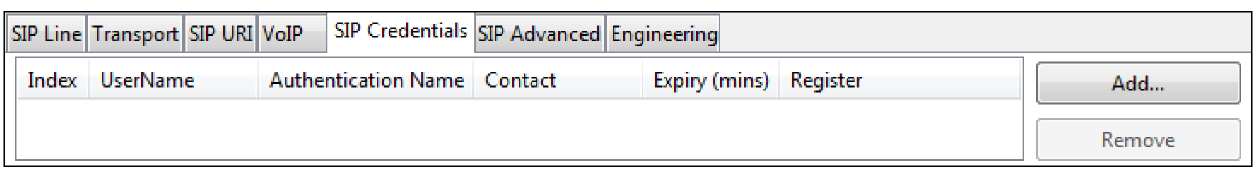
Select the **Transport** tab and set the following:

- Set **ITSP Proxy Address** to the IP address for VTX Connect PABX-IP.
- Set **Use Network Topology Info** to **None** as NAT is not used in this configuration and the Network Topology settings defined in **Section 5.2** are not required.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Send Port** and **Listen Port** to **5060**.

On completion, click the OK button (not shown).

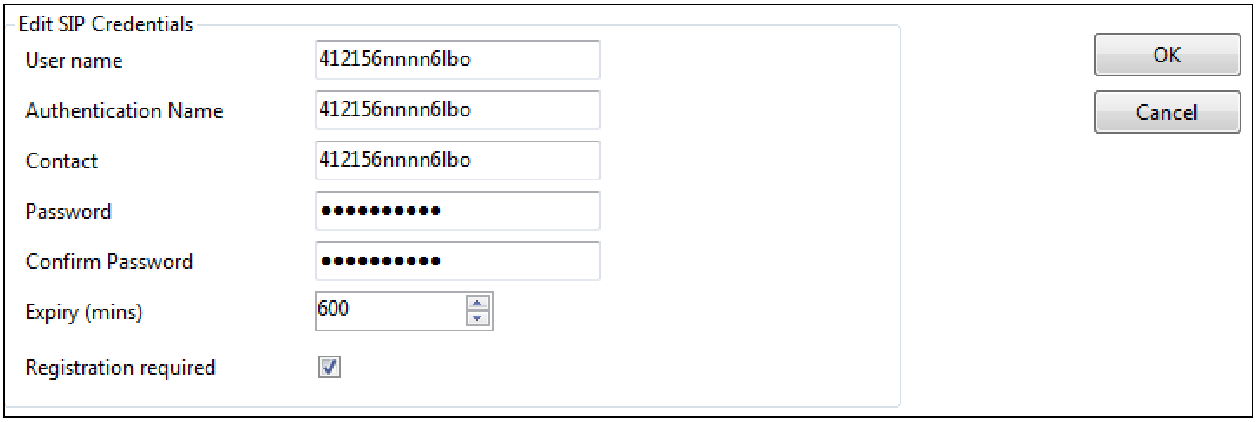


After the SIP line parameters are defined, the SIP credentials used for registration and authorisation on this line must be created. To define SIP credentials, first select the **SIP Credentials** tab. Click the **Add** button and the **New SIP Credentials** area will appear at the bottom of the pane.

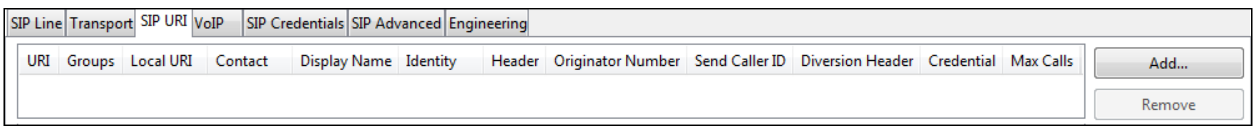


Enter the username and password provided by VTX Services here.

The username can be used for the **User name**, **Authentication Name** and **Contact fields**. The registration timeout is taken from the value provided in the 200 OK Contact header received from the network. The value of **Expiry (mins)** is shown here as an example and reflects the value received from the network during testing. Ensure that the **Registration required** box is checked.



After the SIP credentials are defined, the SIP URIs that Avaya IP Office will receive and transmit on this line must be created. To create a SIP URI entry, first select the **SIP URI** tab. Click the **Add** button and the **New URI** area will appear at the bottom of the pane.

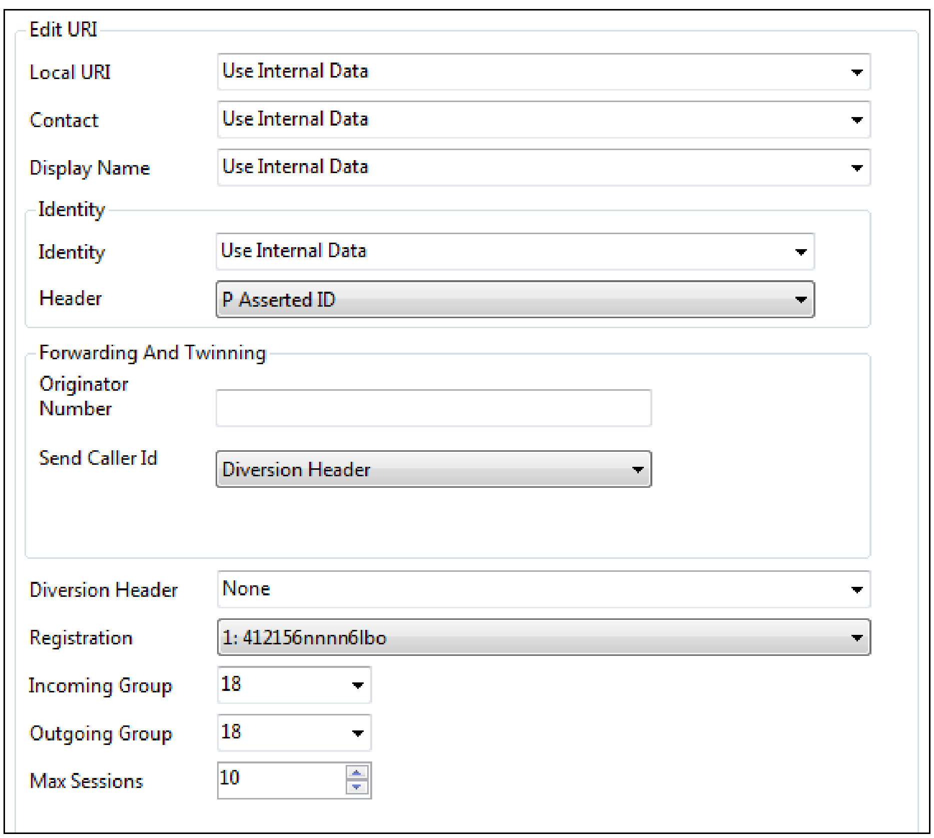


Two SIP URI's are shown in this example, one for calls to and from extensions that have a DDI number assigned to them and the other for calls to services such as Voicemail Collect and the Mobile Twinning FNE.

The SIP URI for calls to and from extensions that have DDI numbers associated with them was created with the parameters shown below.

- Set **Local URI**, **Contact** and **Display Name** to **Use Internal Data**. On incoming calls, this will analyse the Request-Line sent by VTX and match to the SIP settings in the User profile as described in **Section 5.7**. On outgoing calls this will insert the SIP settings in the User profile into the relevant headers in the SIP messages.
- Leave the **Originator Number** for **Forwarding and Twinning** blank so that the originating number is sent as the calling party number. Select **Diversion Number** as the **Send Caller ID** value to ensure that the DDI number assigned to the forwarding extension is sent in the Diversion header.
- Select the previously defined credentials in the **Registration** drop down menu.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. For the compliance test, a new incoming group **18** was defined that was associated to a single line (line 18).
- Associate this line with an unused outgoing line group by entering a line group number in the **Outgoing Group** field. For the compliance test, a new outgoing group **18** was defined that was associated to a single line (line 18)
- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Leave other fields at default values.

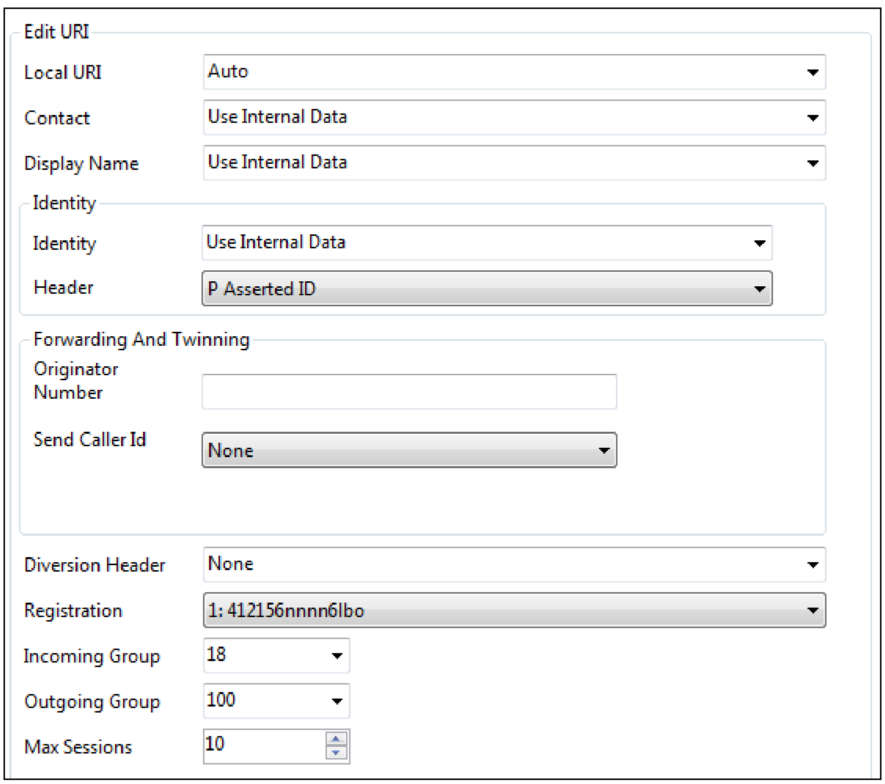
On completion, click the **OK** button (not shown).



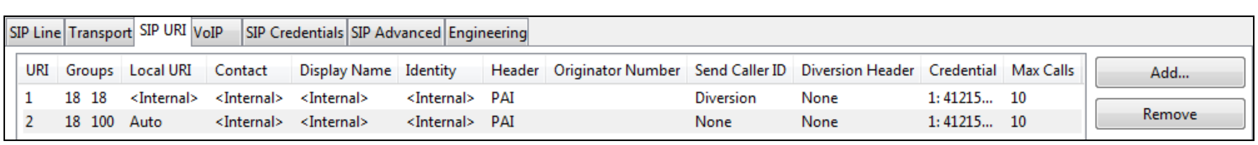
The SIP URI for calls to services such as Voicemail Collect and the Mobile Twinning FNE was created with the parameters shown below:

- Set **Local URI** to **Auto**. This will accept incoming calls that do not match any SIP details defined for the Users, specifically the Voicemail Collect and Mobile FNE services. If a check is required on these calls, enter the specific DDI numbers assigned to the services in this field.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. For the compliance test, a new incoming group **21** was defined that was associated to a single line (line 21).
- Associate this line with an unused outgoing line group as this URI is defined for incoming calls only. A value of **0** was used for compliance testing.
- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Leave other fields at default values.

On completion, click the **OK** button.

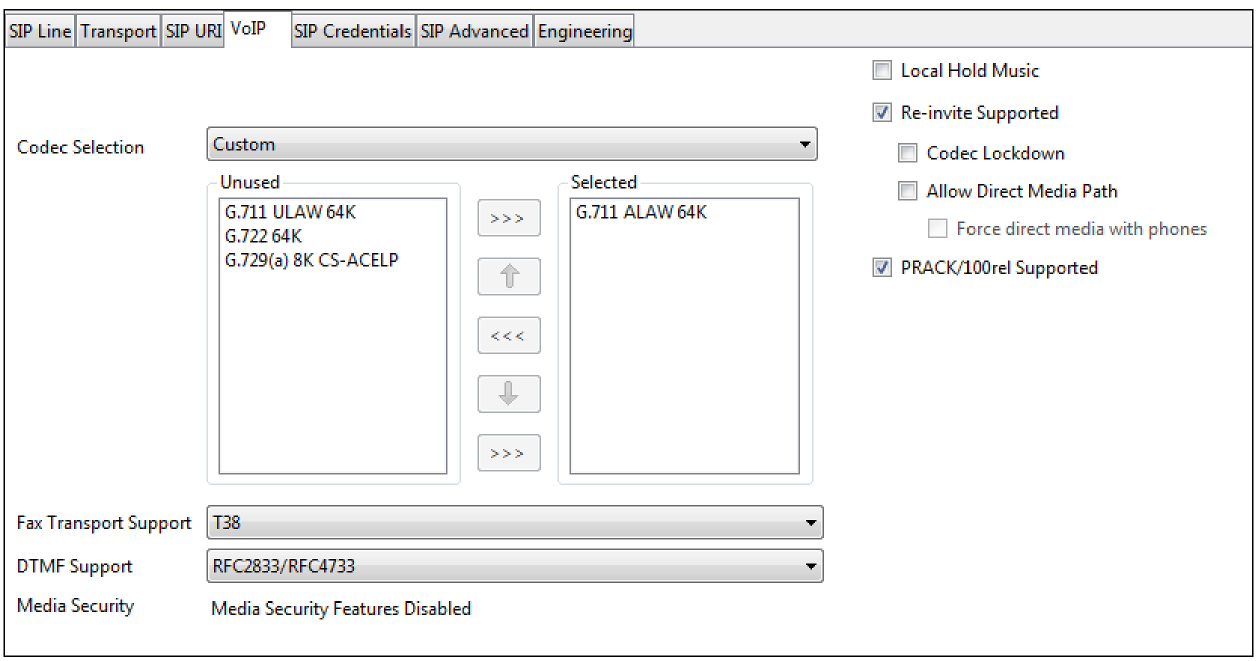


The following screenshot shows the completed configuration:



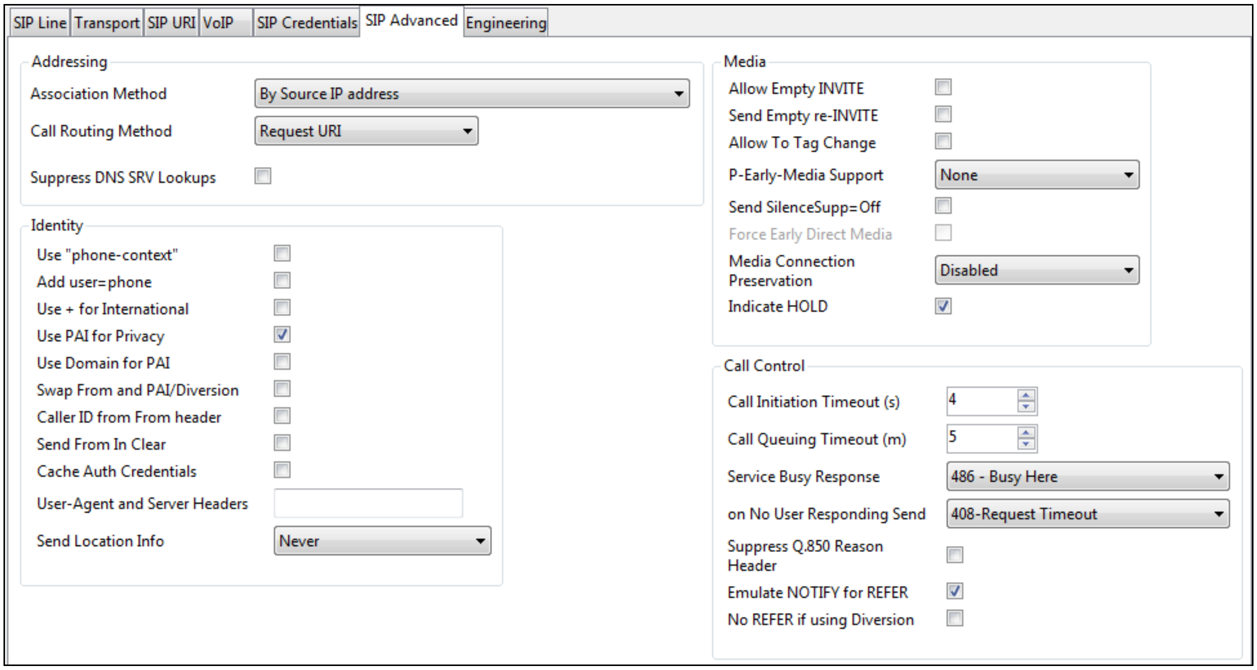
Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- In **Section 5.4**, system default codecs were defined. If any other codec combination is required for this SIP Line, select **Custom** in the **Codec Selection** drop down menu.
- Highlight codecs in the **Unused** box that are to be used on this line and click on the right arrows to move them to the **Selected** box.
- Highlight codecs in the **Selected** box that are not to be used and click on the left arrows to move them to the **Unused** box.
- Highlight codecs in the **Selected** box and use the up and down arrows to change the priority order of the offered codecs if required, for testing with VTX Connect PABX-IP, only **G.711 ALAW 64K** was used.
- Select **T38** in the **Fax Transport Support** drop down menu.
- Select **RFC2833/RFC4733** in the **DTMF Support** drop down menu. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Check the **Re-invite Supported** box to allow for codec re-negotiation in cases where the target of the incoming call or transfer does not support the codec originally negotiated.
- Leave **Allow Direct Media Path** unchecked as direct media cannot be used where the SIP trunk is in a different network to the IP Office endpoints.
- Check the **PRACK/100rel Supported** box if early media is required. This was checked during compliance testing.
- On completion, click the **OK** button (not shown).



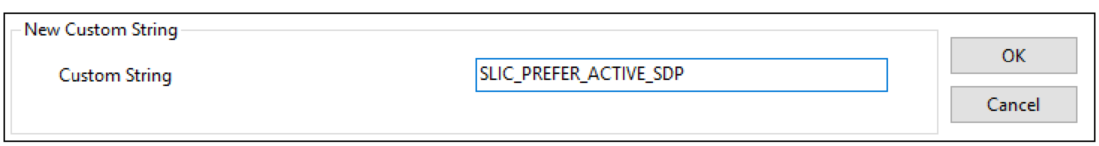
Select the **SIP Advanced** tab and set the following:

- Check the **Use PAI for Privacy** box to send the calling party number for outbound calls with CLI Restricted in the P-Asserted-Identity header.
- Test calls were made with **Indicate HOLD** checked. While it is desirable to indicate in signalling using SIP re-INVITE messages that a call has been placed on hold or taken off hold, it was not necessary for effective operation of the IP Office with VTX Connect PABX-IP.

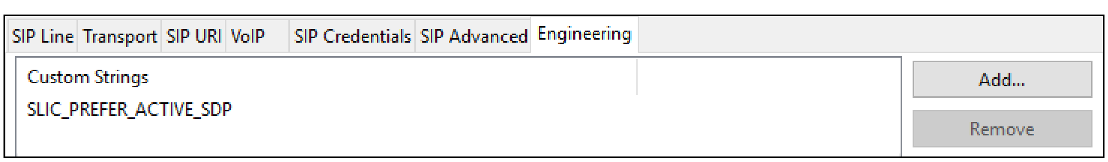


During compliance testing, an issue was found with incorrect allocation of RTP port numbers on incoming T.38 fax calls as described in **Section 2.2**. To change the order of the media lines in the SDP so that the active media line (T.38) is specified first, use a SIP Line Custom String as follows:

- Select the **Engineering** tab and click on **Add** (not shown).
- In the dialogue box, type **SLIC_PREFER_ACTIVE_SDP**.
- Click on **OK**.



The following screenshot shows the completed Custom String:

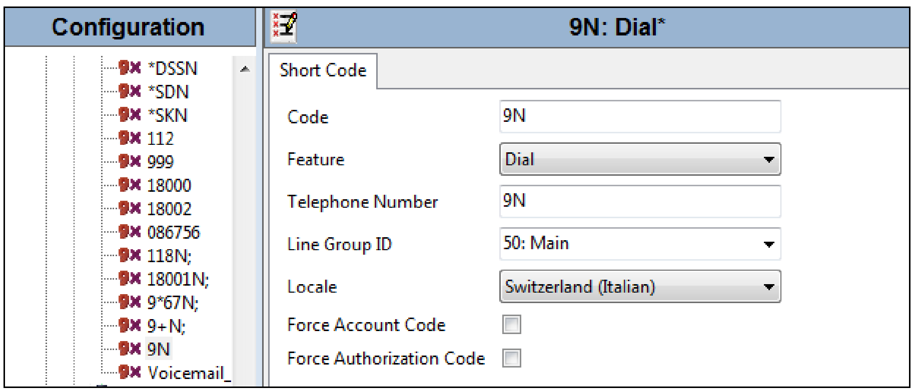


Note: It is advisable at this stage to save the configuration as described in **Section 5.11** to make the Line Group ID defined in **Section 5.5** available.

5.6. Short Codes

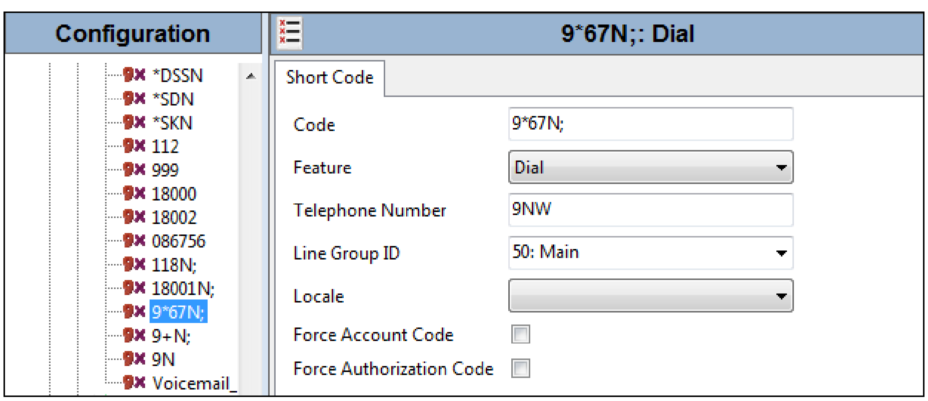
Define a short code to route outbound traffic to the SIP line. To create a short code, right-click **Short Code** in the Navigation pane and select **New**. On the **Short Code** tab in the Details pane, configure the parameters as shown in the example below for public numbers.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon.
- The example shows **9N**; which will be invoked when the user dials 9 followed by a public number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **9N** so that the call is passed to the ARS function with the dialled number unchanged.
- Set the **Line Group Id** to the ARS route number described in **Section 5.10**.
- On completion, click the **OK** button (not shown).



A further example is shown of a short code to route numbers where CLI is to be withheld:

- The **Code** is **9*67N** which is an outbound call prefixed with *67 which indicates that CLI is to be withheld.
- Set **Telephone Number** to **+NW** which removes the access code and the *67 and inserts the dialled number with a "W" suffix that causes Avaya IP Office to withhold the CLI.

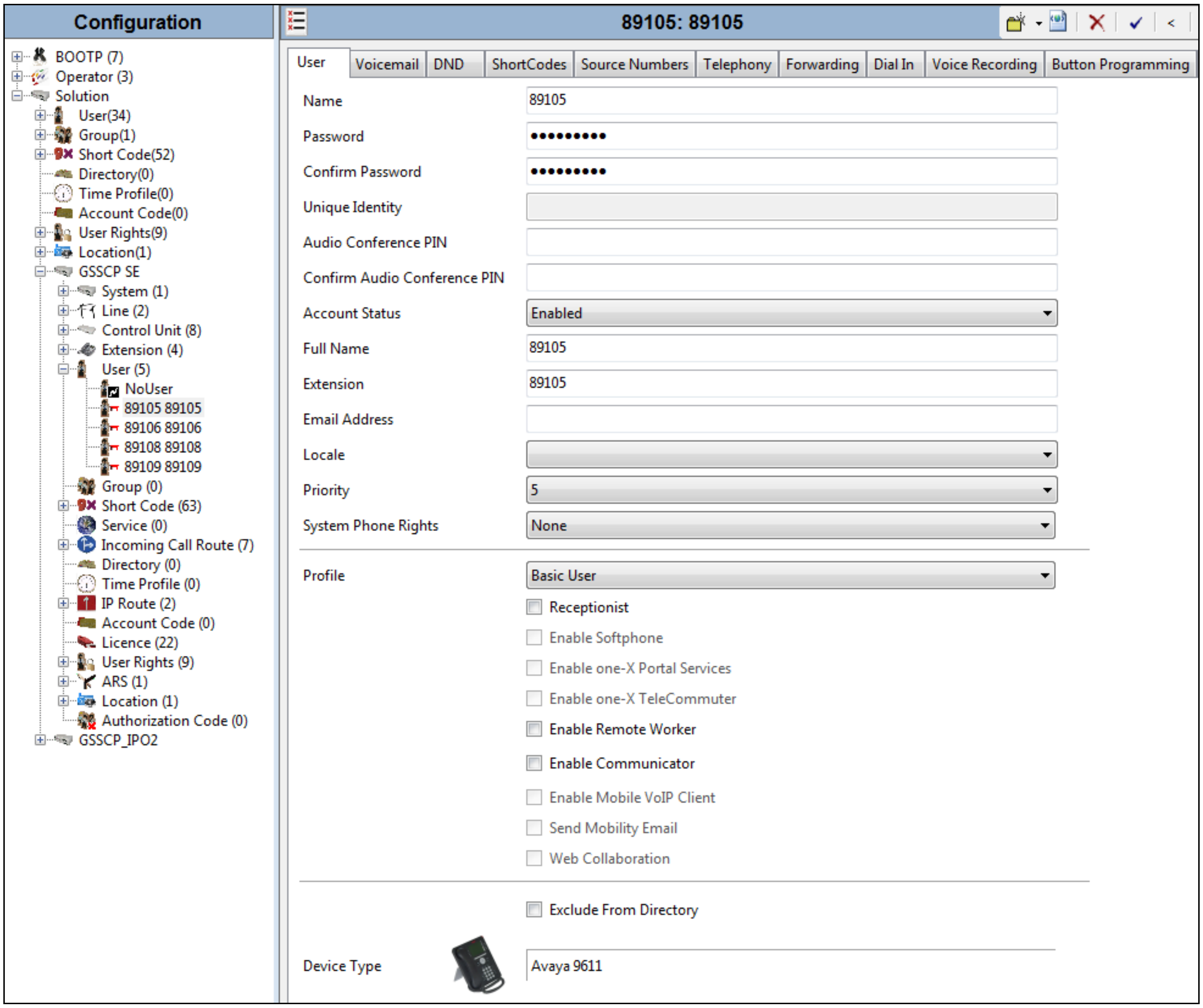


5.7. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.5**. To configure these settings, first navigate to **User** in the Navigation pane. Select the **User** tab if any changes are required.

The following example shows the configuration required for an H.323 Endpoint.

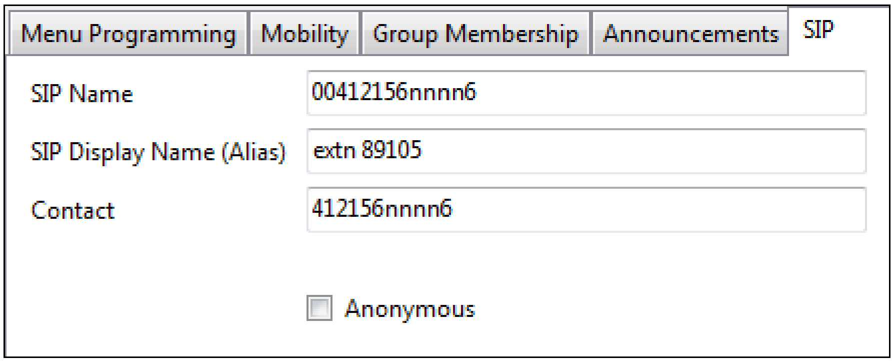
- Change the **Name** of the User if required.
- Set the **Password** and **Confirm Password**.
- Select the required profile from the **Profile** drop down menu. **Basic User** is commonly used; **Power User** can be selected for SIP softphone, WebRTC and Remote Worker endpoints.



SIP endpoints require setting of the **SIP Registrar Enable** as described in **Section 5.2**.

Next select the **SIP** tab in the Details pane. To reach the **SIP** tab click the right arrow on the right hand side of the Details pane until it becomes visible. 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. These fields should be set to the DDI numbers assigned to the enterprise from VTX in international format.

In the example below, one of the DDI numbers in the test range is used, though some of the digits have been obscured. On completion, click the **OK** button (not shown).



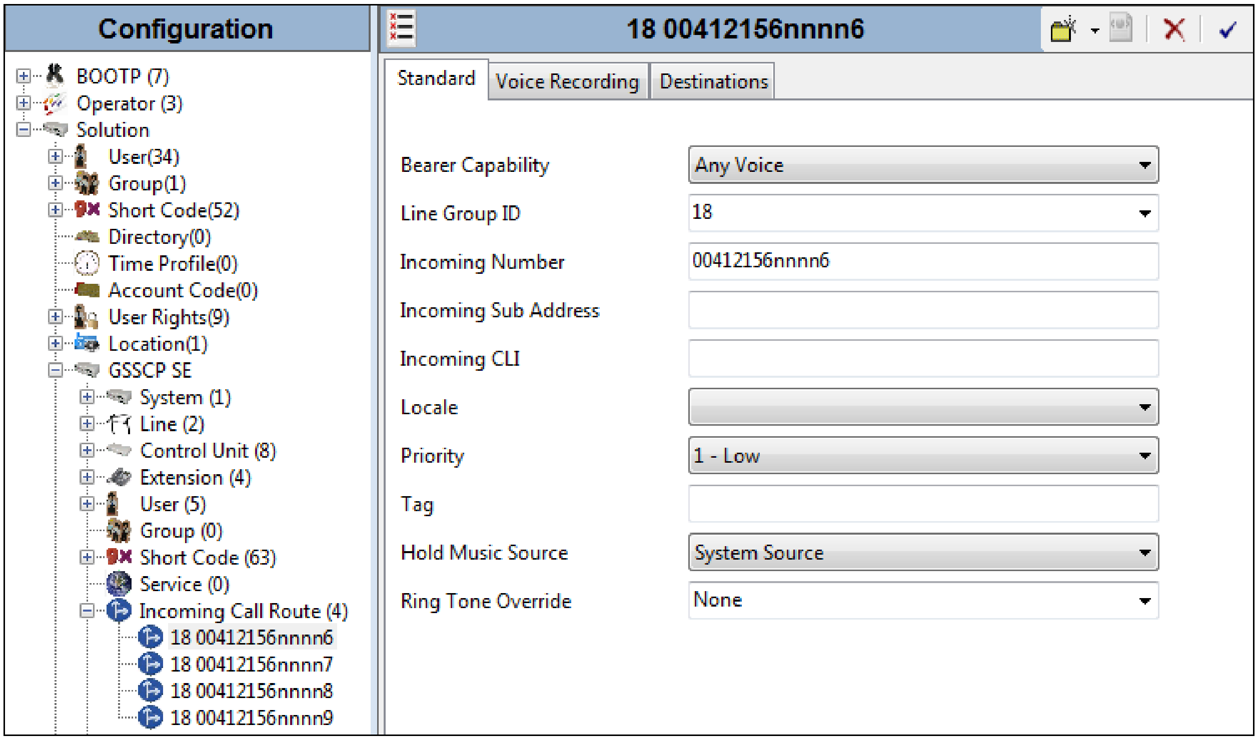
Note: The **Anonymous** box can be used to restrict Calling Line Identity (CLID).

5.8. Incoming Call Routing

An incoming call route maps an inbound DDI number on a specific line to an internal extension. To create an incoming call route, right-click **Incoming Call Route** in the Navigation pane and select **New**, (not shown).

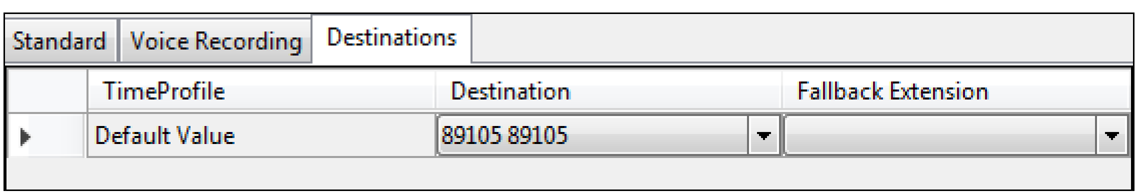
On the **Standard** tab of the Details pane, enter the parameters as shown below:

- Set the **Bearer Capability** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.5**.
- Set the **Incoming Number** to the incoming number that this route should match on. Matching is right to left.
- Default values can be used for all other fields.



Note: A number of digits of the DDI have been obscured. Number format for incoming calls is international.

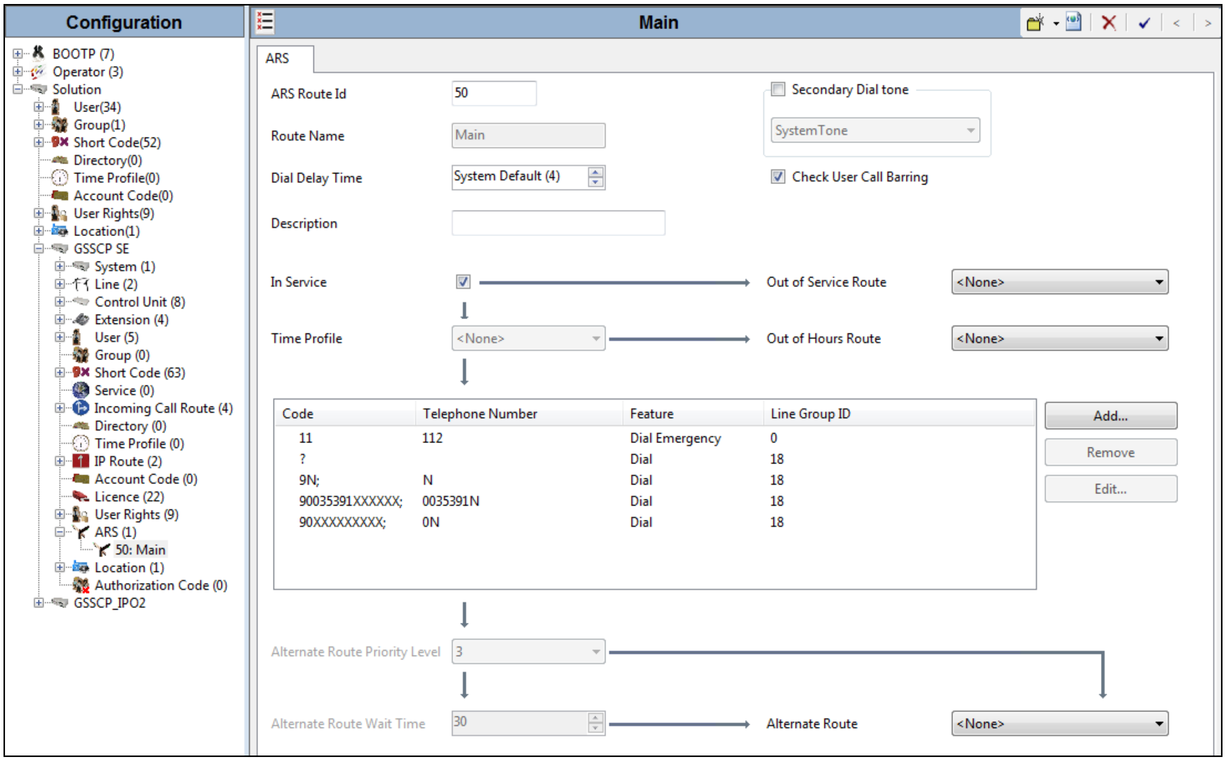
On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number on line 18 are routed to extension 89105.



Note: Calls coming in to destinations not associated with an extension such as Voice Mail and FNE also appear on line 18 in this configuration. The destinations are defined as the short codes for Voicemail Collect and FNE Service.

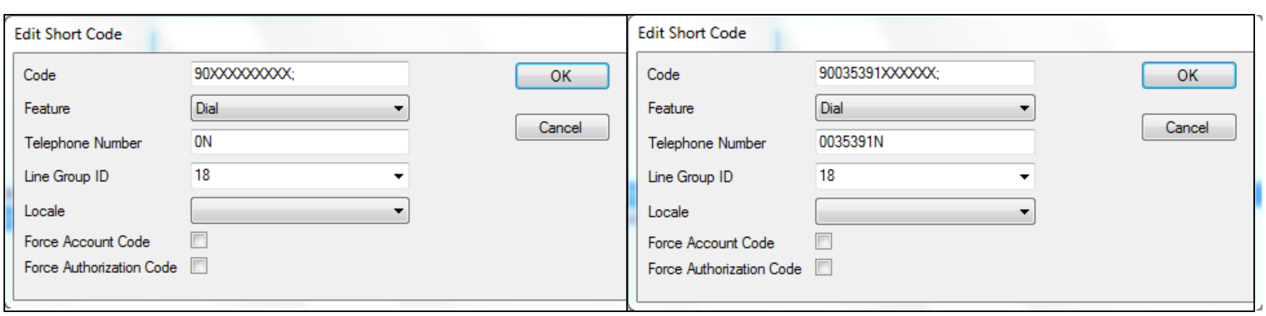
5.9. ARS

The Main ARS route exists by default and requires editing. Select the ARS **Main** route and click on **Add**.



Define numbers as required. An example for national numbers is as follows:

- Define the Short **Code**, the example shows both a 10 digit national number and an international number with country code and city code prefixed with **9** for an outside line. Note that **X** indicates any digit and **;** causes the system to wait for the full number to be dialled.
- Select **Dial** in the **Feature** drop down menu.
- Define the **Telephone Number** without the **9** which removes it and sends the number as dialled. All **X** characters can be replaced with a single **N**.
- Select the **Line Group ID** defined in the SIP Line URI described in **Section 5.5**. During testing this was **18** for the SIP Trunk. Click on **OK**



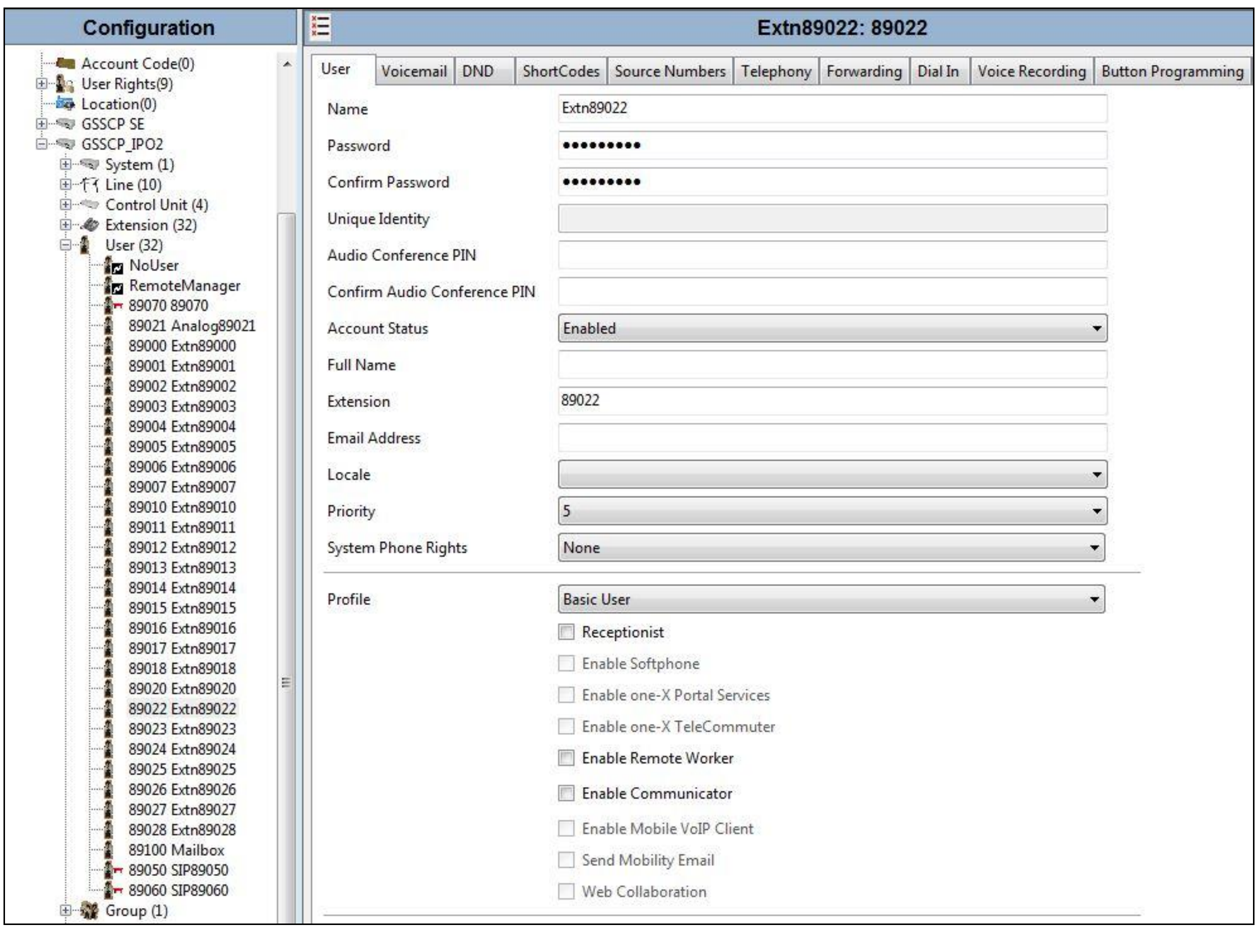
5.10. T.38 Fax

At Release 10, T.38 Fax is supported on IP Office Server Edition when using an IP Office Expansion (500 V2). The VTX Connect PABX-IP testing was carried out using this configuration with only the analogue extension for the fax machine on the Expansion. In this configuration, the T.38 fax settings are configured on the SIP line between the Expansion and the Server.

5.10.1. Analogue User

To configure the settings for the fax User, first navigate to **User** in the Navigation pane for the Expansion. In the test environment, the 500V2 Expansion is called **GSSCP_IPO2**. Select the **User** tab. The following example shows the configuration required for an analogue Endpoint.

- Change the **Name** of the User if required.
- The **Password** and **Confirm Password** fields are set but are not required for analogue endpoints.
- Select the required profile from the **Profile** drop down menu. **Basic User** is sufficient for fax.



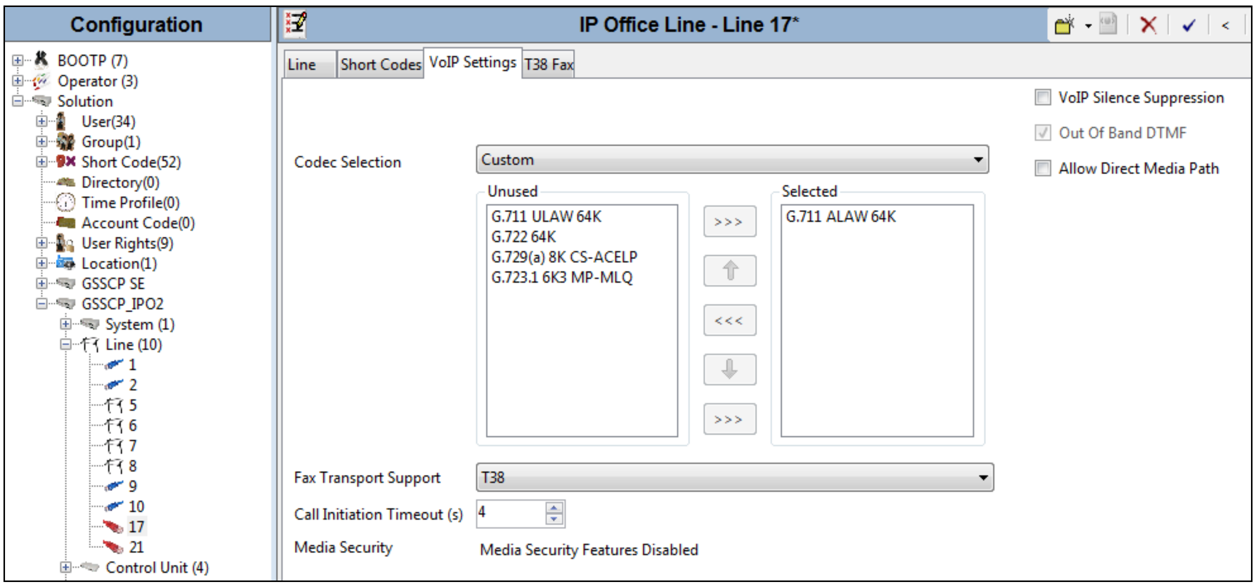
Configure other settings as described in **Section 5.7**

5.10.2. T.38 Fax Settings

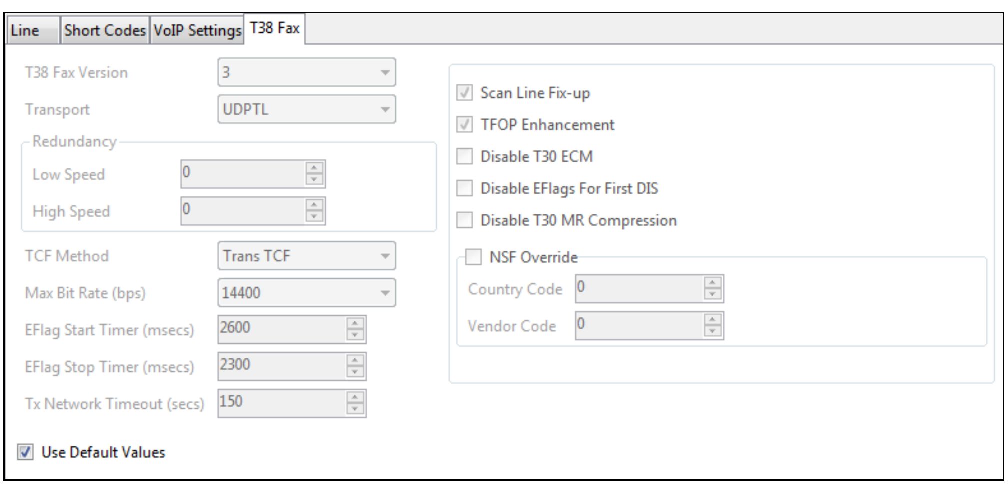
The T.38 Fax settings are defined on the SIP Line between the Expansion and the Server. Note that the VoIP settings for T.38 Fax are required in three places in this configuration:

- The SIP Line for the VTX Connect PABX-IP as described in **Section 5.5**.
- The IP Office Line between the Server and the Expansion on the Expansion.
- The IP Office Line between the Server and the Expansion on the Server.

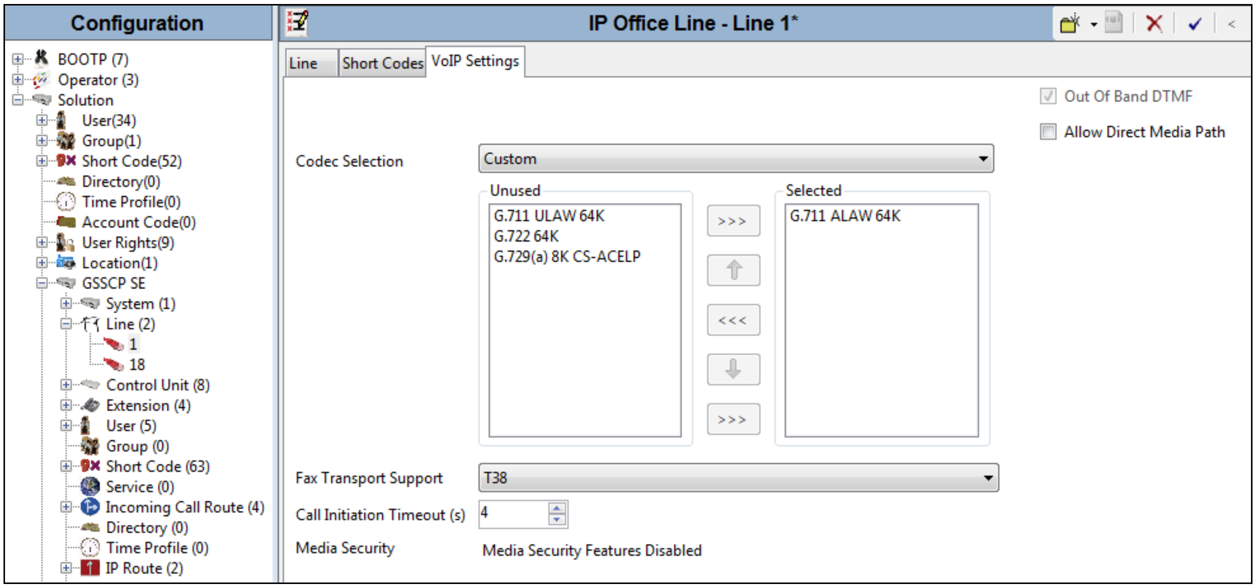
In all the above cases, the **Fax Transport Support** was set to **T38**. The following screenshot shows the VoIP Settings for the IP Office Line between the Server and the Expansion on the Expansion:



The T.38 Fax settings are defined in the T38 Fax tab. This is available on the Expansion only. Default values were used for testing. If changes are required, uncheck **Use Default Values**.



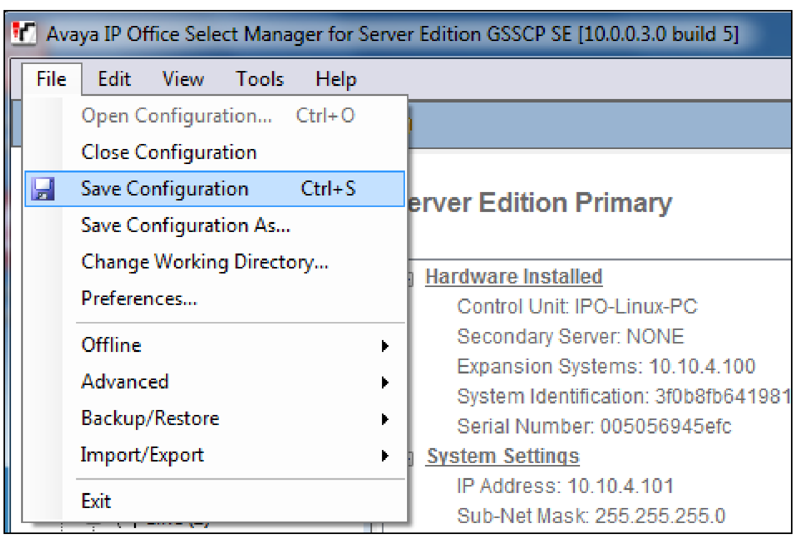
The following shows the **VoIP Settings** tab in the IP Office Line for the Expansion in the Server configuration:



Refer to **Section 5.5.2** for the VoIP Settings on the SIP Line for VTX Connect PABX-IP.

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.

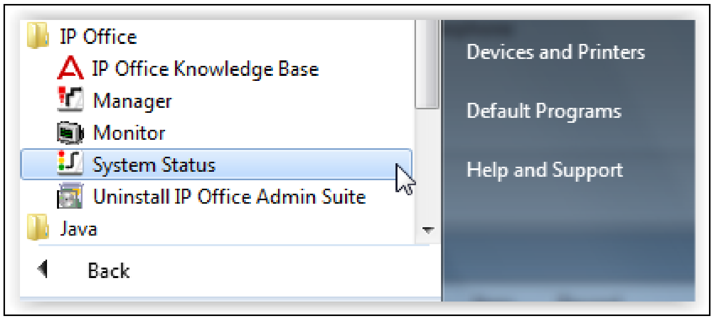


6. Verification Steps

This section includes steps that can be used to verify that the configuration has been done correctly.

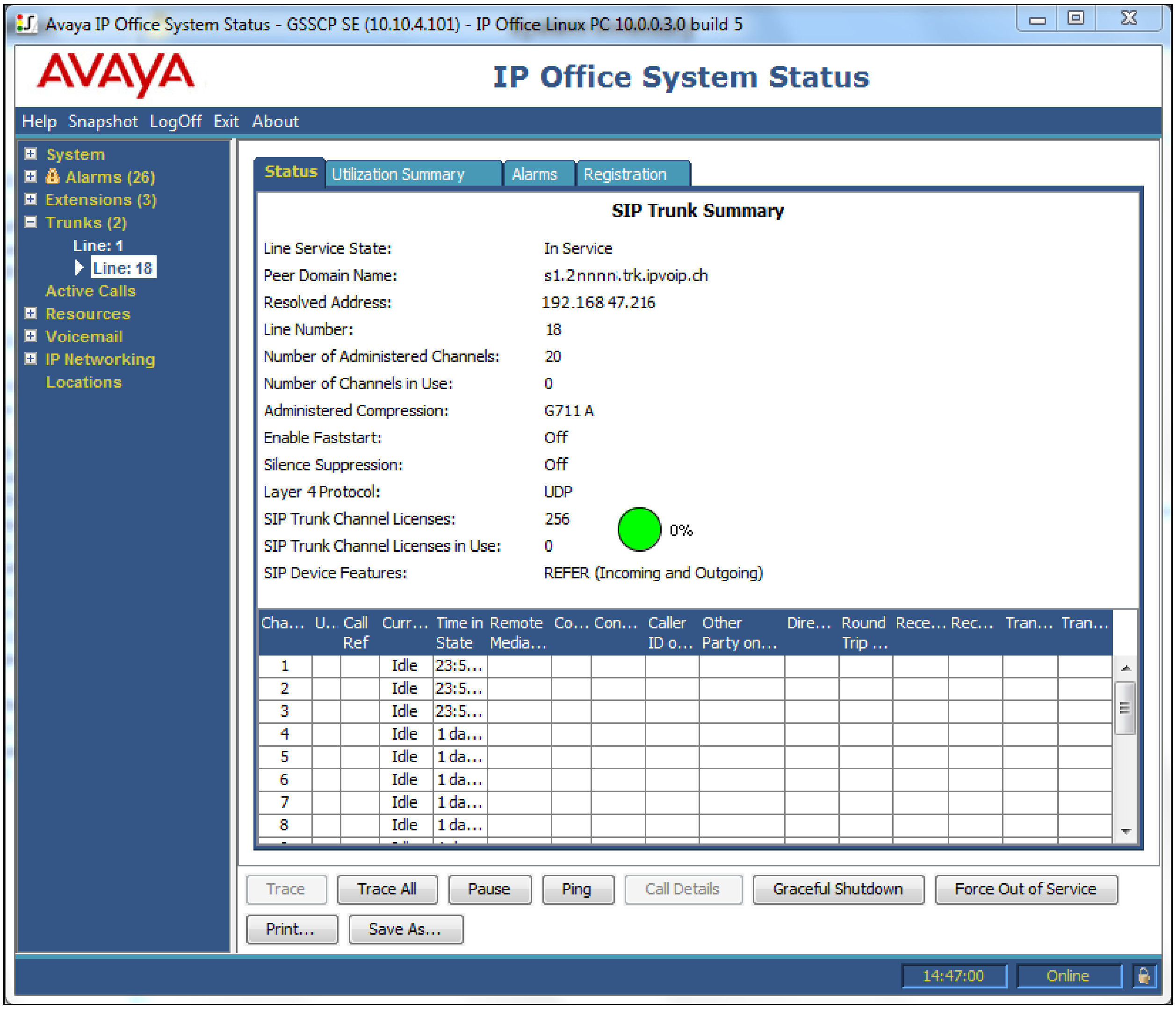
6.1. SIP Trunk status

The status of the SIP trunk can be verified by opening the System Status application. A Windows 7 PC was used for testing and the application was opened by pressing the Start button and selecting **All Programs→ IP Office → System Status**.



Log in to IP Office System Status at the prompt using the **Control Unit IP Address** for the IP Office. The **User Name** and **Password** are the same as those used for IP Office Manager.

From the left hand menu expand **Trunks** and choose the SIP trunk (**21** in this instance). The status window will show the status as being idle and time in state if the Trunk is operational.



7. Conclusion

All tests for VTX Connect PABX-IP were completed. Observations for the testing are listed in **Section 2.2**.

8. Additional References

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

[1] *Avaya IP Office™ Platform Start Here First*, Release 10, August 2016.

[2] *Avaya IP Office™ Platform Server Edition Reference Configuration*, August 2016

[3] *IP Office™ Platform 10.0, Deploying Avaya IP Office™ Platform Servers as Virtual Machines*, Jan 2017

[4] *IP Office™ Platform 10.0, Deploying Avaya IP Office™ Platform IP500 V2*, Jan 2017.

[5] *IP Office™ Platform 10.0 Installing and Maintaining the Avaya IP Office™ Platform Application Server*, Document number 15-601011, Jan 2017.

[6] *Administering Avaya IP Office™ Platform with Web Manager*, Release 10.0, September 2016.

[7] *Administering Avaya IP Office™ Platform with Manager*, Release 10.0, September 2016.

[8] *IP Office™ Platform 10.0 Using System Status*, Document number15-601758, 7th July 2016.

[9] *IP Office™ Platform 10.0 Using IP Office System Monitor*, Document number 15-601019, 25th November 2016.

[10] *Using Avaya Communicator for Windows on IP Office*, Release 10.0, August 2016.

[11] *Avaya Communicator for Web- IP Office™ Platform: User Guide*, October 2016.

[12] *Avaya Communicator for Web- IP Office™ Platform: Administering Guide*, October 2016.

[13] *IP Office™ Platform 10.0 - Third-Party SIP Extension Installation Notes*, June 2016.

[14] *RFC 3261 SIP: Session Initiation Protocol*, http://www.ietf.org/

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.