DevConnect Program

Application Notes for Configuring Avaya IP Office 11.1 with VTX Connect PBX-IP SIP Trunking Service – Issue 1.0

Abstract

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

The VTX Connect PBX-IP SIP Trunk Platform 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 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.

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between the VTX Connect PBX-IP SIP Trunk and Avaya IP Office R11.1. Customers using this Avaya SIP-enabled enterprise solution with VTX Connect PBX-IP 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 R11.1 to connect to the VTX Connect PBX-IP SIP Platform. 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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and the VTX Connect PBX-IP SIP platform do not include use of any specific encryption features.

2.1. Interoperability Compliance Testing

Avaya IP Office was connected to the VTX Connect PBX-IP SIP Trunk. 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, 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 including H.323, 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.
- Incoming and Outgoing PSTN calls to/from Avaya Workplace Client for Windows soft phone.
- Calls using the G.711A and G.729 codecs.
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using T.38 transmissions.
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls.
- Various call types including: local, long distance, international, toll free (outbound) and directory assistance.
- Caller ID presentation and Caller ID restriction.
- User features such as hold and resume, and conference.
- Blind and Consultative call transfer to PSTN.
- Off-net call forwarding and mobile twinning.

2.2. Test Results

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

- No inbound toll-free numbers were tested, however routing of inbound DDI numbers and the relevant number translation was successfully tested.
- Access to Emergency Services was not tested as no test call had been booked by the Service Provider with the Emergency Services Operator.
- 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.

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:
https://www.vtx.ch/fr/support/contact
https://www.vtx.ch/de/support/kontakt

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the VTX Connect PBX-IP SIP Trunk. Located at the enterprise site is an Avaya IP Office Server Edition and Avaya IP Office 500 V2 as an expansion. Endpoints include Avaya 1600 Series IP Telephones (with H.323 firmware), Avaya 9600 Series IP Telephones (with H.323 firmware), Avaya 1140e SIP Telephones, Avaya J179 Series SIP Phones, Avaya 1400 Series Digital Deskphones, Analog Telephone and a fax machine. The site also has a Windows 10 PC running Avaya IP Office Manager to configure the Avaya IP Office as well as Avaya Workplace Client for Windows for softphone testing.

For security purposes, public IP addresses have been changed and any PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

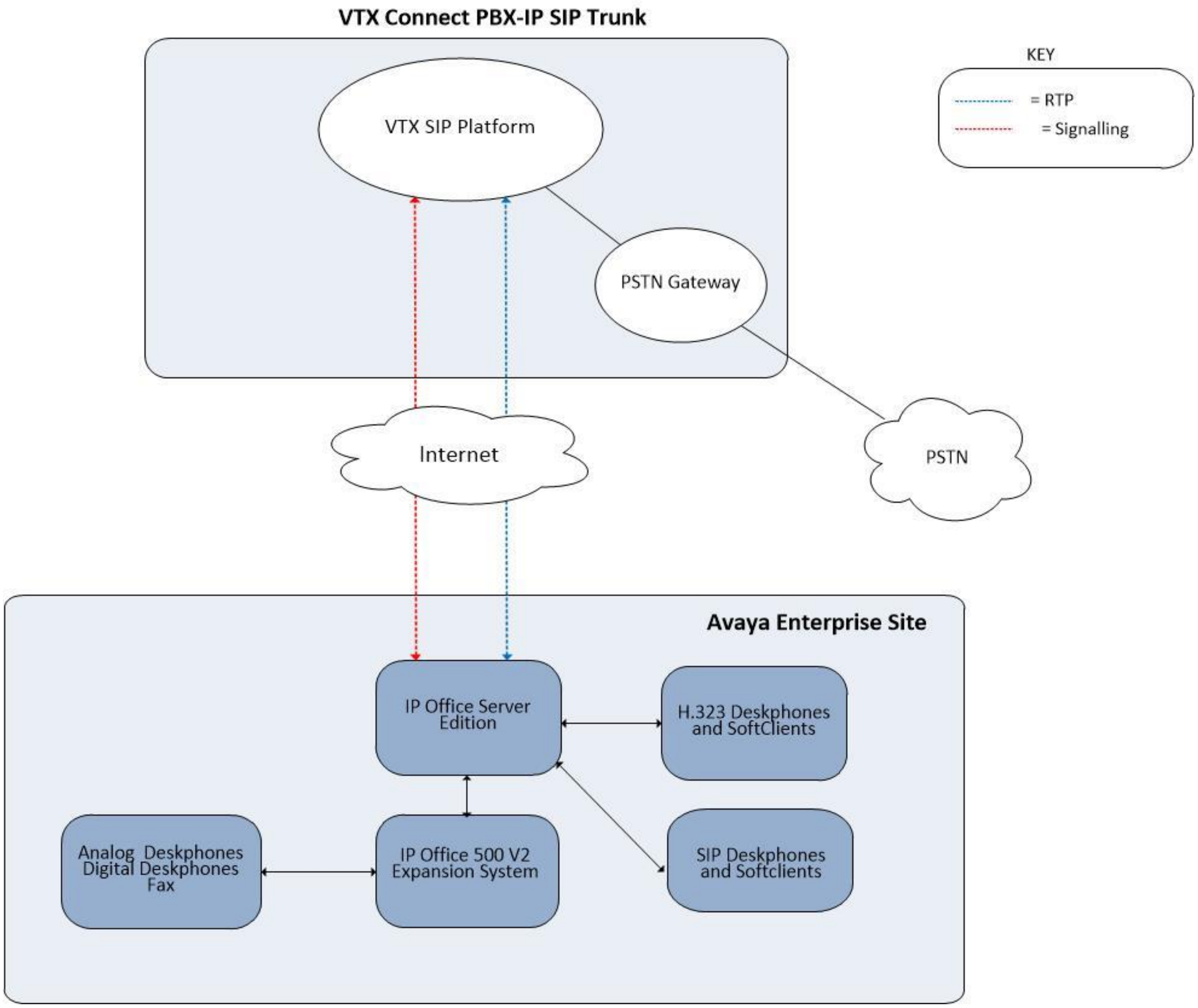


Figure 1: VTX Connect PBX-IP to Avaya IP Office Topology

4. Equipment and Software Validated

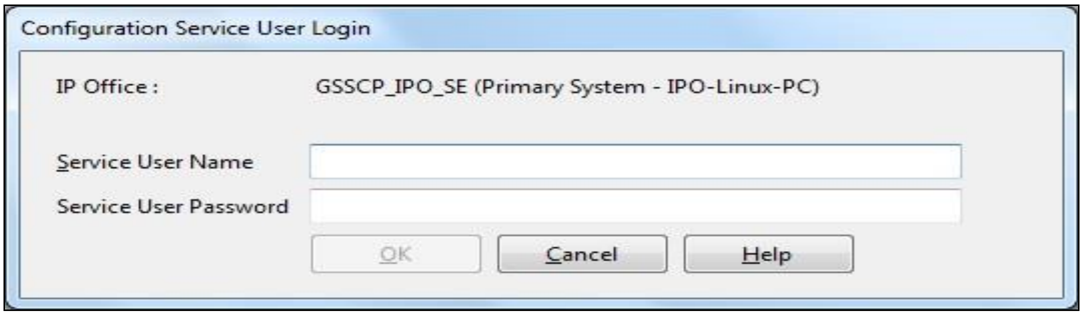
The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya	
Avaya IP Office Server Edition	Version 11.1.3.0.0 build 23
Avaya IP Office 500 V2	Version 11.1.3.0.0 build 23
Avaya Voicemail Pro Client	Version 11.1.3.0.0
Avaya IP Office Manager	Version 11.1.3.0.0 build 23
Avaya 1608 Phone (H.323)	1.3.12
Avaya 9611G Series Phone (H.323)	6.8.3
Avaya 9608 Series Phone (H.323)	6.8.3
Avaya J179 IP Phone (SIP)	4.0.10
Avaya Workplace for Windows (SIP)	3.34.1
Avaya 1140e (SIP)	FW: 04.04.23.00.bin
Avaya 98390 Analogue Phone	N/A
VTX	
VTX CONNECT PBX IP	VTX CONNECT PBX-IP <X> (<X>: depending on number of channels: 4, 8, 15, 30, 60, etc.)
Platform: Communi5, VAS C5	7.4 udp7
SBC: Audiocodes	Mediant VE SBC, Version 7.40A.250.541

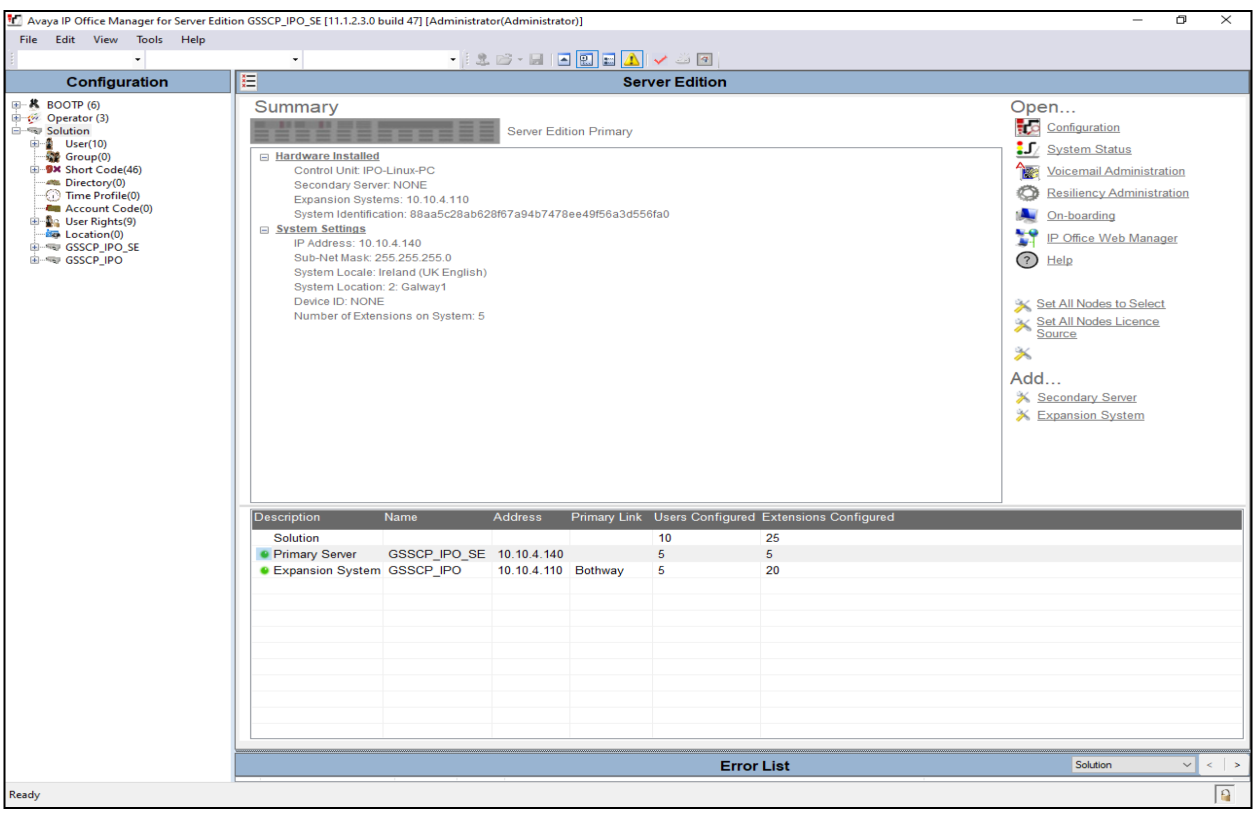
Note – Testing was performed with IP Office Server Edition with Expansion IP Office 500 V2 R11.1. Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition. **Note:** that IP Office 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 the VTX Connect PBX-IP SIP platform. 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**, select the appropriate Avaya IP Office system from the pop-up window and log in with the appropriate credentials.

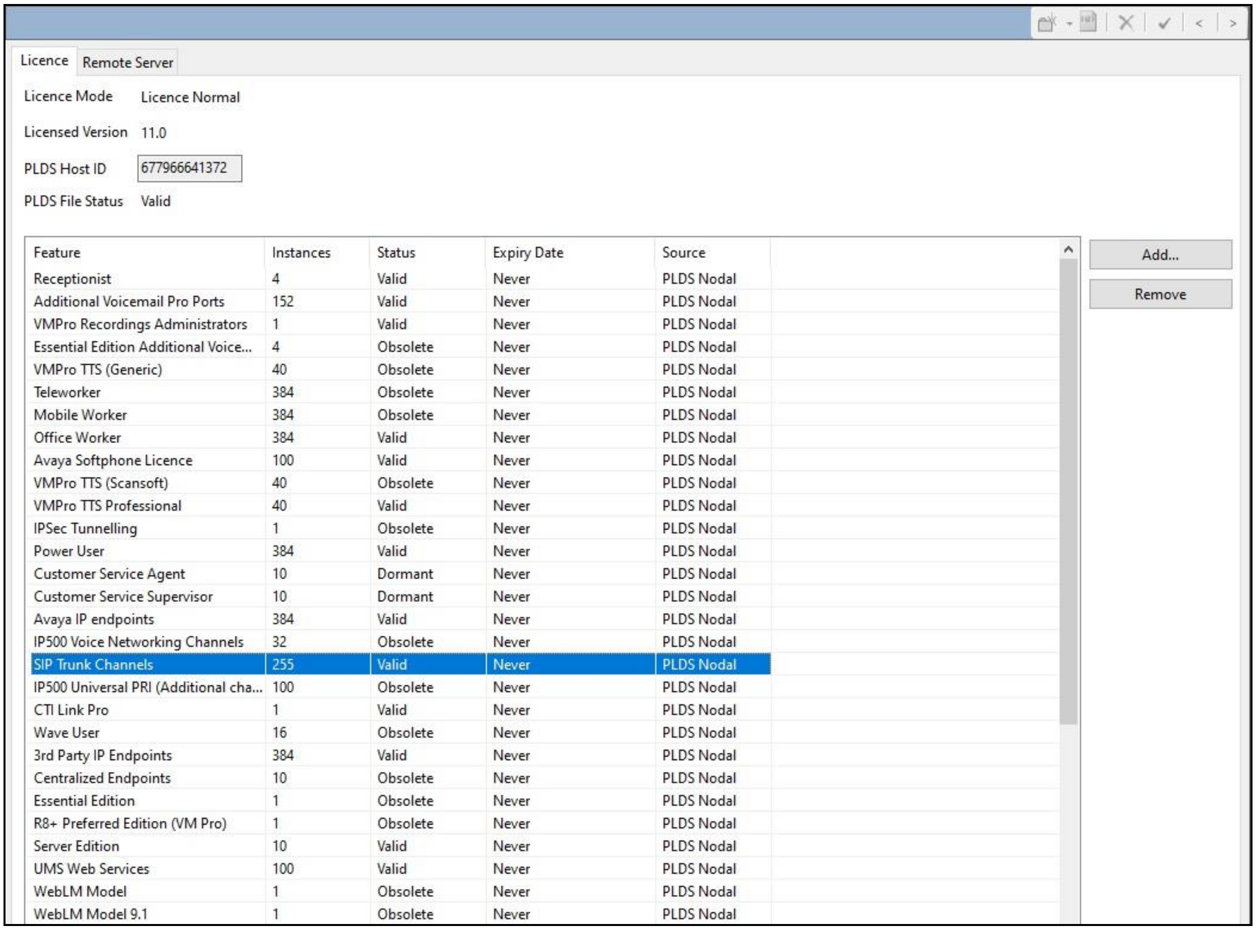


A management window will appear similar to the one in the next section. All the Avaya IP Office configurable components are shown in the left pane known as the Navigation Pane. The pane on the right is the Details Pane. These panes will be referenced throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the Service Provider 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.



Licence | Remote Server

Licence Mode: Licence Normal

Licensed Version: 11.0

PLDS Host ID: 677966641372

PLDS File Status: Valid

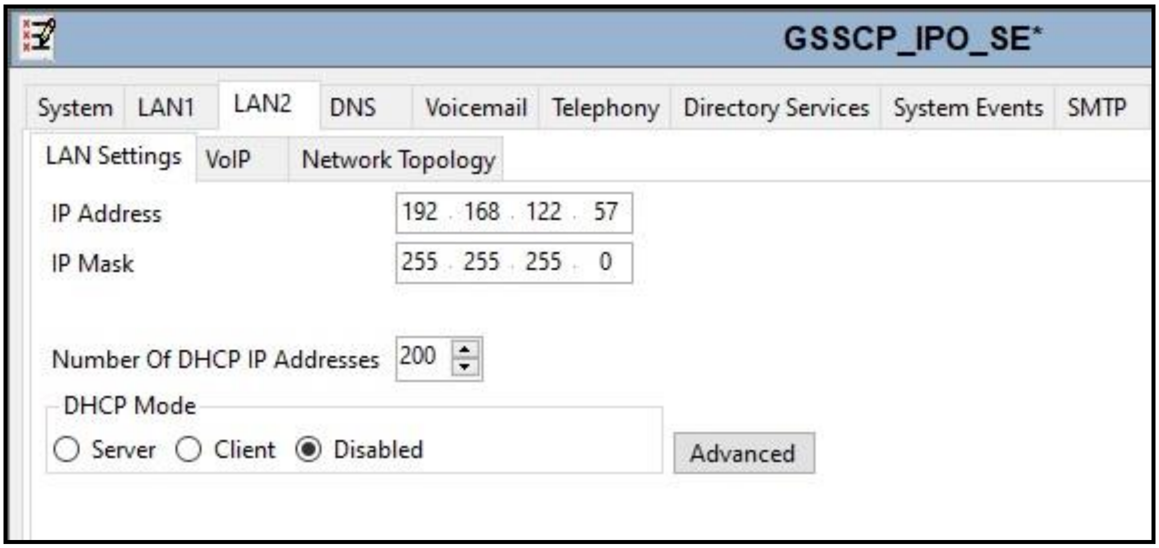
Feature	Instances	Status	Expiry Date	Source
Receptionist	4	Valid	Never	PLDS Nodal
Additional Voicemail Pro Ports	152	Valid	Never	PLDS Nodal
VMPro Recordings Administrators	1	Valid	Never	PLDS Nodal
Essential Edition Additional Voice...	4	Obsolete	Never	PLDS Nodal
VMPro TTS (Generic)	40	Obsolete	Never	PLDS Nodal
Teleworker	384	Obsolete	Never	PLDS Nodal
Mobile Worker	384	Obsolete	Never	PLDS Nodal
Office Worker	384	Valid	Never	PLDS Nodal
Avaya Softphone Licence	100	Valid	Never	PLDS Nodal
VMPro TTS (Scansoft)	40	Obsolete	Never	PLDS Nodal
VMPro TTS Professional	40	Valid	Never	PLDS Nodal
IPSec Tunnelling	1	Obsolete	Never	PLDS Nodal
Power User	384	Valid	Never	PLDS Nodal
Customer Service Agent	10	Dormant	Never	PLDS Nodal
Customer Service Supervisor	10	Dormant	Never	PLDS Nodal
Avaya IP endpoints	384	Valid	Never	PLDS Nodal
IP500 Voice Networking Channels	32	Obsolete	Never	PLDS Nodal
SIP Trunk Channels	255	Valid	Never	PLDS Nodal
IP500 Universal PRI (Additional cha...	100	Obsolete	Never	PLDS Nodal
CTI Link Pro	1	Valid	Never	PLDS Nodal
Wave User	16	Obsolete	Never	PLDS Nodal
3rd Party IP Endpoints	384	Valid	Never	PLDS Nodal
Centralized Endpoints	10	Obsolete	Never	PLDS Nodal
Essential Edition	1	Obsolete	Never	PLDS Nodal
R8+ Preferred Edition (VM Pro)	1	Obsolete	Never	PLDS Nodal
Server Edition	10	Valid	Never	PLDS Nodal
UMS Web Services	100	Valid	Never	PLDS Nodal
WebLM Model	1	Obsolete	Never	PLDS Nodal
WebLM Model 9.1	1	Obsolete	Never	PLDS Nodal

Add... | Remove

5.2. LAN2 Settings

In an Avaya IP Office, the LAN2 tab settings correspond to the Avaya IP Office WAN port (public network side) and the LAN1 tab settings correspond to the LAN port (private network side).

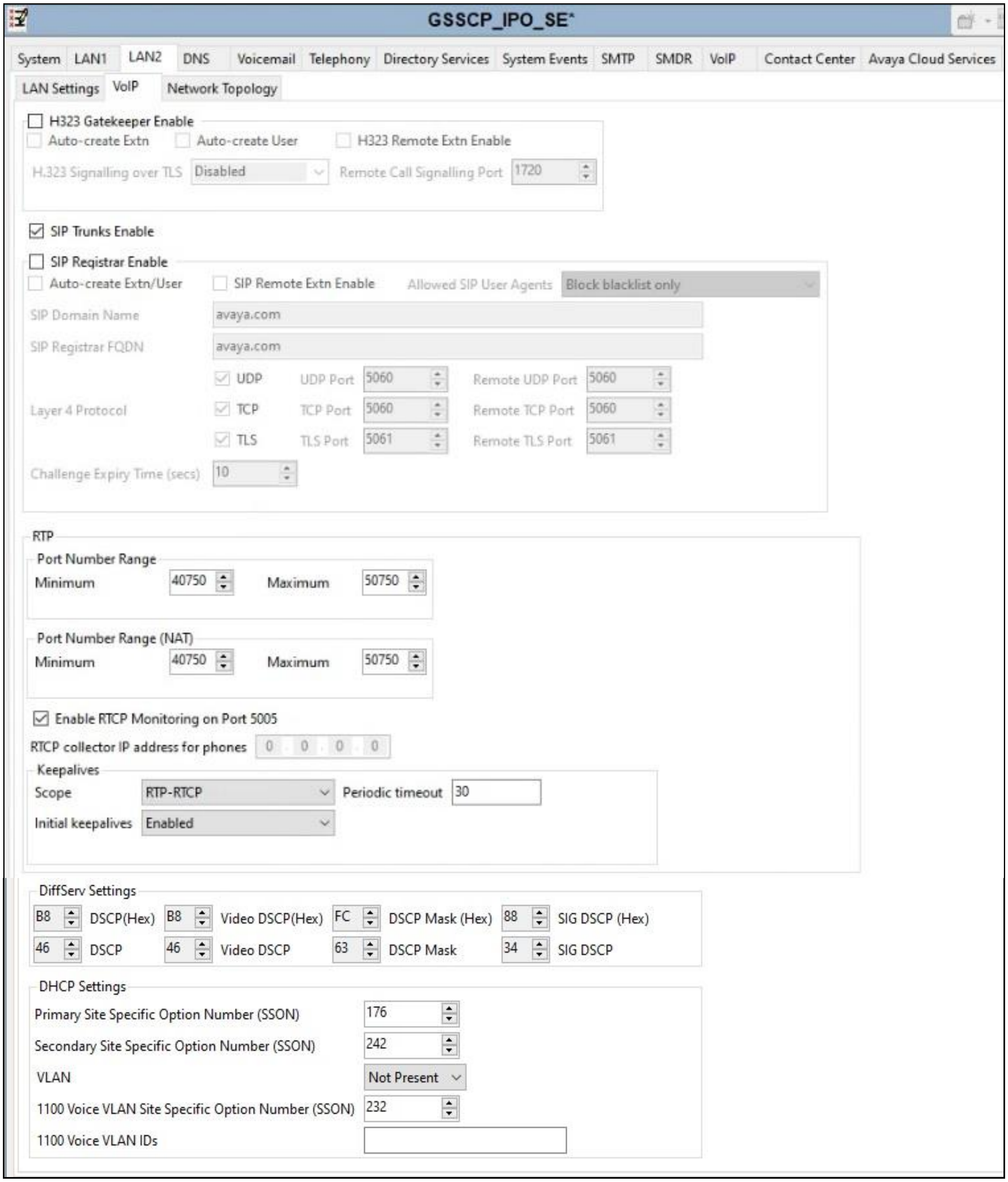
In the test configuration, the LAN2 port was used to connect the Avaya IP Office to the external internet. To access the LAN2 settings, first navigate to **System → GSSCP_IPO_SE** in the Navigation Pane where GSSCP_IPO_SE is the name of the IP Office. Navigate to the **LAN2 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are the public 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 Avaya Communicator along with any other SIP endpoint is to be used, the **SIP Registrar Enable** box must also be checked. The **Domain Name** has been set to the customer premises equipment domain "**avaya.com**". If the **Domain Name** is left at the default blank setting, SIP registrations may use the IP Office LAN2 IP Address. All other parameters shown are default values.

The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Set **Scope** to **RTP-RTCP** and **Initial keepalives** to **Enabled** and **Periodic timeout** to **30**.

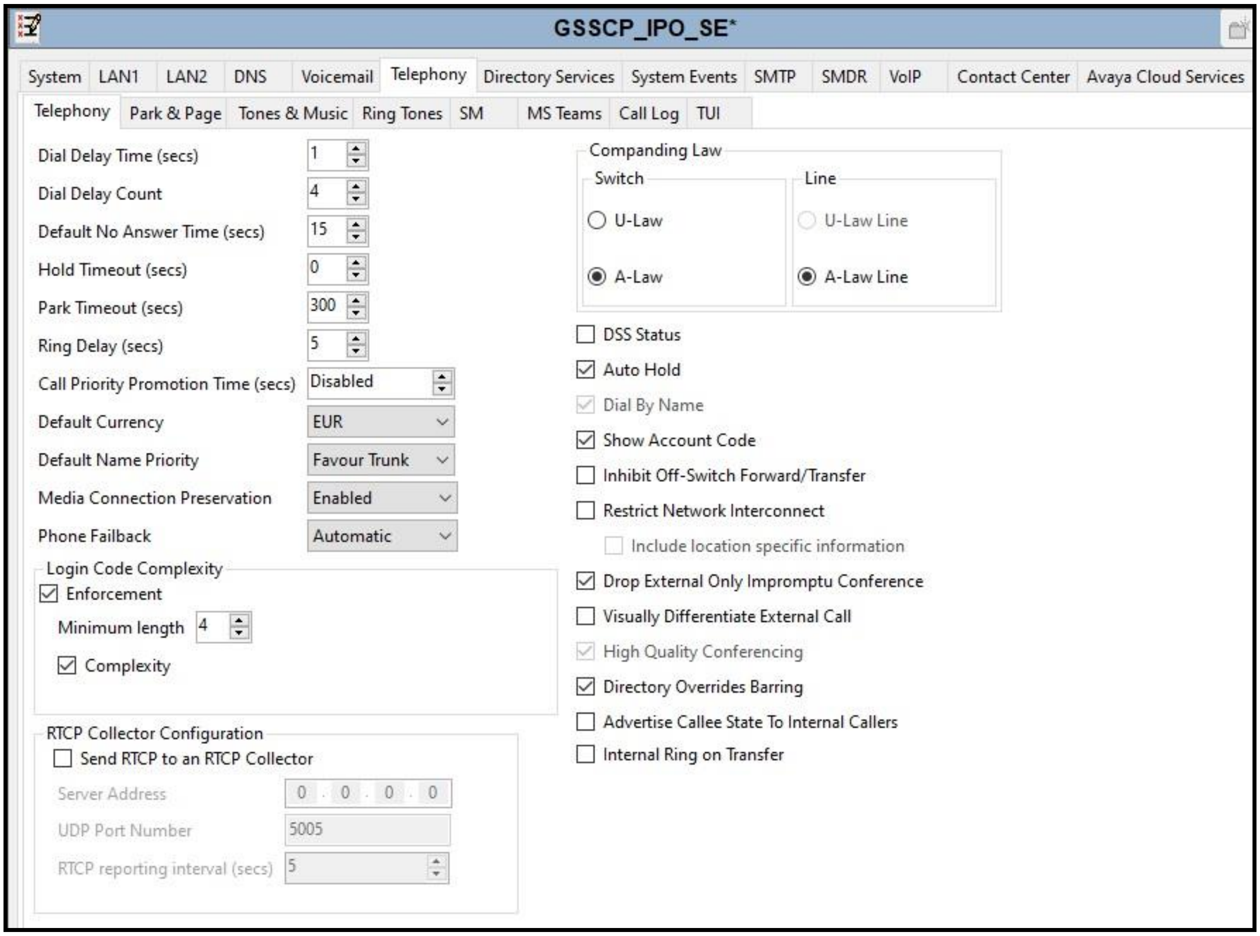
Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signalling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signalling. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).



On the **Network Topology** tab, set the **Firewall/NAT Type** from the pulldown menu to **Open Internet**. With this configuration, the **STUN Server IP Address** and **STUN Port** are not used as NAT was not required for this configuration, therefore resulting in no requirement for a STUN server. The **Use Network Topology Info** in the **SIP Line** was set to **None** in **Section 5.5.2**.Set **Binding Refresh Time (seconds)** to **30**. This value is used to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider. Default values were used for all other parameters. On completion, click the **OK** button (not shown).

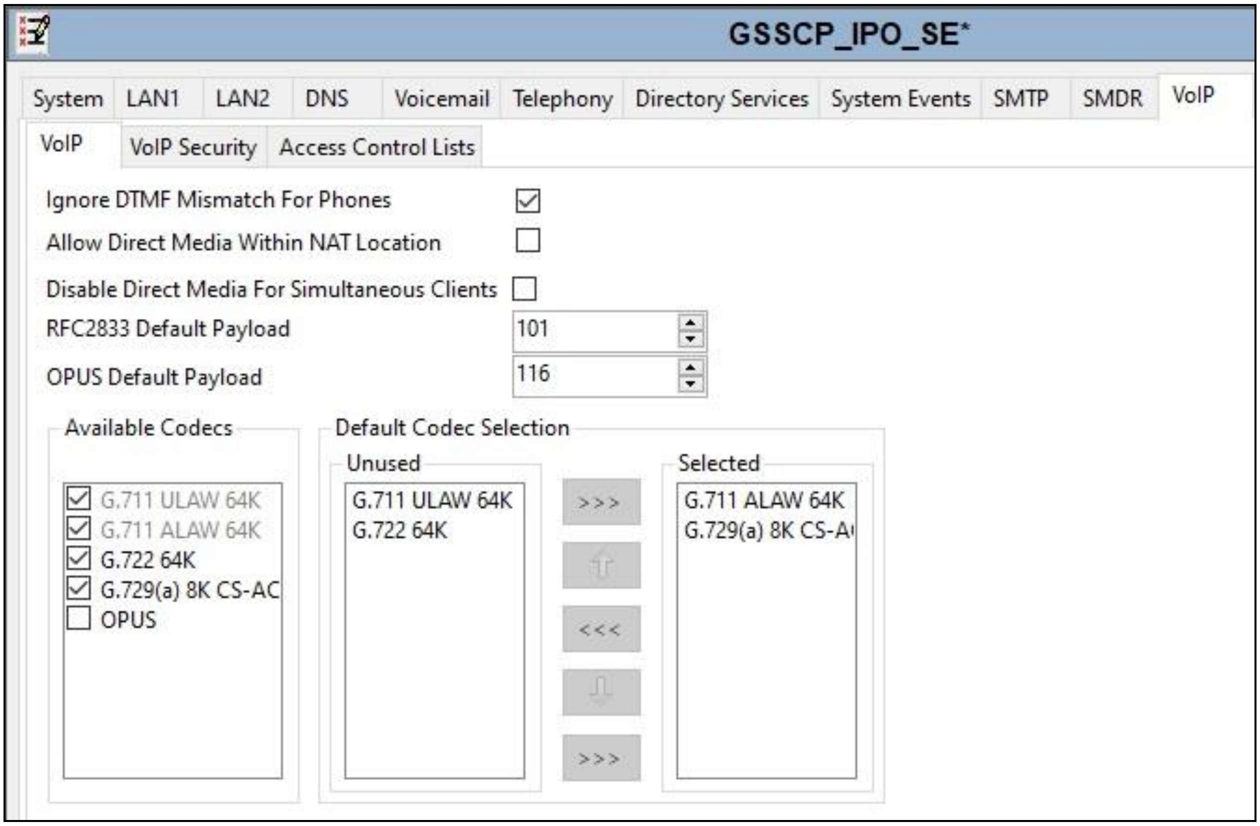
5.3. System Telephony Settings

Navigate to the **Telephony → Telephony** tab on the Details Pane. Choose the **Companding Law** typical for the enterprise location. For Europe, **ALAW** 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. VoIP Settings

Navigate to the **VoIP** tab on the Details Pane. Check the available Codecs boxes as required. 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** is set as the priority codec and **G.729(a) 8K CS-AC** set as the secondary codec as per screenshot below.



5.5. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the VTX Connect PBX-IP SIP platform. 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:

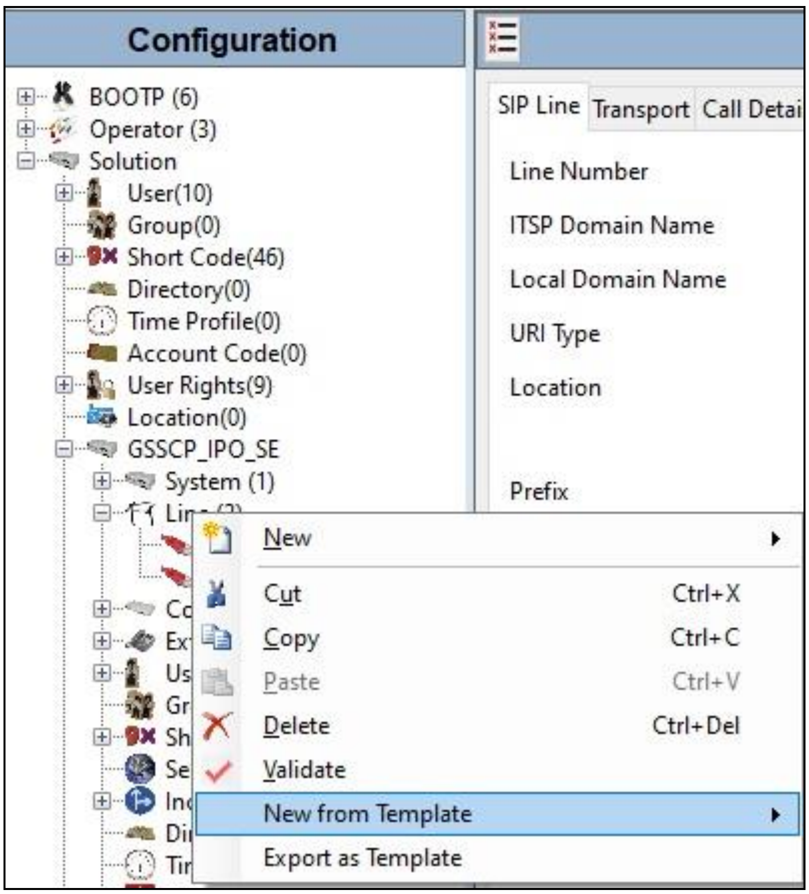
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Section 5.5.2**.

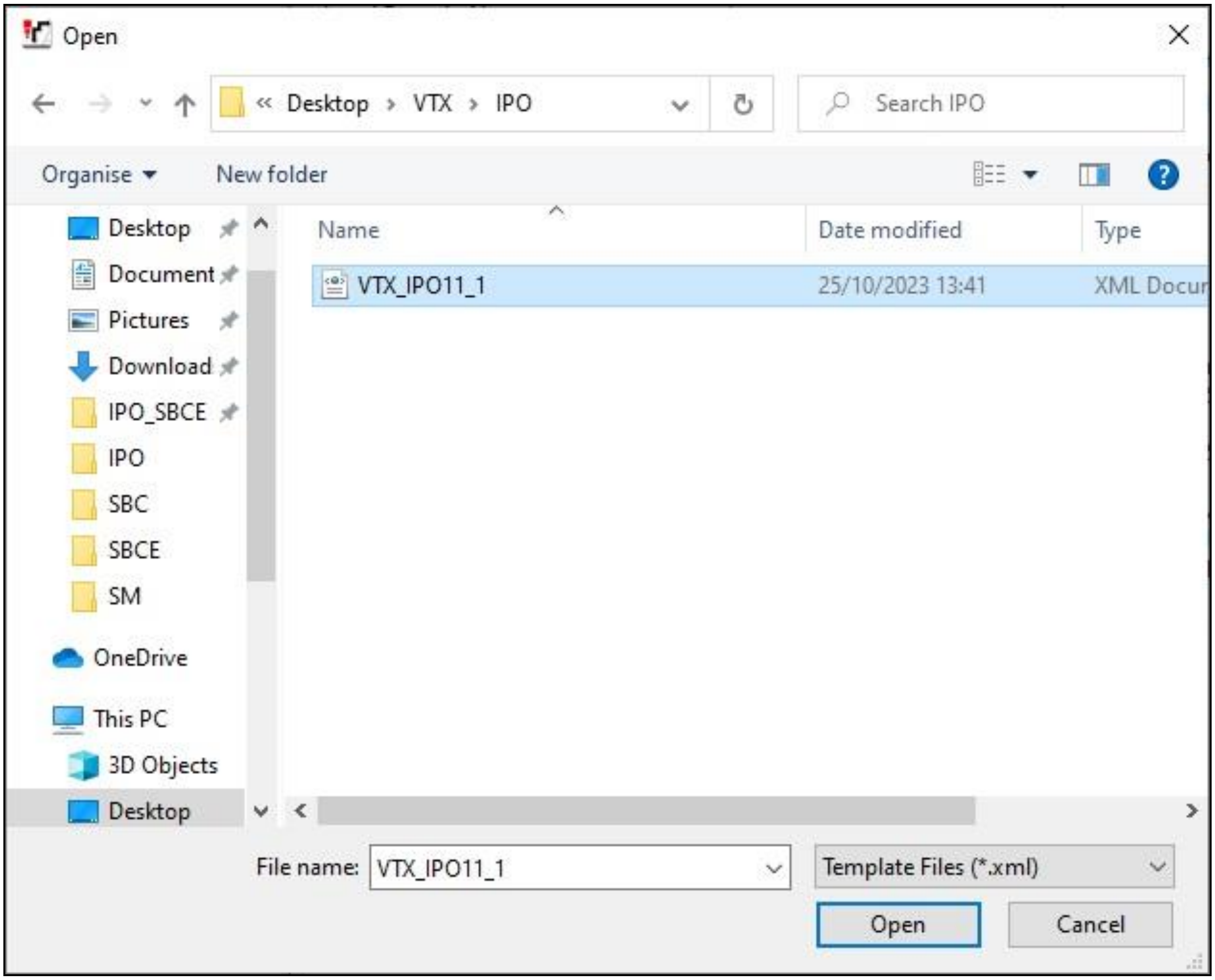
5.5.1. SIP Line From Template

DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500 V2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment.

Copy a previously created template file to a location (e.g., *\temp*) on the same 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**.



Navigate to the directory on the local machine where the template was copied and select the template as required.



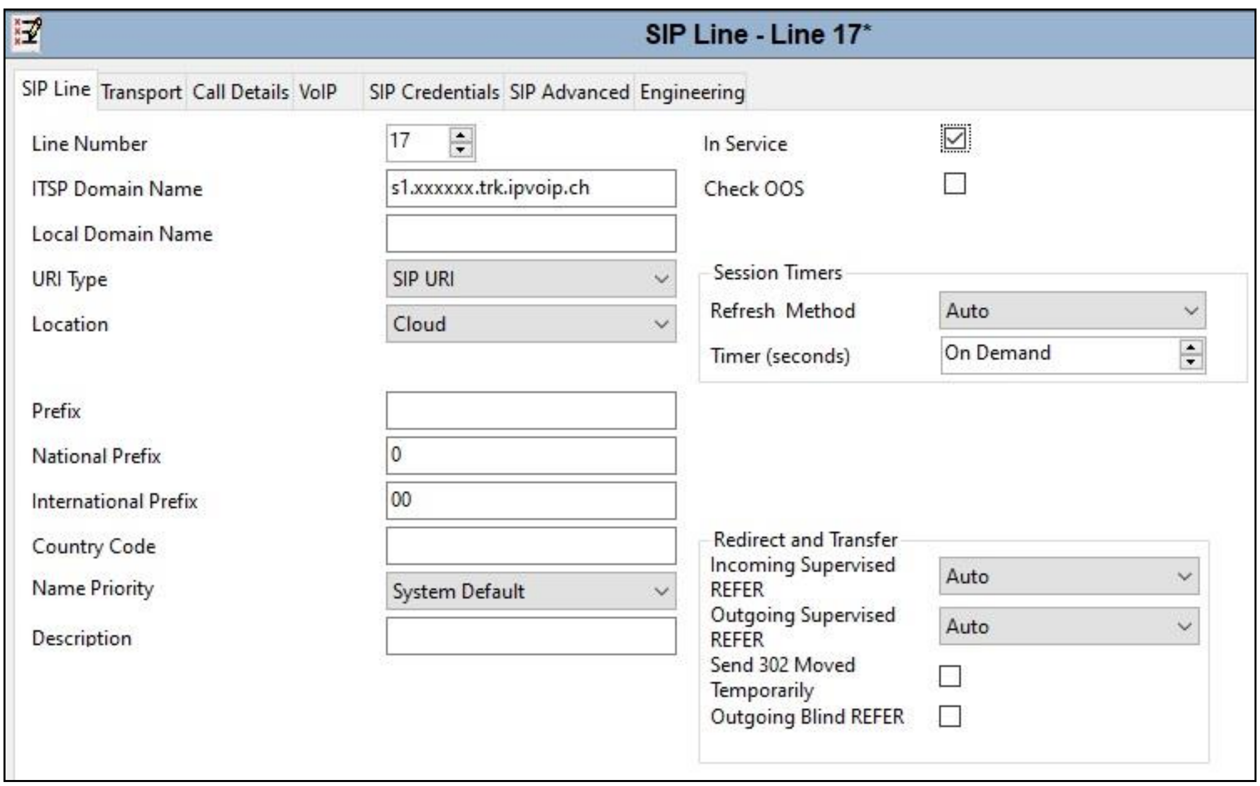
The SIP Line is automatically created and can be verified and edited as required using the configuration described in **Section 5.5.2**.

5.5.2. Manual SIP Line Configuration

On the **SIP Line** tab in the Details Pane, configure the parameters below to connect to the SIP Trunking service.

- 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.
- 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.
- Set **Incoming Supervised REFER** and **Outgoing Supervise REFER** to **Auto**.
- Leave all other fields at default settings.

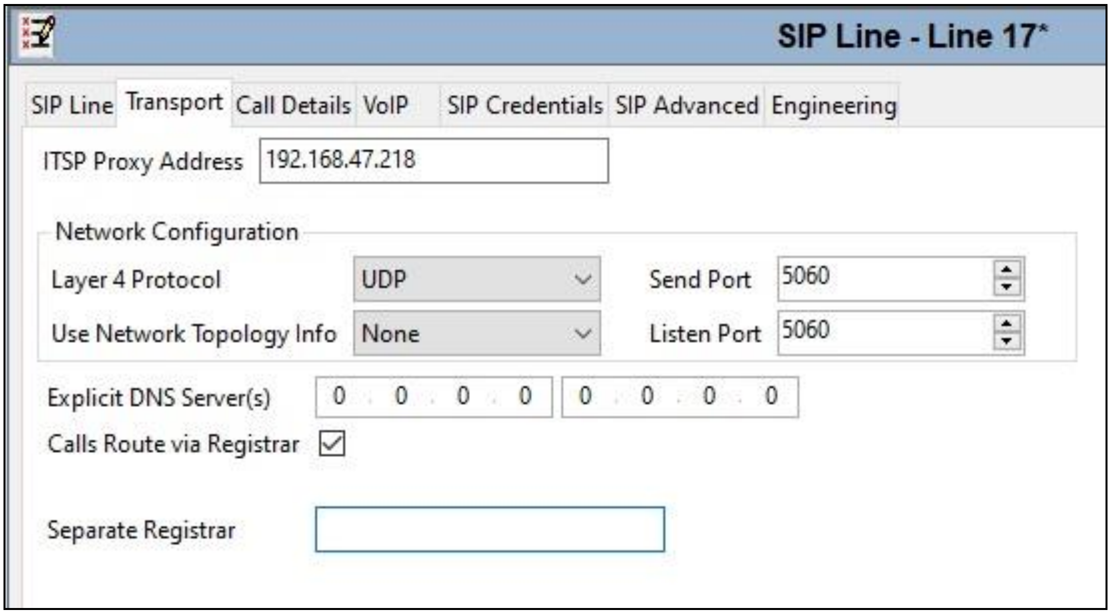
On completion, click the **OK** button (not shown).



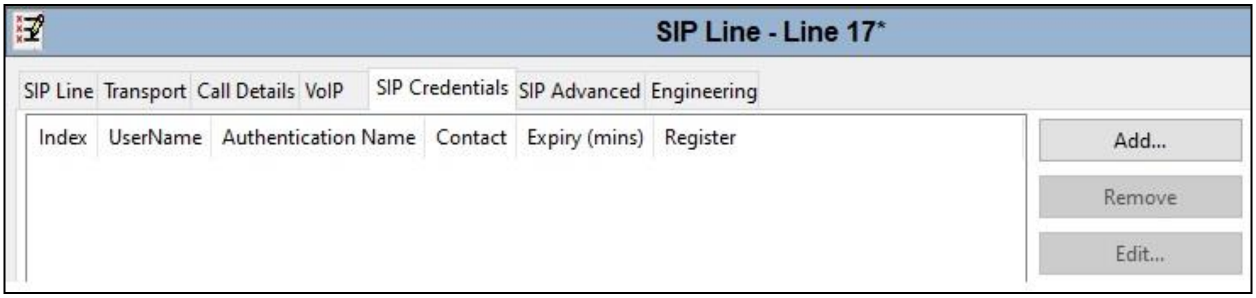
Select the **Transport** tab and set the following:

- 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).

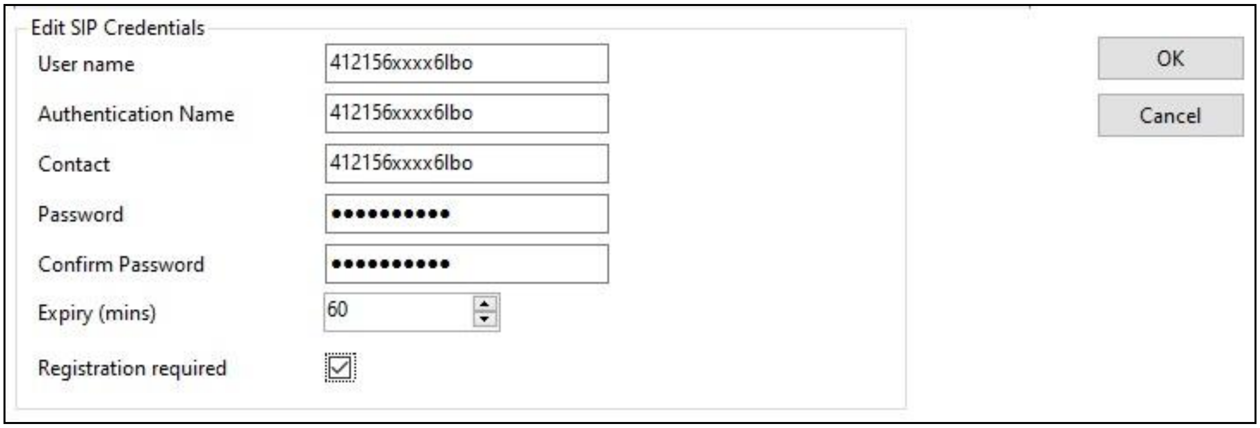


After the SIP line parameters are defined, the SIP credentials used for registration and authorisation on this line must be created. To define SIP credentials, first select the **SIP Credentials** tab. Click the **Add** button and the **New SIP Credentials** area will appear at the bottom of the pane.

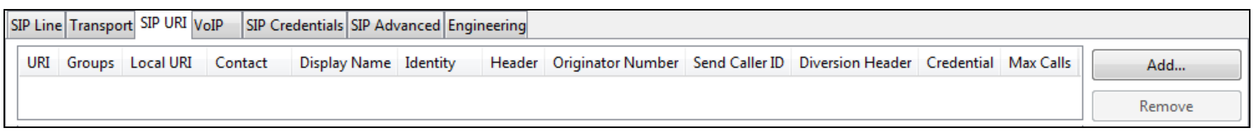


Enter the username and password provided by VTX Services here.

The username can be used for the **User name**, **Authentication Name** and **Contact fields**. The registration timeout value should be provided by the service provider. It can also be 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.

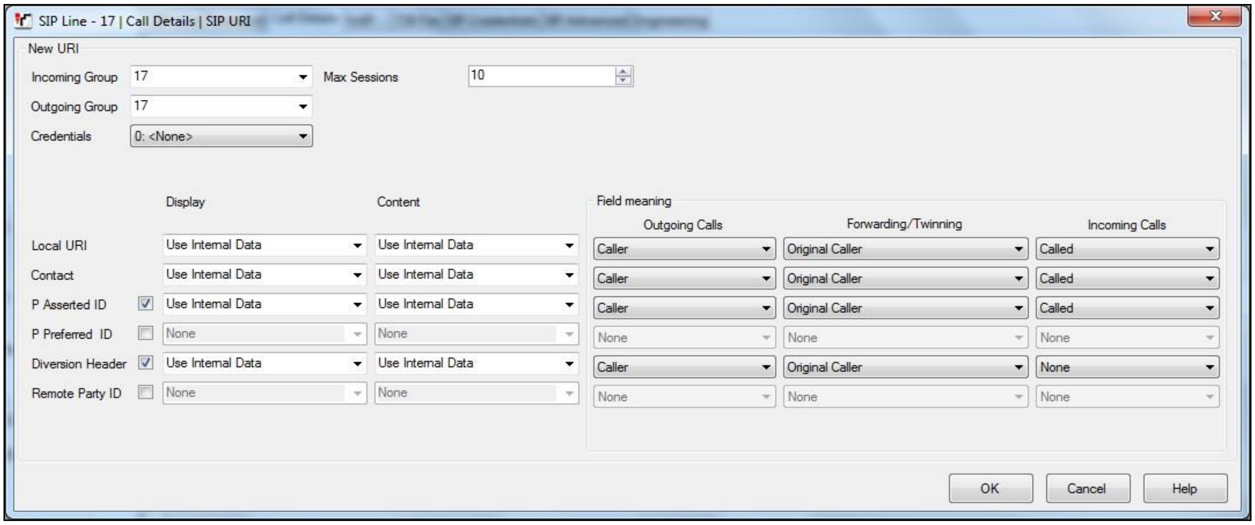


After the SIP credentials are defined, the SIP URIs that Avaya IP Office will receive and transmit on this line must be created. To create a SIP URI entry, first select the **SIP URI** tab. Click the **Add** button and the **New URI** area will appear at the bottom of the pane.

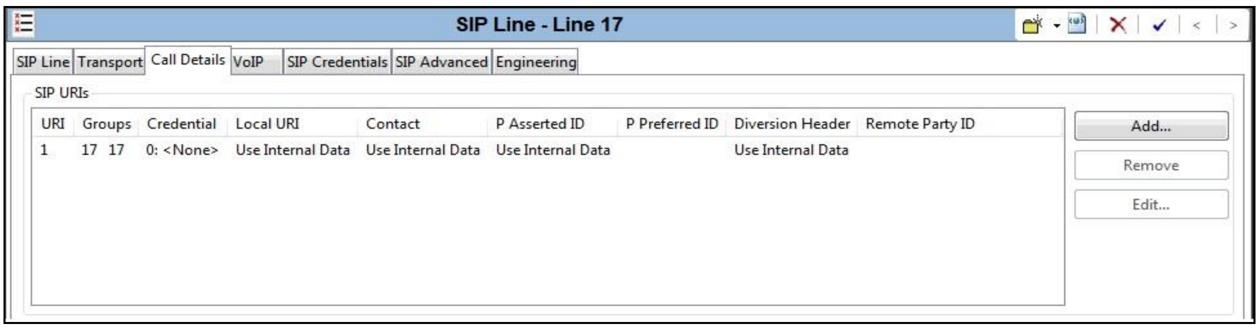


For the compliance test, SIP URI entries were created that matched any number assigned to an Avaya IP Office user. The entry was created with the parameters shown below.

- Set **Incoming Group**. This is the value assigned for incoming calls that's analysed in the Incoming Call Route settings described in **Section 5.8**. In the test environment a value of **17** was used for the VTX Connect PBX-IP SIP platform.
- Set **Outgoing Group**. This is the value assigned for outgoing calls that can be selected directly in the short code settings described in **Section 5.6**. In the test environment a value of **17** was used.
- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Set **Local URI**, **Contact**, **P Asserted ID** and **Diversion Header** to **Use Internal Data** for both the **Display** name and **Content**. 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 **Outgoing Calls**, **Forwarding/Twinning** and **Incoming Calls** at their respective default values of **Caller**, **Original Caller** and **Called** for the **Local URI**, **Contact** and **P Asserted ID** call details.



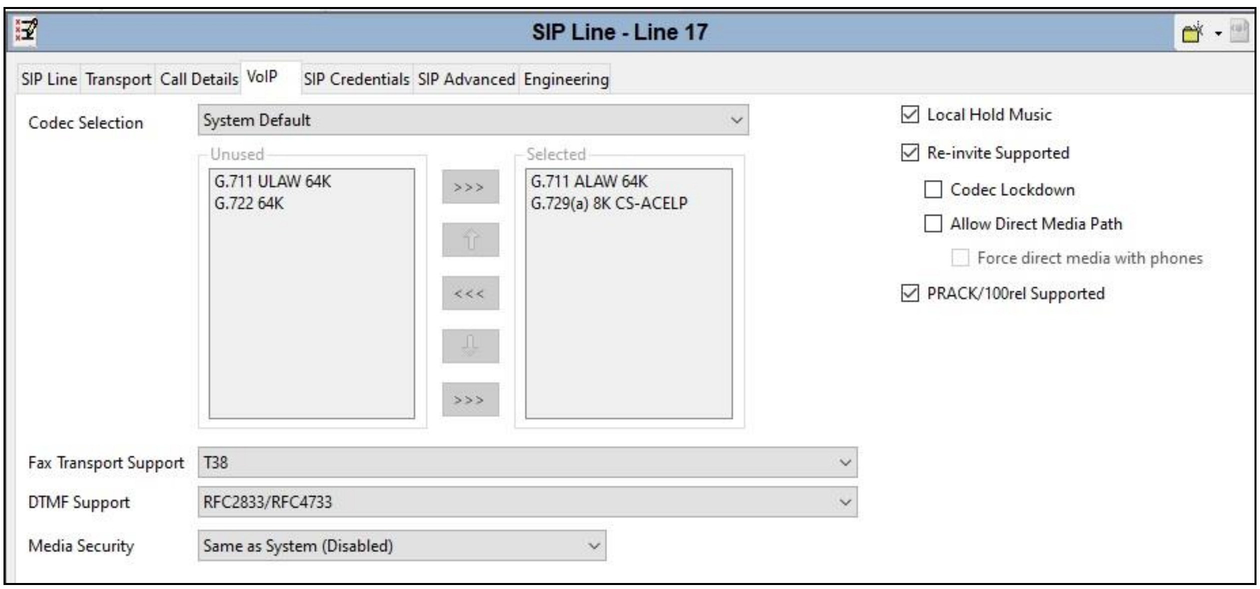
The following screenshot shows the completed configuration:



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

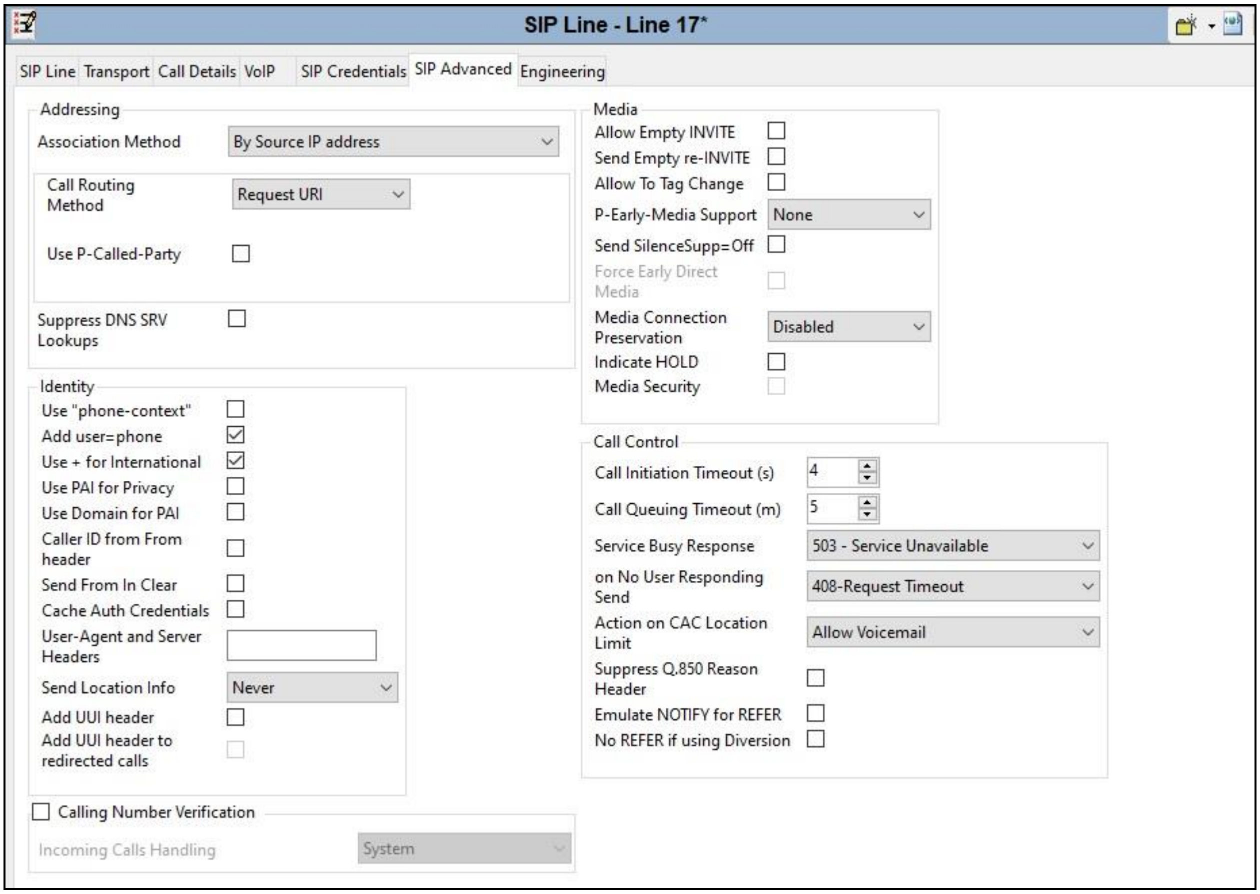
- Select **System Default** from the drop-down menu as system default codecs were already defined in **Section 5.4**.
- Set the **Fax Transport Support** box to **T38** as this is the preferred method of fax transmission for VTX.
- Set the **DTMF Support** field to **RFC2833/RFC4733**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Check the **Local Hold Music** box.
- 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.
- 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).

Default values may be used for all other parameters.



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

- Check the **Add user=phone** box to send SIP parameter user with the value phone to the From and To Headers in outgoing calls.
- Check the **Use + for International** as E.164 numbering is used on the SIP Trunk.
- Default values may be used for all other parameters.



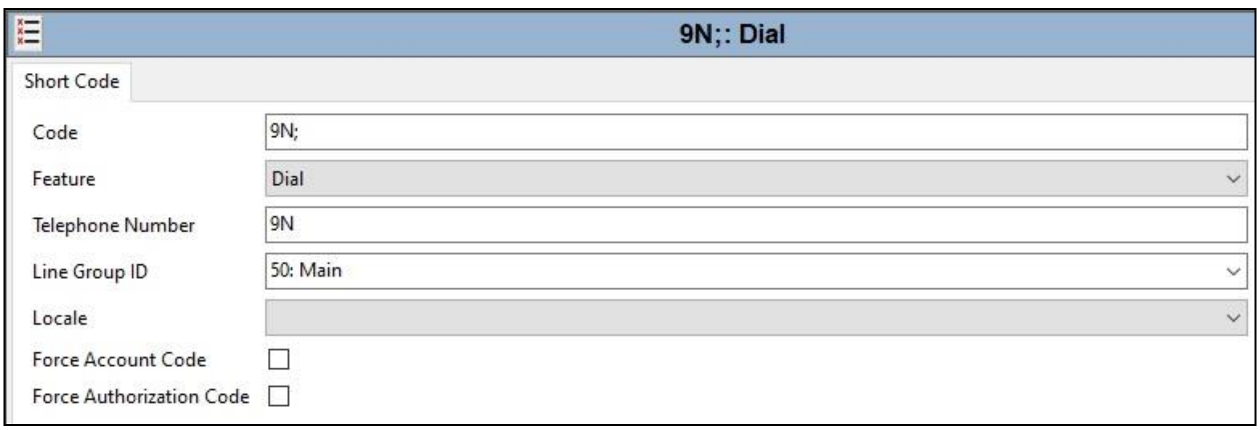
Note: It is advisable at this stage to save the configuration as described in **Section 5.11** to make the Line Group ID defined in **Section 5.5.2** available.

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 required. The example below shows the configuration used during testing for national 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 5.9**.
- On completion, click the **OK** button (not shown).

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

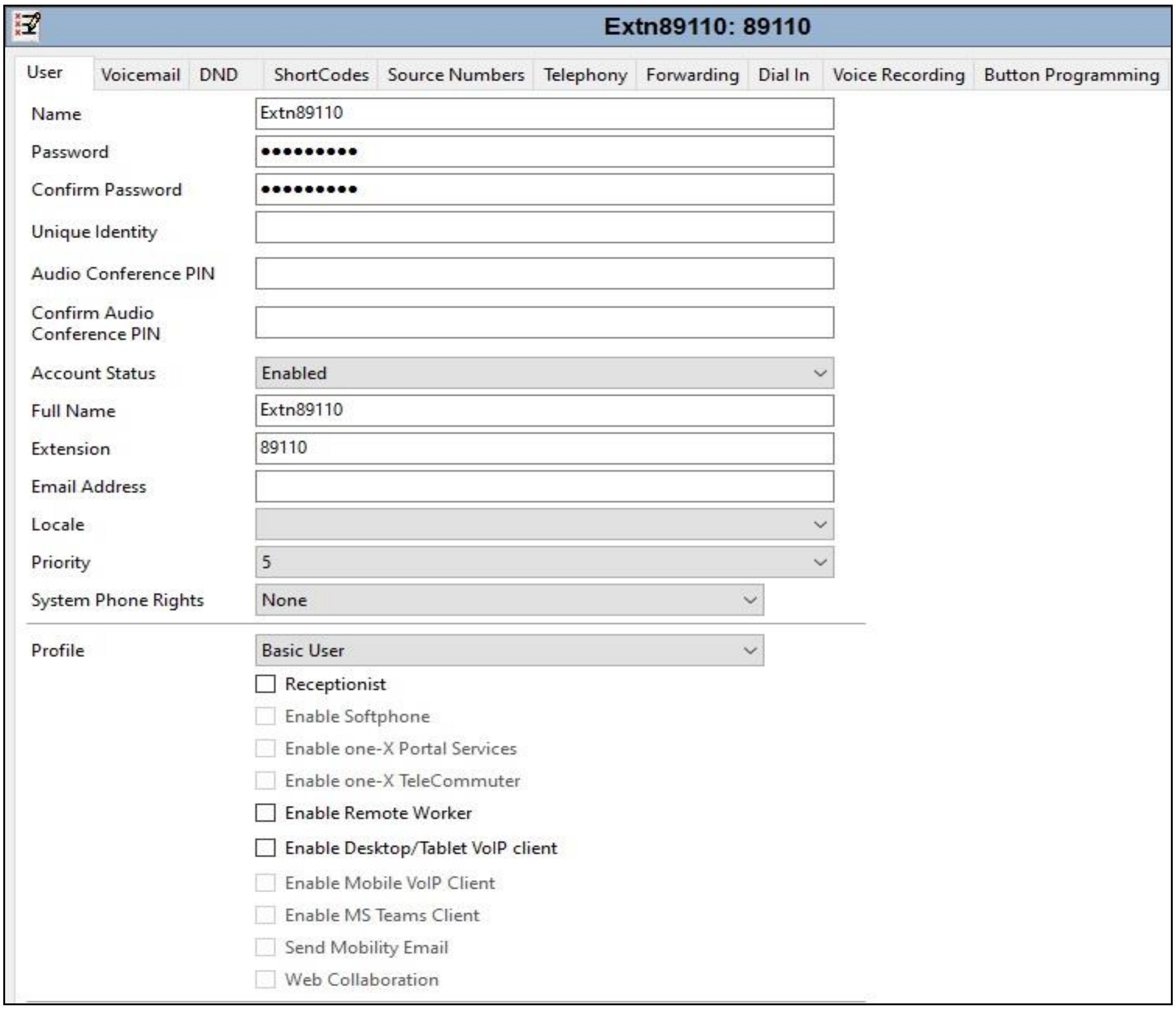


5.7. User

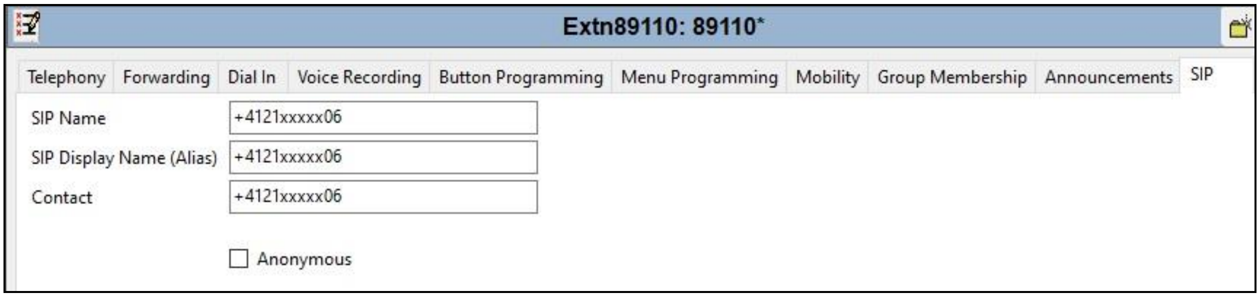
Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.5.2**. To configure these settings, first navigate to **User** in the Navigation Pane. Select the **User** tab if any changes are required.

The following example shows the configuration required for a SIP 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.

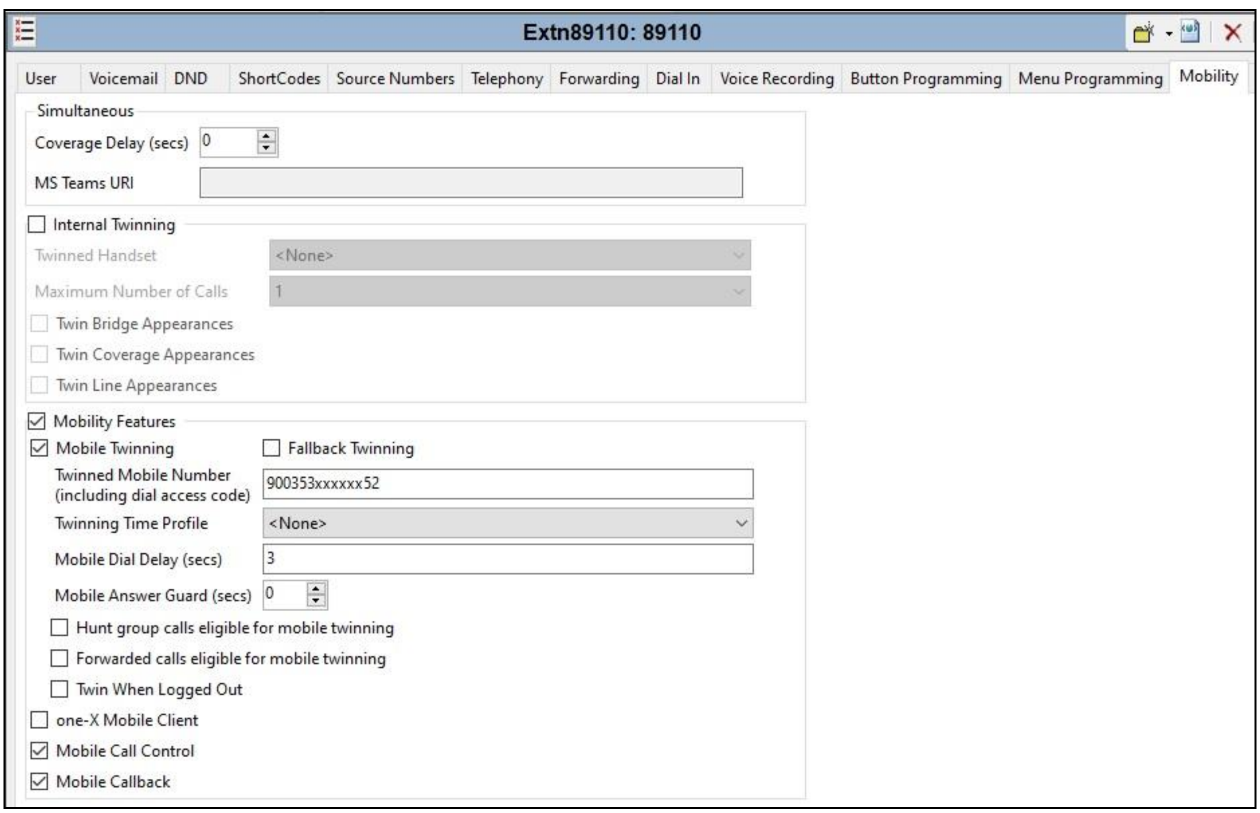


Next, select the **SIP** tab in the Details Pane. To reach the **SIP** tab click the right arrow on the right-hand side of the Details Pane until it becomes visible. The values entered for the SIP **Name** and **Contact** fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls. These allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.5.2)**. As such, these fields should be set to one of the DDI numbers assigned to the enterprise from VTX.



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

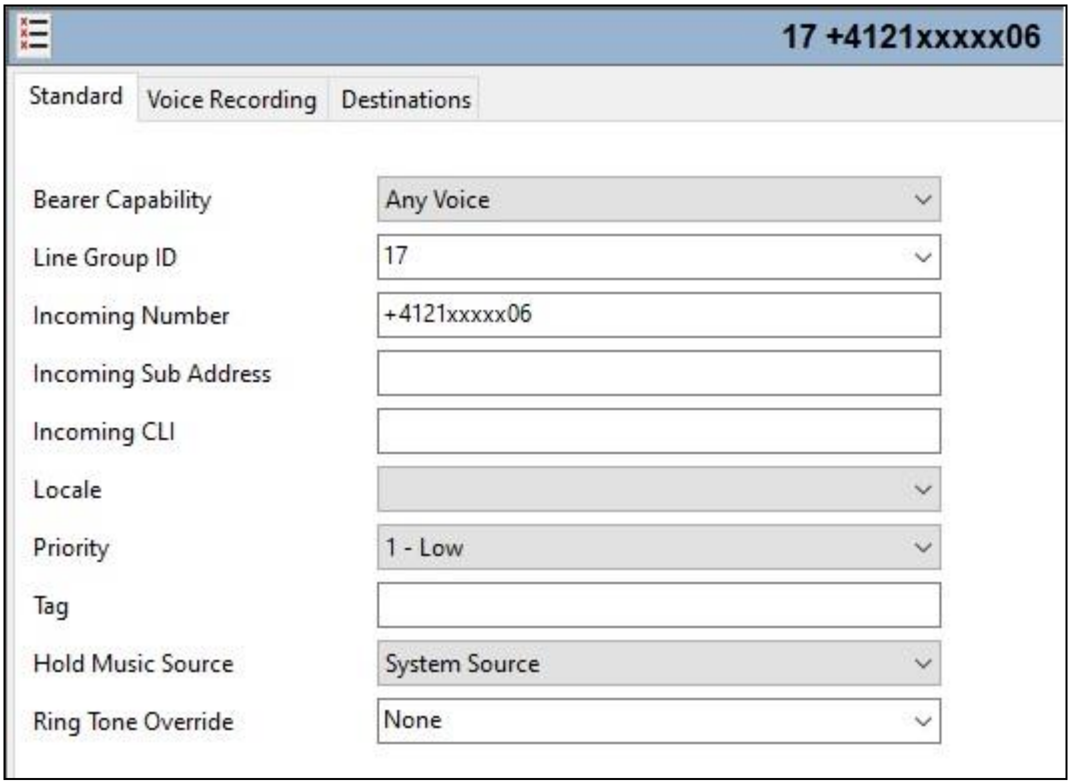
The following screen shows the Mobility tab for user 89110. The **Mobility Features** and **Mobile Twinning** are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone over the SIP Trunk. Other options can be set accordingly to customer requirements.



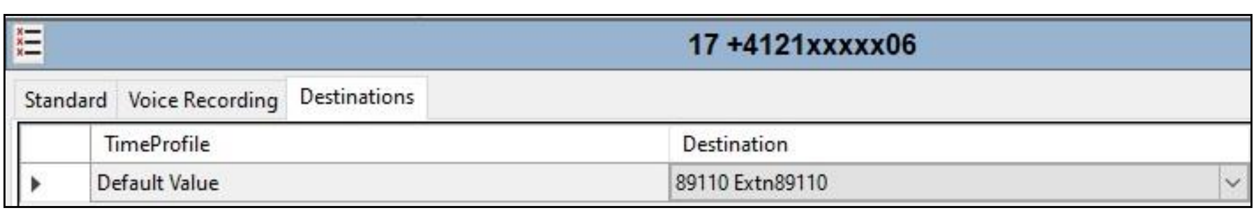
5.8. Incoming Call Routing

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 Routes** in the Navigation Pane and select **New**. 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 5.5.2**.
- 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.

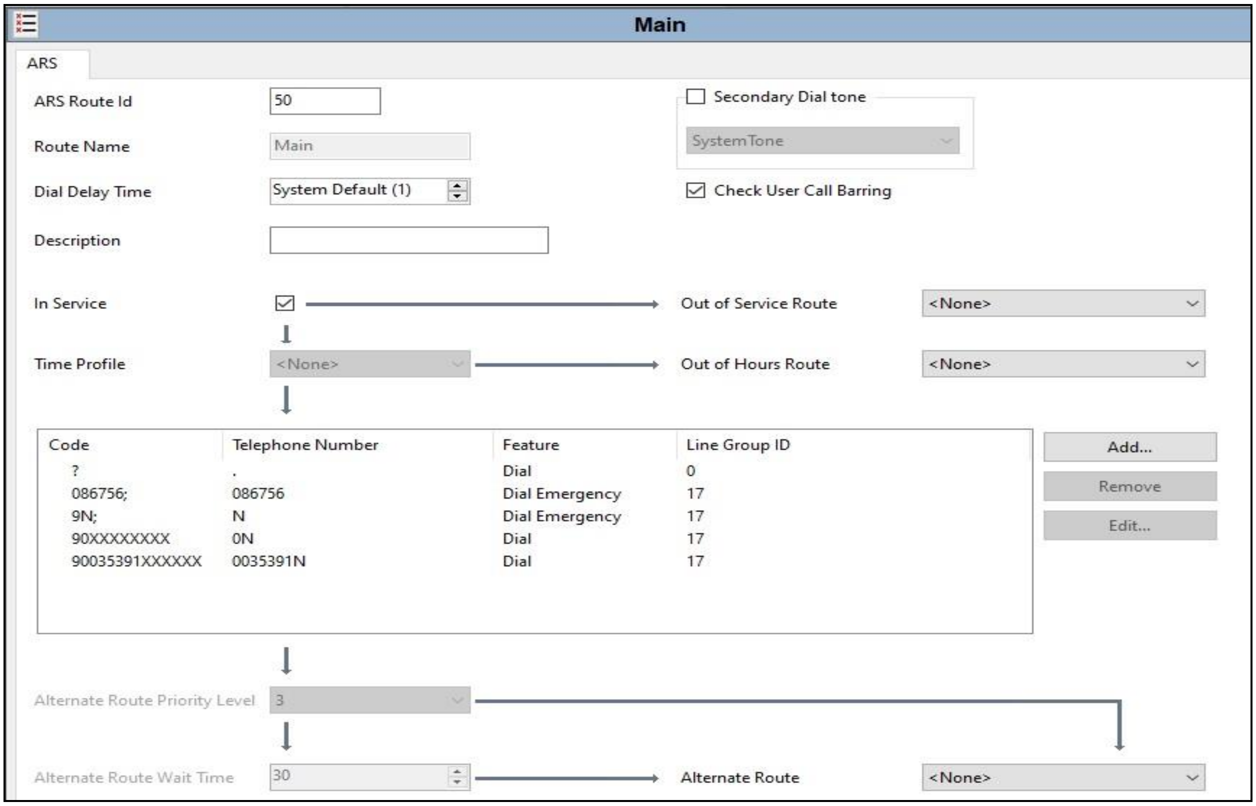


On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number **+4121xxxxx06** on line 17 are routed to extension 89110.



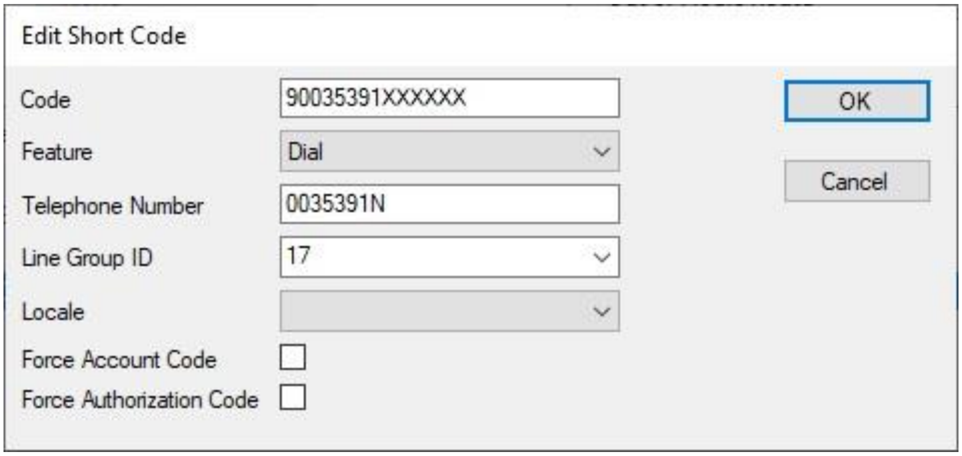
5.9. ARS

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 a 15 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 5.5.2**. During testing this was **17** for the SIP Trunk. Click on **OK**


Edit Short Code
Code
90035391XXXXXX
Feature
Dial
Telephone Number
0035391N
Line Group ID
17
Locale
Force Account Code
Force Authorization Code
OK
Cancel

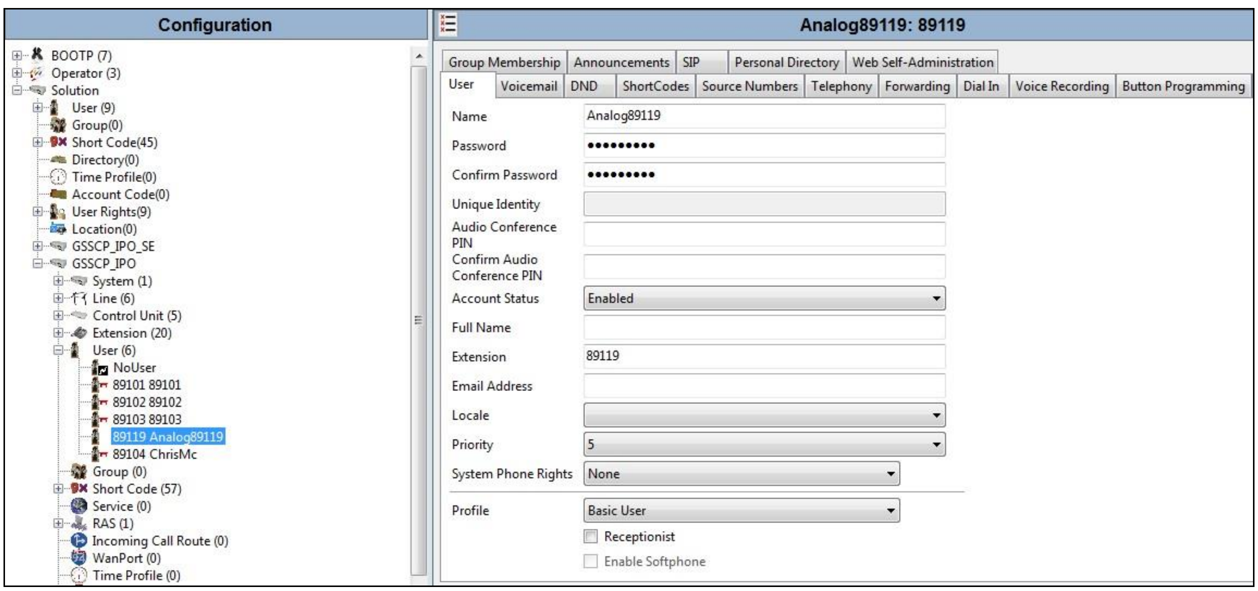
5.10. T38 Fax

At Release 11, both G.711 and T.38 Fax is supported on IP Office Server Edition when using an IP Office Expansion (500 V2). The VTX SIP Trunk 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_IPO**. Select the **User** tab. The following example shows the configuration required for an analog Endpoint.

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



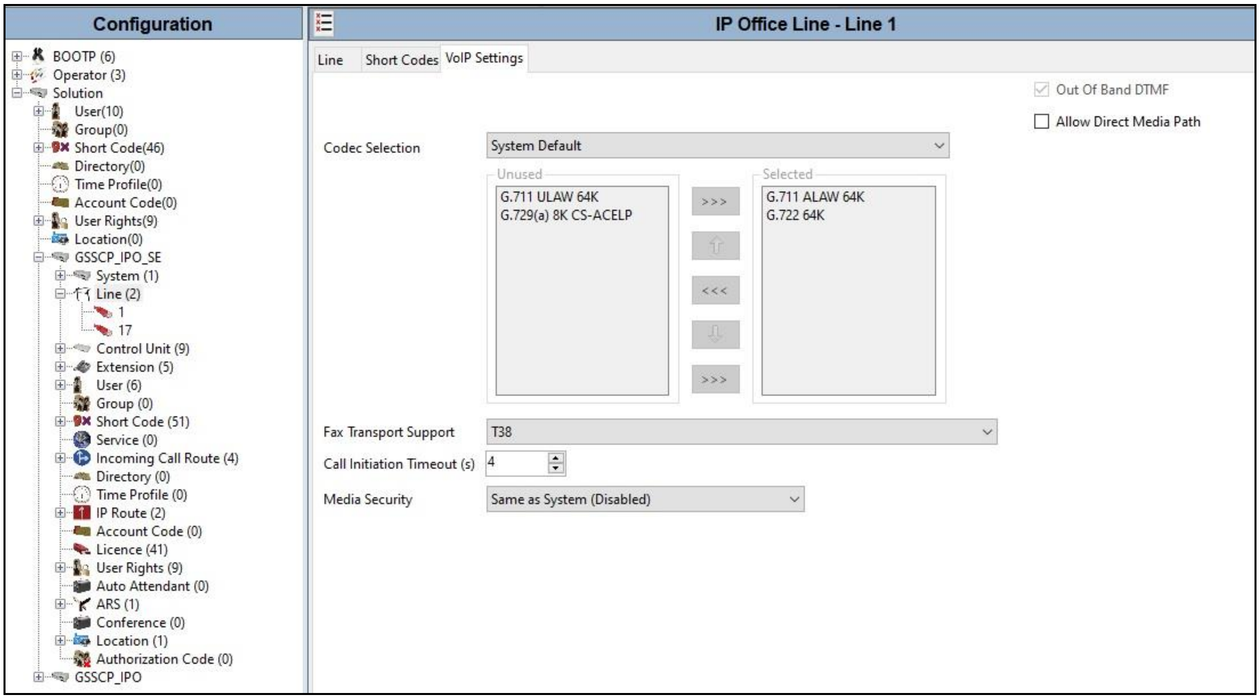
Configure other settings as described in **Section 5.7**.

5.10.2. T.38 Fax Settings

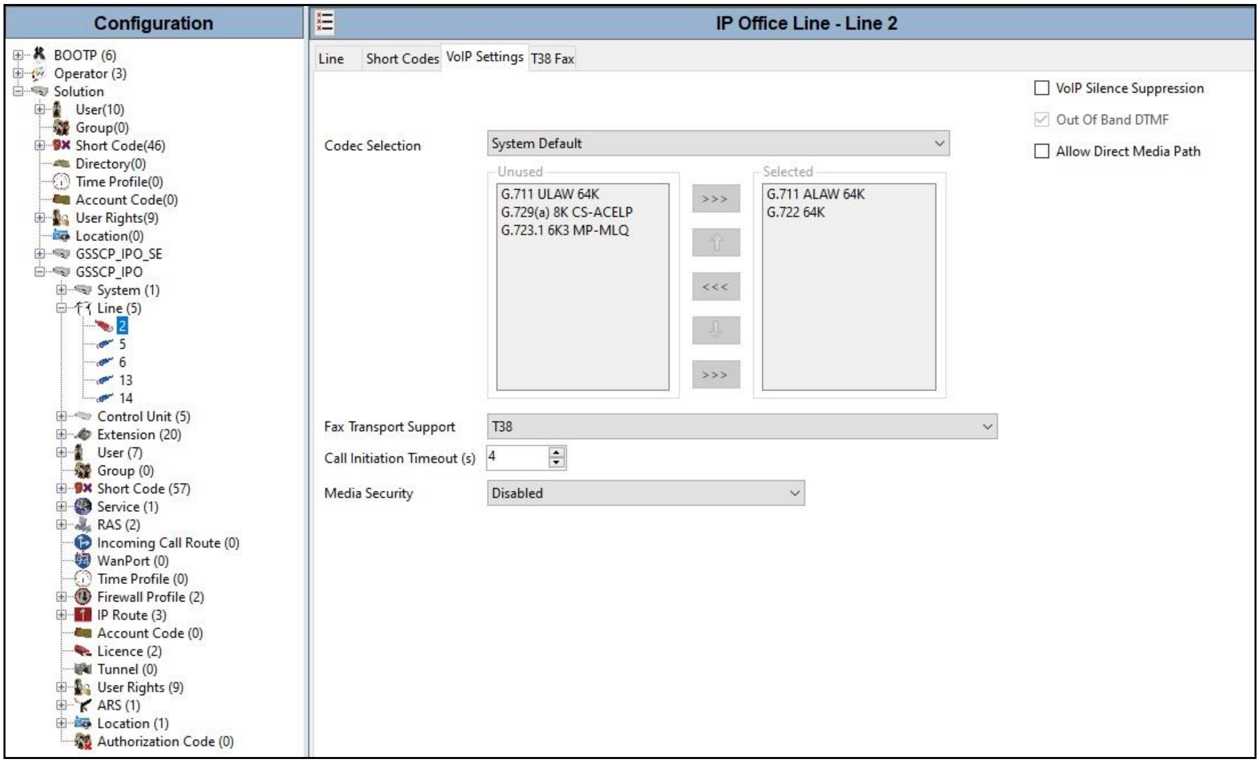
The T.38 Fax settings are defined on the SIP Line between the Expansion and the Server. Note that the VoIP settings for T.38 Fax are required in three places in this configuration:

- The SIP Line for the VTX SIP Trunk as described in **Section 5.5.2**.
- 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.

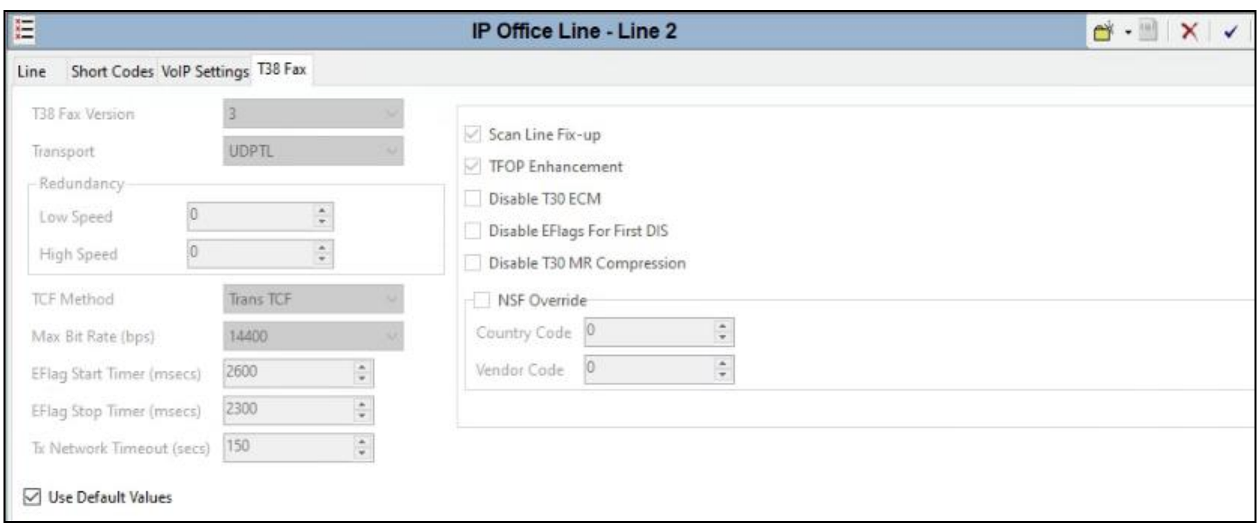
In all the above cases, the **Fax Transport Support** was set to **T38**. The following screenshot shows the VoIP Settings for the IP Office Line between the Server and the Expansion on the Server configuration:



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



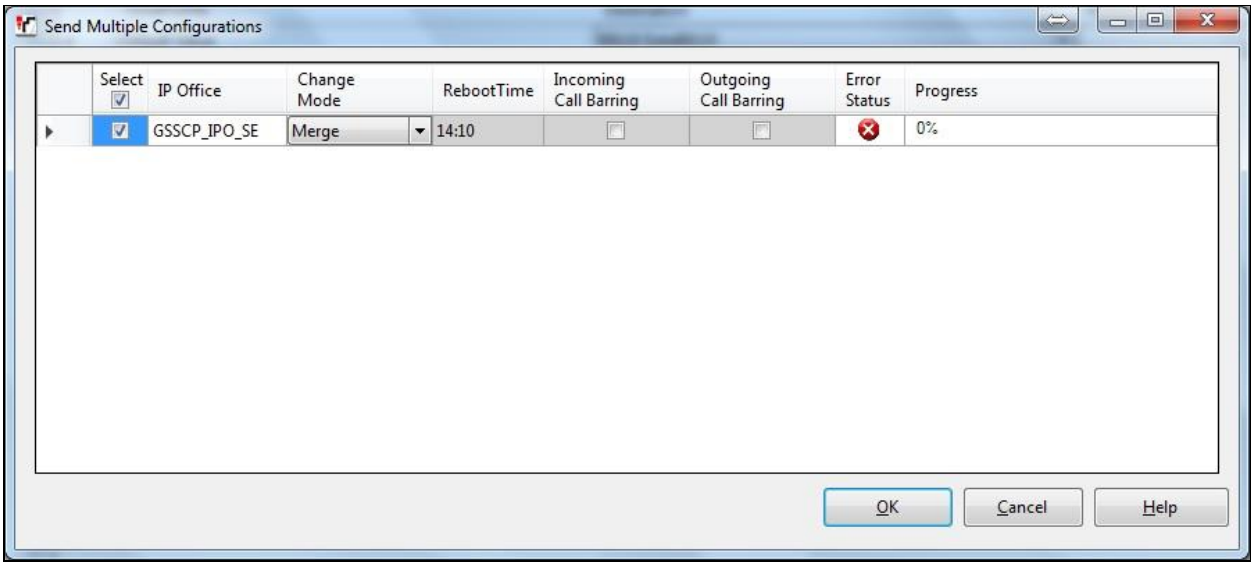
The following shows the T38 Fax tab in the IP Office Line for the Server in the Expansion configuration with **Use Default Values** enabled.



Refer to **Section 5.5.2** for the VoIP Settings on the SIP Line for the VTX SIP Trunk.

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. A screen like the one shown below is displayed where the system configuration has been changed and needs to be saved on the system. **Merge, Reboot, Timed** or **RebootWhen Free** can be selected from the **Change Mode** drop-down menu based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a service disruption. Click **OK** to save the configuration.



6. VTX Connect PBX-IP SIP Trunk Configuration

The configuration of the VTX equipment used to support VTX's SIP platform is outside of the scope of these Application Notes and will not be covered. To obtain further information on VTX equipment and system configuration please contact an authorized VTX representative

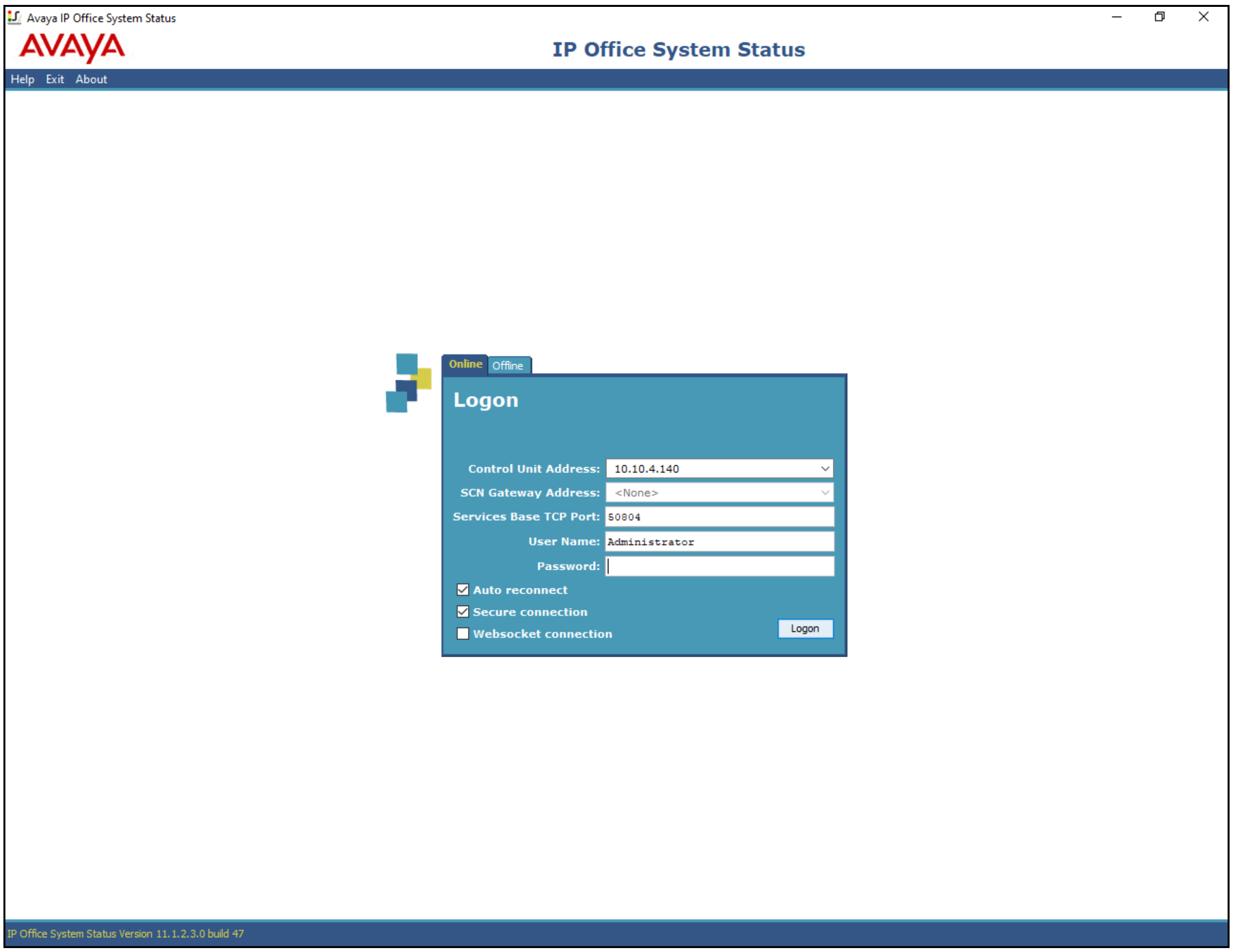
7. Verification Steps

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

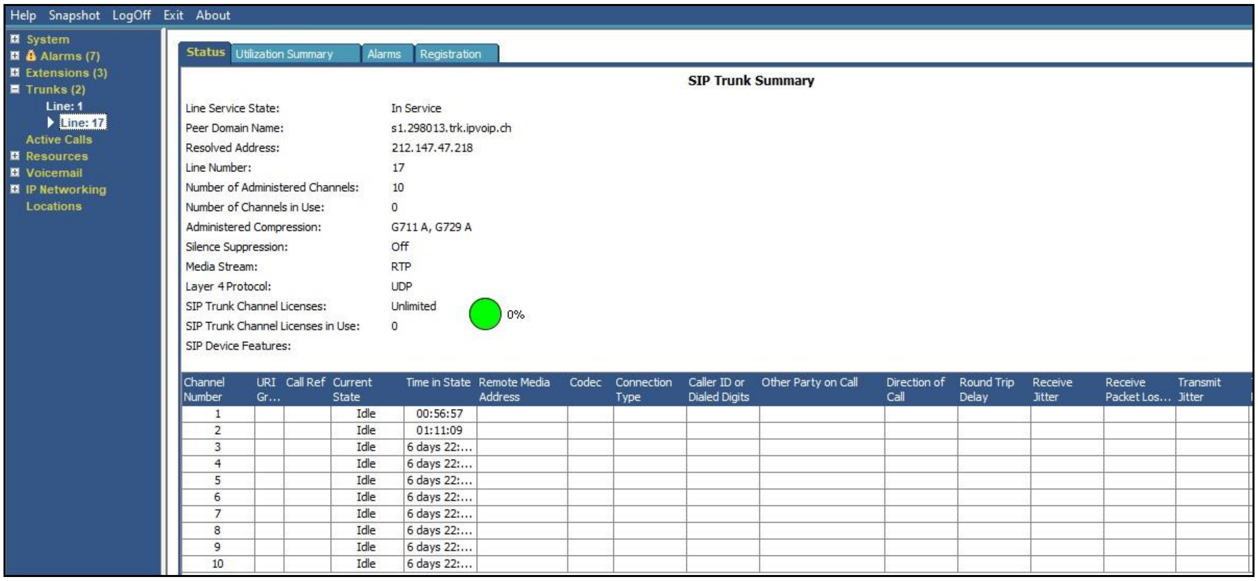
7.1. SIP Trunk status

The status of the SIP trunk can be verified by opening the System Status application. This is found on the PC where IP Office Manager is installed in PC programs under **Start →All Programs →IP Office →System Status** (not shown).

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.

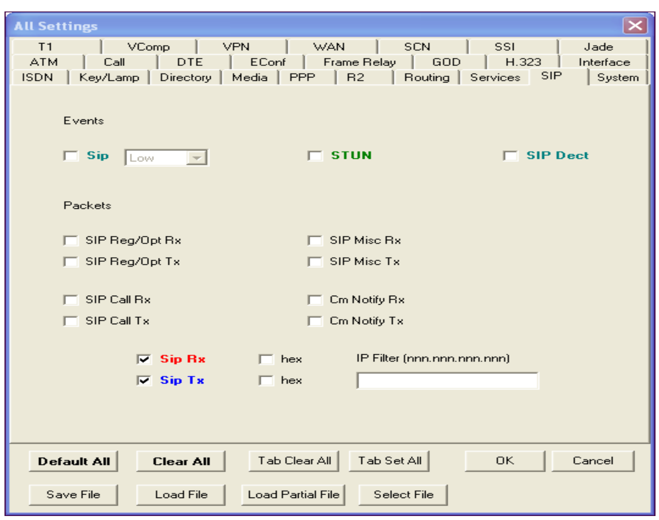


From the left-hand menu expand **Trunks** and choose the SIP trunk (**17** in this instance). The status window will show the status as being idle and time in state if the Trunk is operational.



7.2. Monitor

The Monitor application can also be used to monitor and troubleshoot IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor**. The application allows the monitored information to be customized. To customize, select the button that is third from the right in the screen below, or select **Filters →Trace Options**. The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked. All SIP messages will appear in the trace with the color blue. To customize the color, right-click on **SIP Rx** or **SIP Tx** and select the desired color.



As an example, the following shows a portion of the monitoring window of an OPTIONS message being sent between IP Office LAN1 interface and a J179 SIP IP phone.


Avaya IP Office SysMonitor - [STOPPED] Monitoring 10.10.4.140 (GSSCP_IPO_SE (Server Edition(P))); Log Settings - C:\Users\...\sysmonitorsettings....
File Edit View Filters Status Help
19:02:58 1212511831mS RES6: LinuxMem Total=8170762240 Used=3106082816 Free=5064679424, VMail CPU=01.41% UsedMem=21225472
19:03:00 1212513109mS Sip: (f6918858) SessionTimerTimeout in-dialog OPTIONS for stimulus-dialog: f6918858, session_timer_interval:
19:03:00 1212513109mS Sip: (f6918858) OPTIONS SENT TO 10.10.5.62 14709
19:03:00 1212513109mS SIP Tx: TCP 10.10.4.140:5060 -> 10.10.5.62:14709
OPTIONS sip:89115@10.10.5.62:14709;transport=tcp SIP/2.0
v: SIP/2.0/TCP 10.10.4.140:5060;rport;branch=z9hG4bK9dcdc7da76194a9f41794c17a814fc0d
f: <sip:CCMSInterface@10.10.4.140>;tag=f3b633c57d1d3f56
t: <sip:89115@avaya.com>;tag=63e5381d2ad19e473i3o64z6f3w2q5ar4e1b44_F89115
i: 2_63e5381d-51415ea3c192t2s6r6u2766t26441y_I89115
CSeq: 7101 OPTIONS
m: <sip:CCMSInterface@10.10.4.140:5060;transport=tcp>
Max-Forwards: 70
l: 0
19:03:00 1212513127mS SIP Rx: TCP 10.10.5.62:14709 -> 10.10.4.140:5060
SIP/2.0 200 OK
From: <sip:CCMSInterface@10.10.4.140>;tag=f3b633c57d1d3f56
To: <sip:89115@avaya.com>;tag=63e5381d2ad19e473i3o64z6f3w2q5ar4e1b44_F89115
Call-ID: 2_63e5381d-51415ea3c192t2s6r6u2766t26441y_I89115
CSeq: 7101 OPTIONS
Via: SIP/2.0/TCP 10.10.4.140:5060;branch=z9hG4bK9dcdc7da76194a9f41794c17a814fc0d
User-Agent: Avaya J179 IP Phone 4.0.10.3.2 c81feac884d4
Contact: <sip:89115@10.10.5.62:14709;transport=tcp>;+avaya.gmtoffset="0:00";+avaya.js-ver="1.0";+avaya.model="J1
iptolerance
Accept-Language: en
Content-Length: 0

8. Conclusion

These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office R11.1 and VTX Connect PBX-IP SIP Trunk solution as shown in **Figure 1**.

The reference configuration shown in these Application Notes is representative of a basic enterprise customer configuration and demonstrates Avaya IP Office R11.1 can be configured to interoperate successfully with VTX Connect PBX-IP SIP Trunk service. VTX Connect PBX-IP SIP Trunk service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2**.

9. Additional References

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

[1] *Deploying IP Office as Virtual Servers*, Release 11.1, Nov 2021.
[2] *Deploying IP Office Server Edition Servers*, Release 11.1, Nov 2021.
[3] *Deploying an IP500 V2 IP Office System*, Release 11.1, Jul 2022.
[4] *Administering Avaya IP Office with IP Office Web Manager*, Release 11.1, Nov 2021.
[5] *Administering Avaya IP Office with IP Office Manager*, Release 11.1, Nov 2021.
[6] *Using Avaya IP Office System Status*, Nov 2021.
[7] *Using IP Office System Monitor*, Nov 2021.
[8] *Administrating Voicemail Pro*, Release 11.1, Nov 2021.
[9] *Using Avaya Workplace Client for Windows*, Jul 2022.
[10] *IP Office SIP Phone Installation Notes*, Nov 2021.
[11] *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.