DevConnect Program

Application Notes for Configuring Avaya IP Office Server Edition R12.0 with Swisscom Enterprise SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between Swisscom Enterprise SIP Trunk Service and Avaya IP Office Server Edition R12.0.

Swisscom Enterprise SIP Trunk Service provides PSTN access via a SIP Trunk connected to the Swisscom Voice over Internet Protocol (VoIP) network as an alternative to legacy analogue or digital trunks. Swisscom 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 Swisscom Enterprise SIP Trunk Service and Avaya IP Office Server Edition R12.0.

Avaya IP Office is a versatile communications solution that combines the reliability and ease of a traditional telephony system with the applications and advantages of an IP telephony solution. This converged communications solution can help businesses reduce costs, increase productivity, and improve customer service.

Customers using this Avaya SIP-enabled enterprise solution with Swisscom'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 Server Edition R12.0 to connect to the Swisscom Enterprise SIP Trunk. 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.

2.1. Interoperability Compliance Testing

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 codec.
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using G.711 pass-through 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, call mute, transfer, and conference.
- Off-net call forwarding and mobile twinning.

2.2. Test Results

Interoperability testing of the test configuration was completed with successful results for Swisscom's SIP Trunk service with the following observations:

- During T.38 fax testing, it was observed that when Swisscom sent a reINVITE to negotiate to T.38 fax calls, IP Office responded with a 200OK with 2 x media lines in the SDP. The first media line had an attribute value of "inactive" which made the second media line active. However, Swisscom would respond to the 200OK from IP Office with a BYE and the call was terminated. Swisscom does not support the method in which IP Office negotiates the use of T.38, therefore T.38 fax is not supported on the Swisscom Enterprise SIP Trunk service.
- The Privacy Header as required by Swisscom is not included in the SIP INVITE for outbound calls with Calling Line Identity Restriction (CLIR) when using an IP Office short code (*67 was used in the test configuration). As a workaround, the anonymous button can be enabled on the SIP tab in **Section 5.7** to restrict CLIR and include a Privacy Header as required by Swisscom.
- 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

2.3. Support

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

For technical support on Swisscom products please contact the Swisscom support team:
Email: ent.incident-voice@swisscom.com.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the Swisscom 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 1400 Series Digital Deskphones, Analog 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 Workplace Client for Windows for softphone testing.

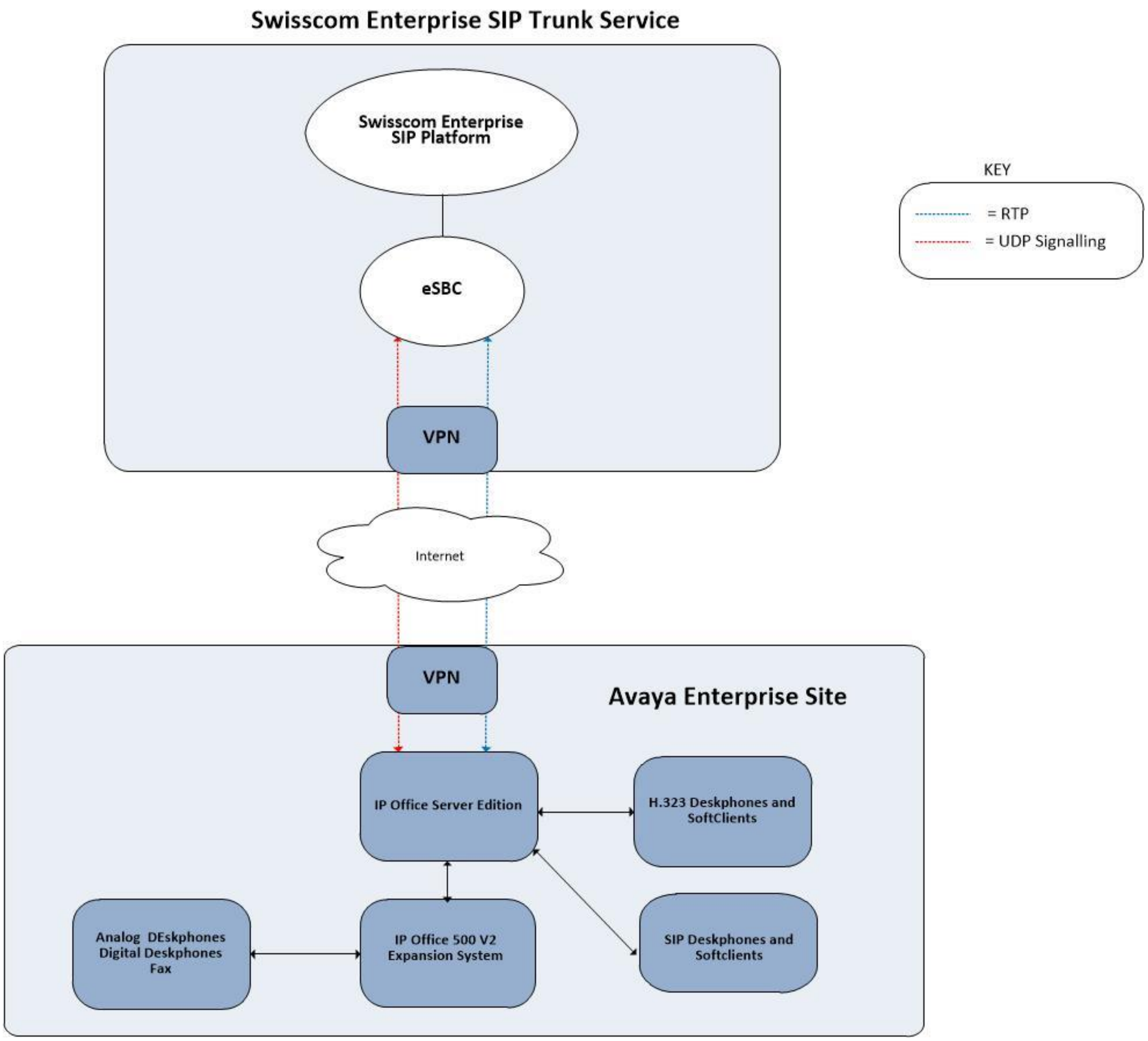


Figure 1: Swisscom Enterprise SIP Trunk to Avaya IP Office Topology

4. Equipment and Software Validated

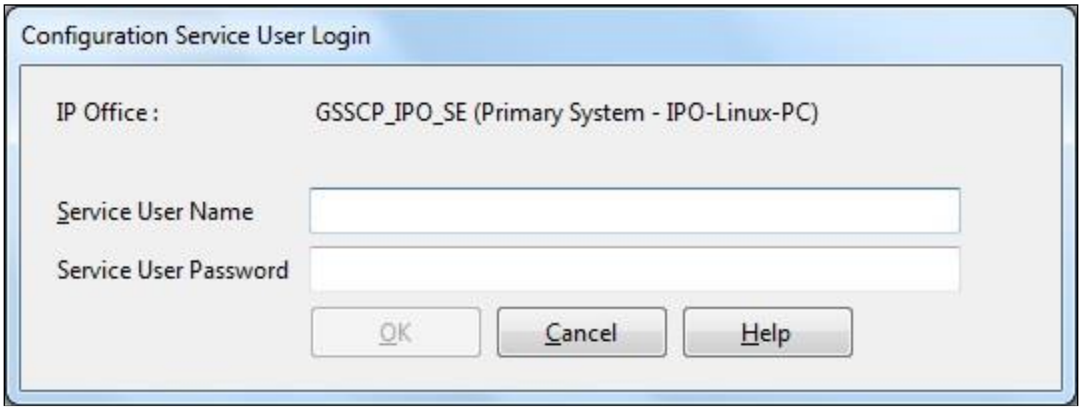
The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya	
Avaya IP Office Server Edition	Version 12.0.0.0.0 build 55
Avaya IP Office 500 V2	Version 12.0.0.0.0 build 55
Avaya Voicemail Pro Client	Version 12.0.0.26
Avaya IP Office Manager	Version 12.0.0.0.0 build 55
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)	3.36.0
Avaya 1140e (SIP)	FW: 04.04.30.00.bin
Avaya 1408 Digital Telephone	R48
Avaya 98390 Analogue Phone	N/A
Swisscom	
eSBC	Cisco IOS XE Software, Version 17.06.04
C-SBC	Oracle SCZ9.1.0
SESM	Ribbon 21.0.26

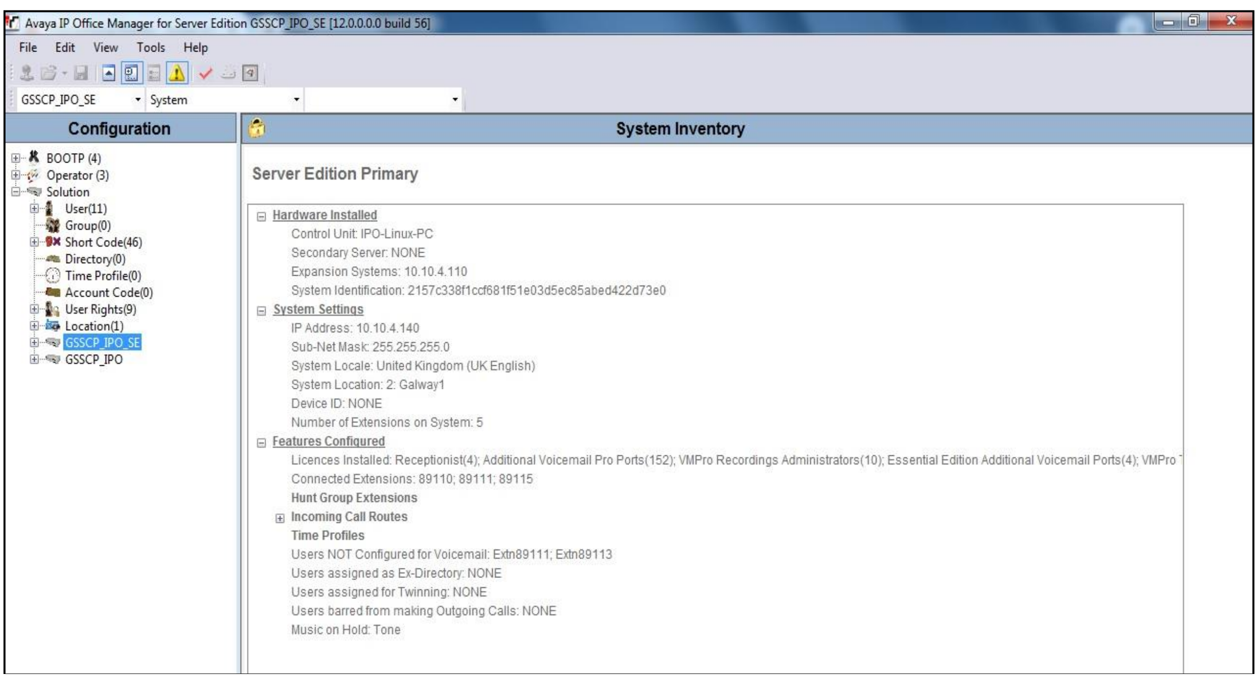
Note – Testing was performed with IP Office Server Edition with 500 V2 Expansion R12.0. Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition. **Note:** that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analogue 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 Swisscom Enterprise SIP service. 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 proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials.

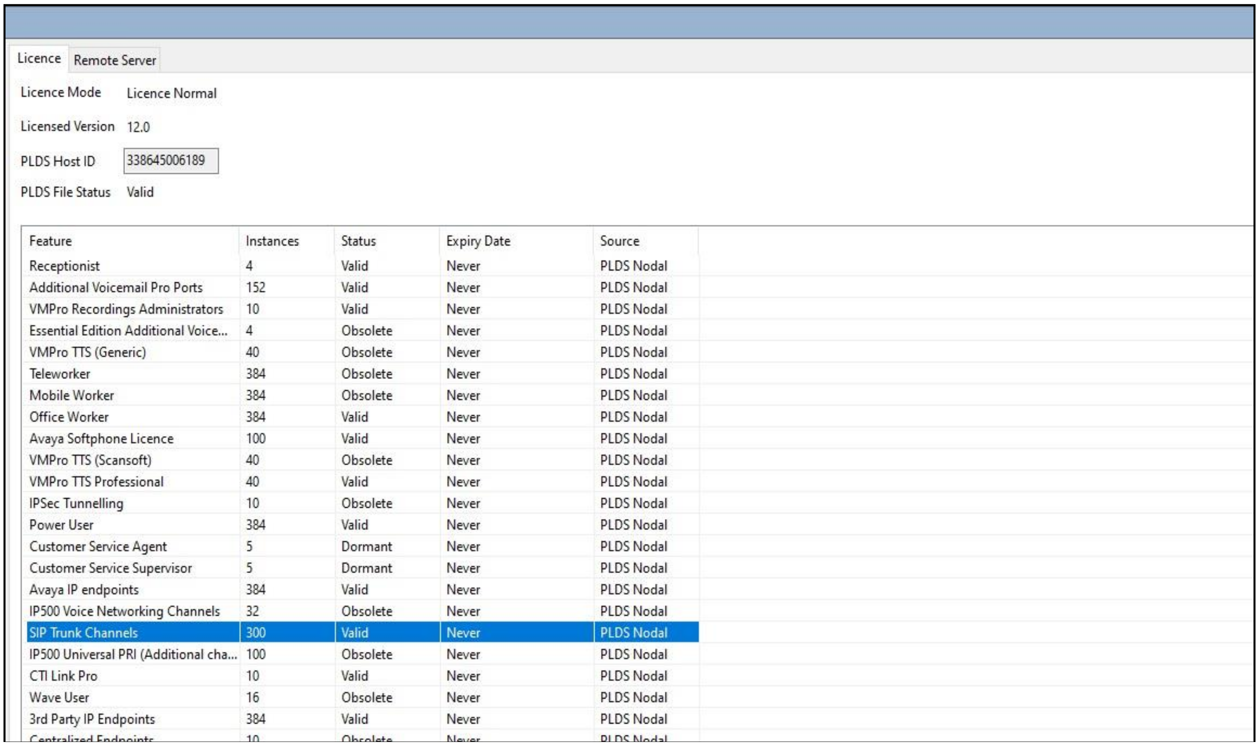


A management window will appear similar to the one in the next section. All the Avaya IP Office configurable components are shown in the left pane known as the Navigation Pane. The pane on the right is the Details Pane. These panes will be referenced throughout the Avaya IP Office configuration. All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



5.1. Verify System Capacity

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



Licence | Remote Server

Licence Mode: Licence Normal

Licensed Version: 12.0

PLDS Host ID: 338645006189

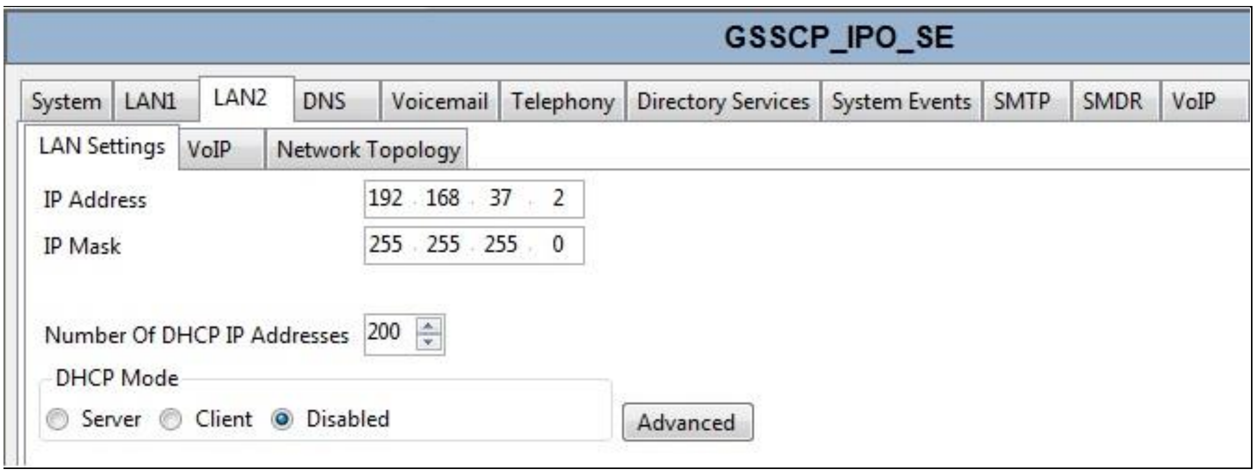
PLDS File Status: Valid

Feature	Instances	Status	Expiry Date	Source
Receptionist	4	Valid	Never	PLDS Nodal
Additional Voicemail Pro Ports	152	Valid	Never	PLDS Nodal
VMPro Recordings Administrators	10	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	10	Obsolete	Never	PLDS Nodal
Power User	384	Valid	Never	PLDS Nodal
Customer Service Agent	5	Dormant	Never	PLDS Nodal
Customer Service Supervisor	5	Dormant	Never	PLDS Nodal
Avaya IP endpoints	384	Valid	Never	PLDS Nodal
IP500 Voice Networking Channels	32	Obsolete	Never	PLDS Nodal
SIP Trunk Channels	300	Valid	Never	PLDS Nodal
IP500 Universal PRI (Additional cha...	100	Obsolete	Never	PLDS Nodal
CTI Link Pro	10	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

5.2. LAN2

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). For the compliance test, the **LAN2** interface was used to the Avaya IP Office to the Swisscom Enterprise SIP platform.

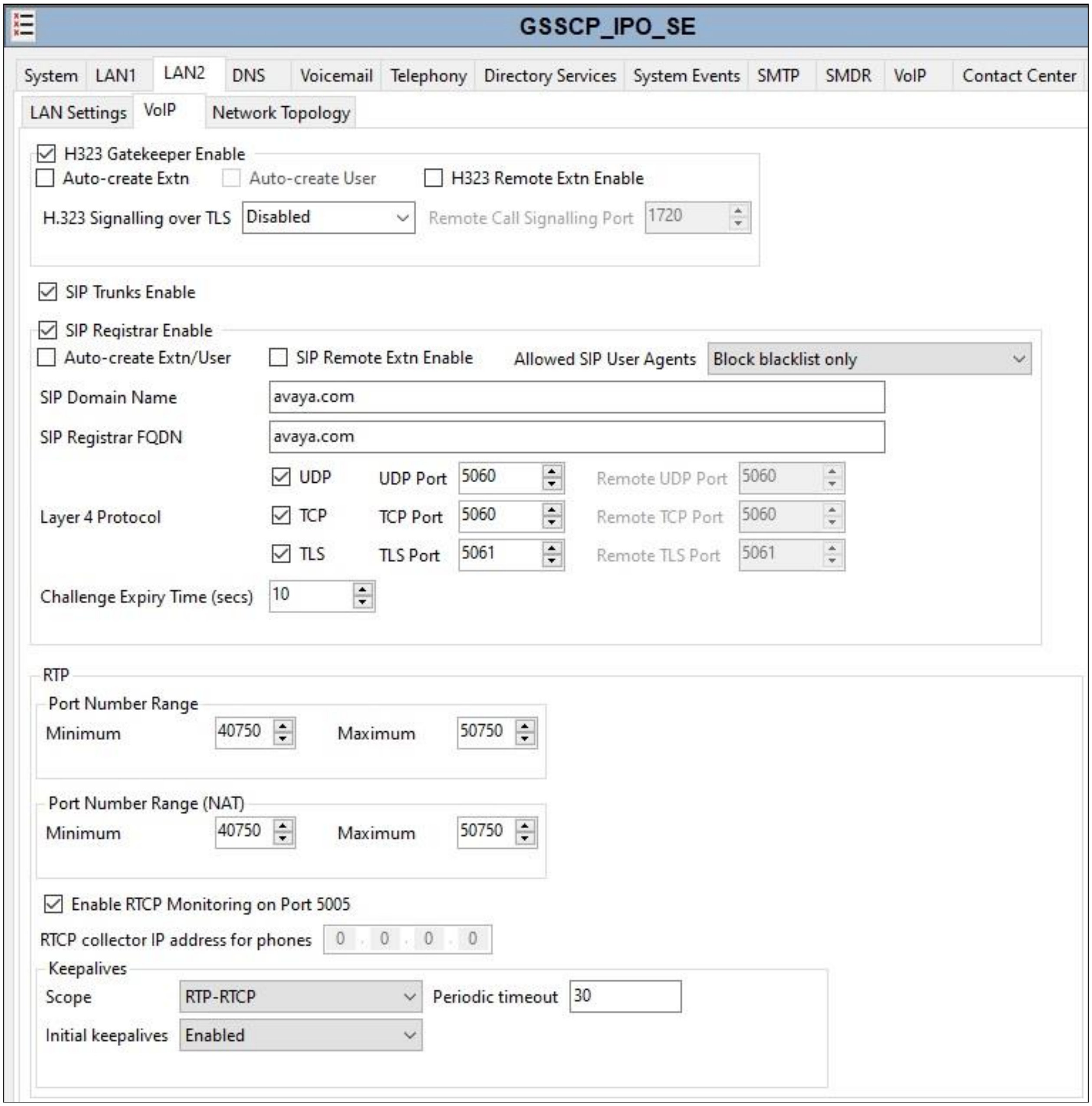
To access the LAN2 settings, first navigate to **System → GSSCP_IPO_SE** in the Navigation Pane where GSSCP_IPO_SE is the name of the IP Office. Navigate to the **LAN1 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are the private interface of the IP Office. All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).



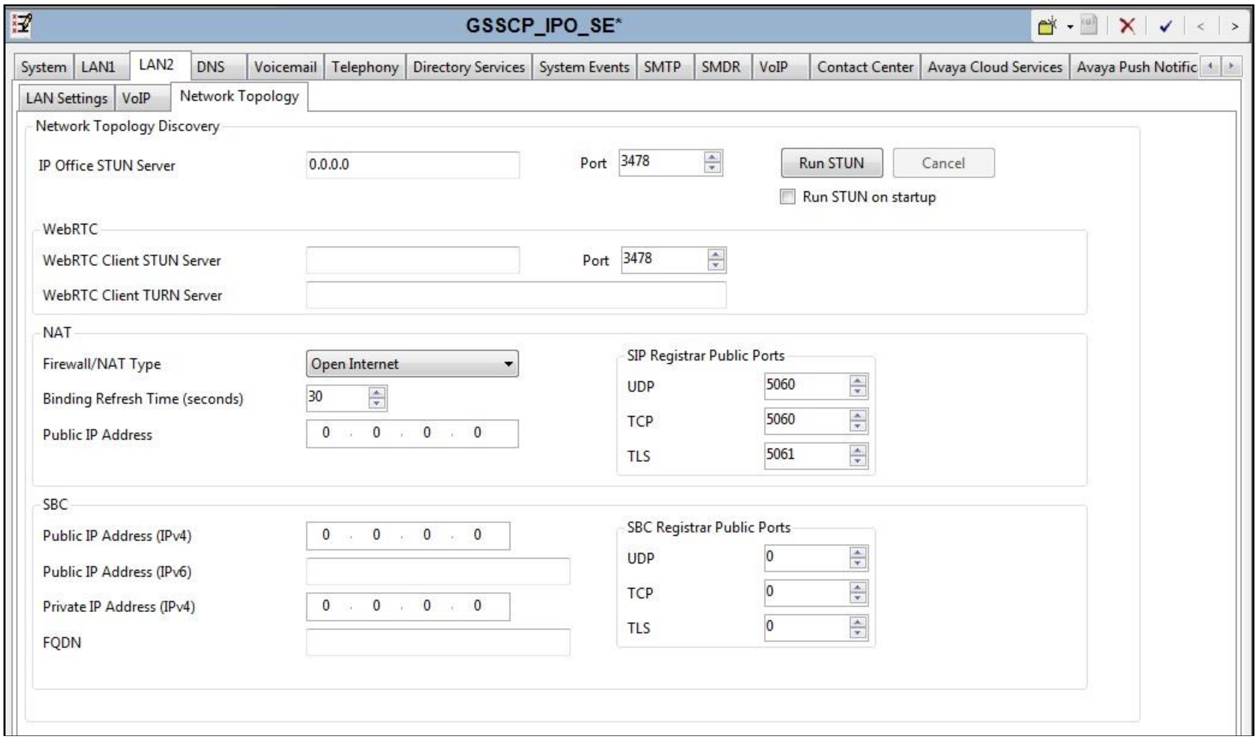
On the **VoIP** tab in the Details Pane, the **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol. Set **H.323 Signalling over TLS** to **Preferred** to allow IP Office endpoints to use TLS for signalling. Check the **SIP Trunks Enable** box to enable 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. 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 LAN1 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).

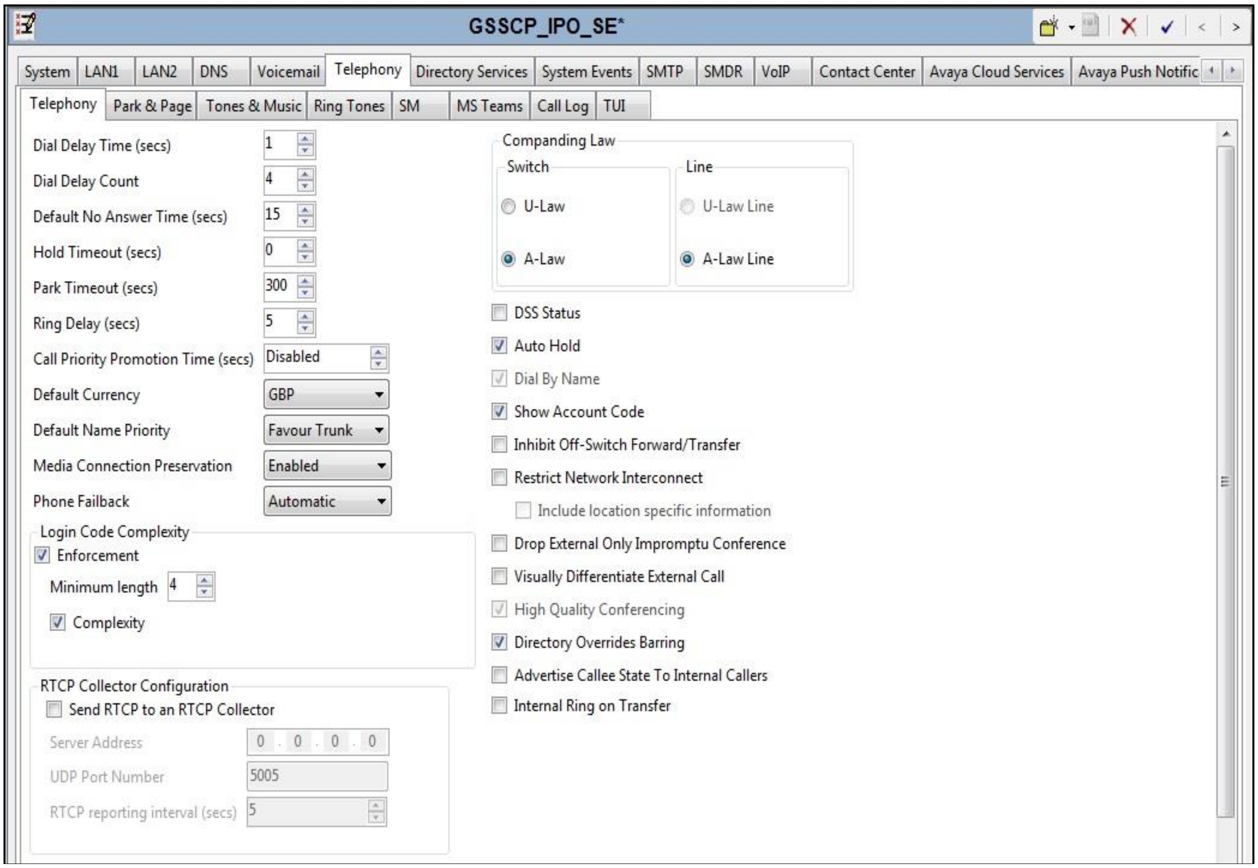


On the **Network Topology** tab, set the **Firewall/NAT Type** from the pulldown menu to **Open Internet**. With this configuration, the **STUN Server IP Address** and **STUN Port** are not used as NAT was not required for this configuration, therefore resulting in no requirement for a STUN server. The **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).



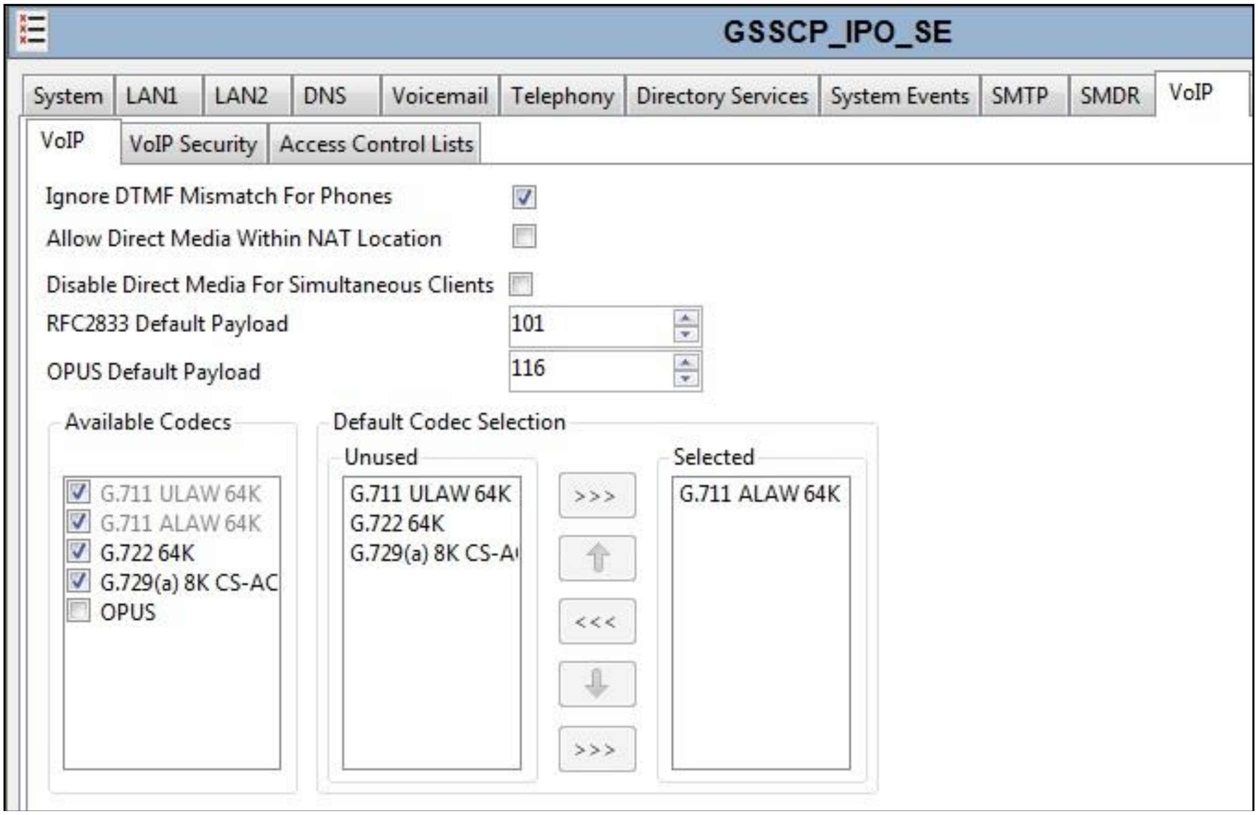
5.3. System Telephony Settings

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 selection.



5.5. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and Swisscom Enterprise 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:

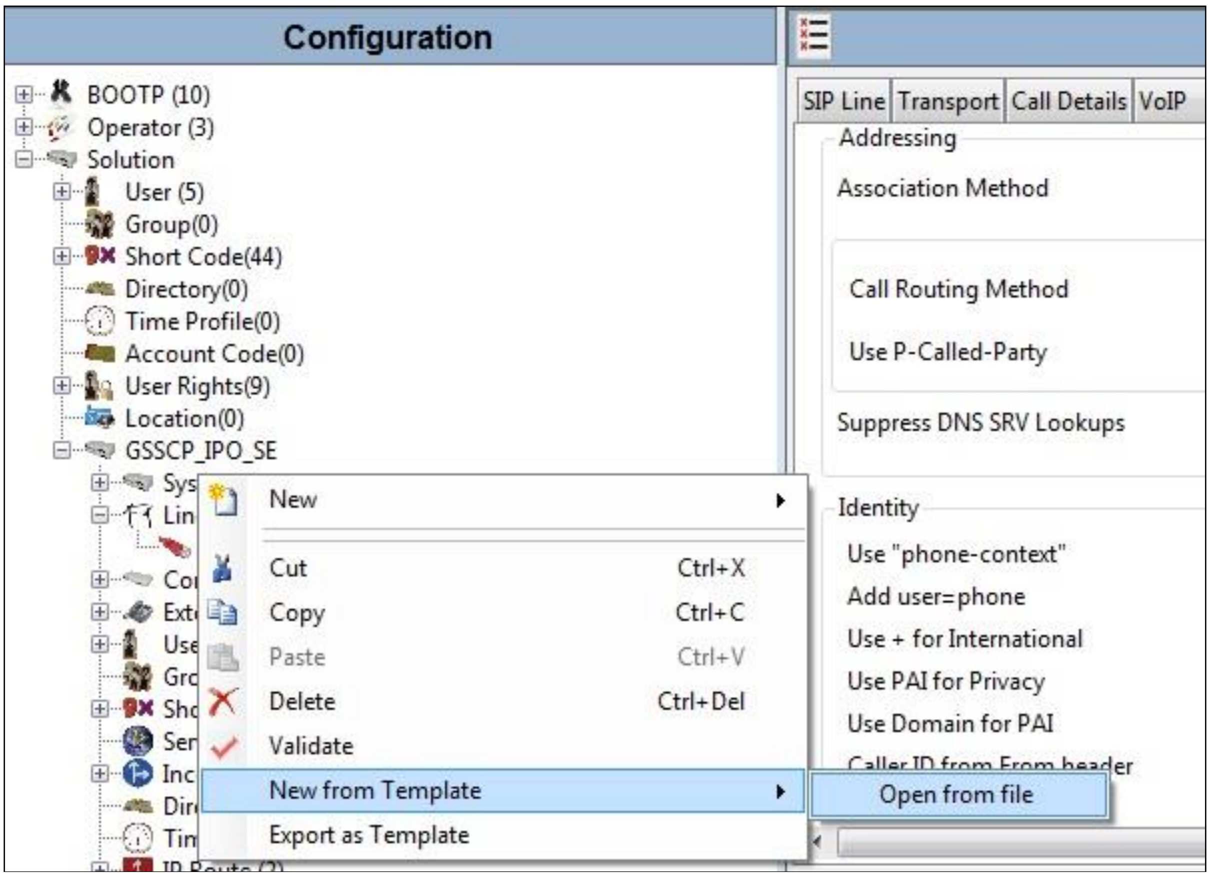
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

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**.

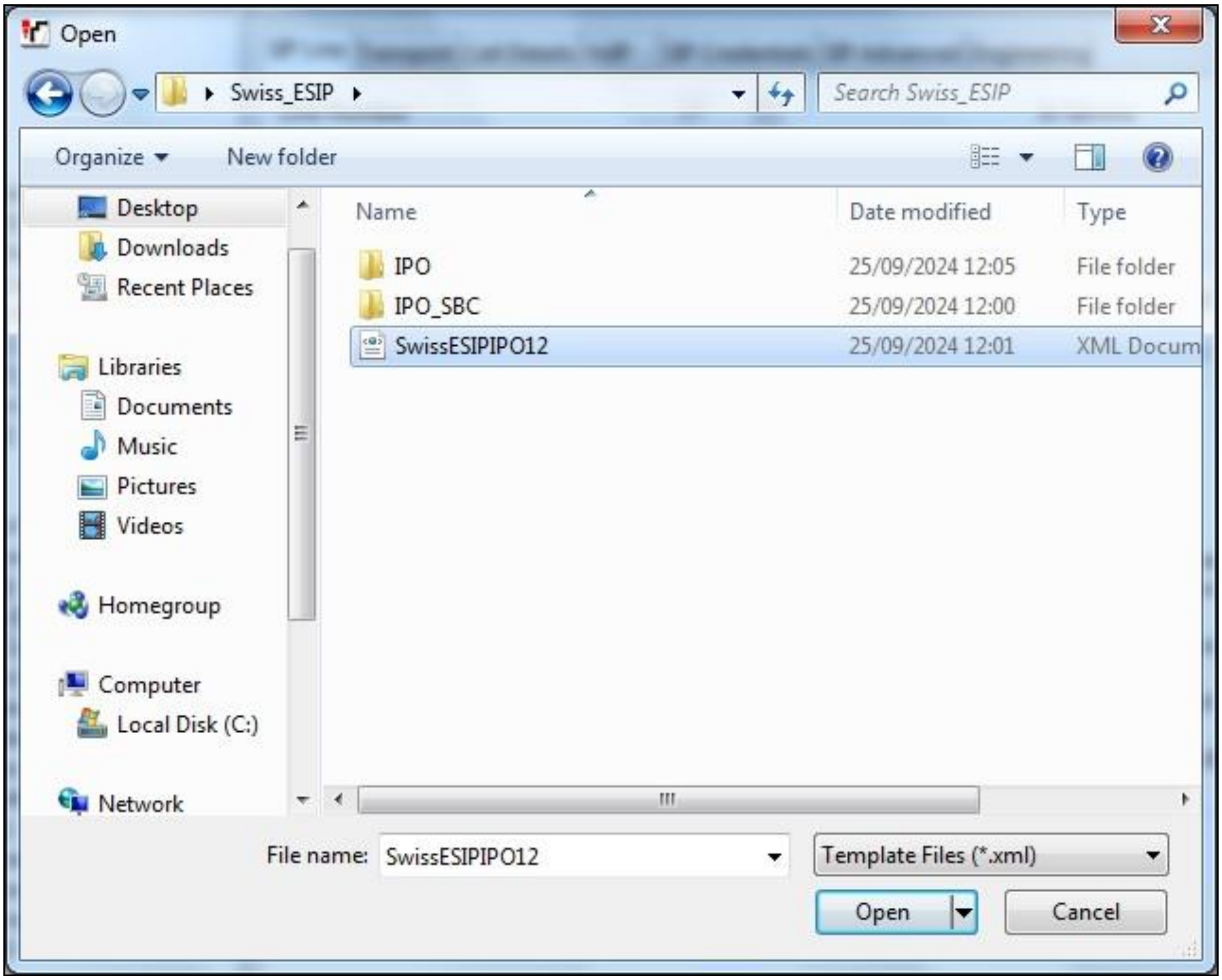
5.5.1. SIP Line From Template

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.



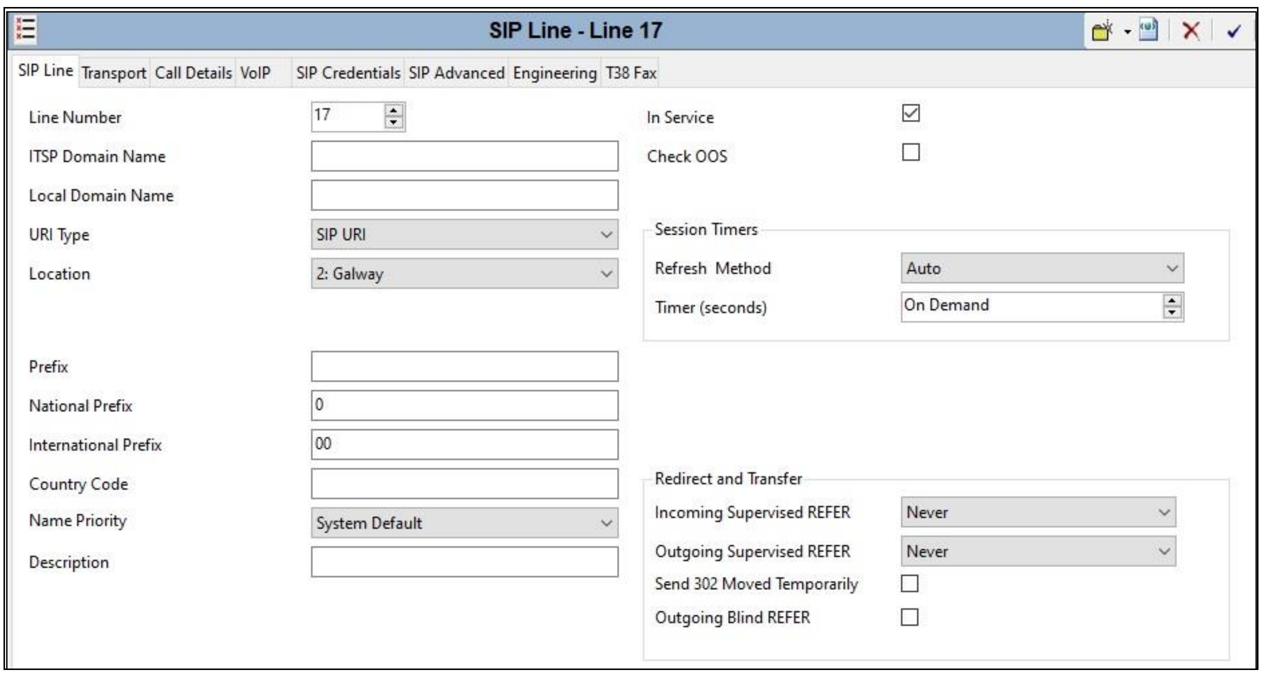
The SIP Line is automatically created and can be verified and edited as required using the configuration described in **Section 5.5.2**.

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 **ITSP Domain Name** to a domain name provider by the Service Provider if required, however no ITSP Domain Name was used in this configuration.
- Set **Location** to that defined for Emergency calls as described in **Section 5.9**.
- Set **National Prefix** to **0** and **International Prefix** to **00** for number conversion as follows: outbound national and international called party numbers are converted to E.164 format; inbound national and international calling party numbers are converted to diallable format.
- Ensure the **In Service** box is checked.
- Leave the **Refresh Method** at the default value of **Auto** which results in re-INVITE being used for Session Refresh.
- Leave **Timer (seconds)** at the default value of **On Demand**. This value allows the Session Refresh interval to be set by the network.
- Set **Incoming Supervised REFER** and **Outgoing Supervise REFER** to **Never**. REFER is not supported by Swisscom Enterprise SIP platform.
- Default values may be used for all other parameters.

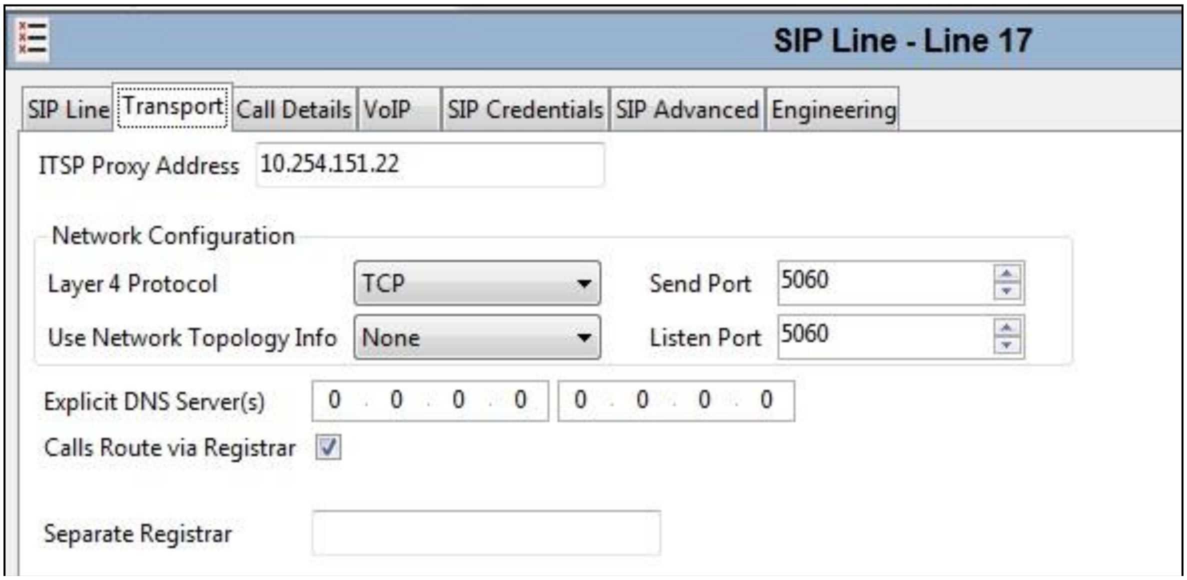
On completion, click the **OK** button (not shown).



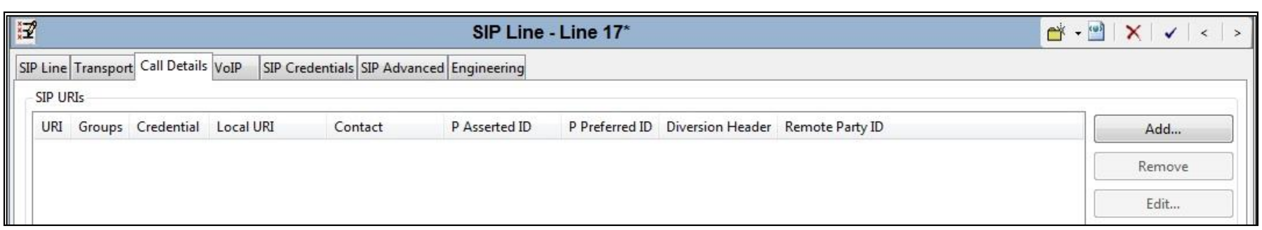
Select the **Transport** tab and set the following:

- Set **ITSP Proxy Address** to the IP Address for Swisscom Enterprise SIP platform.
- Set **Layer 4 Protocol** to **TCP**.
- Set **Send Port** to **5060** and **Listen Port** to **5060**.
- 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.

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



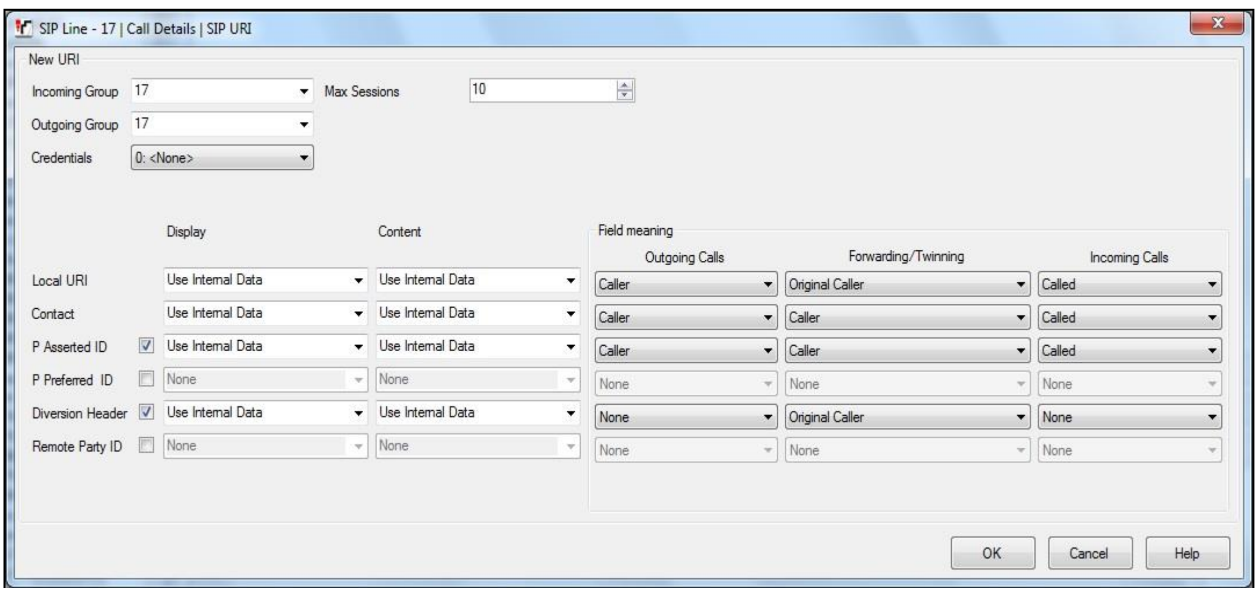
After the SIP line parameters are defined, the SIP URIs that Avaya IP Office will accept on this line must be created. To create a SIP URI entry, select the **Call Details** tab and click on **Add**.



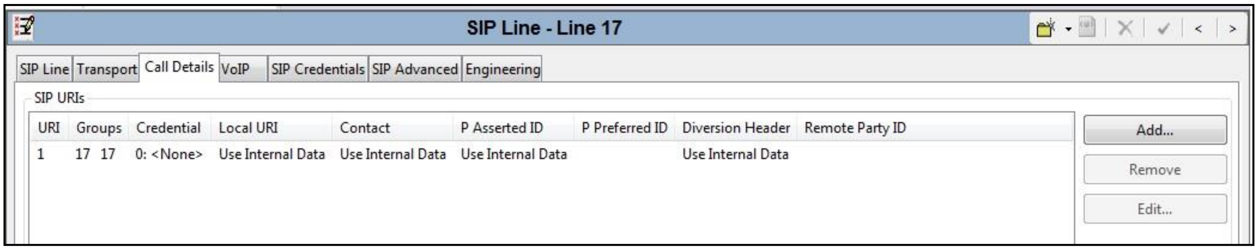
A SIP URI is shown in this example that is used for calls to and from extensions that have a DDI number assigned to them. Additional SIP URI’s may be required for calls to services such as Voicemail Collect and the Mobile Twinning FNE, these would be for incoming calls only.

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 are analysed in the Incoming Call Route settings described in **Section 5.8**. In the test environment a value of **17** was used for the Swisscom.
- 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** and **P Asserted ID** to **Use Internal Data** for both the **Display** name and **Content**. On incoming calls, this will analyse the Request-Line sent by Swisscom 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.
- Set the **Outgoing Calls**, **Forwarding/Twinning** and **Incoming Calls** at their respective values of **Caller**, **Original Caller** and **Called** for the **Local URI** setting call details. Set the **Outgoing Calls**, **Forwarding/Twinning** and **Incoming Calls** at their respective values of **Caller**, **Caller** and **Called** for the **Contact** and **P Asserted ID** setting call details. Set the **Outgoing Calls**, **Forwarding/Twinning** and **Incoming Calls** at their respective values of **None**, **Original Caller** and **None** for the **Diversion Header** setting call details.



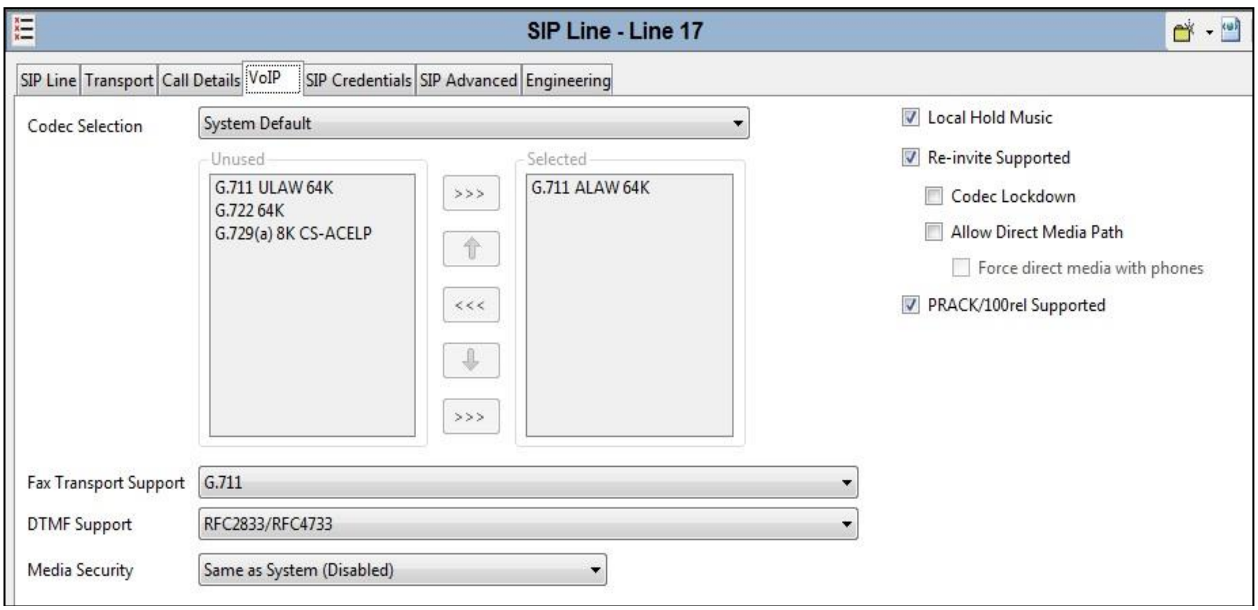
The following screenshot shows the completed configuration:


SIP Line - Line 17
SIP Line
Transport
Call Details
VoIP
SIP Credentials
SIP Advanced
Engineering
SIP URIs
URI
Groups
Credential
Local URI
Contact
P Asserted ID
P Preferred ID
Diversion Header
Remote Party ID
1
17 17
0: <None>
Use Internal Data
Use Internal Data
Use Internal Data
Use Internal Data
Add...
Remove
Edit...

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

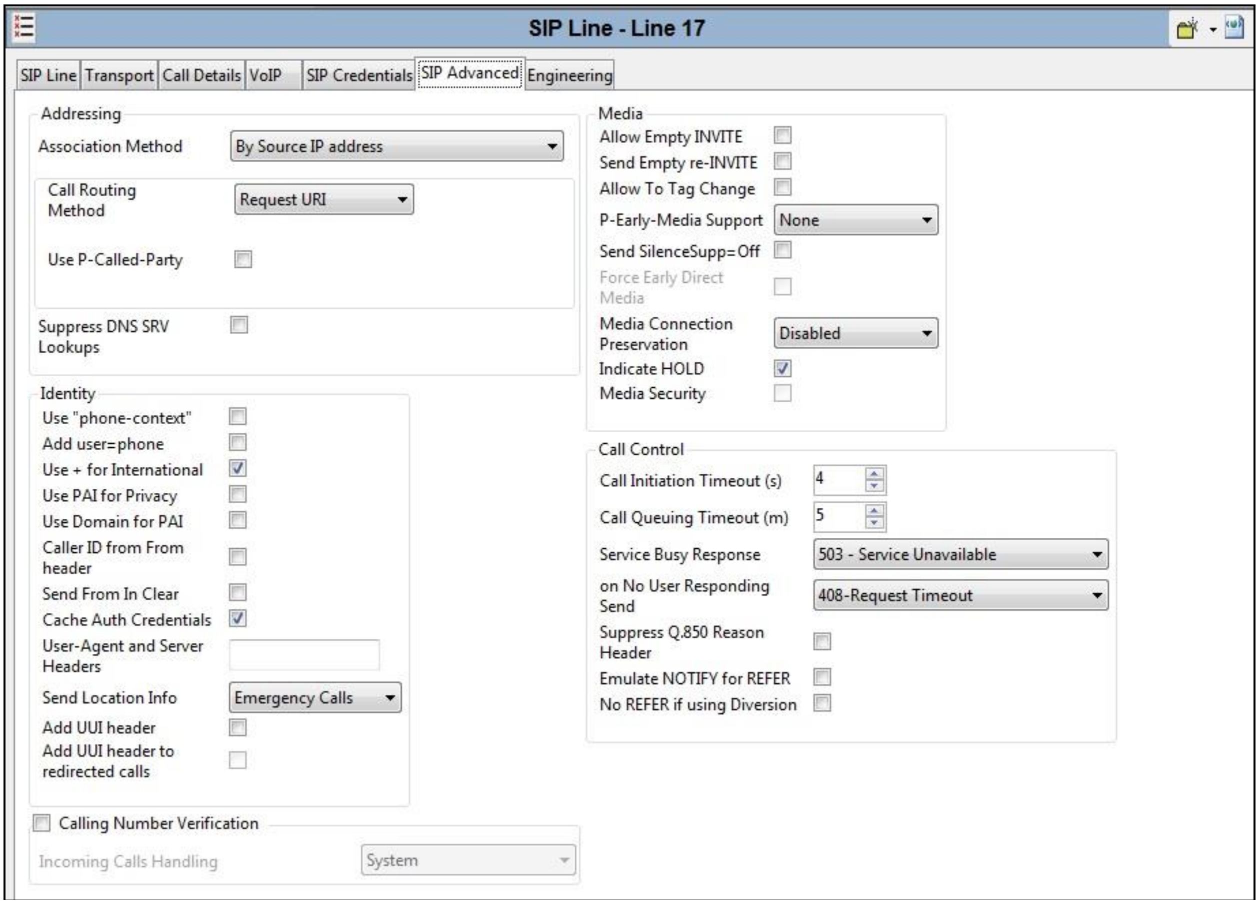
- Select **System Default** from the drop-down menu as system default codecs were already defined in **Section 5.4**.
- Set the **Fax Transport Support** box to **G.711** as this is the preferred method of fax transmission for Swisscom.
- Set the **DTMF Support** field to **RFC2833/RFC4733**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Check **Media Security** to **Same as System (Disabled)**.
- 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 **Use + for International** as E.164 numbering is used on the SIP Trunk.
- Select **Emergency Calls** from the **Send Location Info** drop down menu if required
- Default values may be used for all other parameters.



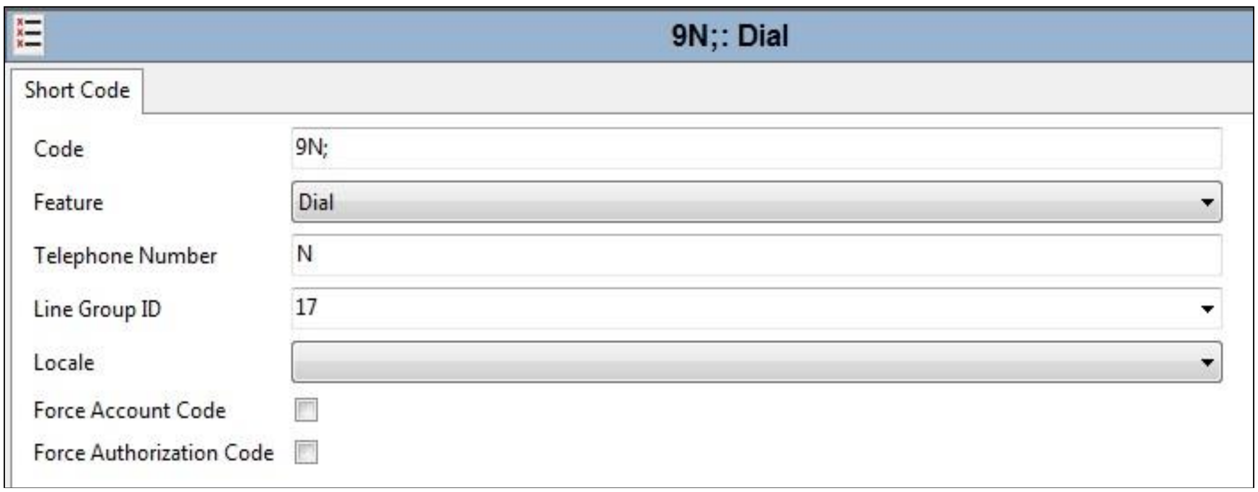
Note: It is advisable at this stage to save the configuration as described in **Section 5.11** to add the Line Group ID defined in **Section 5.5.2** available.

5.6. ShortCodes

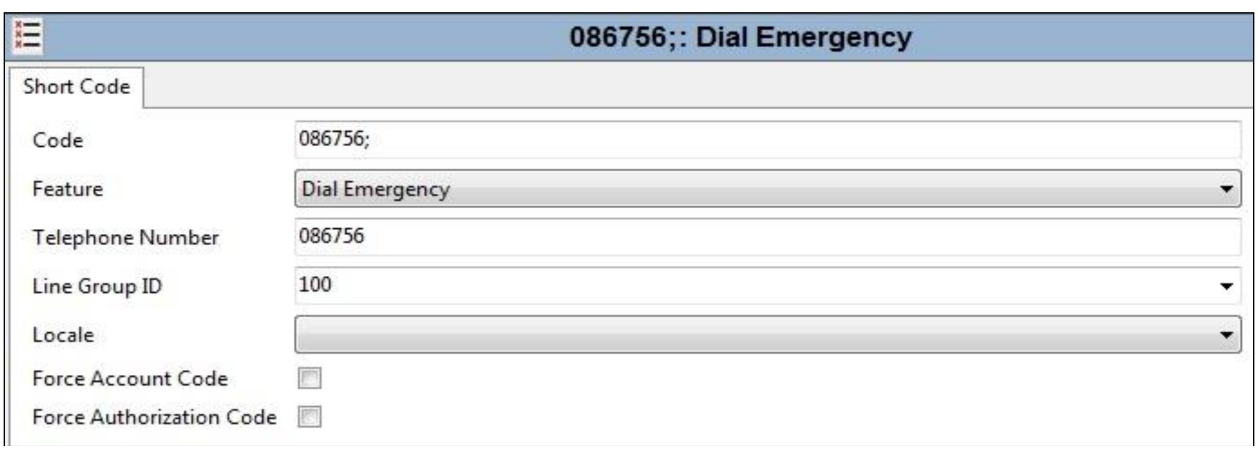
Define a short code to route outbound traffic to the SIP line and route incoming calls from mobility extensions to access Feature Name Extensions (FNE) hosted on IP Office. To create a short code, right-click **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

- 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 the dialled number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N**. The **Telephone Number** field is used to construct the Request URI and To Header in the outgoing SIP INVITE message.
- Set the **Line Group Id** to the outgoing line group number defined on the SIP URI tab on the SIP Line in **Section 5.5.2**.

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



A further example is shown for an emergency number.

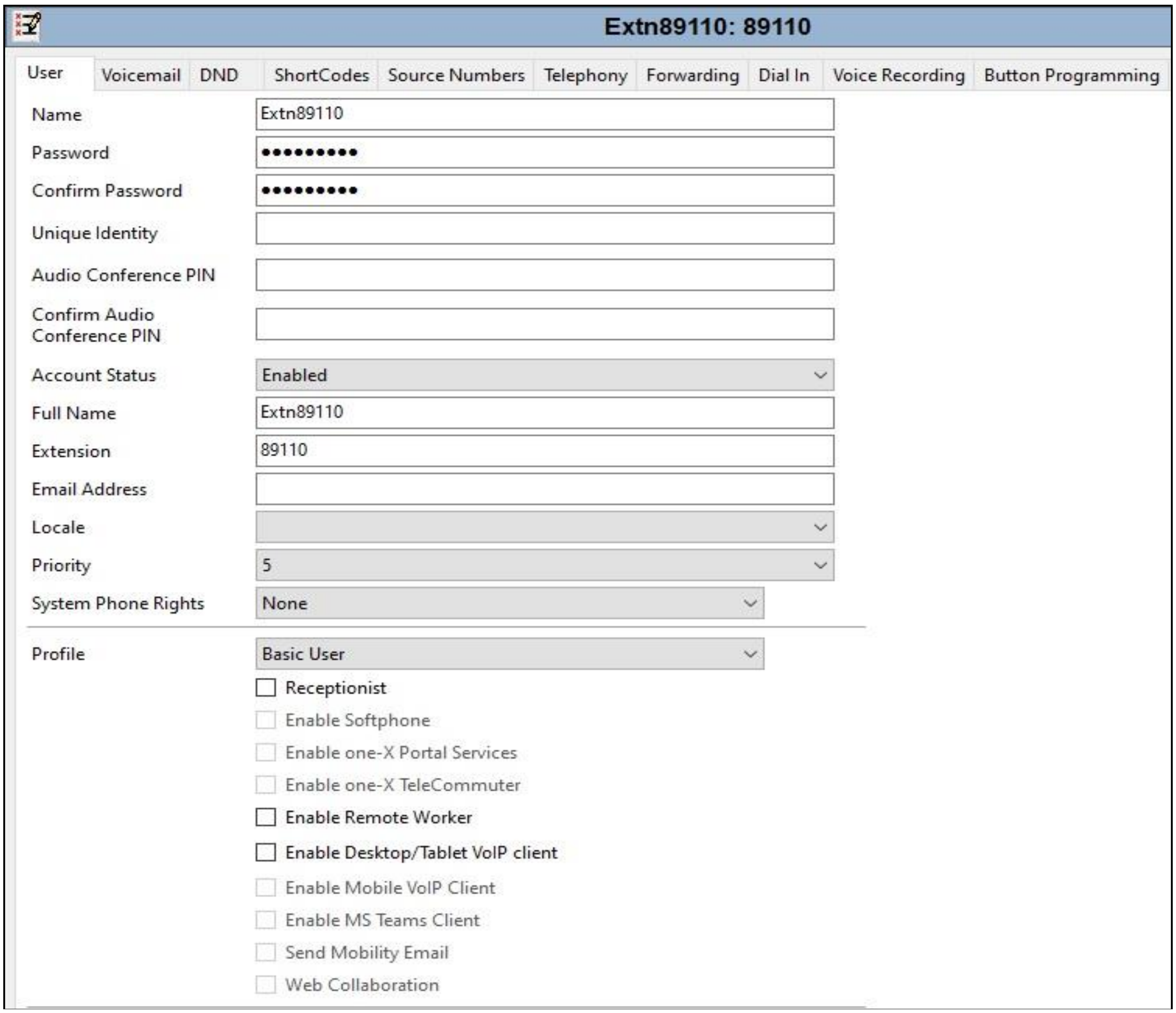


5.7. User

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.

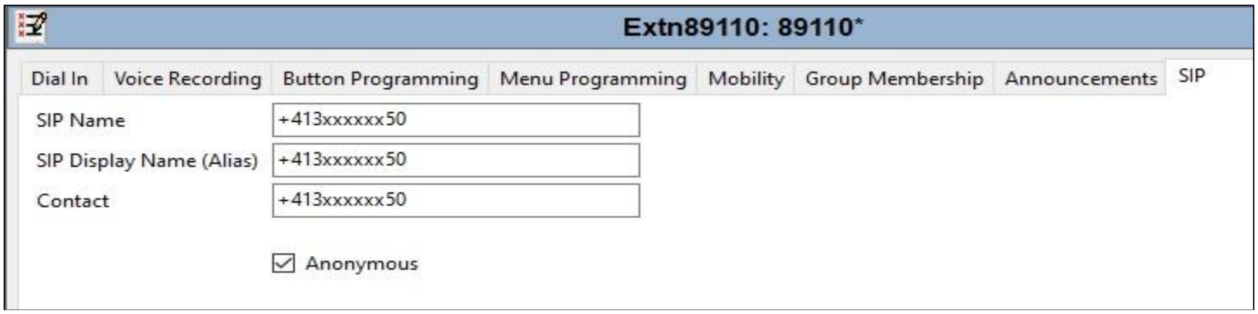
The following example shows the configuration required for a SIP Endpoint.

- 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.



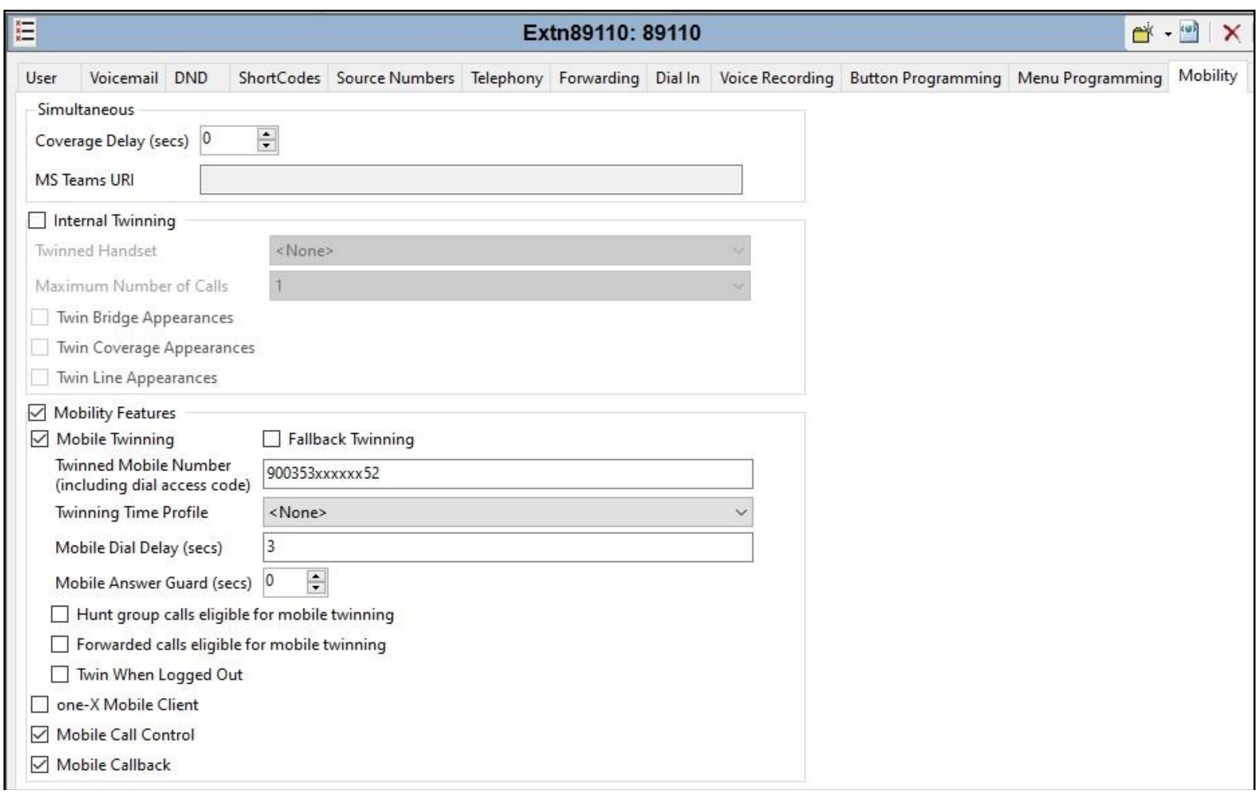
SIP endpoints require setting of the **SIP Registrar Enable** as described in **Section 5.2**.

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 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 Swisscom.



Note: The **Anonymous** box can be used to restrict Calling Line Identity (CLIR) as discussed **Section 2.2**.

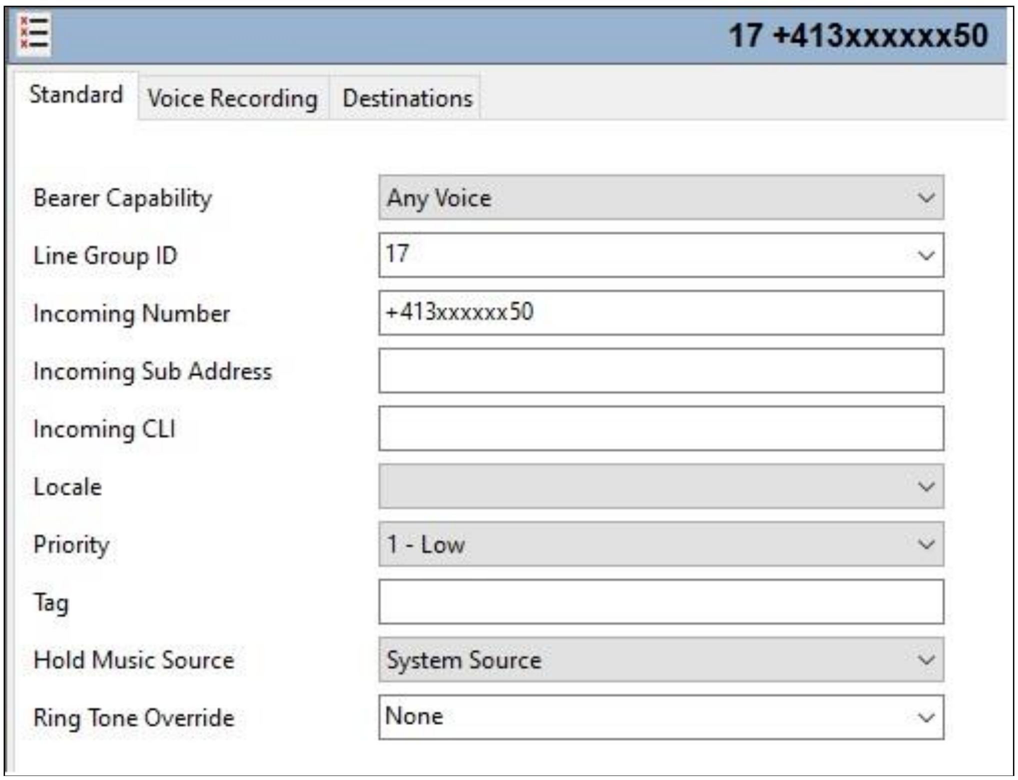
The following screen shows the Mobility tab for user 89110. The **Mobility Features** and **Mobile Twinning** are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone over the SIP Trunk. Other options can be set accordingly to customer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



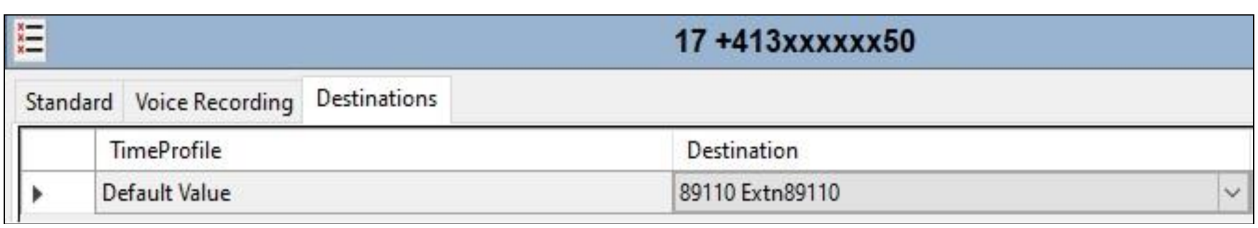
5.8. Incoming Call Routing

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.



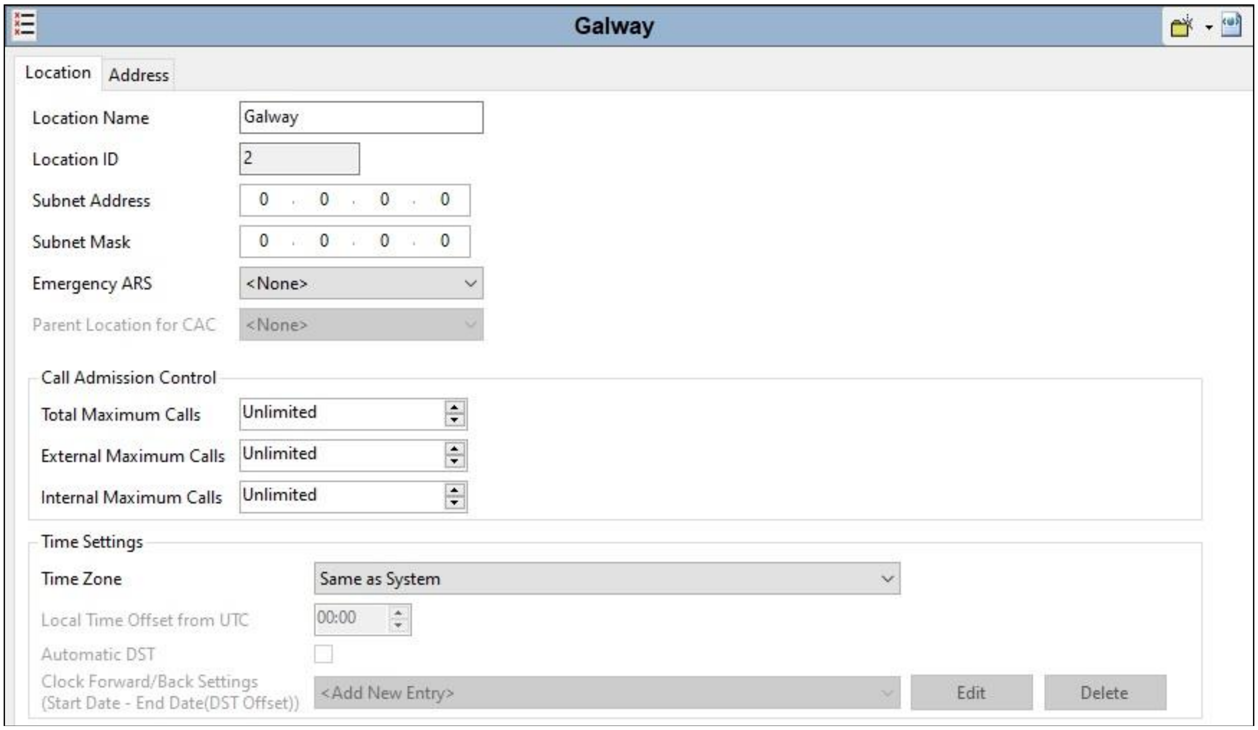
On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number **+413xxxxxx50** on line 17 are routed to extension 89110.



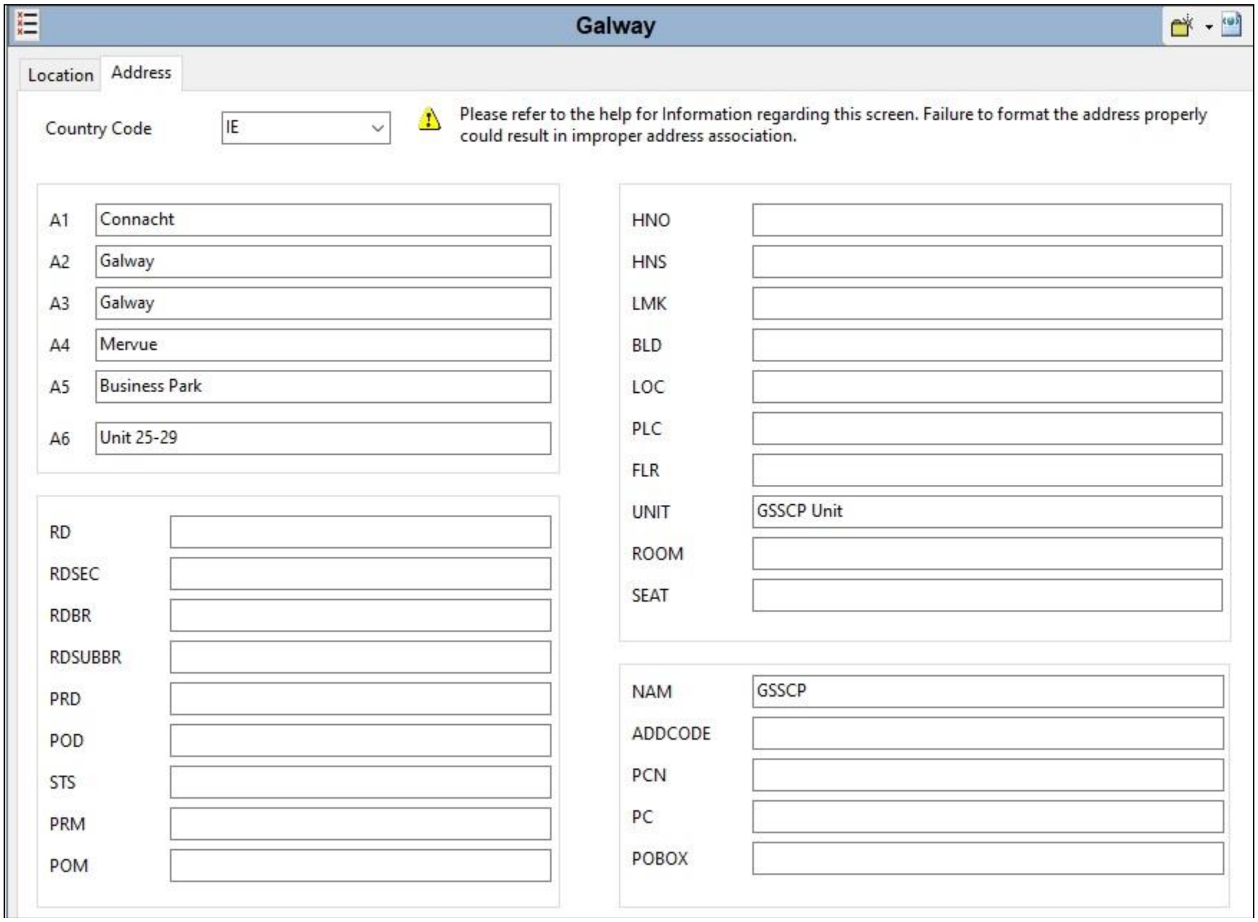
5.9. Location

If Location information is required for calls to Emergency Services, right-click **Location** in the Navigation Pane and select **New**, (not shown). On the **Location** tab of the Details Pane, enter the parameters as required. An example used during testing is shown below:

- Define a **Location Name**.
- Define a **Subnet Address** and **Subnet Mask** as required. In the test environment, there was no differentiation based on subnet.
- In the example, all other fields were left at default values.



Click on the **Address** tab and enter data as required. The following screenshot shows an example used during testing:



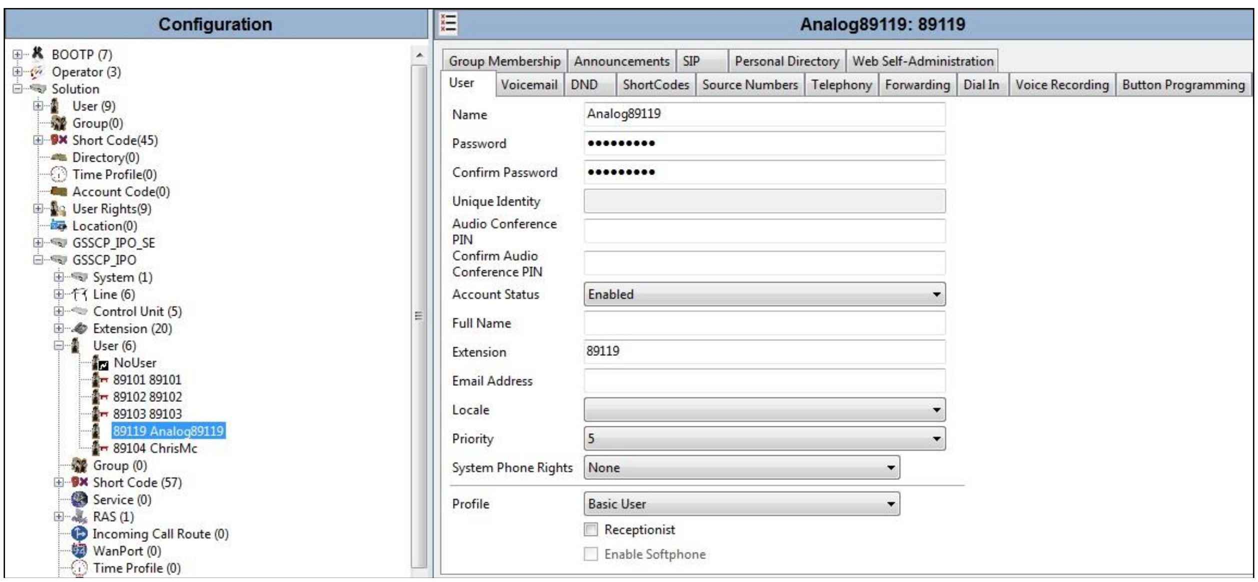
5.10. Fax

At Release 12.0, both G.711 and T.38 Fax is supported on IP Office Server Edition when using an IP Office Expansion (500 V2). The Swisscom Enterprise SIP Trunk testing was carried out using this configuration with only the analog 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. Analog 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.



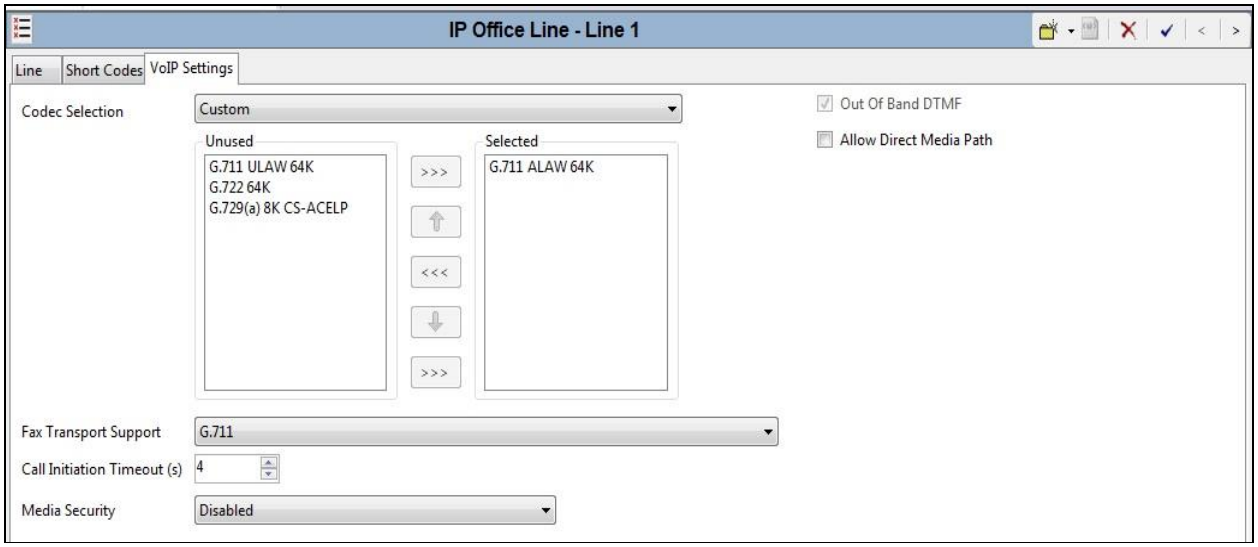
Configure other settings as described in **Section 5.7**.

5.10.2. G.711 Fax Settings

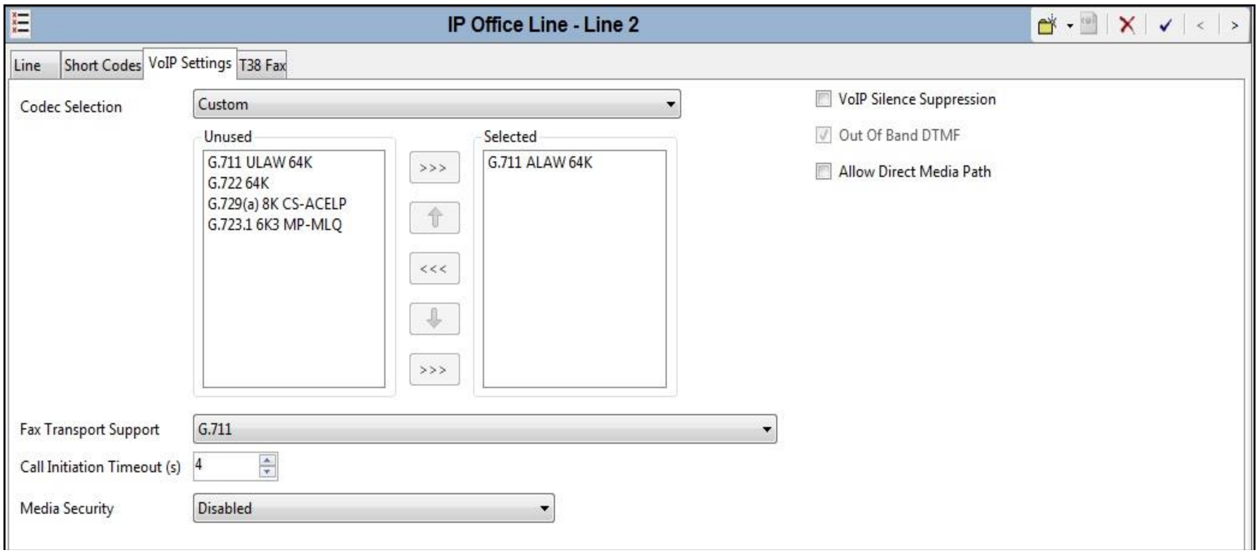
The G.711 Fax settings are defined on the SIP Line between the Expansion and the Server. Note that the VoIP settings for G.711 are required in three places in this configuration:

- The SIP Line for the Swisscom Smart Business Connect SIP platform 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 **G711**. The following screenshot shows the VoIP Settings for the IP Office Line between the Server and the Expansion on the Expansion:



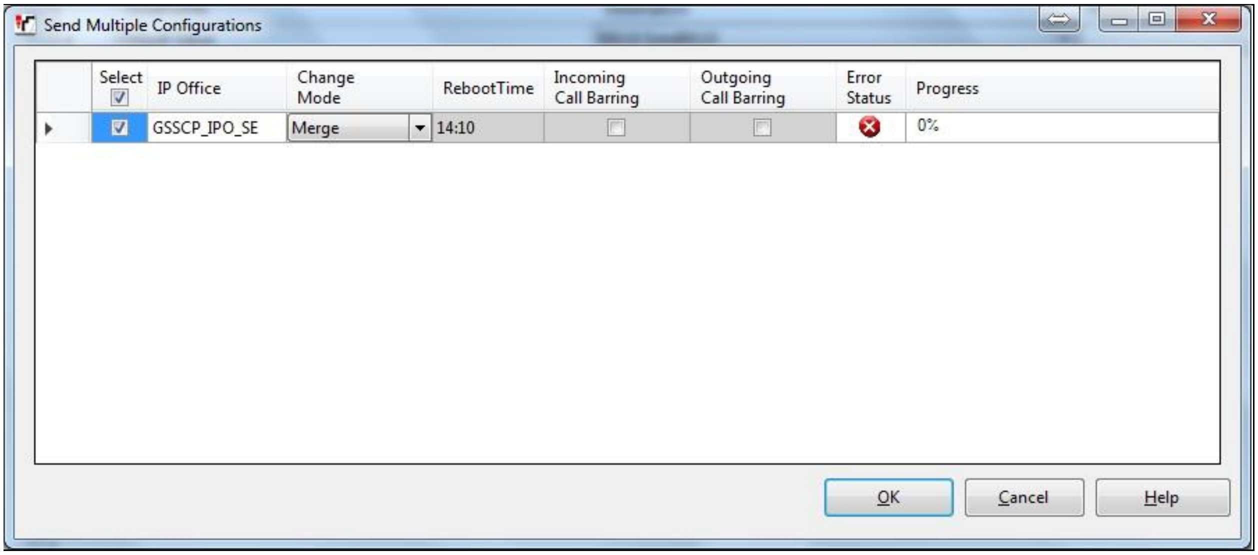
The following shows the **VoIP Settings** tab in the IP Office Line for the Expansion in the Server configuration:



Refer to **Section 5.5.2** for the VoIP Settings on the SIP Line for the Swisscom 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. Configure the Swisscom Equipment

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

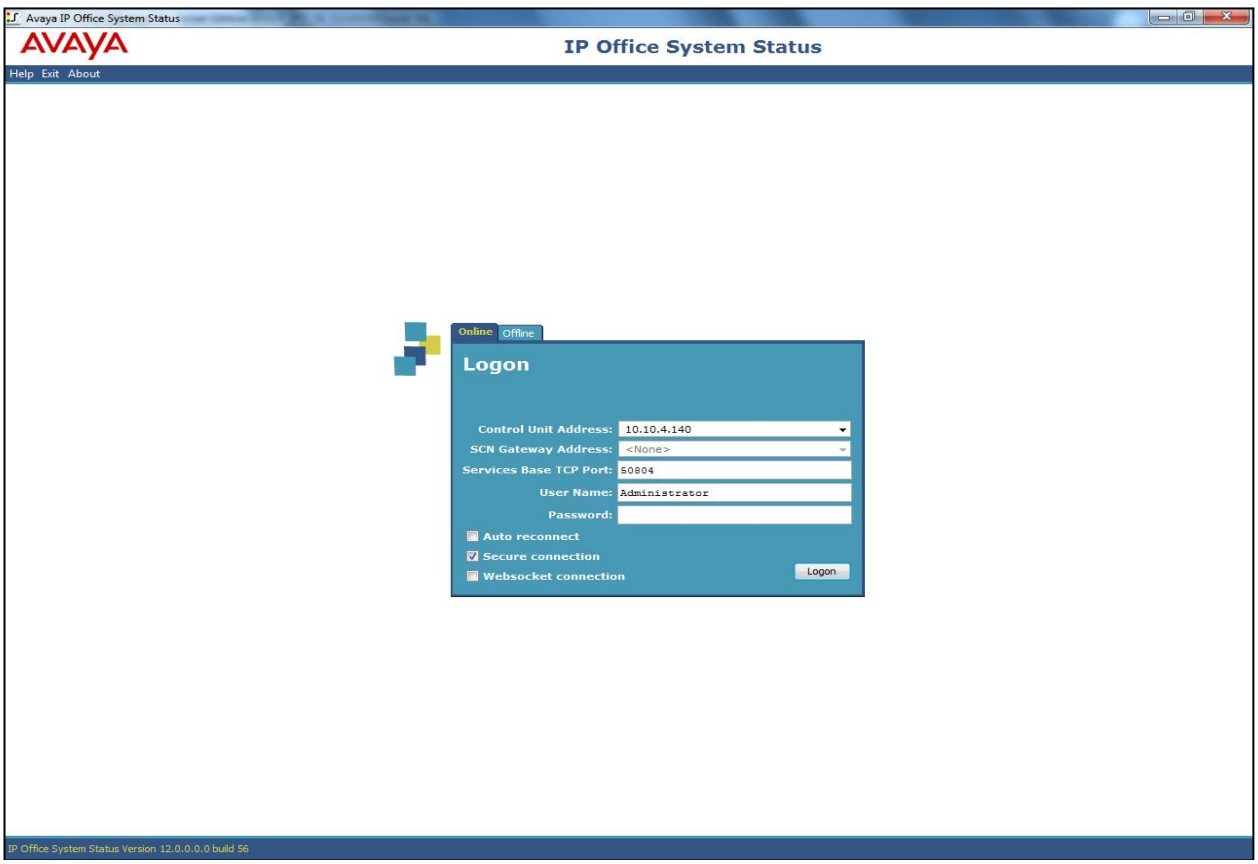
7. Verification Steps

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

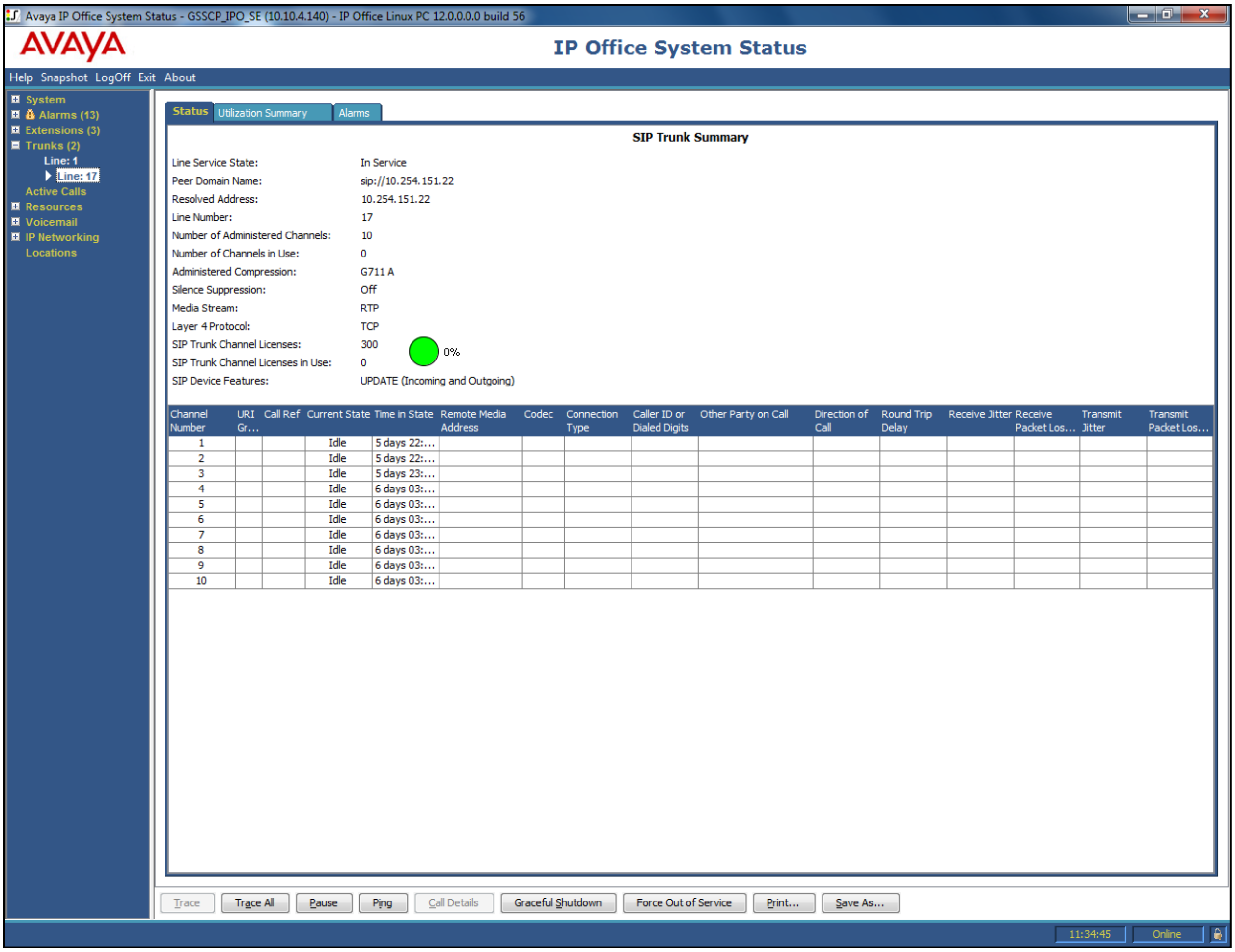
7.1. SIP Trunk Status

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 **Username** and **Password** are the same as those used for IP Office Manager.

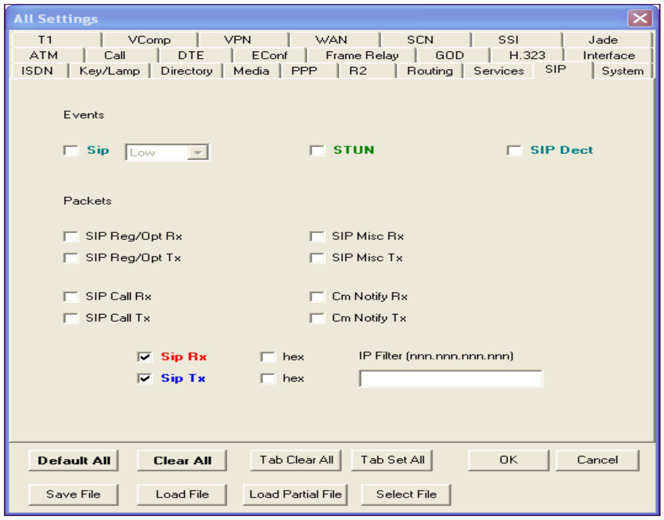


From the left-hand menu expand **Trunks** and choose the SIP trunk (**17** in this instance). The status window will show the status as being idle and time in state if the Trunk is operational.

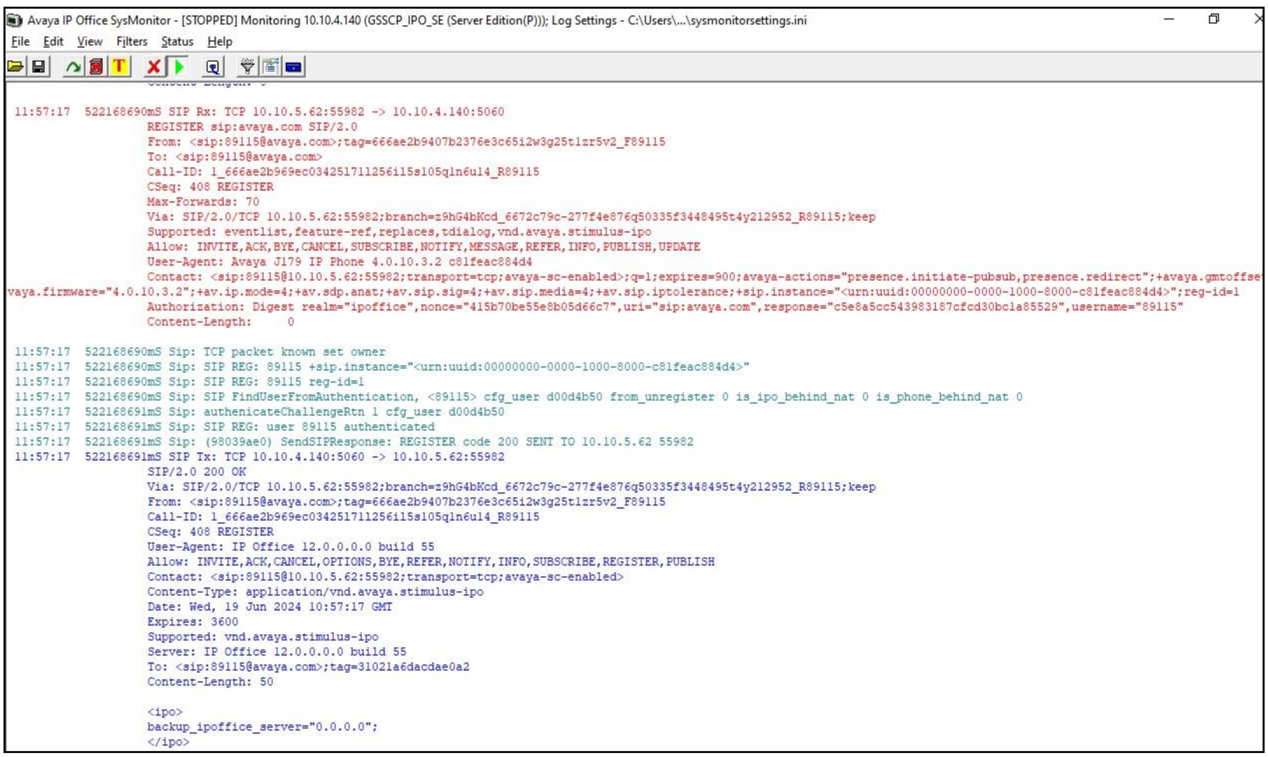


7.2. Monitor

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 REGISTERs being sent between IP Office and a SIP phone.



8. Conclusion

These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office Server Edition R12.0 and Swisscom Enterprise SIP Trunk service 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 Server Edition R12.0 can be configured to interoperate successfully with Swisscom's Enterprise SIP Trunk service. Swisscom's Enterprise SIP Trunk 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 12.0, Apr 2024.
[2] *Deploying IP Office Server Edition Servers*, Release 12.0, Apr 2024.
[3] *Deploying an IP500 V2 IP Office System*, Release 12.0, Apr 2024.
[4] *Administering Avaya IP Office with IP Office Web Manager*, Release 12.0, May 2024.
[5] *Administering Avaya IP Office with IP Office Manager*, Release 12.0, May 2024.
[6] *Using Avaya IP Office System Status*, Apr 2024.
[7] *Using IP Office System Monitor*, Apr 2024.
[8] *Administrating Voicemail Pro*, Release 12.0, May 2024.
[9] *Using Avaya Workplace Client for Windows*, Nov 2023.
[10] *IP Office SIP Phone Installation Notes*, Apr 2024.
[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.