



## **Application Notes for Configuring Avaya IP Office 11.0 with Telenor IPT Multi-User SIP Trunk – Issue 1.0**

### **Abstract**

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between Telenor IPT Multi-User SIP Trunk and Avaya IP Office.

The Telenor IPT Multi-User SIP Trunk Platform provides PSTN access via a SIP trunk connected to the Telenor Voice over Internet Protocol (VoIP) network as an alternative to legacy analogue or digital trunks. Telenor 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 the Telenor IPT Multi-User SIP Trunk and Avaya IP Office.

Customers using this Avaya SIP-enabled enterprise solution with Telenor IPT Multi-User 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 R11.0 to connect to the Telenor IPT Multi-User 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 Telenor IPT Multi-User SIP platform do not include use of any specific encryption features.

## 2.1. Interoperability Compliance Testing

Avaya IP Office was connected to the Telenor IPT Multi-User 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 and analogue telephones at the enterprise. Calls were routed to the enterprise across the SIP trunk from Telenor.
- 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 Telenor.
- Calls using the G.722 and G.711A codecs.
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using G.711 fax transmissions.
- DTMF transmission using RFC 2833 with successful Voice Mail for inbound and outbound calls.
- Inbound and outbound PSTN calls to/from Avaya Equinox Softphone client.
- 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, transfer, and conference.
- 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 Telenor IPT Multi-User SIP Trunk with the following observations:

- T.38 fax is not supported by Telenor and therefore was not tested.
- The Privacy Header is not included in the SIP INVITE for outbound calls with Calling Line Identity (CLIR) when using an IP Office short code (\*67 was used in the test configuration). This is a known issue currently under investigation. As a workaround, the anonymous button can be enabled on the SIP tab in **Section 5.7** to restrict CLIR.
- 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.

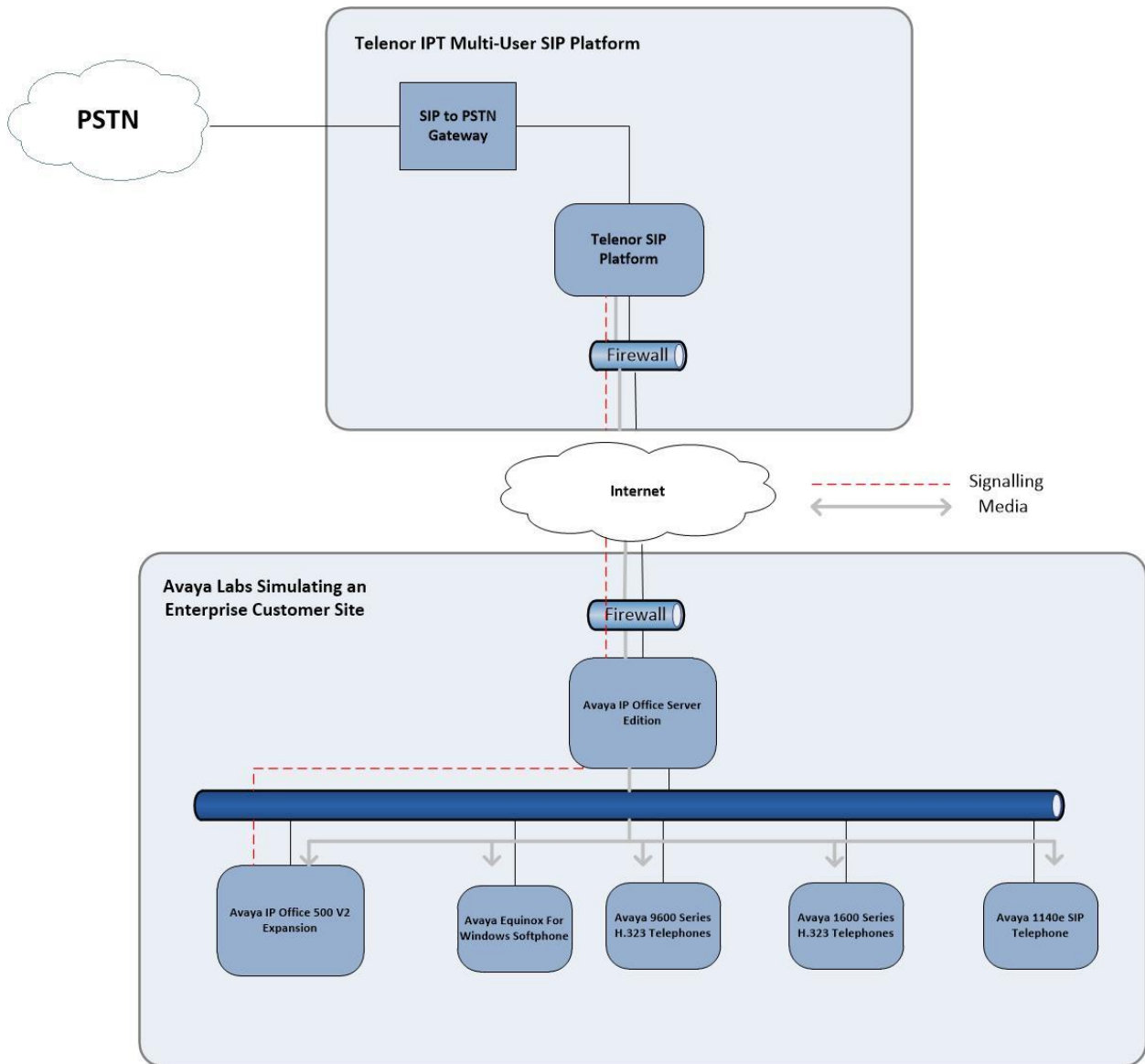
## 2.3. Support

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

For technical support on Telenor products please use the following web link:  
<https://www.telenor.no/bedrift/kundeservice/>

### 3. Reference Configuration

**Figure 1** illustrates the test configuration. The test configuration shows an enterprise site connected to the Telenor IPT Multi-User SIP Trunk. Located at the enterprise site is an Avaya IP Office 500 v2. 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 and Avaya Communicator for Web for mobility 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: Telenor IPT Multi-User SIP Trunk to Avaya IP Office Topology**

## 4. Equipment and Software Validated

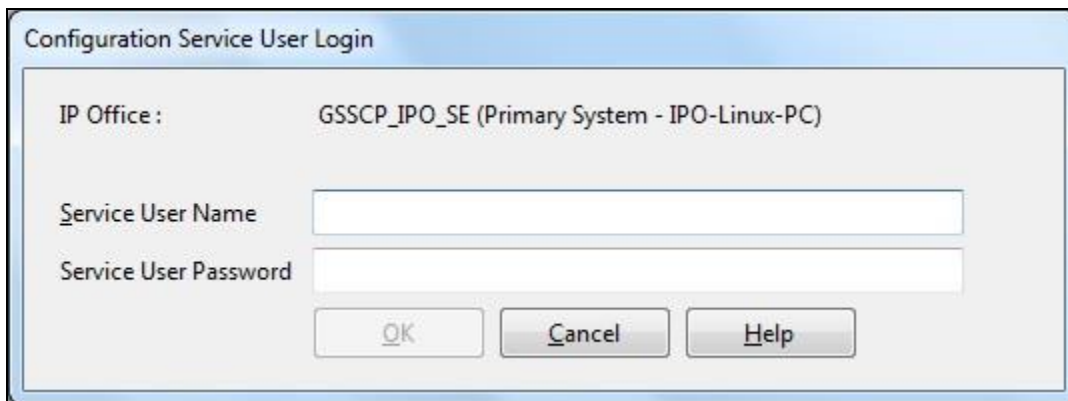
The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
<b>Avaya</b>	
Avaya IP Office Server Edition	Version 11.0.4.0.0 build 74
Avaya IP Office 500 V2	Version 11.0.4.0.0 build 74
Avaya Voicemail Pro Client	Version 11.0.4.0
Avaya IP Office Manager	Version 11.0.4.0.0 build 74
Avaya 1608 Phone (H.323)	1.3.12
Avaya 9611G Series Phone (H.323)	6.8.0
Avaya 9608 Series Phone (H.323)	6.8.0
Avaya Communicator for Equinox (SIP)	3.5.5.113.24
Avaya 1140e (SIP)	FW: 04.04.30.00.bin
Avaya 98390 Analogue Phone	N/A
<b>Telenor</b>	
IPT Version	7.4.227

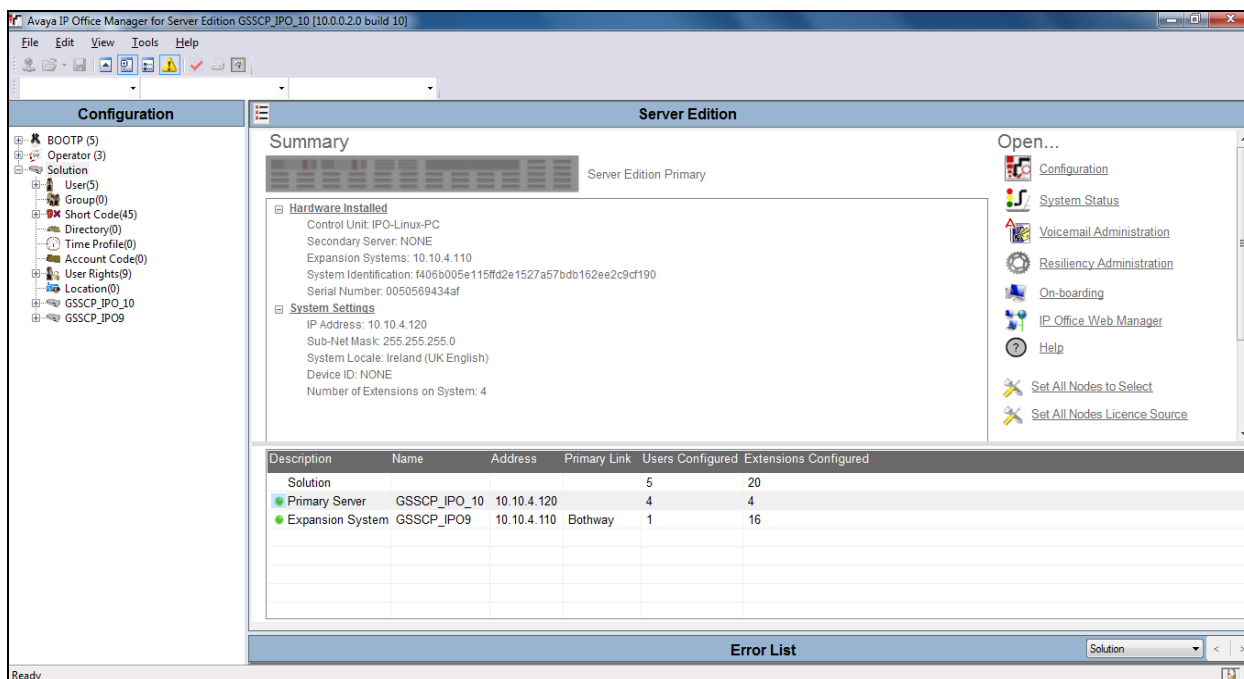
**Note** – Testing was performed with IP Office Server Edition with 500 V2 Expansion R11.0. Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition. **Note:** that IP Office Server 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 the Telenor IPT Multi-User 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 Telenor.

Feature	Instances	Status	Expiry Date	Source
Receptionist	10	Valid	Never	PLDS Nodal
Additional Voicemail Pro Ports	252	Valid	Never	PLDS Nodal
VMPro Recordings Administrators	1	Valid	Never	PLDS Nodal
Office Worker	1000	Valid	Never	PLDS Nodal
VMPro TTS Professional	40	Valid	Never	PLDS Nodal
IPSec Tunnelling	1	Obsolete	Never	PLDS Nodal
Power User	1000	Valid	Never	PLDS Nodal
Customer Service Agent	100	Dormant	Never	PLDS Nodal
Customer Service Supervisor	100	Dormant	Never	PLDS Nodal
Avaya IP endpoints	1000	Valid	Never	PLDS Nodal
<b>SIP Trunk Channels</b>	<b>256</b>	<b>Valid</b>	<b>Never</b>	<b>PLDS Nodal</b>
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	1000	Valid	Never	PLDS Nodal
Server Edition	150	Valid	Never	PLDS Nodal
UMS Web Services	1000	Valid	Never	PLDS Nodal
Avaya Mac Softphone	1000	Valid	Never	PLDS Nodal



## 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).

The screenshot shows the configuration window for 'GSSCP\_IPO\_SE\*'. The 'LAN2' tab is selected, and within it, the 'LAN Settings' sub-tab is active. The 'IP Address' field is set to '192 . 168 . 122 . 47' and the 'IP Mask' field is set to '255 . 255 . 255 . 0'. The 'Number Of DHCP IP Addresses' is set to '200'. Under 'DHCP Mode', the 'Disabled' radio button is selected. An 'Advanced' button is visible on the right side of the form.

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. 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).

The screenshot displays the 'GSSCP\_IPO\_SE\*' configuration window, which is divided into several tabs: System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VoIP, and Contact Center. The 'VoIP' tab is currently selected, and within it, the 'Network Topology' sub-tab is active.

**LAN Settings**

- ☒ H323 Gatekeeper Enable
  - ☐ Auto-create Extn ☐ Auto-create User ☐ H323 Remote Extn Enable
  - H.323 Signalling over TLS: Disabled (dropdown) Remote Call Signalling Port: 1720 (spin box)
- ☒ SIP Trunks Enable
- ☒ SIP Registrar Enable
  - ☐ Auto-create Extn/User ☐ SIP Remote Extn Enable Allowed SIP User Agents: Block blacklist only (dropdown)
  - SIP Domain Name: avaya.com
  - SIP Registrar FQDN: avaya.com
- Layer 4 Protocol:
  - ☒ UDP UDP Port: 5060 Remote UDP Port: 5060
  - ☒ TCP TCP Port: 5060 Remote TCP Port: 5060
  - ☐ TLS TLS Port: 5061 Remote TLS Port: 5061
- Challenge Expiry Time (secs): 10 (spin box)

**RTP**

- Port Number Range:
  - Minimum: 49152 Maximum: 53246
  - Port Number Range (NAT):
    - Minimum: 49152 Maximum: 53246
- ☒ Enable RTCP Monitoring on Port 5005
- RTCP collector IP address for phones: 0 . 0 . 0 . 0
- Keepalives:
  - Scope: RTP-RTCP (dropdown) Periodic timeout: 30 (spin box)
  - Initial keepalives: Enabled (dropdown)

**DiffServ Settings**

B8 (spin box) DSCP (Hex)	B8 (spin box) Video DSCP (Hex)	FC (spin box) DSCP Mask (Hex)	88 (spin box) SIG DSCP (Hex)
46 (spin box) DSCP	46 (spin box) Video DSCP	63 (spin box) DSCP Mask	34 (spin box) SIG DSCP

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 Error! Reference source not found.** Set **Binding Refresh Time (seconds)** to **300**. 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).

The screenshot shows the 'GSSCP\_IPO\_SE\*' configuration window with the 'Network Topology' tab selected. The 'Network Topology Discovery' section contains the following fields and controls:

- STUN Server Address:** An empty text input field.
- STUN Port:** A numeric input field set to 3478.
- Firewall/NAT Type:** A dropdown menu set to 'Open Internet'.
- Binding Refresh Time (seconds):** A numeric input field set to 300.
- Public IP Address:** A field showing '0 . 0 . 0 . 0'.
- Public Port:** A section with three sub-fields: UDP (0), TCP (0), and TLS (0).
- Run STUN on startup:** An unchecked checkbox.

At the bottom right of the configuration area are two buttons: 'Run STUN' and 'Cancel'.

### 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).

The screenshot displays the configuration interface for GSSCP\_IPO\_SE\*. The top navigation bar includes tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony (selected), Directory Services, System Events, SMTP, SMDR, VoIP, Contact Center, and Avaya Cloud Services. Below this, the Telephony section is active, showing sub-tabs for Park & Page, Tones & Music, Ring Tones, SM, Call Log, and TUI. The main configuration area is divided into two columns. The left column contains various telephony parameters: Dial Delay Time (secs) set to 1, Dial Delay Count set to 4, Default No Answer Time (secs) set to 15, Hold Timeout (secs) set to 0, Park Timeout (secs) set to 300, Ring Delay (secs) set to 5, Call Priority Promotion Time (secs) set to Disabled, Default Currency set to EUR, Default Name Priority set to Favour Trunk, Media Connection Preservation set to Enabled, and Phone Failback set to Automatic. Below these is a Login Code Complexity section with checkboxes for Enforcement and Complexity, and a Minimum length of 4. The right column features the Companding Law section, which has two sub-sections: Switch and Line. In the Switch section, the A-Law radio button is selected. In the Line section, the A-Law Line radio button is selected. Below these are several checkboxes: DSS Status (unchecked), Auto Hold (checked), Dial By Name (checked), Show Account Code (checked), Inhibit Off-Switch Forward/Transfer (unchecked), Restrict Network Interconnect (unchecked), Include location specific information (unchecked), Drop External Only Impromptu Conference (checked), and Visually Differentiate External Call (unchecked).

## 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.722 64K** is set as the priority codec and **G.711 ALAW 64K** set as the secondary codec as per screenshot below.

The screenshot shows the 'GSSCP\_IPO\_SE\*' configuration window with the 'VoIP' tab selected. The 'VoIP' sub-tab is also active, showing settings for 'VoIP Security' and 'Access Control Lists'. The 'Ignore DTMF Mismatch For Phones' checkbox is checked, and 'Allow Direct Media Within NAT Location' is unchecked. The 'RFC2833 Default Payload' is set to '101'. Below these are three panels: 'Available Codecs', 'Default Codec Selection', and 'Selected'. The 'Available Codecs' panel lists four codecs with checkboxes: G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, and G.729(a) 8K CS-ACELP. The 'Default Codec Selection' panel has an 'Unused' section with G.711 ULAW 64K and G.729(a) 8K CS-ACELP. The 'Selected' panel lists G.722 64K and G.711 ALAW 64K. Navigation buttons (right arrow, up arrow, left arrow, down arrow, and right arrow) are positioned between the panels.

GSSCP_IPO_SE*										
System	LAN1	LAN2	DNS	Voicemail	Telephony	Directory Services	System Events	SMTP	SMDR	VoIP
VoIP										
VoIP Security Access Control Lists										
Ignore DTMF Mismatch For Phones <input checked="" type="checkbox"/>										
Allow Direct Media Within NAT Location <input type="checkbox"/>										
RFC2833 Default Payload 101										
Available Codecs										
<input checked="" type="checkbox"/> G.711 ULAW 64K										
<input checked="" type="checkbox"/> G.711 ALAW 64K										
<input checked="" type="checkbox"/> G.722 64K										
<input checked="" type="checkbox"/> G.729(a) 8K CS-ACELP										
Default Codec Selection										
Unused										
G.711 ULAW 64K										
G.729(a) 8K CS-ACELP										
>>>										
↑										
<<<										
↓										
>>>										
Selected										
G.722 64K										
G.711 ALAW 64K										

## 5.5. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the Telenor IPT Multi-User 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** Error! Reference source not found..

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

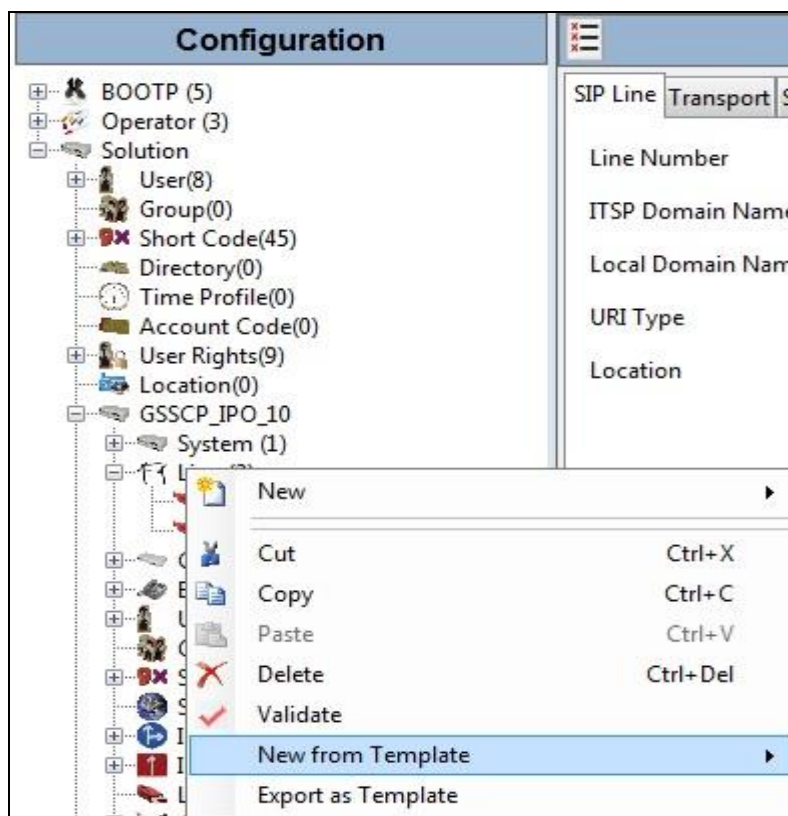
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

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** Error! Reference source not found..

### 5.5.1. SIP Line From Template

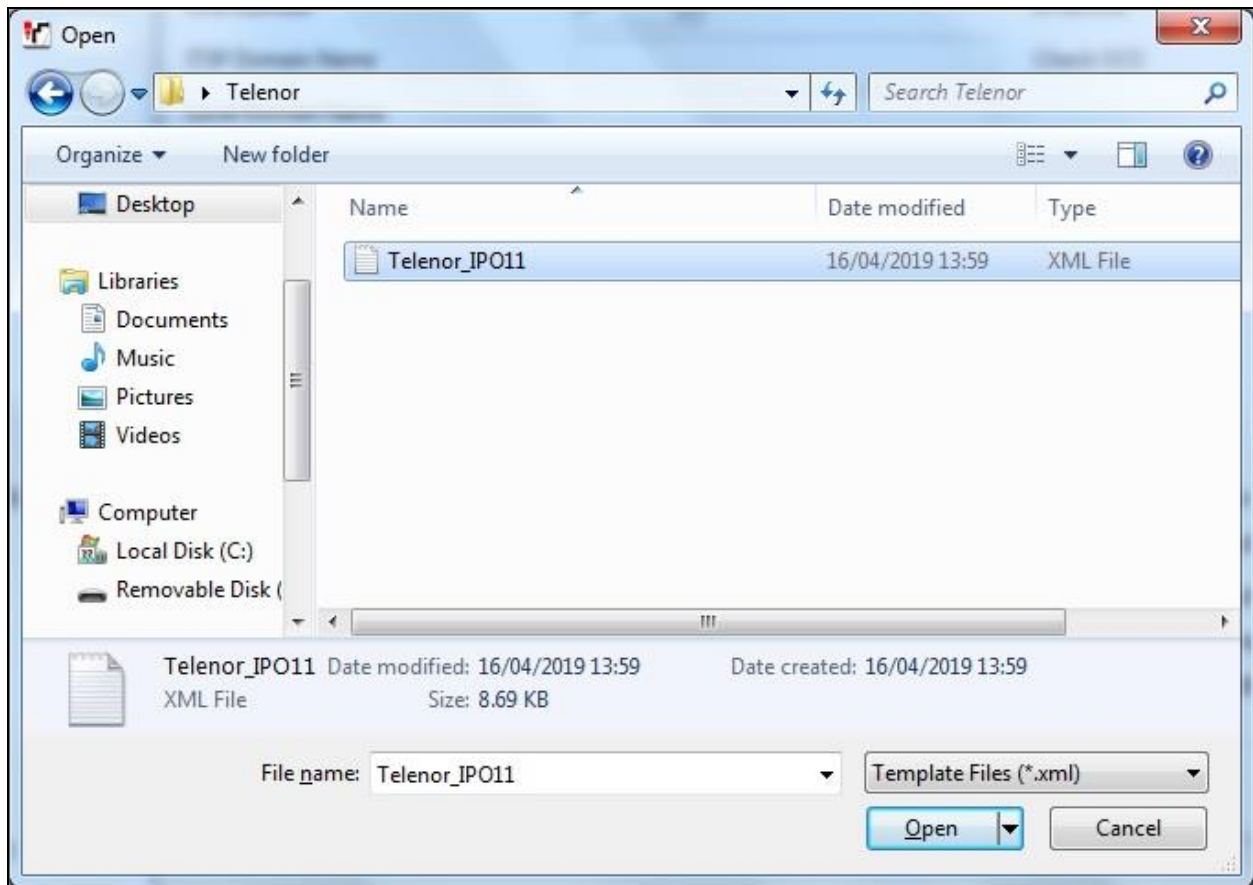
DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500 V2) and 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**.

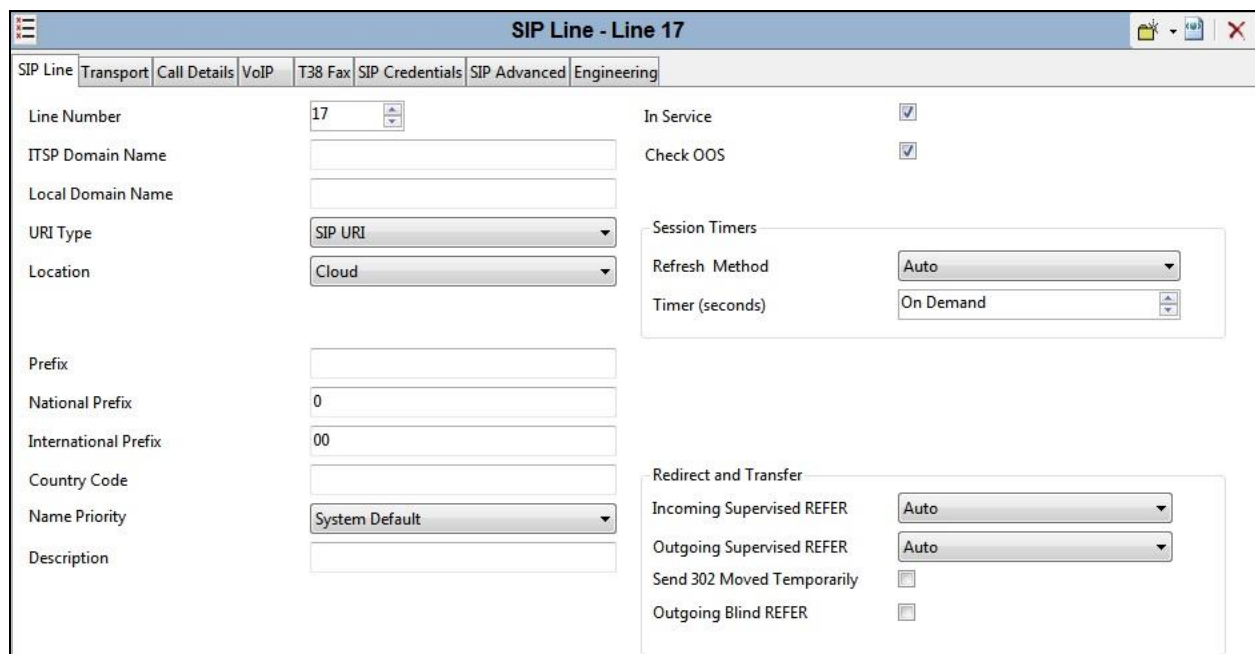


## 5.5.2. Manual SIP Line Configuration

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 **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.
- Ensure the **Check OSS** 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 **Auto**.
- Default values may be used for all other parameters.

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



SIP Line - Line 17	
<b>SIP Line</b>   Transport   Call Details   VoIP   T38 Fax   SIP Credentials   SIP Advanced   Engineering	
Line Number	17
ITSP Domain Name	
Local Domain Name	
URI Type	SIP URI
Location	Cloud
Prefix	
National Prefix	0
International Prefix	00
Country Code	
Name Priority	System Default
Description	
In Service	<input checked="" type="checkbox"/>
Check OSS	<input checked="" type="checkbox"/>
<b>Session Timers</b>	
Refresh Method	Auto
Timer (seconds)	On Demand
<b>Redirect and Transfer</b>	
Incoming Supervised REFER	Auto
Outgoing Supervised REFER	Auto
Send 302 Moved Temporarily	<input type="checkbox"/>
Outgoing Blind REFER	<input type="checkbox"/>

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

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

- Set **ITSP Proxy Address** to the IP address of the Telenor SIP proxy.
- 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).

The screenshot shows the 'SIP Line - Line 17\*' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '192.168.98.136'. Under 'Network Configuration', 'Layer 4 Protocol' is set to 'UDP', 'Use Network Topology Info' is set to 'None', 'Send Port' is '5060', and 'Listen Port' is '5060'. 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0' and '0 . 0 . 0 . 0'. 'Calls Route via Registrar' is checked. 'Separate Registrar' is empty.

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**.

The screenshot shows the 'SIP Line - Line 17\*' configuration window with the 'Call Details' tab selected. The 'SIP URIs' section is visible, showing a table with columns: URI, Groups, Credential, Local URI, Contact, P Asserted ID, P Preferred ID, Diversion Header, and Remote Party ID. To the right of the table are buttons for 'Add...', 'Remove', and 'Edit...'.

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 Telenor IPT Multi-User 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 Telenor 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. This ensures that the original called party number is sent for forwarded calls, though this is not currently working as described in **Section 2.2**.

SIP Line - 17 | Call Details | SIP URI

New URI

Incoming Group: 17 Max Sessions: 10

Outgoing Group: 17

Credentials: 0: <None>

	Display	Content	Field meaning		
			Outgoing Calls	Forwarding/Twinning	Incoming Calls
Local URI	Use Internal Data	Use Internal Data	Caller	Original Caller	Called
Contact	Use Internal Data	Use Internal Data	Caller	Original Caller	Called
P Asserted ID	<input checked="" type="checkbox"/> Use Internal Data	Use Internal Data	Caller	Original Caller	Called
P Preferred ID	<input type="checkbox"/> None	None	None	None	None
Diversion Header	<input checked="" type="checkbox"/> Use Internal Data	Use Internal Data	Caller	Original Caller	None
Remote Party ID	<input type="checkbox"/> None	None	None	None	None

OK Cancel Help

**Note:** If required a SIP URI can be created for calls to services such as the Mobile Twinning FNE: The numbers used for these services may not be associated with a User, so the incoming calls would not match the SIP settings in the User profile as described in **Section 5.7**. In order to match the incoming calls with a SIP URI, the Local URI can be set either to **Auto** which will match any number, or to the specific number used for the service. As this SIP URI would be used for incoming calls only, the **Outgoing Group** is set to an unused value, for example **100**. The following screenshot shows an example:

The following screenshot shows the completed configuration:

URI	Groups	Credential	Local URI	Contact	P Asserted ID	P Preferred ID	Diversion Header	Remote Party ID
1	17	0: <None>	Use Internal Data	Use Internal Data	Use Internal Data		Use Internal Data	
2	17 100	0: <None>	Auto	Auto	Auto			

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 Telenor.
- 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.

The screenshot shows the 'SIP Line - Line 17' configuration window with the 'VoIP' tab selected. The interface includes several tabs: 'SIP Line', 'Transport', 'Call Details', 'VoIP', 'SIP Credentials', 'SIP Advanced', and 'Engineering'. The 'VoIP' tab is active, displaying the following settings:

- Codec Selection:** A dropdown menu set to 'System Default'. Below it are two lists of codecs: 'Unused' (G.711 ULAW 64K, G.722 64K) and 'Selected' (G.711 ALAW 64K, G.729(a) 8K CS-ACELP). Navigation buttons (>>>>, <<<, <=>) are between the lists.
- Fax Transport Support:** A dropdown menu set to 'G.711'.
- DTMF Support:** A dropdown menu set to 'RFC2833/RFC4733'.
- Media Security:** A dropdown menu set to 'Same as System (Disabled)'.
- Checkboxes on the right:**
  - ☒ Local Hold Music
  - ☒ Re-invite Supported
  - ☐ Codec Lockdown
  - ☐ Allow Direct Media Path
  - ☐ Force direct media with phones
  - ☒ PRACK/100rel Supported

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.

The screenshot shows the 'SIP Line - Line 17\*' configuration window with the 'SIP Advanced' tab selected. The window is divided into several sections:

- Addressing:** Association Method is set to 'By Source IP address' and Call Routing Method is set to 'Request URI'. Suppress DNS SRV Lookups is unchecked.
- Identity:** A list of checkboxes includes 'Use "phone-context"' (unchecked), 'Add user=phone' (checked), 'Use + for International' (checked), 'Use PAI for Privacy' (unchecked), 'Use Domain for PAI' (unchecked), 'Caller ID from From header' (unchecked), 'Send From In Clear' (unchecked), 'Cache Auth Credentials' (checked), 'User-Agent and Server Headers' (empty text box), 'Send Location Info' (set to 'Emergency Calls'), 'Add UUI header' (unchecked), and 'Add UUI header to redirected calls' (unchecked).
- Media:** Includes checkboxes for 'Allow Empty INVITE' (unchecked), 'Send Empty re-INVITE' (unchecked), 'Allow To Tag Change' (unchecked), 'P-Early-Media Support' (set to 'None'), 'Send SilenceSupp=Off' (unchecked), 'Force Early Direct Media' (unchecked), 'Media Connection Preservation' (set to 'Disabled'), and 'Indicate HOLD' (unchecked).
- Call Control:** Includes 'Call Initiation Timeout (s)' (4), 'Call Queuing Timeout (m)' (5), 'Service Busy Response' (486 - Busy Here), 'on No User Responding Send' (408-Request Timeout), 'Action on CAC Location Limit' (Allow Voicemail), 'Suppress Q.850 Reason Header' (unchecked), and 'Emulate NOTIFY for REFER' (unchecked).

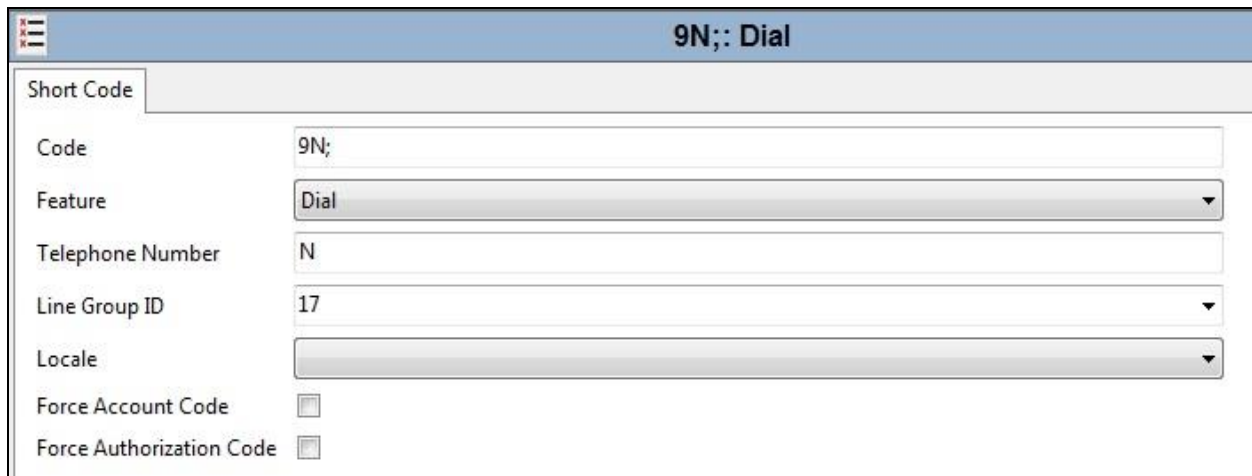
**Note:** It is advisable at this stage to save the configuration as described in **Section 5.10** to make the Line Group ID defined in **Section 5.5.2** available.

## 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 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).



The screenshot shows a configuration window titled "9N;; Dial". The window has a tab labeled "Short Code". Below the tab, there are several fields and checkboxes:

Field	Value
Code	9N;
Feature	Dial
Telephone Number	N
Line Group ID	17
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

## 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 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.

Extn89110: 89110									
Group Membership	Announcements	SIP	Personal Directory	Web Self-Administration					
User	Voicemail	DND	ShortCodes	Source Numbers	Telephony	Forwarding	Dial In	Voice Recording	Button Programming
Name	Extn89110								
Password	••••••••								
Confirm Password	••••••••								
Unique Identity									
Audio Conference PIN									
Confirm Audio Conference PIN									
Account Status	Enabled ▼								
Full Name	Extn89110								
Extension	89110								
Email Address									
Locale	▼								
Priority	5 ▼								
System Phone Rights	None ▼								
Profile	Power User ▼								
<input type="checkbox"/> Receptionist									

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 Telenor.

The screenshot shows the configuration page for 'Ext89110: 89110\*'. The 'SIP' tab is selected. The 'SIP Name' field is set to '+4722xxxx31', the 'SIP Display Name (Alias)' field is set to '+4722xxxx31', and the 'Contact' field is set to '+4722xxxx31'. There is an unchecked checkbox labeled 'Anonymous'.

**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.

The screenshot shows the 'Mobility' configuration page for 'Ext89110: 89110\*'. The 'SIP' tab is selected. The 'Twinned Handset' dropdown is set to '<None>'. The 'Maximum Number of Calls' is set to '1'. The 'Twinned Mobile Number (including dial access code)' is set to '0035389xxxxxxxx1'. The 'Twinned Time Profile' dropdown is set to '<None>'. The 'Mobile Dial Delay (secs)' is set to '3'. The 'Mobile Answer Guard (secs)' is set to '0'. The 'Hunt group calls eligible for mobile twinning' checkbox is unchecked. The 'Forwarded calls eligible for mobile twinning' checkbox is unchecked. The 'Twin When Logged Out' checkbox is unchecked. The 'one-X Mobile Client' checkbox is unchecked. The 'Mobile Call Control' checkbox is checked. The 'Mobile Callback' checkbox is checked.

## 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.

The screenshot shows a configuration window titled "17 +4722xxxxx31". It has three tabs: "Standard", "Voice Recording", and "Destinations". The "Standard" tab is active. The fields and their values are:

Field	Value
Bearer Capability	Any Voice
Line Group ID	17
Incoming Number	+4722xxxxx31
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

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 **+4722xxxxx31** on line 17 are routed to extension 89110.

The screenshot shows the same configuration window, but with the "Destinations" tab active. It displays a table with two columns: "TimeProfile" and "Destination".

TimeProfile	Destination
Default Value	89110 Extn89110

## 5.9. G.711 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 Telenor 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 G.711 fax settings are configured on the SIP line between the Expansion and the Server.

### 5.9.1. 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\_IPO9**. 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.

The screenshot displays the IP Office configuration interface. On the left is the 'Configuration' navigation pane, showing a tree structure with 'User(5)' selected. The main area is titled 'Analog89119: 89119' and contains several tabs: 'Announcements', 'SIP', 'Personal Directory', and 'Web Self-Administration'. The 'SIP' tab is active, showing fields for 'User', 'Voicemail', 'DND', 'ShortCodes', 'Source Numbers', 'Telephony', 'Forwarding', 'Dial In', 'Voice Recording', and 'Button Programming'. The 'User' field is set to 'Analog89119'. The 'Password' and 'Confirm Password' fields are filled with dots. The 'Unique Identity' field is empty. The 'Audio Conference PIN' and 'Confirm Audio Conference PIN' fields are empty. The 'Account Status' dropdown is set to 'Enabled'. The 'Full Name' field is empty. The 'Extension' field is set to '89119'. The 'Email Address' field is empty. The 'Locale' dropdown is set to '5'. The 'Priority' dropdown is set to '5'. The 'System Phone Rights' dropdown is set to 'None'. The 'Profile' dropdown is set to 'Basic User'. There is a checkbox for 'Receptionist' which is unchecked.

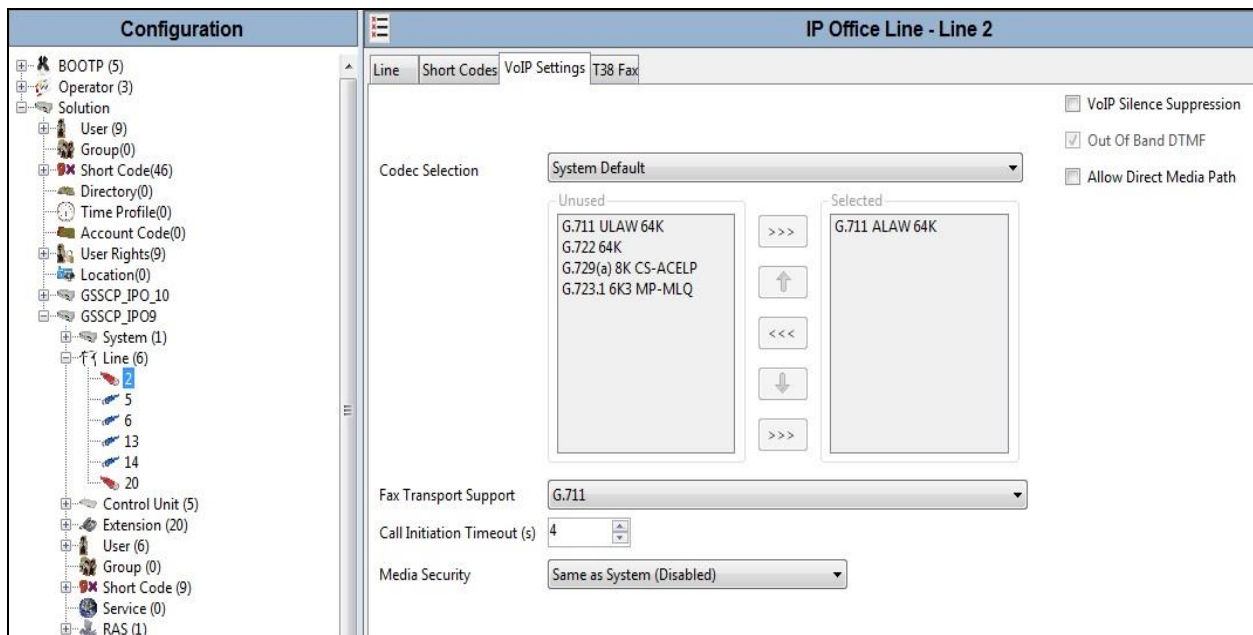
Configure other settings as described in **Section 5.7**.

## 5.9.2. G.711 Fax Settings

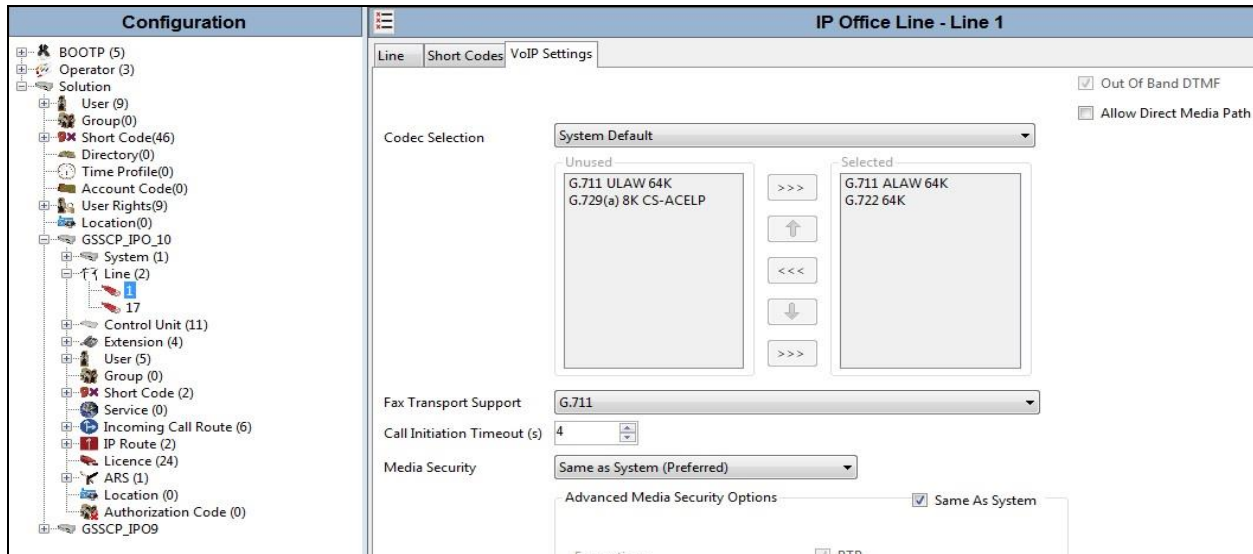
The G.711 Fax settings are defined on the SIP Line between the Expansion and the Server. Note that the VoIP settings for G.711 Fax are required in three places in this configuration:

- The SIP Line for the Telenor 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 **G.711**. The following screenshot shows the VoIP Settings for the IP Office Line between the Server and the Expansion on the Expansion:



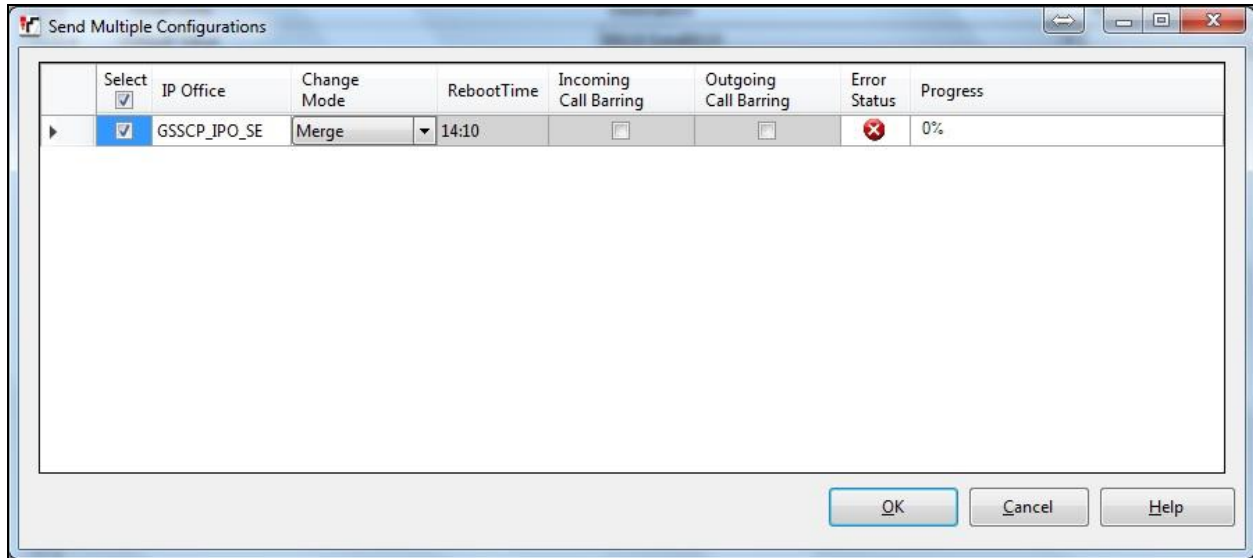
The following shows the **VoIP Settings** tab in the IP Office Line for the Expansion in the Server configuration:



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

## 5.10. 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. Telenor IPT Multi-User SIP Trunk Configuration

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

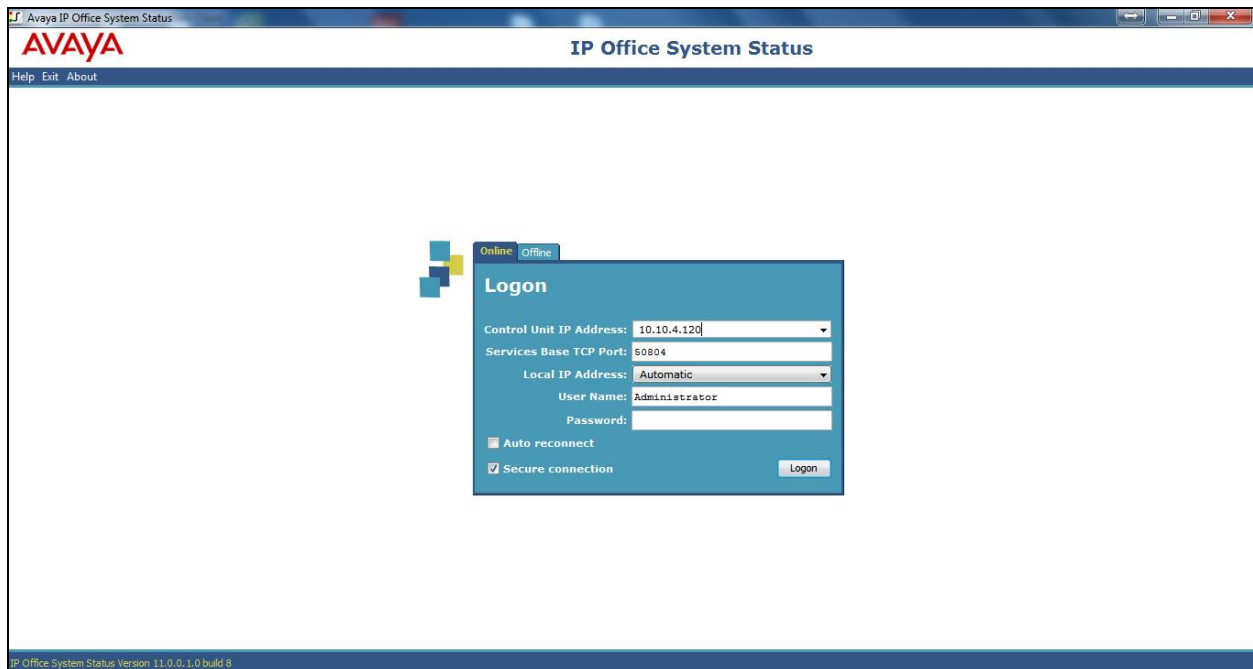
## 7. Verification Steps

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

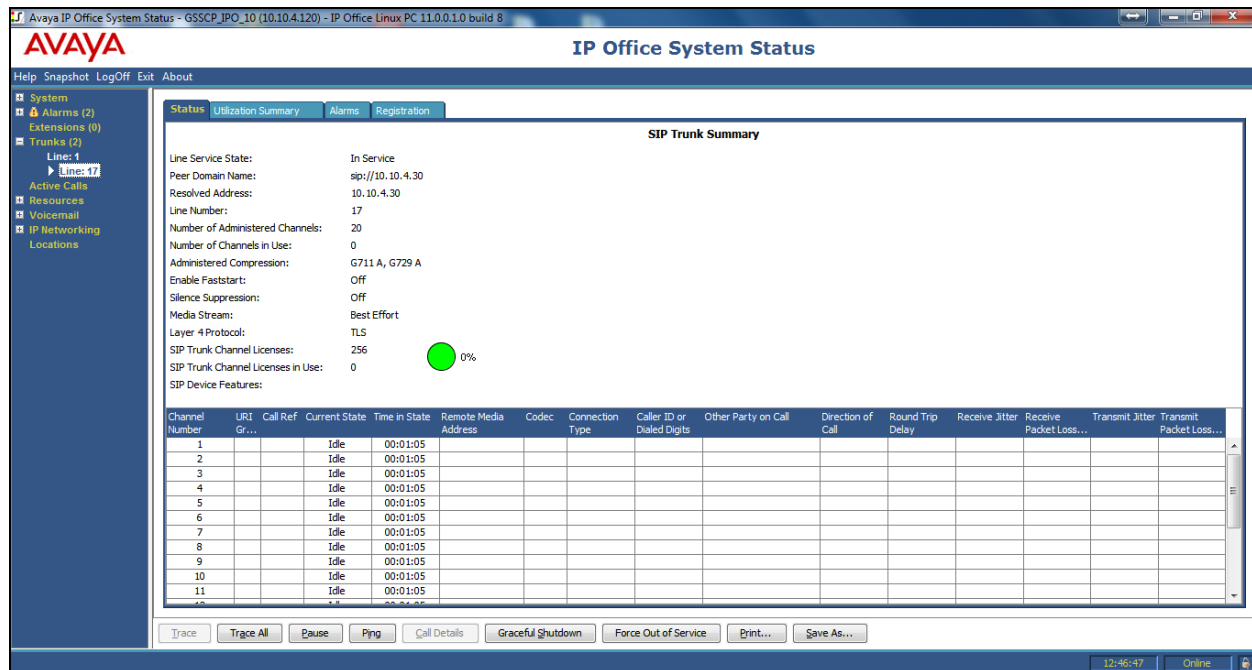
### 7.1. SIP Trunk status

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.

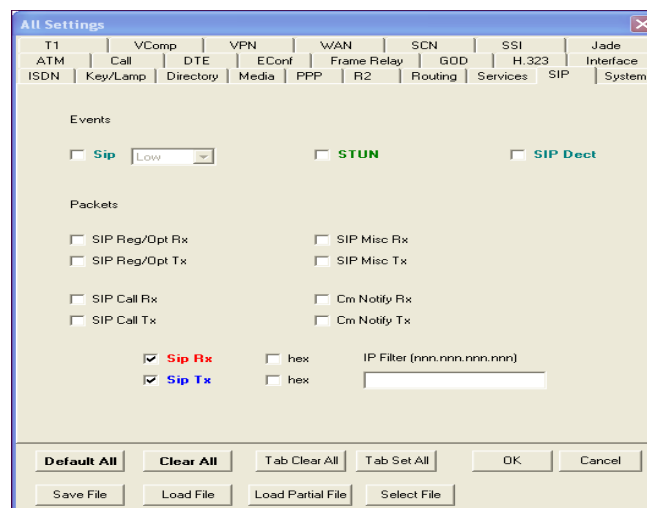


From the left-hand menu expand **Trunks** and choose the SIP trunk (**17** in this instance). The status window will show the status as being idle and time in state if the Trunk is operational.




## 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 and the Service Provider.



The screenshot shows the Avaya IP Office SysMonitor application window. The title bar reads "Avaya IP Office SysMonitor - [STOPPED] Monitoring 10.10.4.120 (GSSCP\_IPO\_10 (Server Edition(P)))". The menu bar includes File, Edit, View, Filters, Status, and Help. The main window displays a log of SIP messages. The messages are as follows:

```
***** SysMonitor v10.1.0.2.0 build 2 [connected to 10.10.4.120 (GSSCP_IPO_10 (Server Edition(P)))] *****
336128685S SIP Rx: TCP 10.10.4.30:43844 -> 10.10.4.120:5060
    OPTIONS sip:avaya.com SIP/2.0
    From: <sip:avaya.com>;tag=1c1904606935
    To: <sip:avaya.com>
    CSeq: 1 OPTIONS
    Call-ID: 07a0401e5c819c50fc33700dd0e04846
    Contact: <sip:10.10.4.30:5060;transport=tcp>
    Record-Route: <sip:10.10.4.30:5060;ipcs-line=2;lr;transport=tcp>
    Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE
    Supported: replaces
    User-Agent: MS00B/v.7.20A.158.056
    Max-Forwards: 69
    Via: SIP/2.0/TCP 10.10.4.30:5060;branch=z9hG4bK-s1632-000939282561-1--s1632-
    Accept: application/sdp, application/simple-message-summary, message/sipfrag
    Content-Length: 0

336128686S Sip: Association found trunk: SIP Line (17)
336128686S Sip: Update SipTCPUser->trunk SIP Line (17)
336128686S Sip: SIPDialog f6e2cdd0 created, dialogs 1 txn_keys 1
336128686S Sip: (f6e2cdd0) SetUnintTransactionCondition to UnInt_None
336128686S Sip: SipTCPUser 8430 has 1 dialog open (AttachDialogToSipTCPUser)
336128686S Sip: SIPDialog::ExtractResponseParamsFromViaHeader remote sent_by: 10.10.4.30:5060 trunk
336128686S Sip: SIPDialog::ExtractResponseParamsFromViaHeader remote sent by transport: SIP/2.0/TCP trunk
336128686S Sip: (f6e2cdd0) SendSIPResponse: OPTIONS code 200 SENT TO 10.10.4.30 43844
336128686S SIP Tx: TCP 10.10.4.120:5060 -> 10.10.4.30:43844
    SIP/2.0 200 OK
    Via: SIP/2.0/TCP 10.10.4.30:5060;branch=z9hG4bK-s1632-000939282561-1--s1632-
    Record-Route: <sip:10.10.4.30:5060;ipcs-line=2;lr;transport=tcp>
    From: <sip:avaya.com>;tag=1c1904606935
    Call-ID: 07a0401e5c819c50fc33700dd0e04846
    CSeq: 1 OPTIONS
    Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,NOTIFY,UPDATE
    Supported: timer
    Server: IP Office 10.1.0.2.0 build 2
    To: <sip:avaya.com>;tag=895dd2b8d0f38743
    Content-Type: application/sdp
    Content-Length: 169

v=0
o=UserA 1712183164 1334060956 IN IP4 10.10.4.120
s=Session SDP
c=IN IP4 10.10.4.120
t=0 0
```

## 8. Conclusion

These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office and Telenor IPT Multi-User 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 can be configured to interoperate successfully with Telenor IPT Multi-User SIP Trunk service. Telenor IPT Multi-User 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] *Avaya IP Office™ Platform Start Here First*, Release 11.0, Apr 2019.
- [2] *Avaya IP Office™ Platform Server Edition Reference Configuration*, Release 11.0, Apr 2019.
- [3] *Deploying IP Office™ Platform Server Edition Solution*, Release 11.0, Apr 2019.
- [4] *IP Office™ Platform 11.0, Deploying IP Office Essential Edition*, Document number 15-601042, Apr 2019.
- [5] *IP Office™ Platform 11.0 Installing and Maintaining the Avaya IP Office™ Platform Application Server*, Document number 15-601011, Apr 2019.
- [6] *Administering Avaya IP Office™ Platform with Web Manager*, Release 11.0, May 2018.
- [7] *Administering Avaya IP Office™ Platform with Manager*, Release 11.0, May 2018.
- [8] *IP Office™ Platform 11.0 Using Avaya IP Office™ Platform System Status*, Document number 15-601758, Apr 2018.
- [9] *IP Office™ Platform 11.0 Using IP Office System Monitor*, Document number 15-601019, May 2018.
- [10] *Using Avaya Equinox for Windows on IP Office*, Release 11.0, Mar 2019.
- [11] *IP Office™ Platform 11.0 - Third-Party SIP Extension Installation Notes*, Apr 2019.
- [12] *Avaya IP Office Knowledgebase*, <http://marketingtools.avaya.com/knowledgebase>

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