



Application Notes for Configuring Avaya IP Office Server Edition R11.1 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 R11.1.

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 at the Avaya Solution and Interoperability Test Lab.

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 R11.1.

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 R11.1 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, G.729 and G.722 codecs.
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using T.38, T38 Fallback and 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 testing it was observed that when an inbound call from a PSTN number terminates on an IP Office user that is call forwarded to another external PSTN number that is not in service or voicemail enabled, the external PSTN will respond to IP Office with a "183 Session Progress" that can contain an announcement "e.g. This number is not in service". However, IP Office responds to the "183 Session Progress" with "180 Ringing" so the PSTN caller that initially made the inbound call does not hear this announcement and will just hear continuous ringback until the Call Queueing Timers expire and busytone is then heard. However, in this particular call scenario, IP Office is behaving as designed as IP Office does not support playing announcements from non-primary targets (forwarding, twinning etc.) as the call is still anchored on IP Office.
- 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 for fax, which consist of IP Office sending a re-INVITE

message with two media lines in the SDP, with the first media line set for audio, with the port set to 0, and the second media line set for T.38, with a valid port number, thus deactivating audio transmission for the call. A SIP Line Custom String (SLIC) was added to the IP Office configuration used during the testing, as shown in **Section 5.5.2**. With the configuration shown in **Section 5.5.2**, IP Office will reverse the order of media line entries in the SDP, so that the active T.38 media line entry appears first, followed by the inactive audio media line entry, with the port set to 0. With the addition of the SIP Line Custom String (SLIC), Swisscom responded successfully to the re-INVITE message sent by Avaya IP Office with "200 OK" and the T.38 fax calls worked properly in both directions.

- The Privacy Header as required by Swisscom 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). 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.

For security purposes, all Service Provider IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, all IP addresses have been changed to a private format and all phone numbers have been obscured beyond the city code.

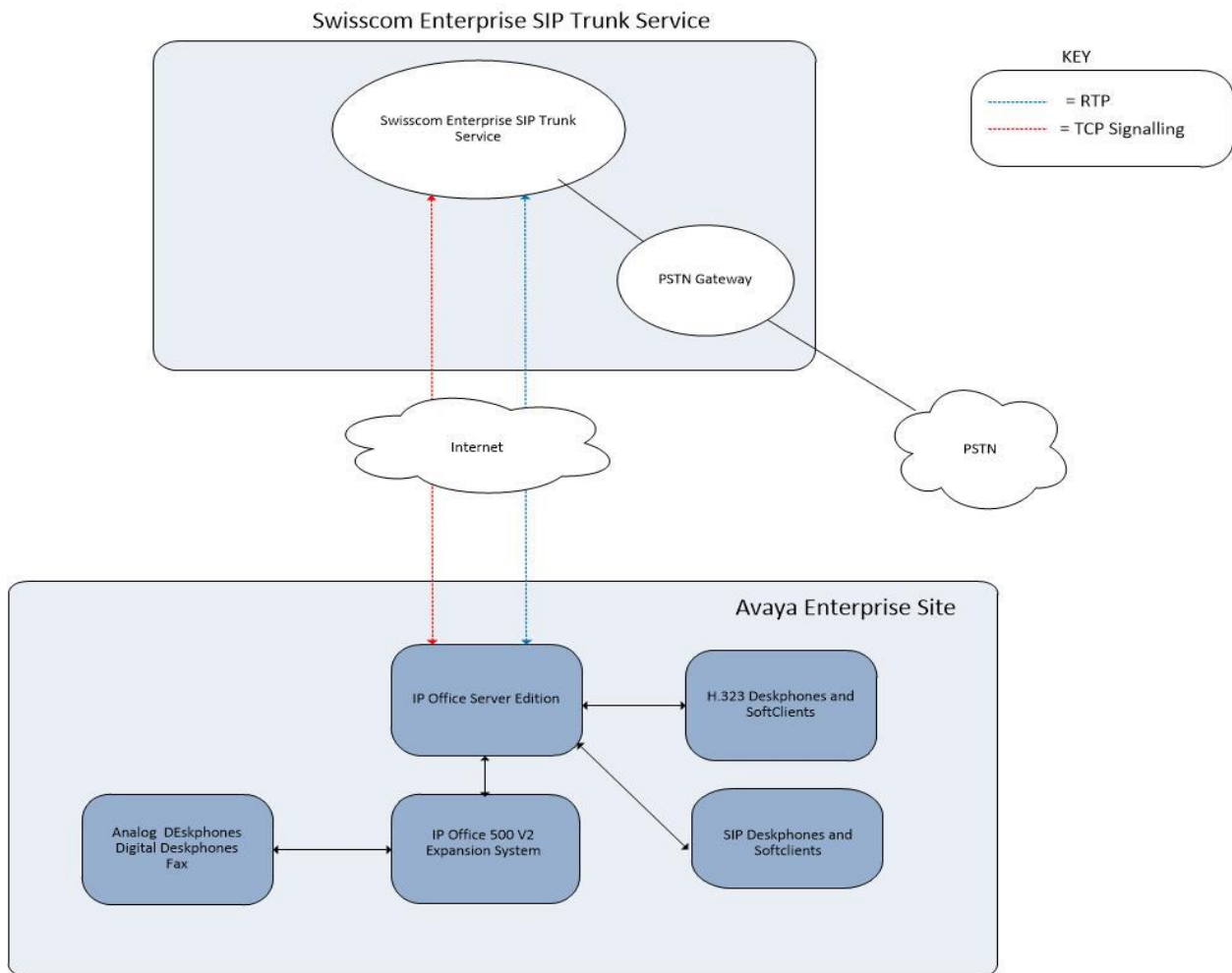


Figure 1: Swisscom Enterprise SIP Trunk to Avaya IP Office Topology

4. Equipment and Software Validated

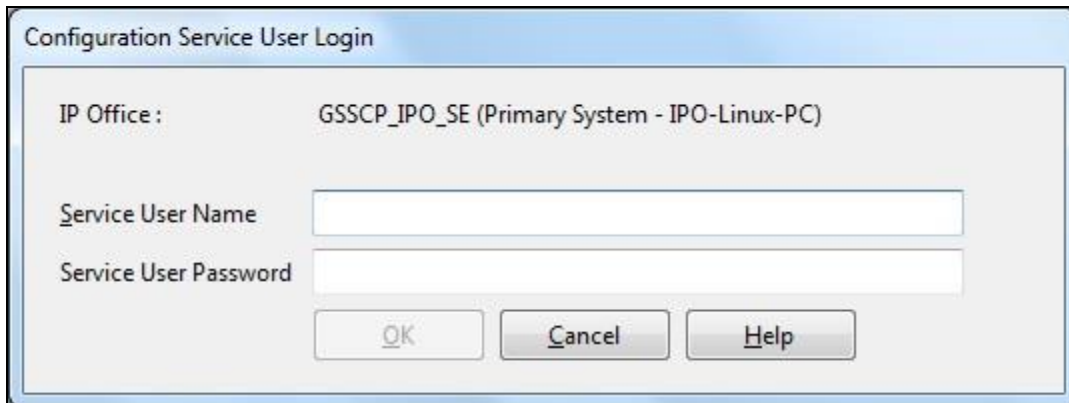
The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya	
Avaya IP Office Server Edition	Version 11.1.1.0.0 build 209
Avaya IP Office 500 V2	Version 11.1.1.0.0 build 209
Avaya Voicemail Pro Client	Version 11.1.1000.152
Avaya IP Office Manager	Version 11.1.1.0.0 build 209
Avaya 1608 Phone (H.323)	1.3.12
Avaya 9611G Series Phone (H.323)	6.8.0
Avaya J179 Series Phone (SIP)	4.0.4.0.10
Avaya Workplace Client for Windows (SIP)	3.17.0.65.16
Avaya 1140e (SIP)	FW: 04.04.23.00.bin
Avaya 1408 Digital Telephone	R48
Avaya Analogue Phone	N/A
Swisscom	
eSBC	Cisco IOS XE Software, Version 17.02.01r
C-SBC	Acme Packet 6300 SCZ8.3.0 MR-1 Patch 8A (Build 366)
SESM	Genband MCP_20.0.3.0_2019-11-17-2346

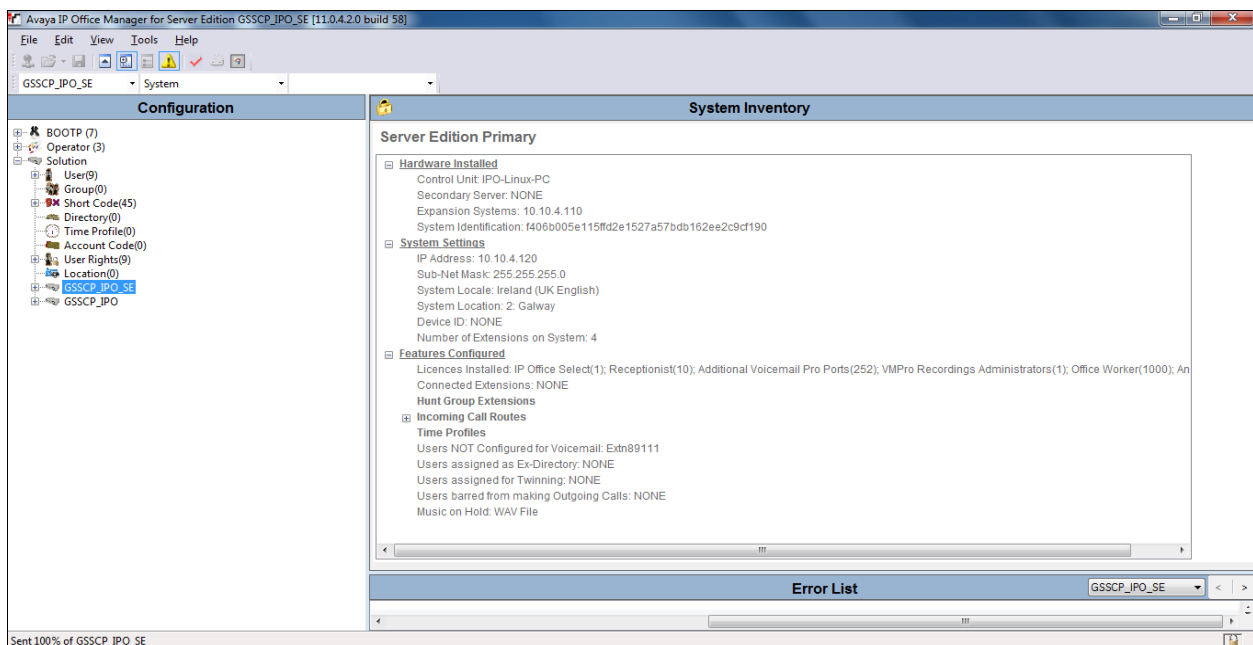
Note – Testing was performed with IP Office Server Edition with 500 V2 Expansion R11.1. Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition. **Note:** that IP Office Server Edition requires an Expansion IP Office 500 V2 to support 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 11.0

PLDS Host ID 213429294550

PLDS File Status Valid

Feature	Instances	Status	Expiry Date	Source
Receptionist	10	Valid	Never	PLDS Nodal
Additional Voicemail Pro Ports	152	Valid	Never	PLDS Nodal
VMPPro Recordings Administrators	10	Valid	Never	PLDS Nodal
Essential Edition Additional Voice...	10	Obsolete	Never	PLDS Nodal
VMPPro 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
VMPPro TTS (Scansoft)	40	Obsolete	Never	PLDS Nodal
VMPPro TTS Professional	40	Valid	Never	PLDS Nodal
IPSec Tunnelling	10	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	128	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

Add...

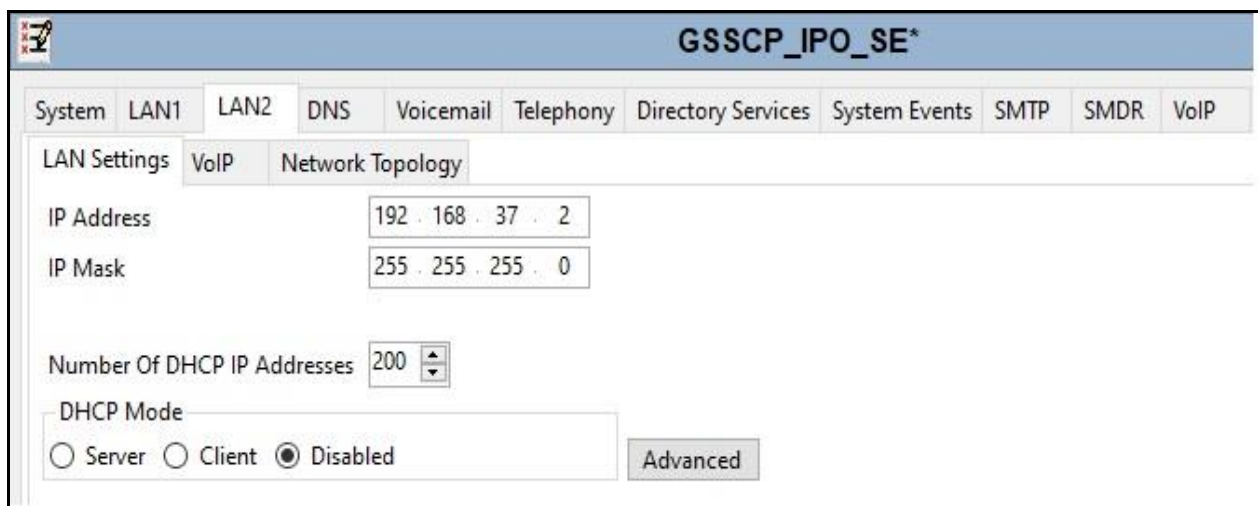
Remove

OK Cancel Help

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).



The screenshot shows the configuration interface for the LAN2 tab of the GSSCP_IPO_SE system. The interface includes a top navigation bar with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, and VoIP. Below this, there are sub-tabs for LAN Settings, VoIP, and Network Topology. The LAN Settings sub-tab is active, showing fields for IP Address (192.168.37.2), IP Mask (255.255.255.0), and Number Of DHCP IP Addresses (200). The DHCP Mode is set to Disabled, with radio buttons for Server, Client, and Disabled. An Advanced button is visible at the bottom right.

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).

The screenshot displays the 'GSSCP_IPO_SE' configuration window. The 'VoIP' tab is selected, and the 'Network Topology' sub-tab is active. The configuration is divided into two main sections: SIP and RTP.

SIP Configuration:

- ☒ 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** (text box)
- SIP Registrar FQDN: **avaya.com** (text box)
- Layer 4 Protocol:
 - ☒ UDP: UDP Port **5060** (spin box), Remote UDP Port **5060** (spin box)
 - ☒ TCP: TCP Port **5060** (spin box), Remote TCP Port **5060** (spin box)
 - ☒ TLS: TLS Port **5061** (spin box), Remote TLS Port **5061** (spin box)
- Challenge Expiry Time (secs): **10** (spin box)

RTP Configuration:

- Port Number Range:
 - Minimum: **40750** (spin box), Maximum: **50750** (spin box)
- Port Number Range (NAT):
 - Minimum: **40750** (spin box), Maximum: **50750** (spin box)
- ☒ Enable RTCP Monitoring on Port 5005
- RTCP collector IP address for phones: **0 . 0 . 0 . 0** (text box)
- Keepalives:
 - Scope: **RTP-RTCP** (dropdown)
 - Periodic timeout: **30** (spin box)
 - Initial keepalives: **Enabled** (dropdown)

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).

The screenshot shows the 'GSSCP_IPO_SE' configuration window with the 'Network Topology' tab selected. The 'Network Topology Discovery' section contains the following settings:

- STUN Server Address:** 0.0.0.0
- STUN Port:** 3478
- Firewall/NAT Type:** Open Internet (selected from a dropdown menu)
- Binding Refresh Time (seconds):** 0
- Public IP Address:** 0 . 0 . 0 . 0
- Public Port:**
 - UDP: 5060
 - TCP: 5060
 - TLS: 5061
- ☐ Run STUN on startup

Buttons for 'Run STUN' and 'Cancel' are located at the bottom right of the configuration area.

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 'GSSCP_IPO_SE*' configuration window, specifically the 'Telephony' tab. The interface is organized into several sections:

- System Tab Bar:** Includes System, LAN1, LAN2, DNS, Voicemail, **Telephony**, Directory Services, System Events, SMTP, SMDR, VoIP, and Contact Center.
- Telephony Sub-Tabs:** Includes Telephony, Park & Page, Tones & Music, Ring Tones, SM, Call Log, and TUI.
- Left Column Settings:**
 - Dial Delay Time (secs): 1
 - Dial Delay Count: 4
 - Default No Answer Time (secs): 15
 - Hold Timeout (secs): 0
 - Park Timeout (secs): 300
 - Ring Delay (secs): 5
 - Call Priority Promotion Time (secs): Disabled
 - Default Currency: EUR
 - Default Name Priority: Favour Trunk
 - Media Connection Preservation: Enabled
 - Phone Failback: Automatic
- Login Code Complexity:**
 - ☒ Enforcement
 - Minimum length: 4
 - ☒ Complexity
- RTCP Collector Configuration:**
 - ☐ Send RTCP to an RTCP Collector
 - Server Address: 0 . 0 . 0 . 0
 - UDP Port Number: 5005
 - RTCP reporting interval (secs): 5
- Companding Law:**
 - Switch:** ☐ U-Law, ☒ A-Law
 - Line:** ☐ U-Law Line, ☒ A-Law Line
- Other Settings:**
 - ☐ DSS Status
 - ☒ Auto Hold
 - ☒ Dial By Name
 - ☒ Show Account Code
 - ☐ Inhibit Off-Switch Forward/Transfer
 - ☐ Restrict Network Interconnect
 - ☐ Include location specific information
 - ☒ Drop External Only Impromptu Conference
 - ☐ Visually Differentiate External Call
 - ☒ High Quality Conferencing
 - ☒ Directory Overrides Barring
 - ☐ Advertise Callee State To Internal Callers
 - ☐ Internal Ring on Transfer

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, **G.729(a) 8K CS-ACELP** set as the secondary codec and **G.722 64K** as the third codec selection.

The screenshot shows the 'GSSCP_IPO_SE' configuration window with the 'VoIP' tab selected. The 'VoIP' sub-tab is also active, showing options for 'VoIP Security' and 'Access Control Lists'. Under 'VoIP Security', there are checkboxes for 'Ignore DTMF Mismatch For Phones' (checked) and 'Allow Direct Media Within NAT Location' (unchecked). The 'RFC2833 Default Payload' is set to '101'. Below these are two main sections: 'Available Codecs' and 'Default Codec Selection'. The 'Available Codecs' list includes four items, all checked: 'G.711 ULAW 64K', 'G.711 ALAW 64K', 'G.722 64K', and 'G.729(a) 8K CS-AC'. The 'Default Codec Selection' section has two columns: 'Unused' and 'Selected'. The 'Unused' column contains 'G.711 ULAW 64K'. The 'Selected' column contains 'G.711 ALAW 64K', 'G.729(a) 8K CS-A', and 'G.722 64K'. Between the columns are five buttons: '>>>', an up arrow, '<<<', a down arrow, and '>>>'.

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			Default Codec Selection							
<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-AC			<div>Unused</div> <div>G.711 ULAW 64K</div> <div>>>> ↑ <<< ↓ >>></div> <div>Selected</div> <div>G.711 ALAW 64K G.729(a) 8K CS-A G.722 64K</div>							

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:

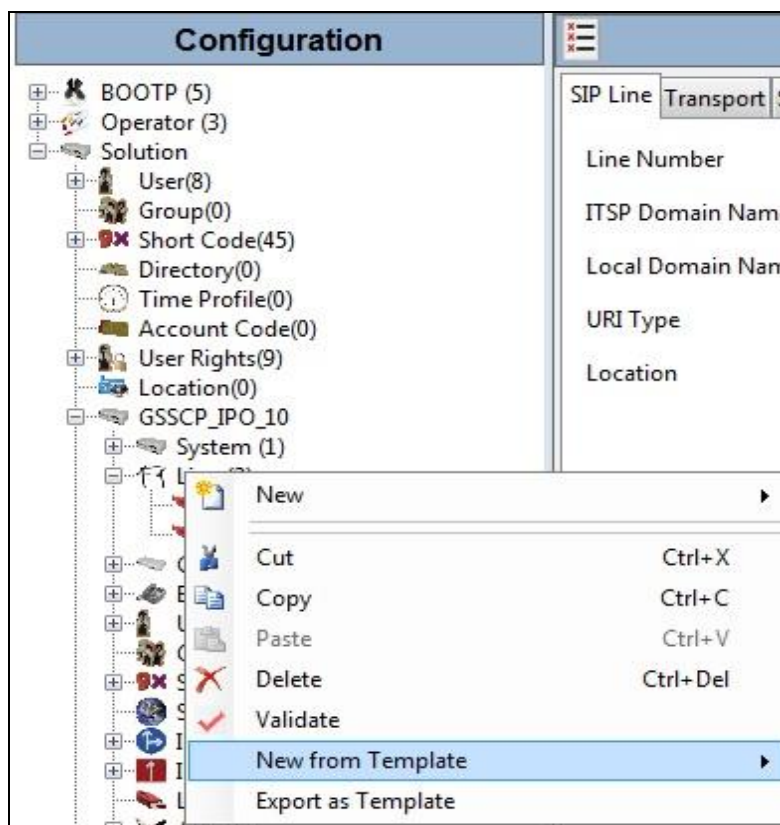
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

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**.

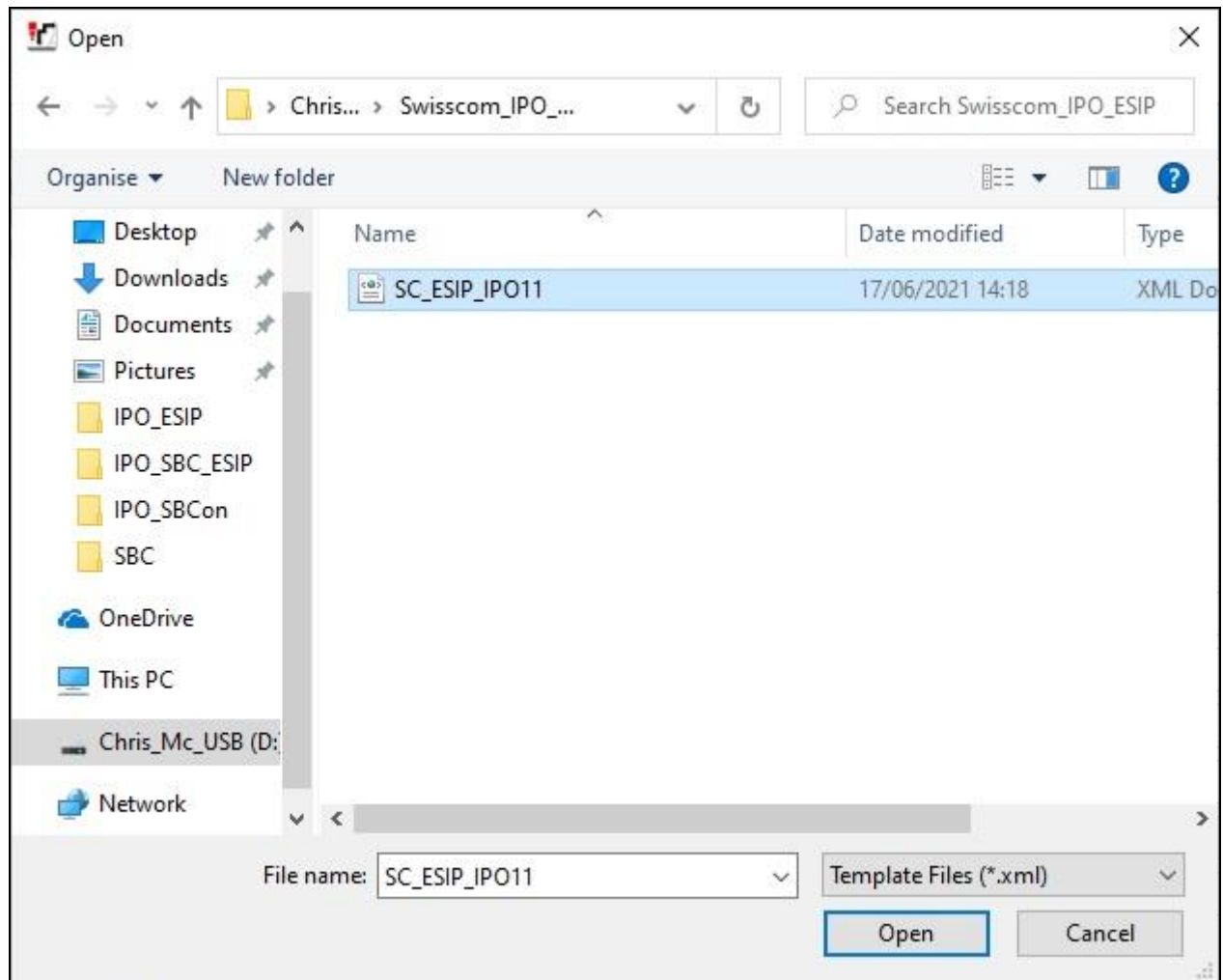
5.5.1. SIP Line From Template

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).



SIP Line - Line 17	
SIP Line Transport Call Details VoIP SIP Credentials SIP Advanced Engineering T38 Fax	
Line Number	17
ITSP Domain Name	
Local Domain Name	
URI Type	SIP URI
Location	2: Galway
Prefix	
National Prefix	0
International Prefix	00
Country Code	
Name Priority	System Default
Description	
In Service	<input checked="" type="checkbox"/>
Check OOS	<input type="checkbox"/>
Session Timers	
Refresh Method	Auto
Timer (seconds)	On Demand
Redirect and Transfer	
Incoming Supervised REFER	Never
Outgoing Supervised REFER	Never
Send 302 Moved Temporarily	<input type="checkbox"/>
Outgoing Blind REFER	<input type="checkbox"/>

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).

The screenshot shows the 'SIP Line - Line 17*' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '192.168.151.22'. Under 'Network Configuration', 'Layer 4 Protocol' is set to 'TCP', 'Send Port' is '5060', 'Use Network Topology Info' is set to 'None', and 'Listen Port' is '5060'. 'Explicit DNS Server(s)' are set to '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. There are 'Add...', 'Remove', and 'Edit...' buttons on the right.

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'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 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.
- 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.

SIP Line - 17 | Call Details | SIP URI

New URI

Incoming Group: 17 Max Sessions: 10

Outgoing Group: 17

Credentials: 1: +413xxxxxx80

	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

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	
2	17 2	0: <None>	Auto	Auto				

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 **T38** 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 **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.

The screenshot shows the 'SIP Line - Line 17*' configuration window with the 'VoIP' tab selected. The window has a title bar and a menu bar with options: SIP Line, Transport, Call Details, VoIP, SIP Credentials, SIP Advanced, and Engineering. The main content area is divided into several sections. On the left, under 'Codec Selection', there is a dropdown menu set to 'System Default'. Below this are two lists: 'Unused' containing 'G.711 ULAW 64K' and 'Selected' containing 'G.711 ALAW 64K', 'G.729(a) 8K CS-ACELP', and 'G.722 64K'. Between these lists are four buttons: '>>>', '<<<', '<<<', and '>>>'. Below the codec lists are three dropdown menus: 'Fax Transport Support' set to 'T38', 'DTMF Support' set to 'RFC2833/RFC4733', and 'Media Security' set to 'Disabled'. On the right side of the window, there are five checkboxes: 'Local Hold Music' (checked), 'Re-invite Supported' (checked), 'Codec Lockdown' (unchecked), 'Allow Direct Media Path' (unchecked), and 'PRACK/100rel Supported' (checked). Below the 'Allow Direct Media Path' checkbox is a sub-option 'Force direct media with phones' which is also unchecked.

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.
- Select **Emergency Calls** from the **Send Location Info** drop down menu if required
- Default values may be used for all other parameters.

SIP Line - Line 17

SIP Line | Transport | Call Details | VoIP | SIP Credentials | **SIP Advanced** | Engineering

Addressing

Association Method: By Source IP address

Call Routing Method: Request URI

Use P-Called-Party: ☐

Suppress DNS SRV Lookups: ☐

Identity

Use "phone-context": ☐

Add user=phone: ☒

Use + for International: ☒

Use PAI for Privacy: ☐

Use Domain for PAI: ☐

Caller ID from From header: ☐

Send From In Clear: ☐

Cache Auth Credentials: ☒

User-Agent and Server Headers:

Send Location Info: Emergency Calls

Add UUI header: ☐

Add UUI header to redirected calls: ☐

Media

Allow Empty INVITE: ☐

Send Empty re-INVITE: ☐

Allow To Tag Change: ☐

P-Early-Media Support: None

Send SilenceSup=Off: ☐

Force Early Direct Media: ☐

Media Connection Preservation: Disabled

Indicate HOLD: ☐

Call Control

Call Initiation Timeout (s): 4

Call Queuing Timeout (m): 5

Service Busy Response: 503 - Service Unavailable

on No User Responding Send: 408-Request Timeout

Suppress Q.850 Reason Header: ☐

Emulate NOTIFY for REFER: ☐

No REFER if using Diversion: ☐

Select the Engineering tab and add the SLIC custom string. Select **Add** button and add a custom string “**SLIC_PREFER_ACTIVE_SDP**” as shown in the screenshot. This custom string will help to overcome the T.38 fax issue as described in **Section 2.2**.

The screenshot shows the 'SIP Line - Line 17*' configuration window. The 'Engineering' tab is selected. In the 'Custom Strings' section, a list contains the string 'SLIC_PREFER_ACTIVE_SDP'. To the right of this list are three buttons: 'Add...', 'Remove', and 'Edit...'. At the bottom, a 'New Custom String' dialog is open, showing a text field with 'SLIC_PREFER_ACTIVE_SDP' and 'OK'/'Cancel' buttons.

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).

The screenshot shows a configuration window titled "9N;; Dial". It has a "Short Code" tab selected. The fields are as follows:

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"/>

A further example is shown for an emergency number.

The screenshot shows a configuration window titled "086756;; Dial Emergency". It has a "Short Code" tab selected. The fields are as follows:

Field	Value
Code	086756;
Feature	Dial Emergency
Telephone Number	086756
Line Group ID	100
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 5.5.2**. To configure these settings, first navigate to **User** in the Navigation Pane. Select the **User** tab if any changes are required.

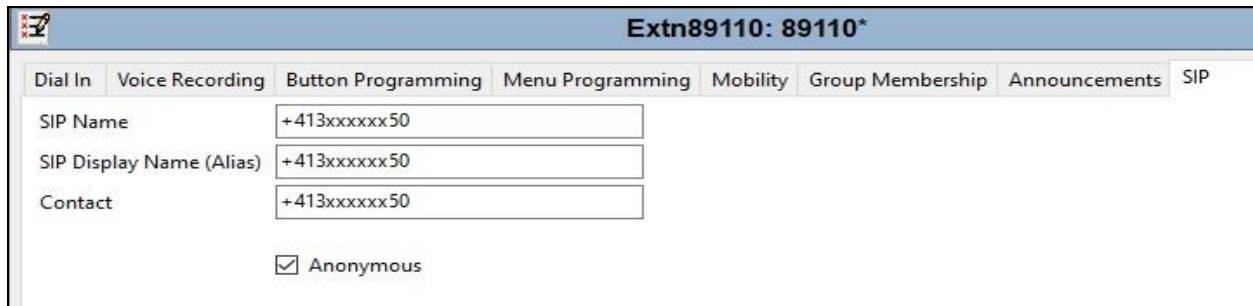
The following example shows the configuration required for a SIP Endpoint.

- 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.

Extn89110: 89110	
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	Basic User

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.



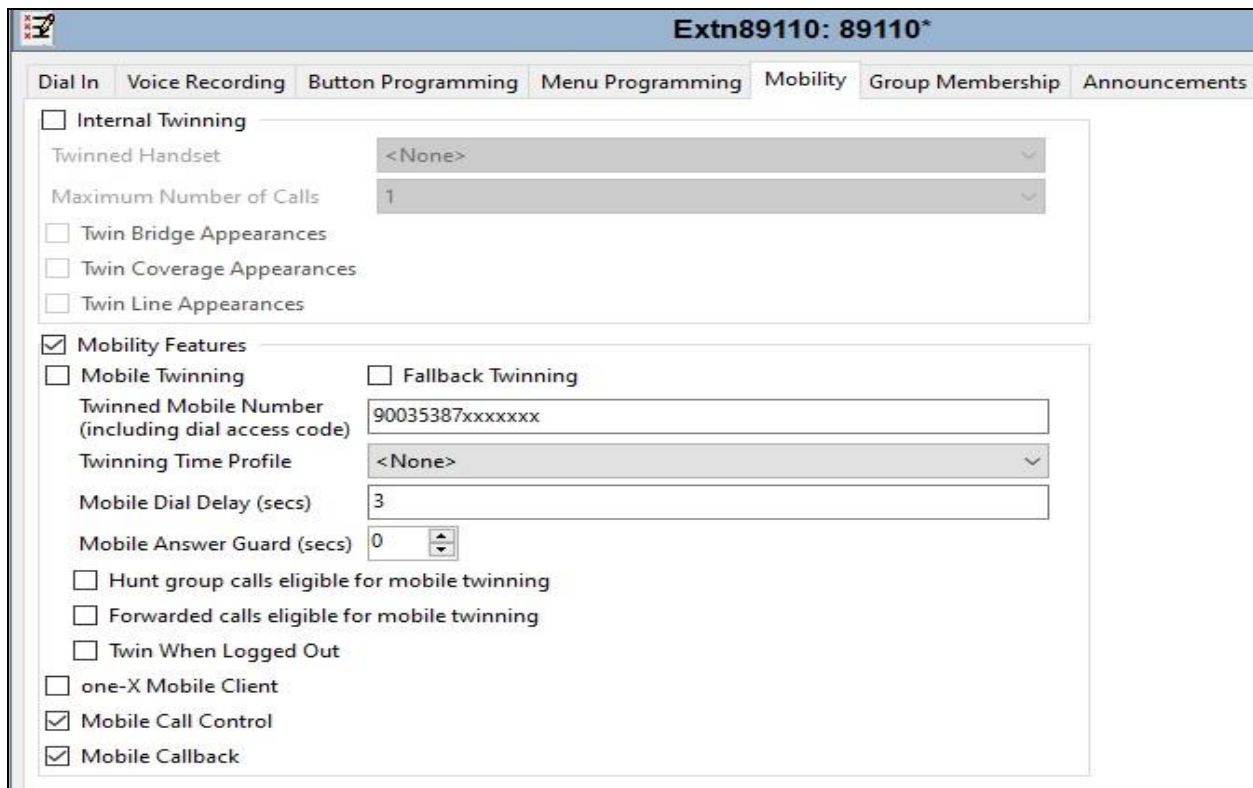
The screenshot shows the configuration page for 'Ext89110: 89110*'. The 'SIP' tab is selected. The fields are as follows:

Field	Value
SIP Name	+413xxxxxx50
SIP Display Name (Alias)	+413xxxxxx50
Contact	+413xxxxxx50

There is a checkbox labeled 'Anonymous' which is checked.

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.



The screenshot shows the configuration page for 'Ext89110: 89110*'. The 'Mobility' tab is selected. The configuration is as follows:

- ☐ Internal Twinning
 - Twinned Handset: <None>
 - Maximum Number of Calls: 1
 - ☐ Twin Bridge Appearances
 - ☐ Twin Coverage Appearances
 - ☐ Twin Line Appearances
- ☒ Mobility Features
 - ☐ Mobile Twinning
 - Twinned Mobile Number (including dial access code): 90035387xxxxxx
 - Twining Time Profile: <None>
 - Mobile Dial Delay (secs): 3
 - Mobile Answer Guard (secs): 0
 - ☐ Hunt group calls eligible for mobile twinning
 - ☐ Forwarded calls eligible for mobile twinning
 - ☐ Twin When Logged Out
 - ☐ Fallback Twinning
 - ☐ one-X Mobile Client
 - ☒ Mobile Call Control
 - ☒ Mobile Callback

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 +413xxxxxx50". It has three tabs: "Standard", "Voice Recording", and "Destinations". The "Standard" tab is active. It contains the following fields:

Bearer Capability	Any Voice
Line Group ID	17
Incoming Number	+413xxxxxx50
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 +**413xxxxxx50** 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. 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.

The screenshot displays the 'Galway' configuration window with the 'Location' tab selected. The 'Address' sub-tab is also visible. The form contains the following fields and values:

Field	Value
Location Name	Galway
Location ID	2
Subnet Address	0 . 0 . 0 . 0
Subnet Mask	0 . 0 . 0 . 0
Emergency ARS	<None>
Parent Location for CAC	<None>

Call Admission Control

Setting	Value
Total Maximum Calls	Unlimited
External Maximum Calls	Unlimited
Internal Maximum Calls	Unlimited

Time Settings

Setting	Value
Time Zone	Same as System
Local Time Offset from UTC	00:00
Automatic DST	<input type="checkbox"/>
Clock Forward/Back Settings (Start Date - End Date(DST Offset))	<Add New Entry>

Buttons: Edit, Delete

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

The screenshot displays the 'Galway' application window with the 'Address' tab selected. The form includes a 'Country Code' dropdown set to 'IE' and a warning message. Below this, there are two main sections of input fields. The left section contains fields labeled A1 through A6, RD, RDSEC, RDBR, RDSUBBR, PRD, POD, STS, PRM, and POM. The right section contains fields labeled HNO, HNS, LMK, BLD, LOC, PLC, FLR, UNIT, ROOM, SEAT, NAM, ADDCODE, PCN, PC, and POBOX. The 'UNIT' field is populated with 'GSSCP Unit' and the 'NAM' field is populated with 'GSSCP'.

Galway	
Location Address	
Country Code	IE
Please refer to the help for Information regarding this screen. Failure to format the address properly could result in improper address association.	
A1	Connacht
A2	Galway
A3	Galway
A4	Mervue
A5	Business Park
A6	Unit 25-29
RD	
RDSEC	
RDBR	
RDSUBBR	
PRD	
POD	
STS	
PRM	
POM	
HNO	
HNS	
LMK	
BLD	
LOC	
PLC	
FLR	
UNIT	GSSCP Unit
ROOM	
SEAT	
NAM	GSSCP
ADDCODE	
PCN	
PC	
POBOX	

5.10. Fax

At Release 11.1, 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 T38 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.

The screenshot displays the IP Office configuration interface. On the left, the 'Configuration' tree shows a hierarchy starting with 'Solution', followed by 'User (9)', 'Group(0)', 'Short Code(45)', 'Directory(0)', 'Time Profile(0)', 'Account Code(0)', 'User Rights(9)', 'Location(0)', 'GSSCP_IPO_SE', 'GSSCP_IPO', 'System (1)', 'Line (6)', 'Control Unit (5)', 'Extension (20)', and 'User (6)'. The 'User (6)' folder is expanded, showing 'NoUser', '89101 89101', '89102 89102', '89103 89103', '89119 Analog89119' (selected), and '89104 ChrisMc'. The right pane shows the configuration for 'Analog89119: 89119'. The 'User' tab is active, showing fields for Name, Password, Confirm Password, Unique Identity, Audio Conference PIN, Confirm Audio Conference PIN, Account Status (set to 'Enabled'), Full Name, Extension (89119), Email Address, Locale, Priority (5), System Phone Rights (None), and Profile (Basic User). There are also checkboxes for 'Receptionist' and 'Enable Softphone'.

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

5.10.2. T38 Fax Settings

The T38 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 Swisscom Enterprise 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 Expansion:

The screenshot shows the 'IP Office Line - Line 1*' configuration window with the 'VoIP Settings' tab selected. The window contains the following settings:

- Line**: Short Codes VoIP Settings
- Codec Selection**: System Default (dropdown)
- Unused**: G.711 ULAW 64K
- Selected**: G.711 ALAW 64K, G.729(a) 8K CS-ACELP, G.722 64K
- Out Of Band DTMF**: ☒
- Allow Direct Media Path**: ☐
- Fax Transport Support**: T38 (dropdown)
- Call Initiation Timeout (s)**: 4 (spinner)
- Media Security**: Same as System (Preferred) (dropdown)

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

The screenshot displays the 'IP Office Line - Line 2' configuration window, specifically the 'VoIP Settings' tab. The window has a title bar with a menu icon and a help icon. Below the title bar are four tabs: 'Line', 'Short Codes', 'VoIP Settings' (which is active), and 'T38 Fax'. The main content area is divided into several sections. On the left, there is a 'Codec Selection' section with a dropdown menu set to 'System Default'. Below this are two lists: 'Unused' and 'Selected'. The 'Unused' list contains 'G.711 ULAW 64K' and 'G.723.1 6K3 MP-MLQ'. The 'Selected' list contains 'G.711 ALAW 64K', 'G.729(a) 8K CS-ACELP', and 'G.722 64K'. Between these lists are five buttons: '>>>', an up arrow, '<<<', a down arrow, and '>>>'. To the right of the codec lists are three checkboxes: 'VoIP Silence Suppression' (unchecked), 'Out Of Band DTMF' (checked), and 'Allow Direct Media Path' (unchecked). Below the codec lists is a 'Fax Transport Support' dropdown menu set to 'T38'. Below that is a 'Call Initiation Timeout (s)' field with a value of '4' and a spinner. At the bottom is a 'Media Security' dropdown menu set to 'Same as System (Preferred)'.

IP Office Line - Line 2

Line Short Codes VoIP Settings T38 Fax

Codec Selection System Default

Unused

G.711 ULAW 64K
G.723.1 6K3 MP-MLQ

>>>
↑
<<<
↓
>>>

Selected

G.711 ALAW 64K
G.729(a) 8K CS-ACELP
G.722 64K

VoIP Silence Suppression
Out Of Band DTMF
Allow Direct Media Path

Fax Transport Support T38

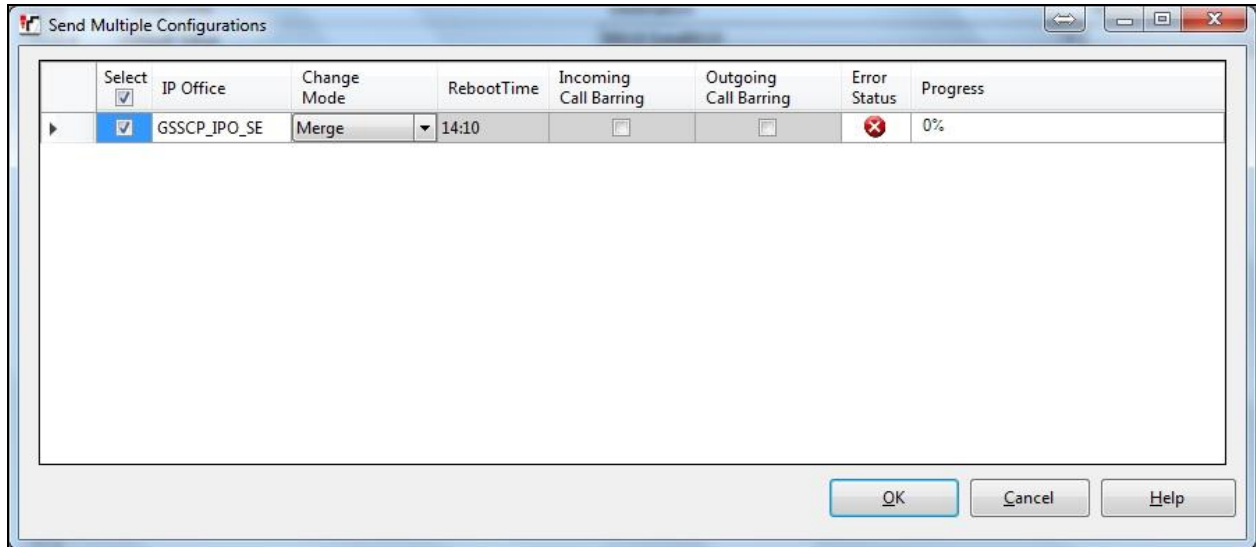
Call Initiation Timeout (s) 4

Media Security Same as System (Preferred)

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

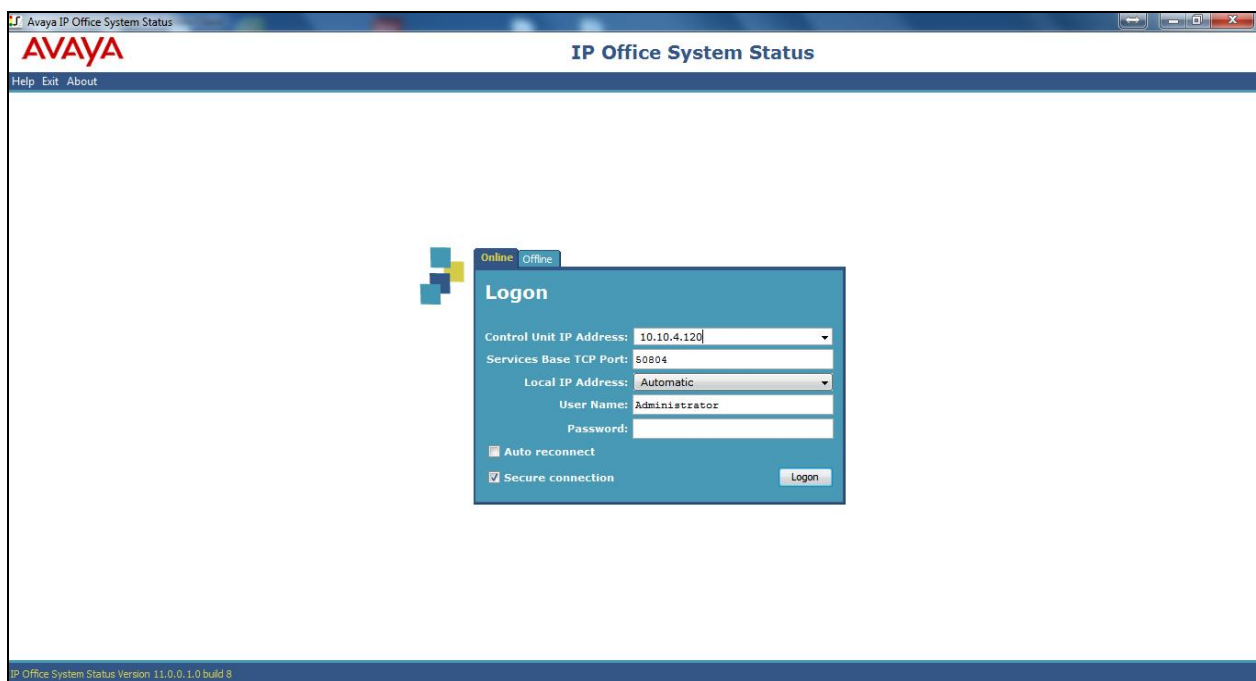
7. Verification Steps

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

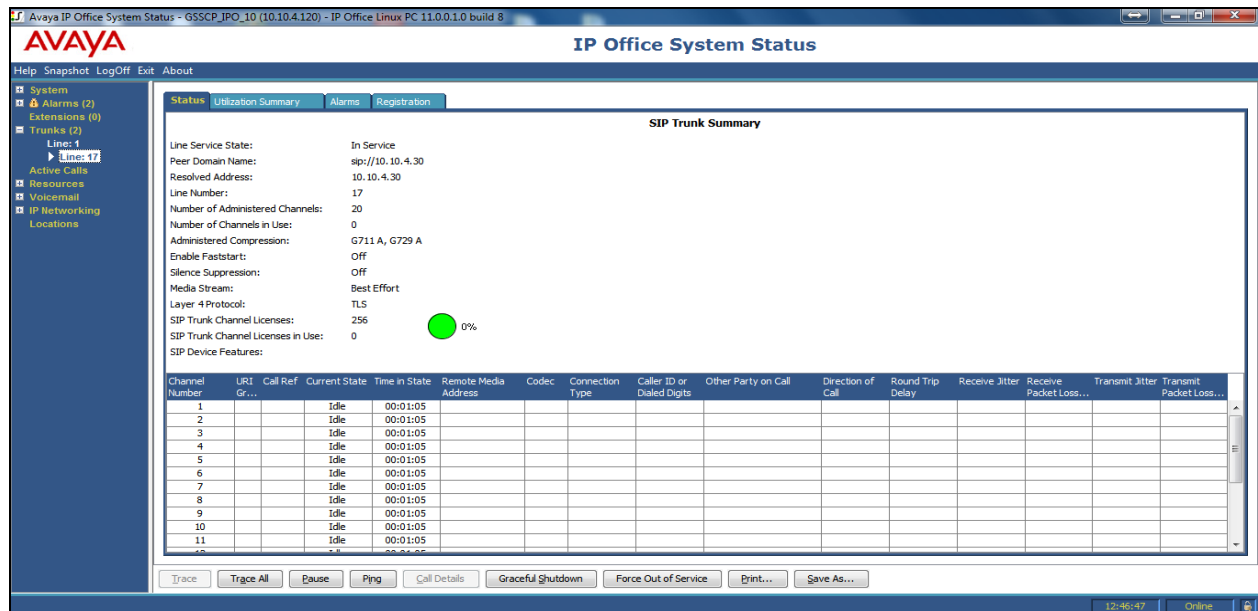
7.1. SIP Trunk Status

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

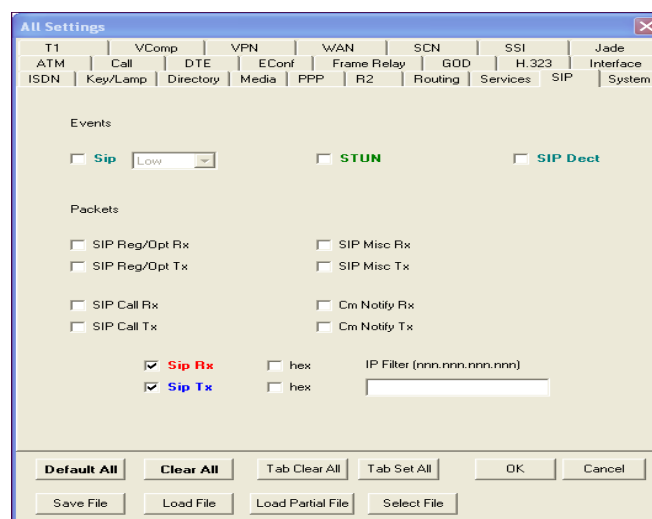


From the left-hand menu expand **Trunks** and choose the SIP trunk (**17** in this instance). The status window will show the status as being idle and time in state if the Trunk is operational.

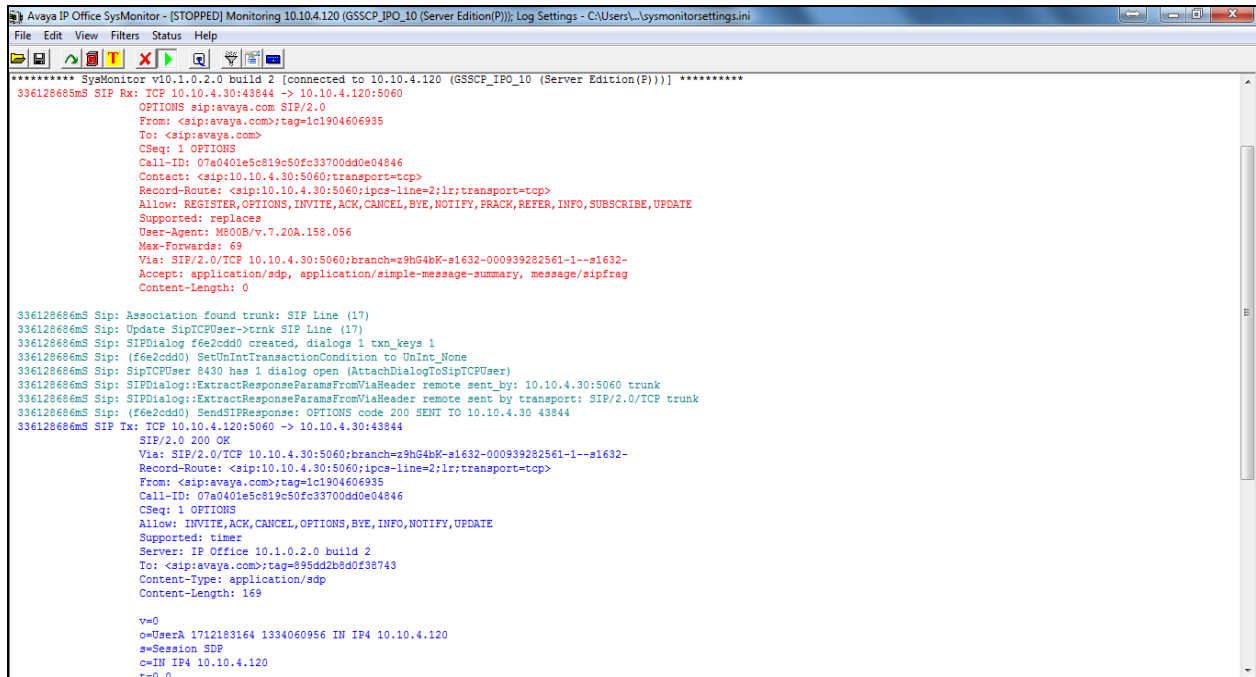


7.2. Monitor

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 a OPTIONS being sent between IP Office and the Service Provider.



```
Avaya IP Office SysMonitor - [STOPPED] Monitoring 10.10.4.120 (GSSCP_IPO_10 (Server Edition(P))); Log Settings - C:\Users\...\sysmonitorsettings.ini
File Edit View Filters Status Help

***** SysMonitor v10.1.0.2.0 build 2 [connected to 10.10.4.120 (GSSCP_IPO_10 (Server Edition(P)))] *****
336128686S 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: MG00B/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 Server Edition R11.1 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 R11.1 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] *Avaya IP Office™ Platform Start Here First*, Release 11.1, Mar 2021.
- [2] *Avaya IP Office™ Platform Server Edition Reference Configuration*, Release 11.1, Mar 2021.
- [3] *Deploying IP Office™ Platform Server Edition Solution*, Release 11.1, Mar 2021.
- [4] *IP Office™ Platform 11.1, Deploying IP Office Essential Edition*, Mar 2021.
- [5] *IP Office™ Platform 11.1 Installing and Maintaining the Avaya IP Office™ Platform Application Server*, Mar 2021.
- [6] *Administering Avaya IP Office™ Platform with Web Manager*, Release 11.1, Mar 2021.
- [7] *Administering Avaya IP Office™ Platform with Manager*, Release 11.1, Mar 2021.
- [8] *IP Office™ Platform 11.1 Using Avaya IP Office™ Platform System Status*, Mar 2021.
- [9] *IP Office™ Platform 11.1 Using IP Office System Monitor*, Mar 2021.
- [10] *Using Avaya Workplace Client for Windows on IP Office*, Feb 2021.
- [11] *IP Office™ Platform 11.1 - Third-Party SIP Extension Installation Notes*, Mar 2021.
- [12] *Avaya IP Office Knowledgebase*, <http://marketingtools.avaya.com/knowledgebase>

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