



## Avaya Solution & Interoperability Test Lab

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# Application Notes for Configuring Telia Norway SIP Trunking with Avaya IP Office Server Edition Release 11.1 - Issue 1.0

### Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider Telia Norway and Avaya IP Office Server Edition Release 11.1.

Telia Norway SIP Trunking Service provides PSTN access via a SIP trunk between the enterprise and the Telia Norway network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

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.

Telia Norway is a member of the Avaya DevConnect Service Provider program. 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 Telia Norway and Avaya IP Office Server Edition solution. In the sample configuration, the Avaya IP Office Server Edition solution consists of the Primary Server running the Avaya IP Office Server Edition Linux software Release 11.1, Avaya IP Office Server Edition Expansion System (IP500 V2), Avaya Voicemail Pro, WebRTC and one-X Portal services enabled, Avaya Communicator for Web, Avaya Workplace for Windows, Avaya H.323 and Avaya SIP Deskphones, digital and analog endpoints.

The Telia Norway service referenced within these Application Notes is designed for business customers. The service enables local and long distance PSTN calling via standards-based SIP trunks as an alternative to legacy analog or digital trunks, without the need for additional TDM enterprise gateways and the associated maintenance costs.

## 2. General Test Approach and Test Results

The general test approach was to simulate an enterprise site in the Solution & Interoperability Test Lab by connecting IP Office to Telia Norway's SIP Trunking service across the public internet. The configuration in **Figure 1** was used to exercise the features and functionality tests 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 this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya and Telia Norway 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:

- SIP OPTIONS queries and responses
- Incoming PSTN calls to various phone types. Phone types included Avaya H.323, Avaya 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. Phone types included Avaya H.323, Avaya 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
- Inbound and outbound PSTN calls from/to the Avaya Communicator for Web with basic telephony transfer feature
- Inbound and outbound PSTN calls from/to the Avaya Workplace for Windows (SIP)
- Inbound and outbound long hold time call stability
- Various call types including: local, international call, and Emergency 112 services
- SIP transport TLS/SRTP and Port 6004 between Telia Norway and the simulated Avaya enterprise site
- Codec G.711A, G.729A, G.711MU
- Caller number/ID presentation
- Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls
- DTMF transmission using RFC 2833
- Voicemail navigation for inbound and outbound calls
- Telephony features such as hold and resume, transfer, and conference
- G.711 pass-through fax
- Off-net call forwarding
- Twinning to mobile phones on inbound calls

Items not supported or not tested included the following:

- Telia Norway does not support Registration/Authentication
- Telia Norway does not support the outbound operator assisted call
- Telia Norway supports outbound/inbound toll-free service in production, however it is not available in their test lab during the compliance testing

## 2.2. Test Results

Interoperability testing of Telia Norway was completed with successful results for all test cases with the exception of the observation described below:

- The anonymous outbound call failed. Avaya IP Office's design for anonymous outbound calls, sends the SIP Invite including "Privacy: id" and the FROM header with the URI format: "From: "Anonymous" sip:anonymous@anonymous.invalid". Telia Norway rejected the call because Telia only accepted the SIP Invite including "Privacy:id" and the From header with USER.URI = user/number". In order to make this call work, Telia Norway did implement to check for "secret" in P-Asserted- Identity header instead of From header.

## 2.3. Support

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

For technical support on Telia Norway SIP Trunking, contact Telia Norway at <http://telia.no>.

### 3. Reference Configuration

**Figure 1** illustrates the test configuration. The test configuration shows an enterprise site connected to the Telia Norway network through the public internet. For confidentiality and privacy purposes, actual public IP addresses used in this testing have been masked out and replaced with fictitious IP addresses throughout the document.

The Avaya components used to create the simulated customer site includes:

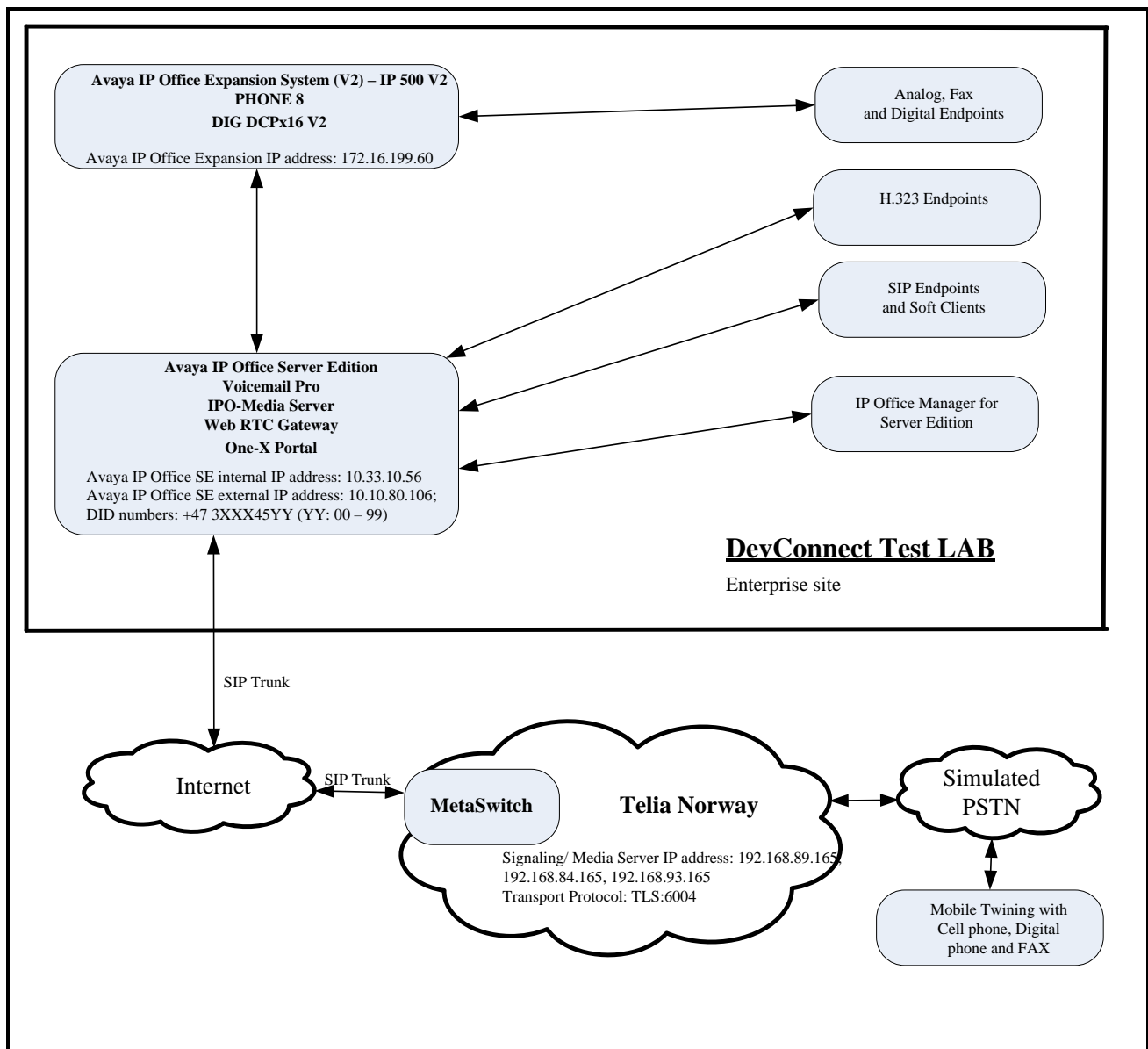
- IP Office Server Edition Primary Server
- IP Office Voicemail Pro
- IP Office Server Edition Expansion System (IP500 V2)
- WebRTC and one-X Portal services
- Avaya 96x1 Series IP Deskphones (H.323)
- Avaya 11x0 Series IP Deskphones (SIP)
- Avaya J129 IP Deskphones (SIP)
- Avaya 1408 Digital phones
- Avaya Analog phones
- Avaya Communicator for Web
- Avaya Workplace for Windows (SIP)

The Primary Server consists of a Dell PowerEdge R640 server, running the Avaya IP Office Server Edition Linux software Release 11.1. Avaya Voicemail Pro runs as a service on the Primary Server. The LAN1 port of the Primary Server (Eth0) is connected to the enterprise LAN (Private network) while the LAN2 port is connected to the public network. The SIP trunk to the Telia Norway system is connected to LAN2 port of the Avaya IP Office Server Edition.

The optional Expansion System (IP500 V2) is used for the support of digital, analog, fax, and additional IP stations. It consists of an Avaya IP Office IP500V2 with the MOD DGTL STA16 expansion module which provides connections for 16 digital stations, the PHONE 8 card which provides connections for 8 analog stations, as well as a 64-channel VCM (Voice Compression Module) for supporting VoIP codecs.

A separate Windows 10 Enterprise PC runs Avaya IP Office Server Edition Manager to configure and administer Avaya IP Office Server Edition system.

Mobility Twinning is configured for some of the Avaya IP Office Server Edition users so that calls to these user's phones will also ring and can be answered at configured mobile phones.



**Figure 1 - Test Configuration for Avaya IP Office Server Edition with Telia Norway SIP Trunk Service**

Inbound calls from the service provider via the SIP trunk arrive to the Server Edition Primary Server, where Incoming Call Routes are checked to determine the call destination. In the event that the destination of the incoming call is an endpoint in the Expansion System (IP500 V2), the call is sent via the Small Community Network (SCN) H.323 trunk (IP Office Line) to the expansion IP500V2 for routing to the final endpoint. This SCN H.323 trunk is automatically created during the initial process of addition of the Expansion System to the IP Office Server Edition solution.

Similarly, outbound calls from the enterprise to the PSTN are routed via the SIP trunk to the Telia Norway network. Calls originated from extensions registered to the Primary Server are routed

directly to Telia Norway, while calls originated from extensions on the Expansion System are sent to the Primary Server via SCN H.323 trunk, before being routed to Telia Norway via the SIP trunk.

For the purposes of the compliance test, Avaya IP Office Server Edition users dialed a short code of 6 + N digits to send digits across the SIP trunk to Telia Norway. The short code of 6 was stripped off by Avaya IP Office Server Edition but the remaining N digits were sent unaltered to Telia Norway. For the compliance test, outbound calls to Canadian numbers within the North American Numbering Plan (NANP) were tested. The user would dial 11 (1 + 10) digits in E.164 Format. For these NANP calls, Avaya IP Office would send 11 digits (E.164 Format) in the Request URI and the To field of an outbound SIP INVITE message, and it was configured to send 10 digits (E.164 Format) in the From field. For inbound calls, Telia Norway sent 10 digits (E.164 Format) in the Request URI and the To field of inbound SIP INVITE messages.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and Avaya IP Office Server Edition, such as a data firewall. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes. However, it should be noted that SIP and SRTP traffic between the service provider and Avaya IP Office Server Edition must be allowed to pass through these devices.

## 4. Equipment and Software Validated

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

Component	Version
<b>Avaya</b>	
Avaya IP Office Server Edition solution <ul style="list-style-type: none"> <li>▪ Primary Server Dell PowerEdge R640 – IPO-Linux-PC</li> <li>▪ IPO-Media Server</li> <li>▪ Voicemail Pro</li> <li>▪ Web RTC Gateway</li> <li>▪ one-X Portal</li> <li>▪ IP Office Manager for Server Edition</li> <li>▪ IP Office Expansion System (V2) – IP 500 V2</li> <li>▪ IP Office Analogue - PHONE 8</li> <li>▪ IP Office Digital - DIG DCPx16 V2</li> </ul>	11.1.2.3.0 build 47 11.1.2.3.0 build 47 11.1.2.3.0 build 3 11.1.2.3.0 build 2 11.1.2.3.0 build 6 11.1.2.3.0 build 47 11.1.2.3.0 build 47 11.1.2.3.0 build 47 11.1.2.3.0 build 47
Avaya 1140E IP Deskphone (SIP)	04.04.33
Avaya 9641G IP Deskphone (H323)	6.8.5.1
Avaya 9621G IP Deskphone (H323)	6.8.5.1
Avaya J129 IP Deskphone (SIP)	4.0.7.1.5
Avaya Communicator for Web	1.0.20.1722
Avaya Workplace Client for Windows	3.28.0.73
Avaya 1408D Digital Deskphone	R48
Avaya Analog Deskphone	N/A
VentaFax	7.10.258.664
<b>Telia Norway</b>	
Metaswitch Perimeta SBC	V4.9.30

**Table 1: Equipment and Software Tested**

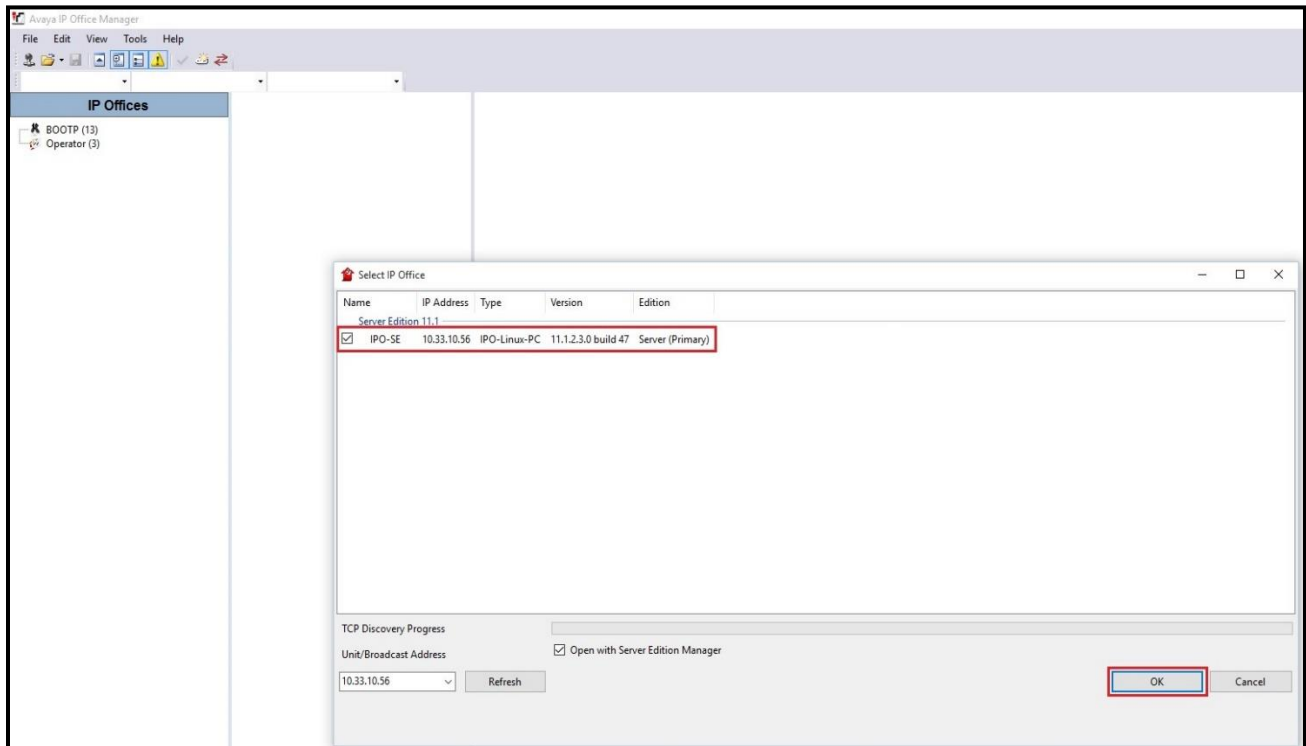
**Note** – Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2, and also when deployed with all configurations of IP Office Server Edition. IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.



## 5. Configure Avaya IP Office Server Edition Solution

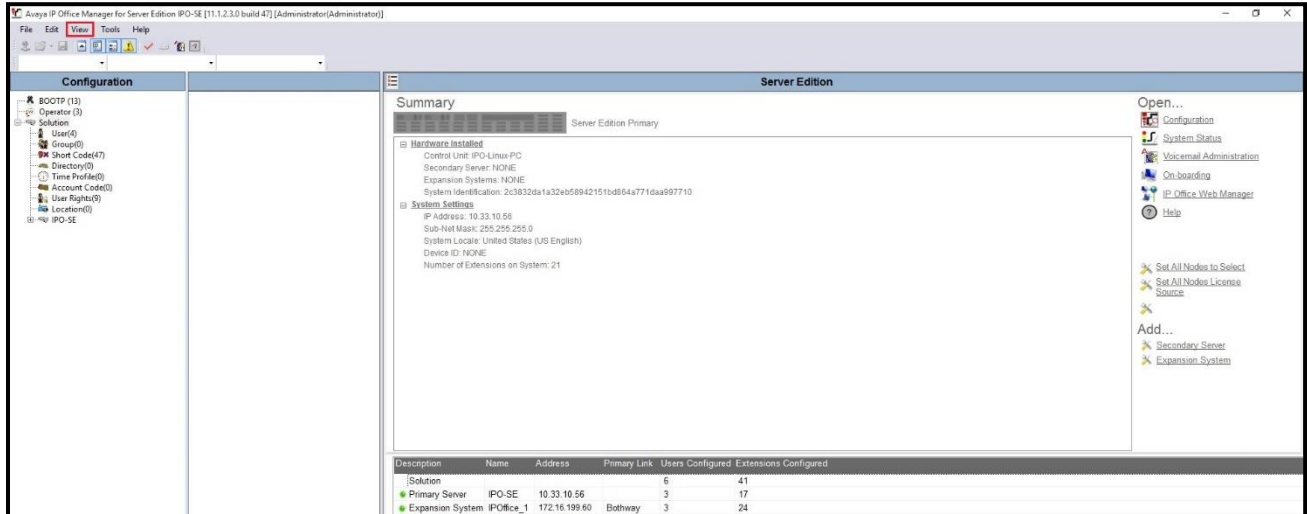
This section describes the Avaya IP Office Server Edition solution configuration necessary to support connectivity to the Telia Norway. It is assumed that the initial installation and provisioning of the Server Edition Primary Server and Expansion System has been previously completed and therefore is not covered in these Application Notes. For information on these installation tasks, refer to the Additional References **Section 9**.

This section describes the Avaya IP Office Server Edition configuration to support connectivity to Telia Norway system. Avaya IP Office Server Edition is configured through the Avaya IP Office Server Edition Manager PC application. From a PC running the Avaya IP Office Server Edition Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office Server Edition system from the pop-up window. Log in using appropriate credentials.



**Figure 2 – Avaya IP Office Server Edition Selection**

The appearance of the Avaya IP Office Server Edition Manager can be customized using the **View** menu. In the screens presented in this section, it includes the system inventory of the servers and links for administration and configuration tasks.



**Figure 3 – Avaya IP Office Server Edition View Menu**

## 5.1. Licensing

The configuration and features described in these Application Notes require the Avaya IP Office Server Edition system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

Licenses for an Avaya IP Office Server Edition solution are based on a combination of centralized licensing done through the Avaya IP Office Server Edition Primary Server, and server specific licenses that are entered into the configuration of the system requiring the feature. SIP Trunk Channels are centralized licenses, and they are entered into the configuration of the Primary Server. Note that when centralized licenses are used to enable features on other systems, such as SIP trunk channels, the Primary Server allocates those licenses to the other systems only after it has met its own license needs. To verify that there is a SIP Trunk Channels license with sufficient capacity, select **Solution** → **IPO-SE** → **License** on the Navigation pane and SIP Trunk Channels in the Group pane. Confirm that there is a valid license with sufficient “Instances” (trunk channels) in the Details pane.

Feature	Instances	Status	Expiration Date	Source
Receptionist	4	Valid	Never	PLDS Nodal
Additional Voicemail Pro Ports	152	Valid	Never	PLDS Nodal
VMPro Recordings Administrators	1	Valid	Never	PLDS Nodal
Essential Edition Additional Voice...	4	Obsolete	Never	PLDS Nodal
VMPro TTS (Generic)	40	Obsolete	Never	PLDS Nodal
Teleworker	384	Obsolete	Never	PLDS Nodal
Mobile Worker	384	Obsolete	Never	PLDS Nodal
Office Worker	384	Valid	Never	PLDS Nodal
Avaya Softphone Licence	100	Valid	Never	PLDS Nodal
VMPro TTS (Scansoft)	40	Obsolete	Never	PLDS Nodal
VMPro TTS Professional	40	Valid	Never	PLDS Nodal
IPSec Tunnelling	1	Obsolete	Never	PLDS Nodal
Power User	384	Valid	Never	PLDS Nodal
Avaya IP endpoints	384	Valid	Never	PLDS Nodal
IP500 Voice Networking Channels	32	Obsolete	Never	PLDS Nodal
<b>SIP Trunk Channels</b>	<b>128</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

Figure 4 – Avaya IP Office Server Edition License

## 5.2. TLS Management

For the compliance test, the signaling on the SIP trunk between IP Office and the Telia Norway was secured using TLS. Testing was done using identity certificate signed by a local certificate authority, Avaya Aura<sup>®</sup> System Manager. This signed certificate was added to Avaya IP Office (on IP Office, navigate to **File** → **Advanced** → **Security Settings**). Avaya also provided the CA certificate to Telia Norway to install on their system. On the other hand, Telia Norway also provided the root CA certificate to install on Avaya IP Office. The generation and installation of these certificates are beyond the scope of these Application Notes. The certificates can be added or viewed on IP Office in the following manner.

To add the certificates on IP Office, navigate to **File** → **Advanced** → **Security Settings**. Log in with the appropriate security credentials (not shown). In the Security Settings window, navigate to **Security** → **System** and select the **Certificates** tab.

- In **Identity Certificate** section, click **Set** button to import the Avaya IP Office signed certificate.
- In **Trusted Certificate Store**, click **Add** button to install the Telia Norway root CA certificate.

To view the certificates currently installed on IP Office, navigate to **File** → **Advanced** → **Security Settings**. Log in with the appropriate security credentials (not shown). In the Security Settings window, navigate to **Security** → **System** and select the **Certificates** tab.

- In **Identity Certificate** section, click **View** button to view the Avaya IP Office signed certificate.
- In **Trusted Certificate Store**, select the certificate and click **View** button to view the Telia Norway root CA certificate.

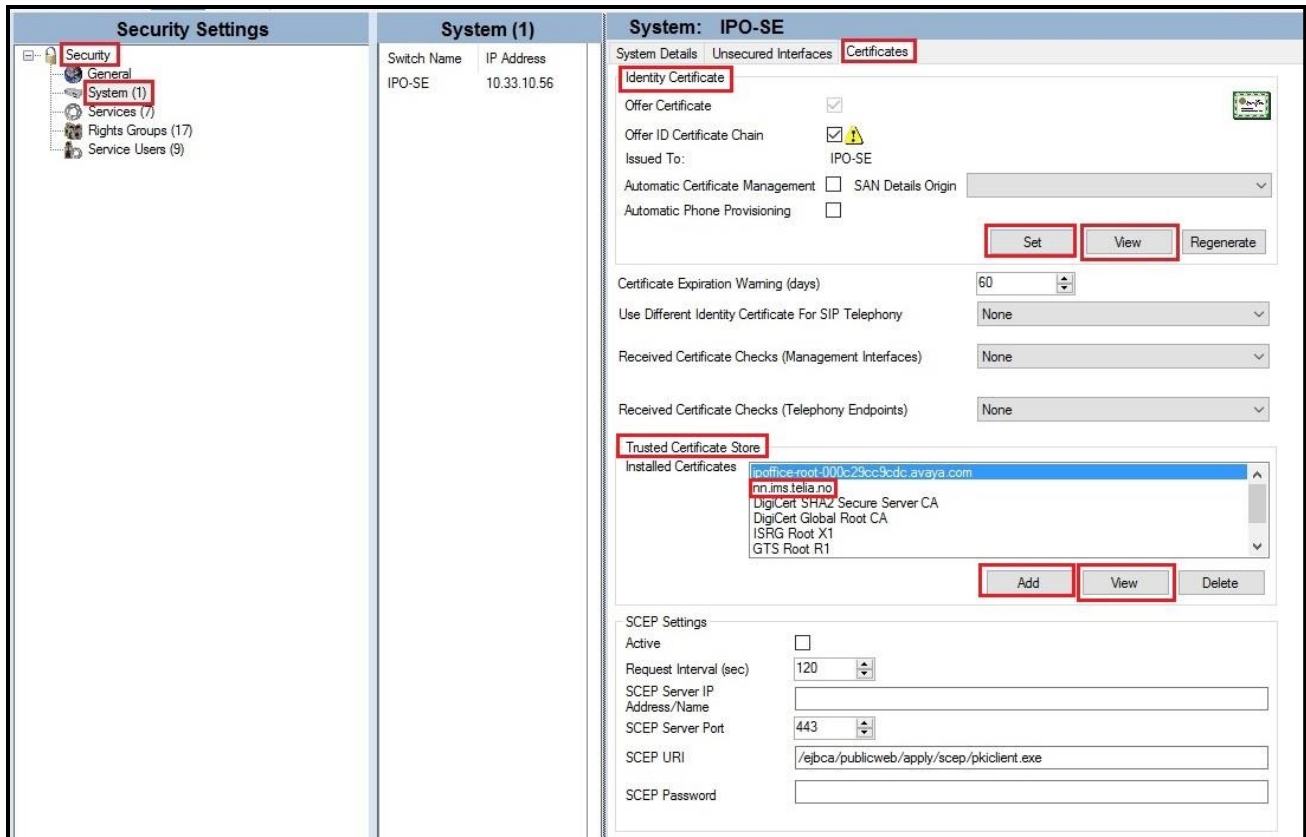


Figure 5 – Avaya IP Office Server Edition TLS Certificate

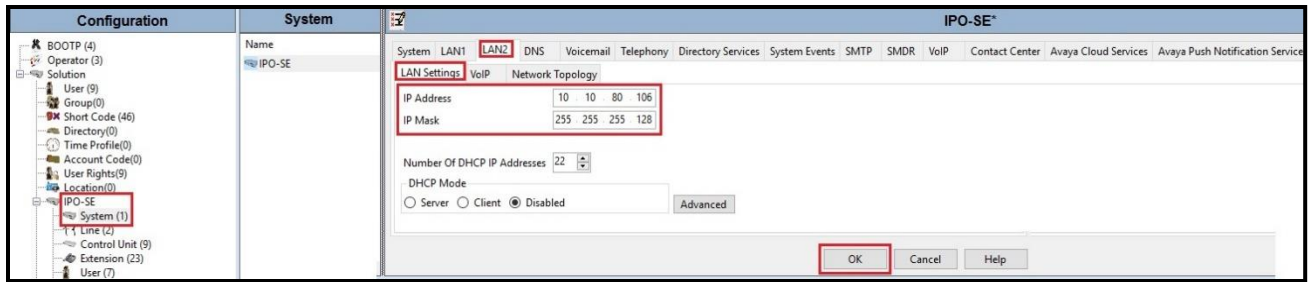
### 5.3. System Settings

Configure the necessary system settings.

#### 5.3.1. System – LAN Tab

In the sample configuration, LAN2 on the Primary Server was used, and LAN1 on the Expansion System was used. Note: The LAN1 port of the Primary Server (Eth0) is connected to the enterprise LAN (Private network) and will not be discussed in this document. The **IPO-SE** was used as the Primary Server name and **IPOffice\_1** was used as the Expansion System name.

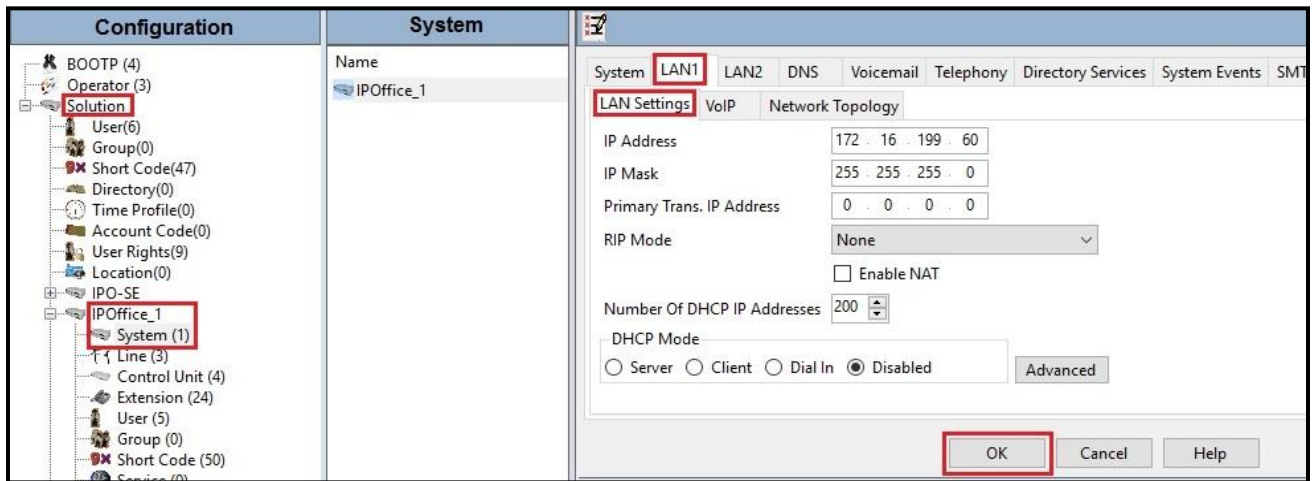
To configure the LAN2 settings on the Primary Server, complete the following steps. Navigate to **IPO-SE** → **System (1)** in the Navigation and Group Panes and then navigate to the **LAN2** → **LAN Settings** tab in the Details Pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office Server Edition LAN2 port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements. Click **OK** to submit the change.



**Figure 6 - Avaya IP Office Primary Server LAN2 Settings**



To configure the LAN1 settings tab for the Expansion System, navigate to **Solution** → **IPOffice\_1** → **System (1)** in the Navigation and Group Panes and then navigate to the **LAN1** → **LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields should be populated with the values assigned during the Expansion System initial installation process. Verify the configuration or modify the values if needed. While DHCP was disabled during the compliance test, this parameter should be set according to customer requirements. Other settings were left at their default values. Click **OK** to submit the change.



**Figure 8 - Avaya IP Office Expansion Server Settings**

The **VoIP** tab for LAN1 in the Expansion System (not shown) can be configured using the same values previously described for the **VoIP** tab in the Primary Server.



### 5.3.2. System – Telephony Tab

Navigate to **Solution** → **IPO-SE** → **System (1)** in the Navigation and Group Panes (not shown) and then navigate to the **Telephony** → **Telephony** tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location. For Norway area, **A-Law** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfers to the PSTN via the service provider across the SIP trunk. The Hold Timeout (sec) field controls how long calls remain on hold before being alerted to the user and should be set based on the customer's requirement. Set **Default Name Priority** to **Favor Trunk** to have IP Office display the name provided in the Caller ID from the SIP trunk. Defaults were used for all other settings. Click **OK** to submit the changes.

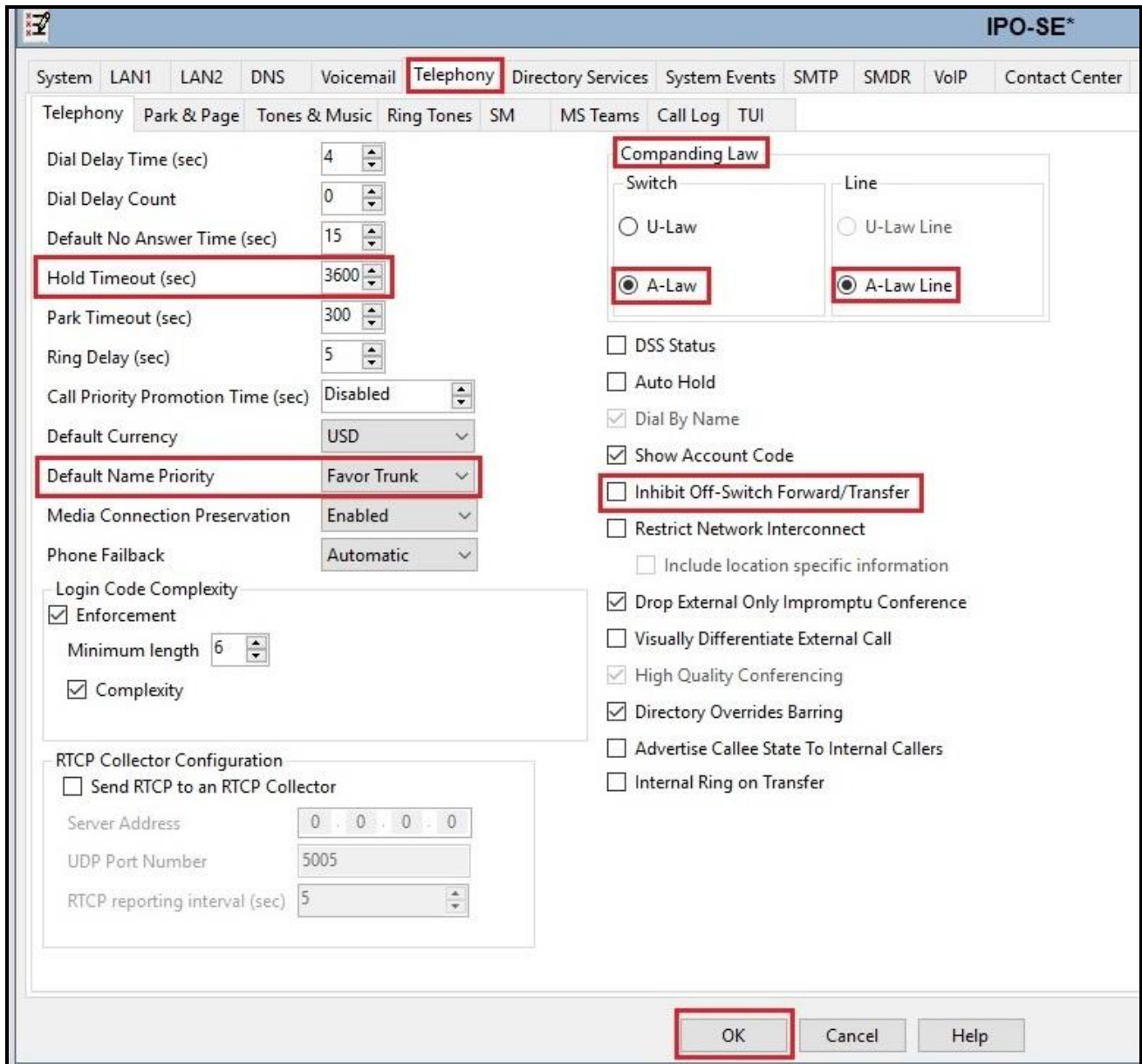
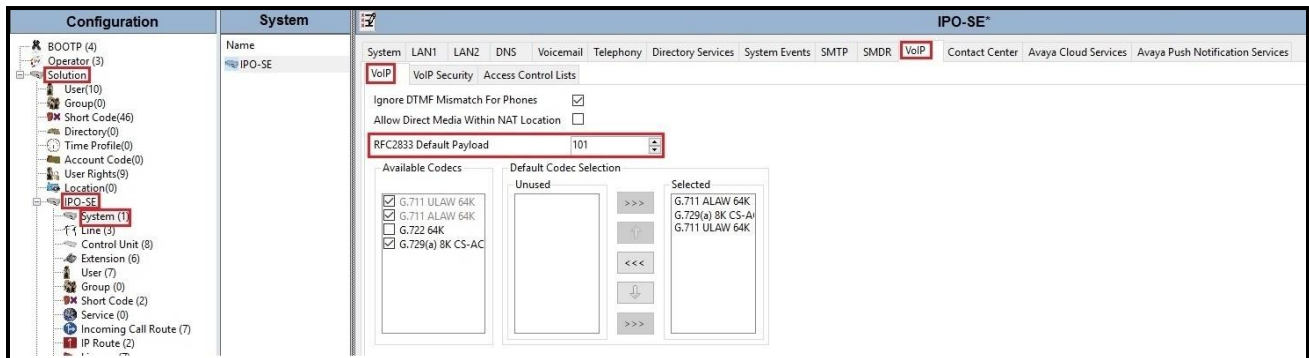


Figure 9 - Avaya IP Office Primary Server Telephony

Navigate to **Solution** → **IPOffice\_1** → **System (1)** (not shown) and repeat the steps above to configure the **Telephony** settings for the Expansion System.

### 5.3.3. System – VoIP Tab

Navigate to **Solution** → **IPO-SE** → **System (1)** in the Navigation and Group Panes and then navigate to the **VoIP** tab in the Details Pane. Leave the **RFC2833 Default Payload** as the default value of **101**. The buttons between the two lists can be used to move codecs between the **Unused** and **Selected** lists, and to change the order of the codecs in the **Selected** codecs list. By default, all IP lines and phones (SIP and H.323) will use the system default codec selection shown here, unless configured otherwise for a specific line or extension. The example below shows the codecs used for IP phones (SIP and H.323), the system's default codecs and order were used. Click **OK** (Not shown) to submit the changes.



**Figure 10 - Avaya IP Office Primary Server VoIP**

**Note:** The codec selections defined under this section (VoIP – VoIP tab) are the codecs selected for the IP phones/extensions. The codec selections defined under **Section 5.5.2** (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk).

Secure Real-Time Transport Protocol (SRTP) refers to the application of additional encryption and or authentication to VoIP calls (SIP and H.323). SRTP can be applied between telephones, between ends of an IP trunk or in various other combinations.

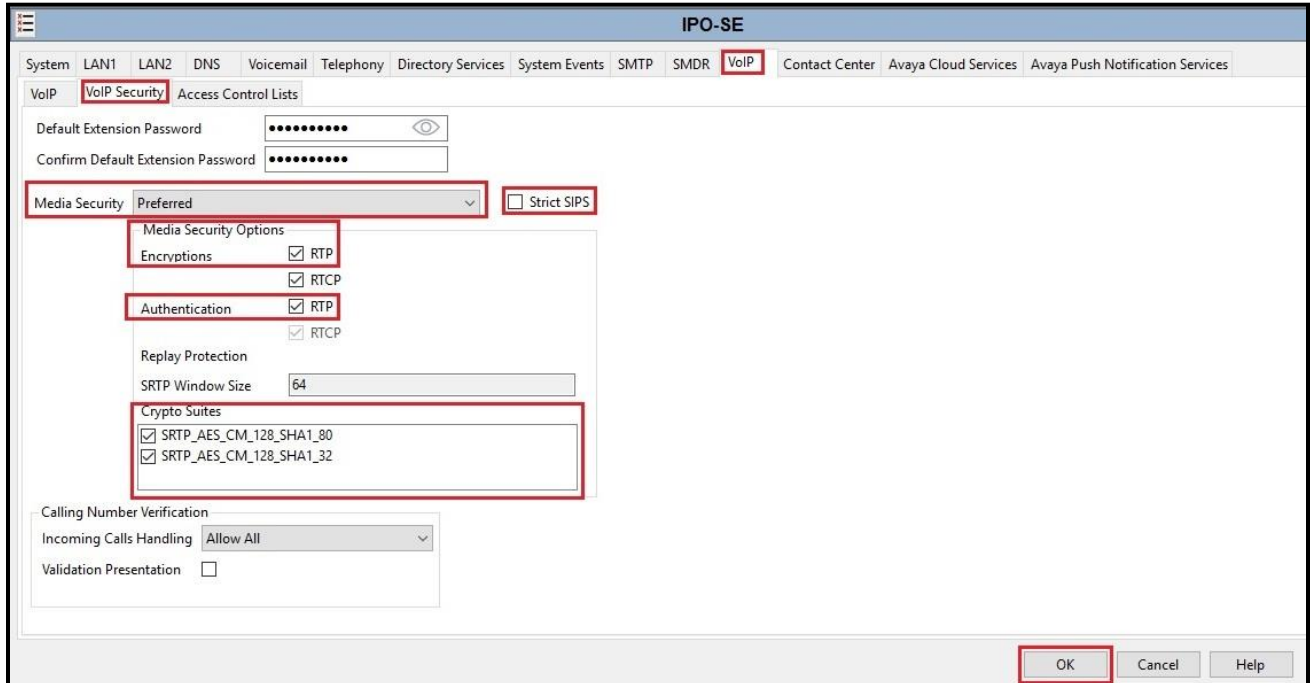
Configuring the use of SRTP at the system level is done on the **VoIP Security** tab using the Media Security setting. The options are:

- Disabled (default)
- Preferred
- Enforced

Note: When enabling SRTP on the system, the recommended setting is **Preferred**. In this scenario, IP Office uses SRTP if supported by the far-end, otherwise uses RTP. If the **Enforced** setting is used, and SRTP is not supported by the far-end, the call is not established.

To configure the use of SRTP, navigate to **Solution** → **IPO-SE** → **System (1)** in the Navigation and Group Panes and then navigate to **VoIP** → **VoIP Security** tab on the Details pane. Set the **Media** drop-down menu to **Preferred** to have IP Office attempt use encrypted RTP for devices that support it and fall back to RTP for devices that do not support encryption.

- Verify **Strict SIPS** is not checked
- Under **Media Security Options**, select **RTP** for the **Encryptions** and **Authentication** fields
- Under **Crypto Suites**, select **SRTP\_AES\_CM\_128\_SHA1\_80** and **SRTP\_AES\_CM\_128\_SHA1\_32**
- Click **OK** to commit



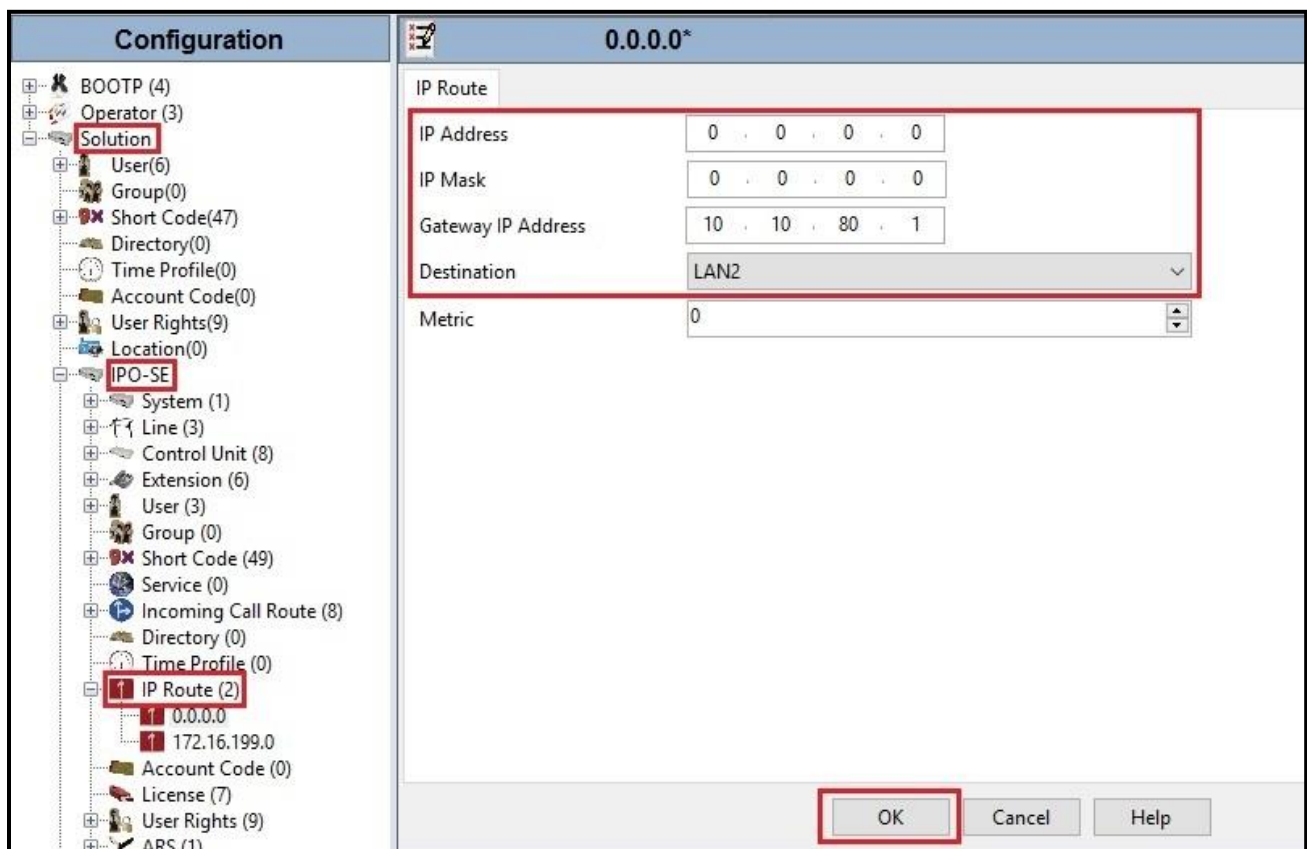
**Figure 11 - Avaya IP Office Primary Server VoIP Security**

## 5.4. IP Route

Create an IP route to specify the IP address of the gateway or router where the IP Office needs to send the packets in order to route calls to Telia Norway.

To create an IP route for the Primary system, navigate to **Solution** → **IPO-SE** → **IP Route**, right-click on **IP Route** and select **New** (Not shown). The values used during the compliance test are shown below:

- Set the **IP Address** and **IP Mask** to **0.0.0.0** to make this the default route
- Set **Gateway IP Address** to the IP address of the gateway/router used to route calls to the public network, e.g., **10.10.80.1**
- Set **Destination** to **LAN2** from the pull-down menu
- Click **OK** to commit



**Figure 12 - Avaya IP Office Primary Server IP Route**

To create an IP route for the Expansion system, navigate to **Solution** → **IPOffice\_1** → **IP Route**, right-click on **IP Route** and select **New** (Not shown). The values used during the compliance test are shown below:

- Set the **IP Address** and **IP Mask** to **0.0.0.0** to make this the default route

- Set **Gateway IP Address** to the IP address of the gateway/router used to route calls to the private network, e.g., **172.16.199.1**
- Set **Destination** to **LAN1** from the pull-down menu
- Click **OK** to commit



**Figure 13 - Avaya IP Office Expansion Server IP Route**

## 5.5. Administer SIP Line

A SIP Line is needed to establish the SIP connection between Avaya IP Office Server Edition and Telia Norway system. 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 Avaya IP Office Server Edition 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

- SIP Advanced Engineering

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**.

For the compliance test, SIP Line 17 was used as trunk for both outgoing and incoming calls.

### 5.5.1. Create SIP Line from an XML Template

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

Create a new folder in a location where Avaya IP Office Server Edition Manager is installed (e.g., C:\Telia Norway\Template). Copy the template file to this folder and rename the template file to **TN-IPO11\_1.xml** (for SIP Line 17).

Create the SIP Trunk from the template, from the Primary server, right-click on **Line** in the Navigation Pane, then navigate to **New from Template** → **Open from file**.

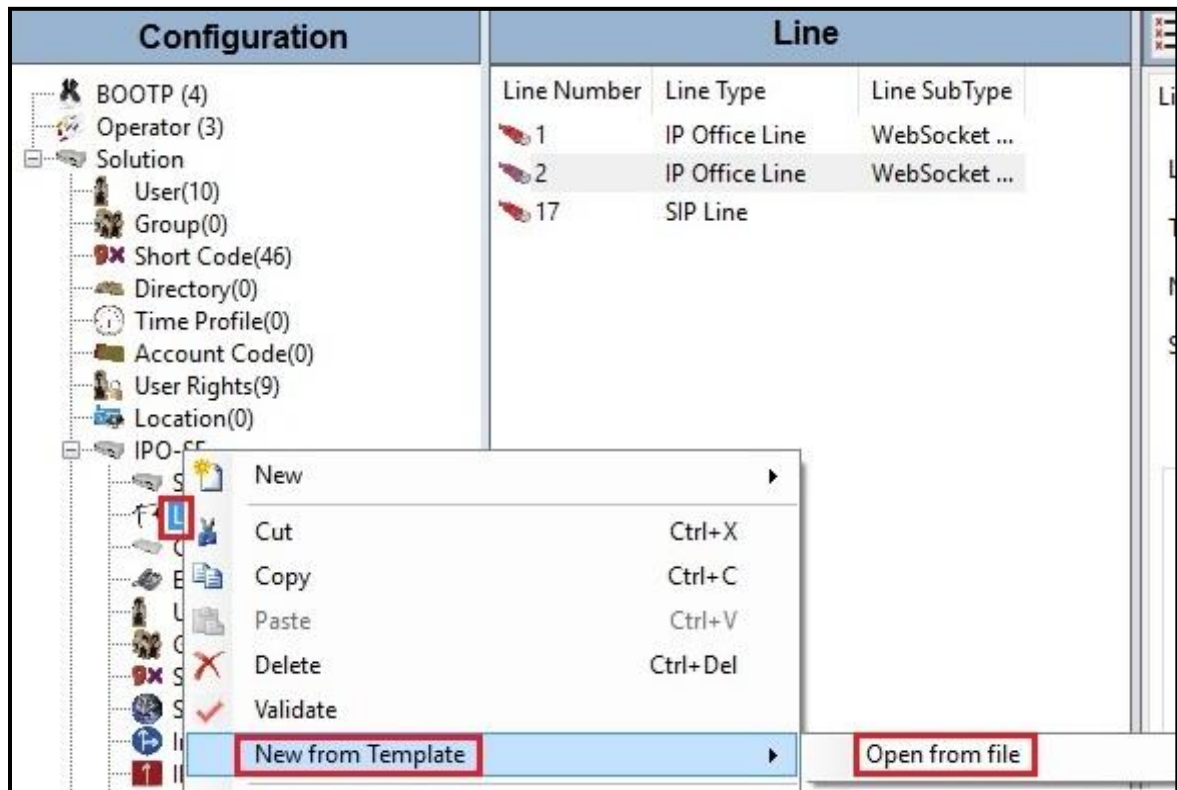
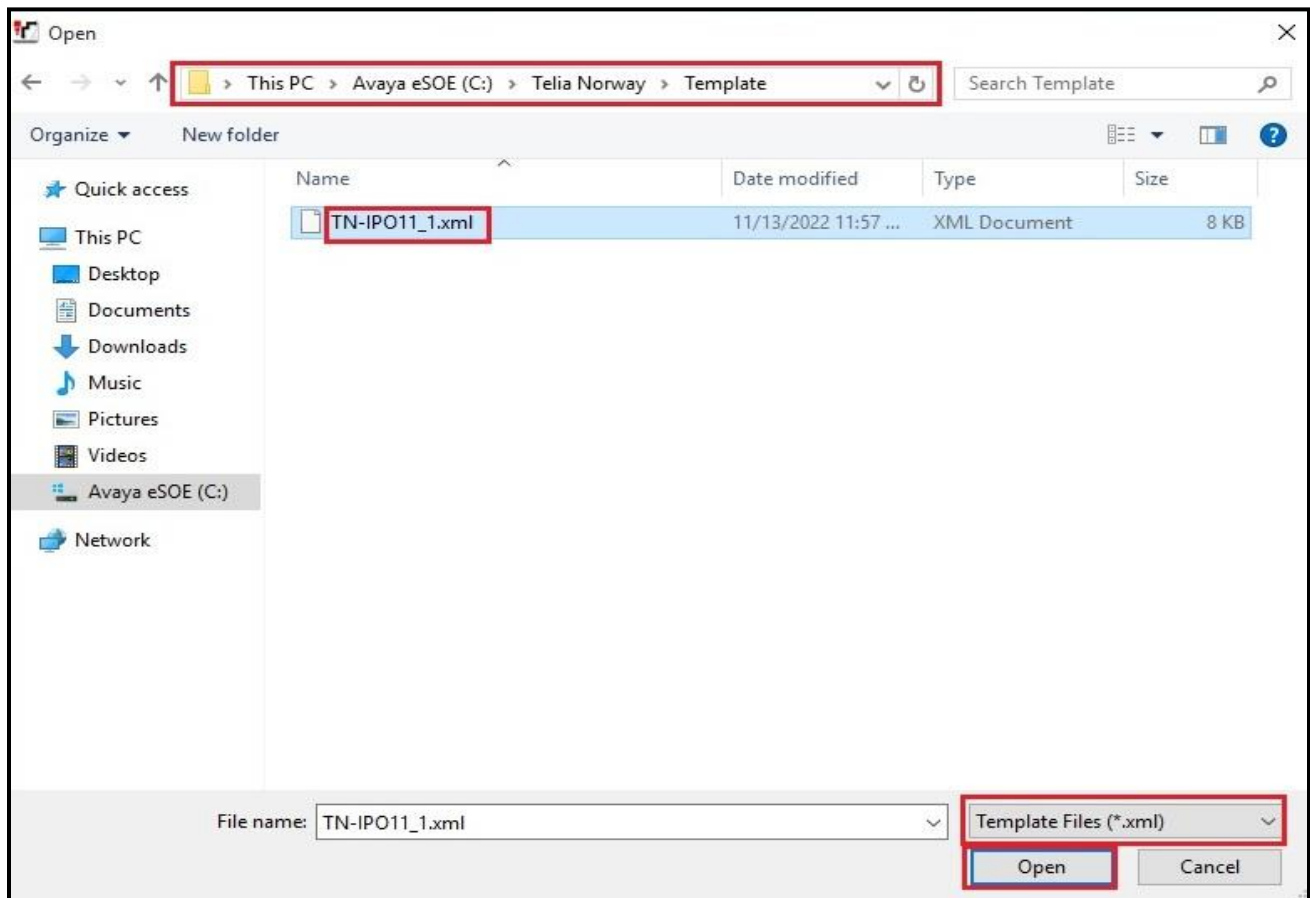


Figure 14 – Create SIP Line from an XML Template



Select the **Template Files (\*.xml)** and select the copied template at folder (e.g., C:\Telia Norway\Template). Click **Open** button to create a SIP line from template.



**Figure 15 – Create SIP Line from directory**

A pop-up window below will appear stating success (or failure). Then click **OK** to continue.



**Figure 16 – Create SIP Line from Template successfully**

Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Section 5.5.2**.

## 5.5.2. Create SIP Line Manually

To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New** → **SIP Line** (not shown).

On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Select available **Line Number: 17**
- Set **ITSP Domain Name** to the Telia Norway domain for production service. This field is used to specify the default host part of the SIP URI in the To and R-URI fields for outgoing calls
- Leave **Local Domain Name** to **blank**
- Check the **In Service** and **Check OOS** boxes
- Set **URI Type** to **SIP URI**
- For **Session Timers**, set **Refresh Method** to **Auto** with **Timer (sec)** to **On Demand**
- Set **Name Priority** to **Favor Trunk**. As described in **Section 5.3.2**, the **Default Name Priority** parameter may retain the default **Favor Trunk** setting or can be configured to **Favor Directory**. As shown below, the default **Favor Trunk** setting was used in the reference configuration
- For **Redirect and Transfer**, set **Incoming Supervised REFER** and **Outgoing Supervised Auto**
- Default values may be used for all other parameters
- Click **OK** to commit then press **Ctrl + S** to save

Configuration	Line	SIP Line - Line 17
BOOTP (13) Operator (3) Solution User(4) Group(0) Short Code(47) Directory(0) Time Profile(0) Account Code(0) User Rights(9) Location(0) IPG-SE System (1) Line (1) Control Unit (9) Extension (21) User (5) Group (0) Short Code (49) Service (0) Incoming Call Route (4) Directory (0) Time Profile (0) IP Route (2) Account Code (0) License (30) User Rights (9) Auto Attendant (0) ARS (1) Conference (0)	Line Number: 17 Line Type: SIP Line Line SubType:	<b>SIP Line</b> Transport: Call Details: VoIP SIP Credentials: SIP Advanced: Engineering Line Number: 17 ITSP Domain Name: cust.3e8t533yh4mf7uj.net Local Domain Name: URI Type: SIP URI Location: Cloud Prefix: National Prefix: International Prefix: Country Code: Name Priority: Favor Trunk Description: In Service: <input checked="" type="checkbox"/> Check OOS: <input checked="" type="checkbox"/> Session Timers Refresh Method: Auto Timer (sec): On Demand Redirect and Transfer Incoming Supervised REFER: Auto Outgoing Supervised REFER: Auto Send 302 Moved Temporarily: <input type="checkbox"/> Outgoing Blind REFER: <input type="checkbox"/> OK Cancel Help

Figure 17 – SIP Line Configuration

On the **Transport** tab in the Details Pane, configure the parameters as shown below:

- The **ITSP Proxy Address** was set to the IP addresses of Telia Norway signaling servers: **192.168.89.165, 192.168.93.165, 192.168.84.165** as shown in **Figure 1**. These are the SIP Proxy addresses used for outgoing SIP calls
- In the **Network Configuration** area, **TLS** was selected as the **Layer 4 Protocol** and the **Send Port** was set to **6004**
- The **Use Network Topology Info** parameter was set to **None**. The **Listen Port** was set to **6004**. Note: For the compliance testing, the **Use Network Topology Info** field was set to **None**, since no NAT was using in the test configuration. In addition, it was not necessary to configure the **System → LAN2 → Network Topology** tab for the purposes of SIP trunking. If a NAT is used between Avaya IP Office and the other end of the trunk, then the **Use Network Topology Info** field should be set to the LAN interface (**LAN2**) used by the trunk and the **System → LAN2 → Network Topology** tab needs to be configured with the details of the NAT device
- The **Calls Route via Registrar** was unchecked as Telia Norway did not support the dynamic Registration on the SIP Trunk
- Other parameters retain default values
- Click **OK** to commit then press Ctrl + S to save

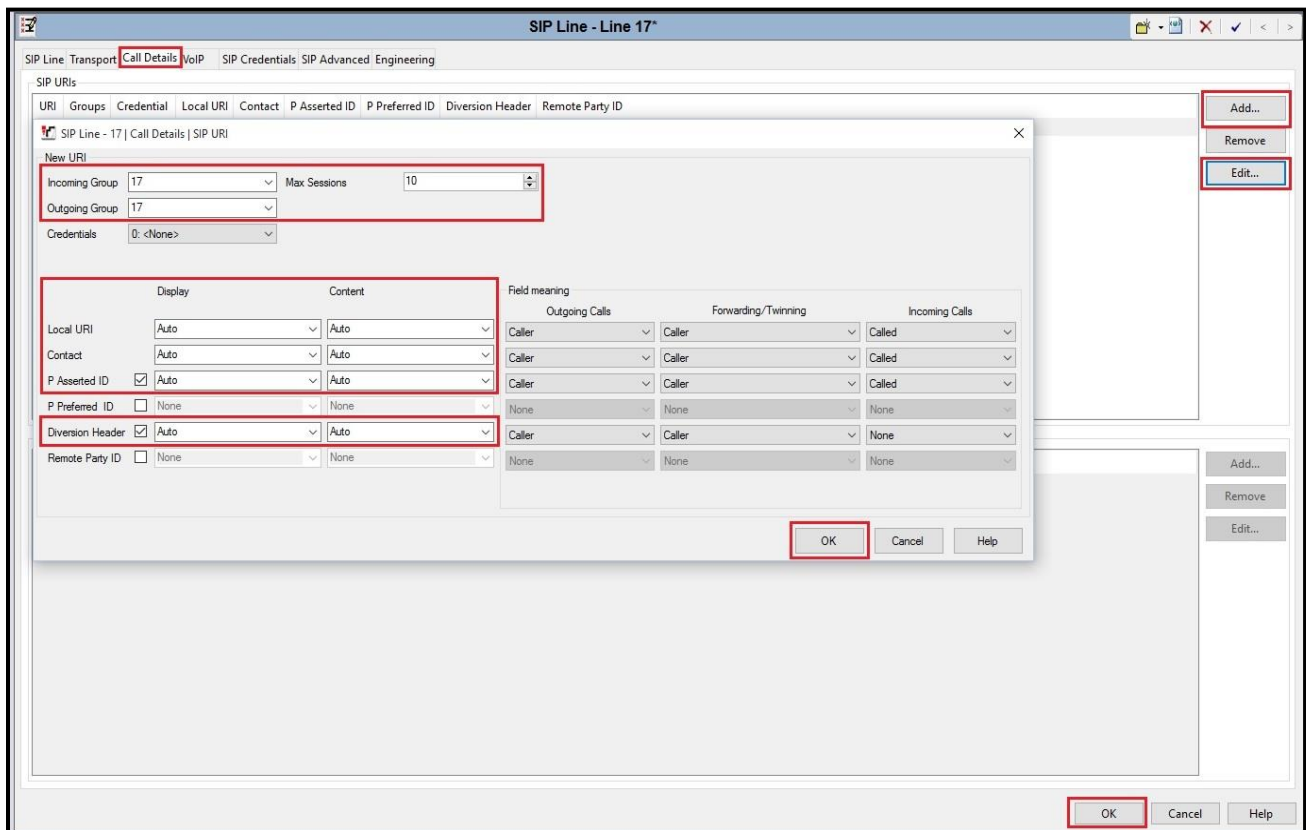
The screenshot shows the 'SIP Line - Line 17\*' configuration window. The 'Transport' tab is selected. The 'ITSP Proxy Address' field contains '192.168.89.165 192.168.93.165 19'. The 'Network Configuration' section is expanded, showing 'Layer 4 Protocol' set to 'TLS', 'Send Port' set to '6004', 'Use Network Topology Info' set to 'None', and 'Listen Port' set to '6004'. The 'Calls Route via Registrar' checkbox is unchecked. The 'Explicit DNS Server(s)' field contains two IP addresses: '0 . 0 . 0 . 0' and '0 . 0 . 0 . 0'. The 'Separate Registrar' field is empty. The 'OK' button is highlighted with a red box.

**Figure 18 – SIP Line Transport Configuration**

The SIP URI entry must be created to match any DID number assigned to an Avaya IP Office user and Avaya IP Office will route the calls on this SIP line. Select the **Call Details** tab; click the **Add** button and the **New Channel** area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top, and click **Edit...** button. In the example screen below, a previously configured entry is edited

A SIP URI entry was created that matched any DID number assigned to an Avaya IP Office user. The entry was created with the parameters shown below:

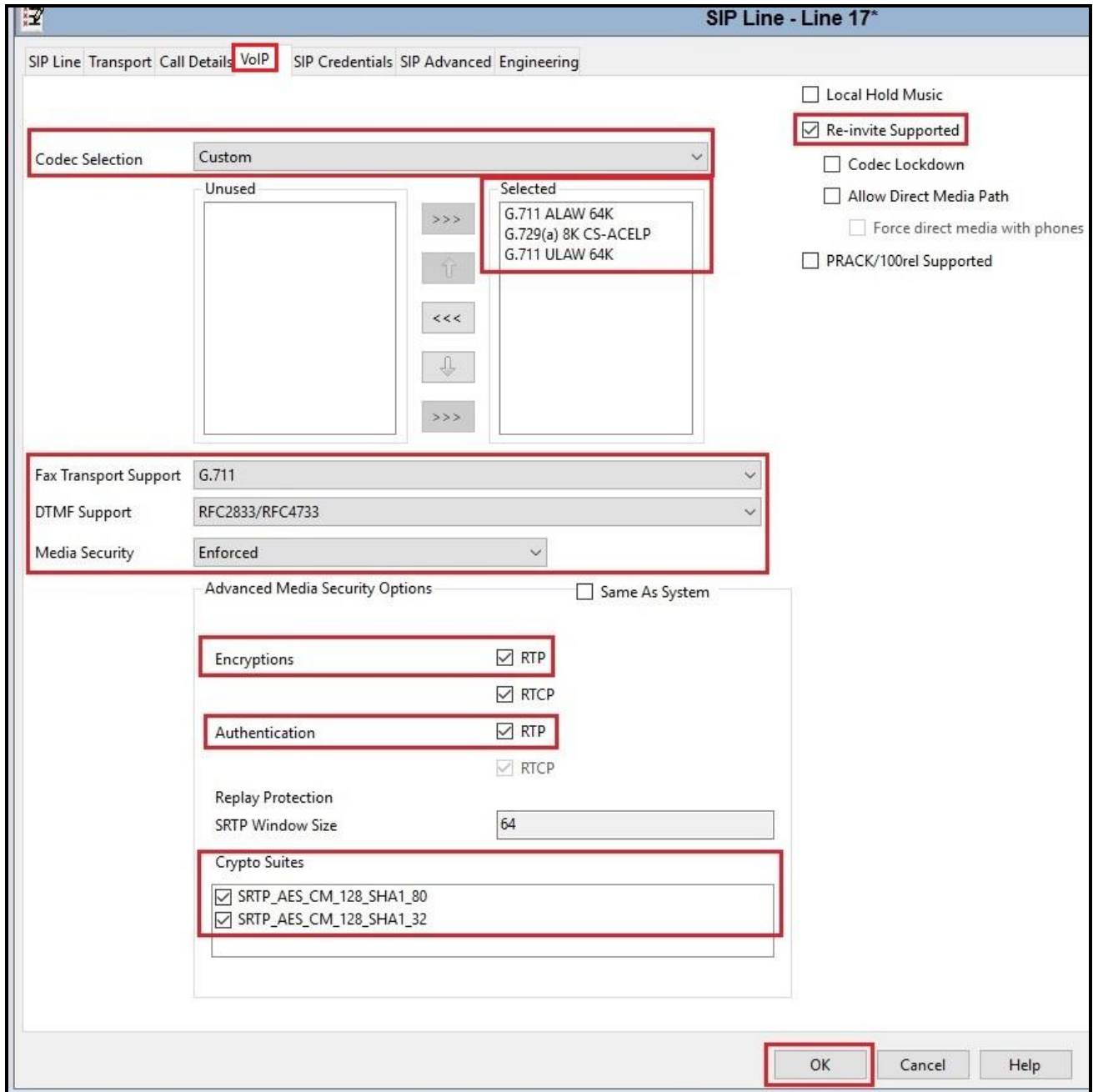
- Associate this SIP line with an incoming line group in the **Incoming Group** field and an outgoing line group in the **Outgoing Group** field. This line group number will be used in defining incoming and outgoing call routes for this line. For the compliance test, a new line group **17** was defined that only contains this line (line 17)
- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern
- Check **P Asserted ID** and **Diversion Header** options
- Set the **Local URI**, **Contact**, **P Asserted ID** and **Diversion Header** fields to the values shown in the screenshot below
- Click **OK** to submit the changes



**Figure 19 – SIP Line Call Details Configuration**

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

- The **Codec Selection** can be selected by choosing **Custom** from the pull-down menu, allowing an explicit ordered list of codecs to be specified. The **G.711 ALAW 64K**, **G.729(a) 8K CS-ACELP** and **G.711 ULAW 64K** codecs are selected. Avaya IP Office Server Edition supports these codecs, which are sent to the Telia Norway, in the Session Description Protocol (SDP) offer
- Check the **Re-invite Supported** box
- Set **Fax Transport Support** to **G.711** from the pull-down menu
- Set the **DTMF Support** to **RFC2833/RFC4733** from the pull-down menu. This directs Avaya IP Office Server Edition to send DTMF tones using RTP events messages as defined in RFC2833 and RFC4733
- Select **Media Security** as **Enforced**. Note: Telia Norway required to send SAVP, therefore Media Security should be selected as Enforced, not Preferred.
- Under **Advanced Media Security Options**, select **RTP** for the **Encryptions** and **Authentication** fields
- Under **Crypto Suites**, select **SRTP\_AES\_CM\_128\_SHA1\_80** and **SRTP\_AES\_CM\_128\_SHA1\_32**
- Default values may be used for all other parameters
- Click **OK** to submit the changes



**Figure 20 – SIP Line VoIP Configuration**

## 5.6. IP Office Line in Primary System

In IP Office Server Edition systems, IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. To edit an existing IP Office Line, select **Line** in the Navigation pane, and select the appropriate line to be configured in the Group pane.

To verify the IP Office line connecting the Primary System to the Expansion System, select **Line** on the navigation pane of Primary System and select the IP Office Line on the Group pane (line 2 on the screen below). Make note of the **Outgoing Group ID 99999** on the Details pane. The **Address of Gateway** is Avaya IP Office Expansion System LAN1 IP address **172.16.199.60**.

The screenshot displays the Avaya IP Office configuration interface. The left pane shows the navigation tree with 'Line' selected under 'IP Office Line'. The middle pane shows a table of lines:

Line Number	Line Type	Line SubType
1	IP Office Line	WebSocket ...
2	IP Office Line	WebSocket ...
17	SIP Line	

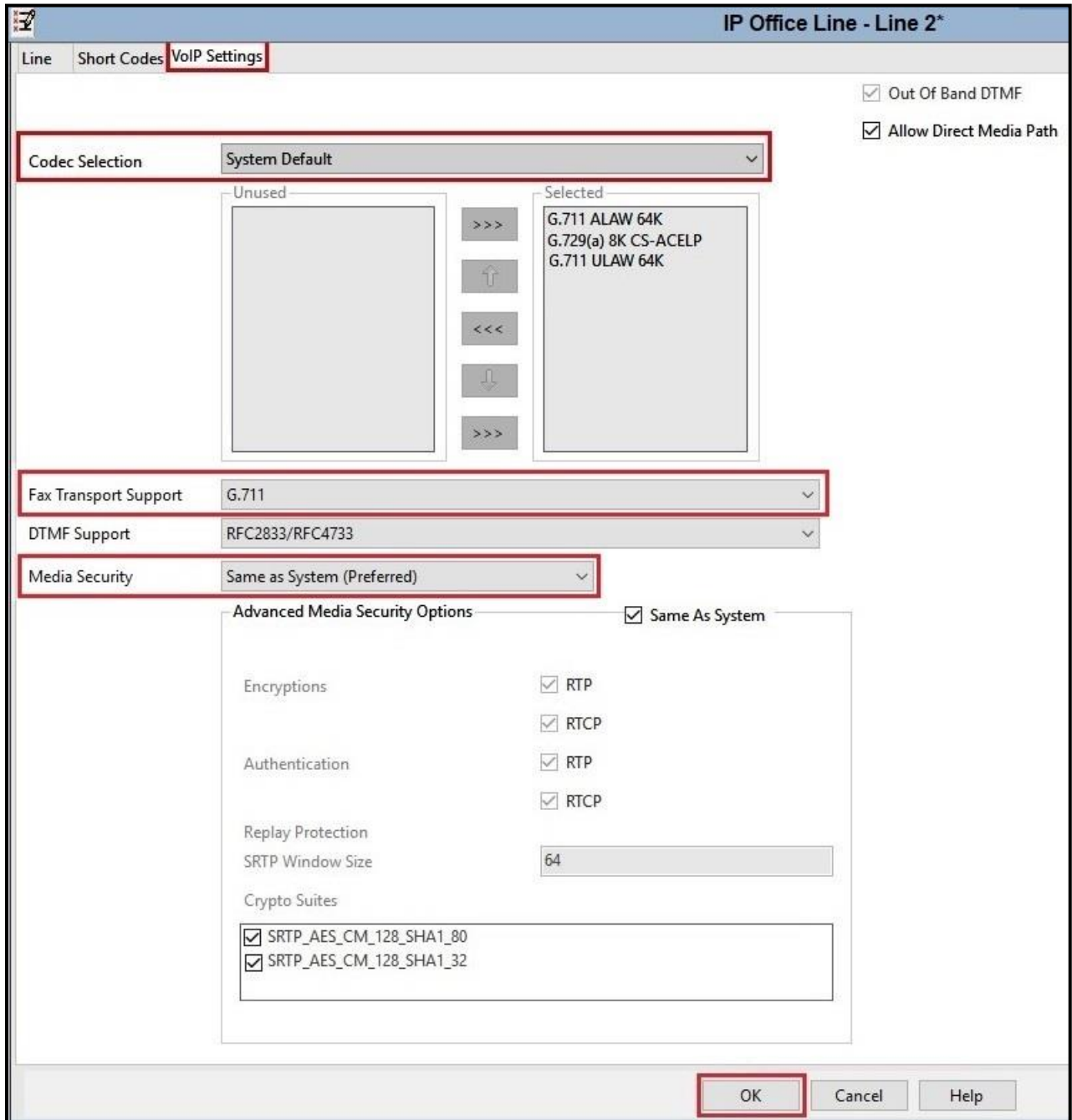
The right pane shows the configuration for 'IP Office Line - Line 2'. Key settings are highlighted with red boxes:

- Line Number: 2
- Transport Type: WebSocket Server
- Networking Level: SCN
- Security: Medium
- Gateway Address: 172 . 16 . 199 . 60
- Outgoing Group ID: 99999
- Number of Channels: 250
- Outgoing Channels: 250

SCN Resiliency Options are also visible, including checkboxes for 'Supports Resiliency', 'Backs up my IP phones', 'Backs up my hunt groups', and 'Backs up my IP DECT phones'.

Figure 21 – IP Office Line for Primary System

To verify the **VoIP Settings** of the IP Office line connecting the Primary System to the Expansion System, select **VoIP Settings** tab. The selected codecs are **G.711 ALAW 64K**, **G.729(a) 8K CS-ACELP** and **G.711 ULAW 64K**. Select **Fax Transport Support** to **G.711** (This setting should be as same as the VoIP settings in SIP line of Primary System and the VoIP settings in IP Office Line of Expansion System). Under **Media Security** verify **Same as System (Preferred)** is selected. Default values may be used for all other parameters. Click **OK** to submit the changes.



**Figure 22 – IP Office Line for Primary System VoIP Settings**



## 5.7. IP Office Line in Expansion System

To verify the IP Office line connecting the Expansion System to the Primary System, select Expansion Line on the navigation pane and select the IP Office Line on the Group pane (line 17 on the screen below). Make note of the **Outgoing Group ID 99999** on the Details pane. The **Address of Gateway** is Avaya IP Office Server Edition LAN1 IP address **10.33.10.56**.

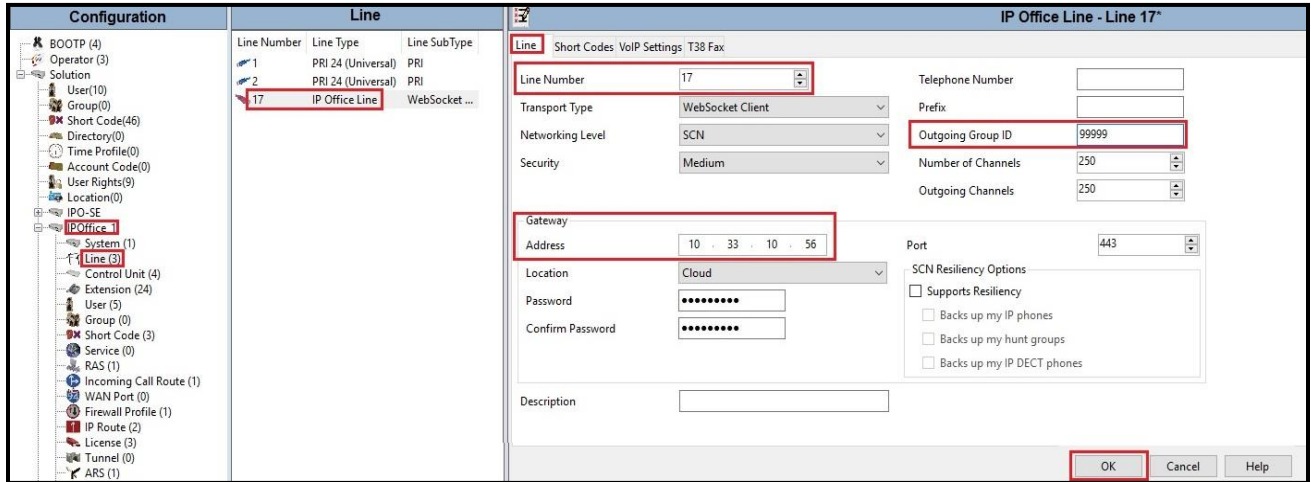


Figure 23 – IP Office Line for Expansion System

To verify the **VoIP Settings** of the IP Office line connecting the Expansion System to the Primary Server, select **VoIP Settings** tab. The selected codecs are **G.711 ALAW 64K**, **G.729(a) 8K CS-ACELP** and **G.711 ULAW 64K** as System Default. Select **Fax Transport Support** to **G.711** (This setting should be as same as the VoIP settings in SIP line and IP Office Line of Primary System). Default values may be used for all other parameters. Click **OK** to submit the changes.

The screenshot shows the configuration interface for an IP Office Line. The title bar reads "IP Office Line - Line 17\*". Below the title bar, there are tabs for "Line", "Short Codes", "VoIP Settings", and "T38 Fax". The "VoIP Settings" tab is active. On the right side, there are three checkboxes: "VoIP Silence Suppression" (unchecked), "Out Of Band DTMF" (checked), and "Allow Direct Media Path" (checked). The main configuration area is divided into two columns: "Unused" and "Selected". The "Selected" column contains three codecs: "G.711 ALAW 64K", "G.729(a) 8K CS-ACELP", and "G.711 ULAW 64K". Below the codec selection, there are five buttons: ">>>", "↑", "<<<", "↓", and ">>>". The "Fax Transport Support" dropdown is set to "G.711". The "Call Initiation Timeout (s)" is set to 4. The "Media Security" dropdown is set to "Same as System (Preferred)".

**Figure 24 – IP Office Line for Expansion Server VoIP Settings**

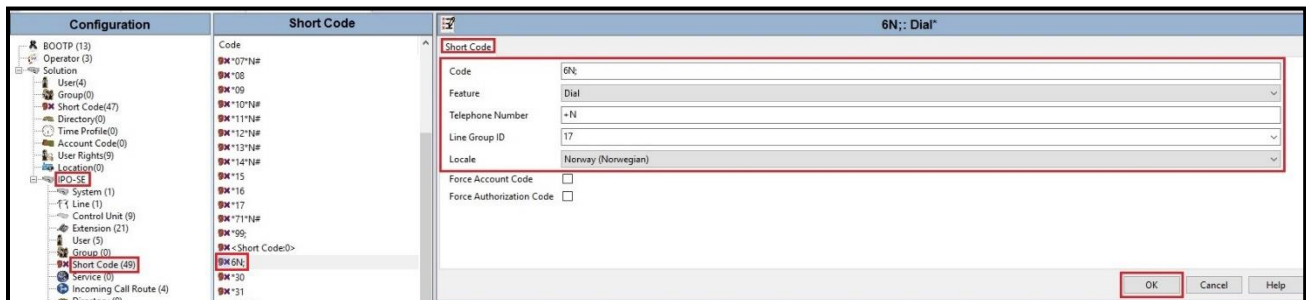
## 5.8. Outbound Short Code

Define a short code to route outbound traffic on the SIP line to Telia Norway. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane and select **New** (not shown). On the **Short Code** tab in the Details Pane, configure the parameters for the new short code to be created.

The screen below shows the details of the previously administered “**6N;**” short code for Primary System used in the test configuration.

Navigate to **Solution → IPO-SE → Short Code**, right-click on **Short Code** and select **New**.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **6N;**, this short code will be invoked when the user dials 6 followed by any number
- Set **Feature** to **Dial**. This is the action that the short code will perform
- Set **Telephone Number** to **+N**. The value **N** represents the number dialed by the user. Note: Use the specific **W** in front of **+N** for restricting all outbound calls
- Set the **Line Group ID** to the **Outgoing Group 17** defined on the **Call Details** tab on the **SIP Line** in **Section 5.5.2**. This short code will use this line group when placing the outbound call
- Set the **Locale** to **Norway (Norwegian)**
- Default values may be used for all other parameters
- Click **OK** to submit the changes



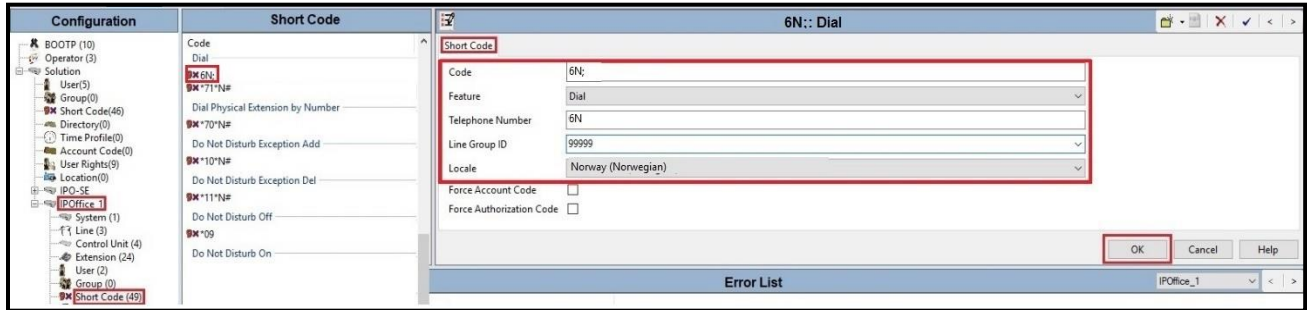
**Figure 25 – Short Code 6N for Primary Server**

The screen below shows the details of the previously administered “**6N;**” short code for Expansion System used in the test configuration.

Navigate to **Solution → IPOffice\_1 → Short Code**, right-click on **Short Code** and select **New**

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **6N;**, this short code will be invoked when the user (using Avaya analog or digital phones) dials 6 followed by any number
- Set **Feature** to **Dial**. This is the action that the short code will perform
- Set **Telephone Number** to **6N**

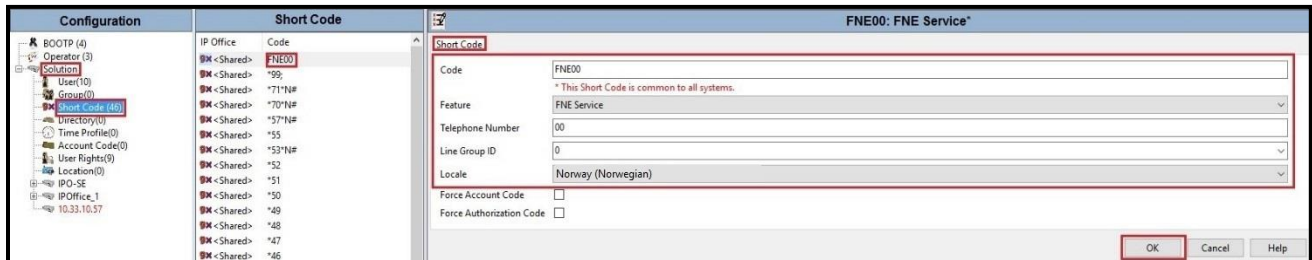
- Set the **Line Group ID** to **99999** defined on the **Outgoing Group ID** of the IP Office line connecting the Expansion System to the Primary System. This short code will use this line group when placing the outbound call via Avaya IP Office Server Edition Primary Server
- Default values may be used for all other parameters
- Click **OK** to submit the changes



**Figure 26 – Short Code 6N for Expansion System**

The feature of incoming calls from mobility extension to idle-appearance FNE (Feature Name Extension) is hosted by Avaya IP Office Server Edition. The Short Code **FNE00** was configured with following parameters:

- For **Code** field, enter FNE feature code as **FNE00** for dial tone
- Set **Feature** to **FNE Service**
- Set **Telephone Number** to **00**
- Set **Line Group ID** to **0**
- Set the **Locale** to **Norway (Norwegian)**
- Default values may be used for other parameters
- Click **OK** to submit the changes



**Figure 27 – Short Code FNE**

## 5.9. 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 select **User** in the left Navigation Pane, then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is **473XXX4500**. Select the **User** tab in the Details pane.

Note: When **Auto** is selected for the **Local URI**, **Contact** and **Diversion Header** parameters (See **Section 5.5.2 - Call Detail** tab), the information in the Incoming Call Route (See **Section 5.10**) is used to populate the SIP From and Contact headers for outbound calls.

The screenshot displays the configuration interface for a user. On the left is a navigation tree with 'User' selected. The center pane shows a list of users, with '473XXX4500' selected. The right pane shows the configuration details for this user, with the 'User' tab active. The configuration includes fields for Name, Password, Confirm Password, Unique Identity, Conference PIN, Confirm Audio, Conference PIN, Account Status, Full Name, Extension, Email Address, Locale, Priority, System Phone Rights, Profile, and Device Type. The 'Profile' section contains several checkboxes for enabling various services like Softphone, Portal Services, Remote Worker, etc. The 'User Rights' section shows 'User data' selected.

473XXX4500: 473XXX4500	
User	Voicemail DND Short Codes Source Numbers Telephony Forwarding Dial In Voice Recording Button Programming Menu Programming Mobility
Name	473XXX4500
Password	*****
Confirm Password	*****
Unique Identity	
Conference PIN	
Confirm Audio	
Conference PIN	
Account Status	Enabled
Full Name	473XXX4500
Extension	473XXX4500
Email Address	
Locale	Norway (Norwegian)
Priority	5
System Phone Rights	None
Profile	Power User
<input type="checkbox"/> Receptionist	
<input checked="" type="checkbox"/> Enable Softphone	
<input type="checkbox"/> Enable one-X Portal Services	
<input type="checkbox"/> Enable one-X TeleCommuter	
<input type="checkbox"/> Enable Remote Worker	
<input type="checkbox"/> Enable Desktop/Tablet VoIP client	
<input type="checkbox"/> Enable Mobile VoIP Client	
<input type="checkbox"/> Enable MS Teams Client	
<input type="checkbox"/> Send Mobility Email	
<input type="checkbox"/> Web Collaboration	
<input type="checkbox"/> Exclude From Directory	
Device Type	Unknown IP handset
User Rights	
User Rights view	User data

Figure 28 – User Configuration – User Tab

To configure the restricted outbound call for a user by using specific **W** in the Short Code, first select **User** in the left Navigation Pane, then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is **473XXX4500**. Select the **Short Codes** tab in the Details pane.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **6N;**, this short code will be invoked when the user dials 6 followed by any number
- Set **Feature** to **Dial**. This is the action that the short code will perform
- Set **Telephone Number** to **W+N**. The value **N** represents the number dialed by the user. Note: Use the specific **W** in front of **+N** for restricting outbound calls for a user
- Set the **Line Group ID** to the **Outgoing Group 17** defined on the **Call Details** tab on the **SIP Line** in **Section 5.5.2**. This short code will use this line group when placing the outbound call
- Set the **Locale** to **Norway (Norwegian)**
- Default values may be used for all other parameters
- Click **OK** to submit the changes

The screenshot shows a web-based configuration interface for user 473XXX4500. The 'Short Codes' tab is active. A table lists existing short codes, but the 'New Short Code' section is highlighted with a red box. This section contains the following fields:

Code	6N;
Feature	Dial
Telephone Number	W+N
Line Group ID	17
Locale	Norway (Norwegian)
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

On the right side of the form, there are buttons for 'Add...', 'Remove', 'Edit...', 'OK', and 'Cancel'. The 'OK' button is highlighted with a red box.

**Figure 29 – User Configuration – Short Code tab**

One of the H.323 IP Deskphones at the enterprise site uses the Mobile Twinning feature. The following screen shows the **Mobility** tab for **User 473XXX4500**. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case **61613XXX5096**. Check **Mobile Call Control** to allow incoming calls from mobility extension to access FNE00 (Defined in **Section 5.8**). Other options can be set according to customer requirements.

The screenshot displays the configuration interface for User 473XXX4500, specifically the Mobility tab. The interface includes several sections and options:

- Simultaneous:** Coverage Delay (secs) is set to 0. MS Teams URI is empty.
- Internal Twinning:** Unchecked. Twinned Handset is set to <None>. Maximum Number of Calls is set to 1. Options for Twin Bridge Appearances, Twin Coverage Appearances, and Twin Line Appearances are unchecked.
- Mobility Features:** Checked. This section includes:
  - Mobile Twinning:** Checked. Twinned Mobile Number (including dial access code) is 61613XXX5096. Twinning Time Profile is <None>. Mobile Dial Delay (sec) is 2.
  - Fallback Twinning:** Unchecked.
- Mobile Answer Guard (sec):** 0.
- Hunt group calls eligible for mobile twinning:** Unchecked.
- Forwarded calls eligible for mobile twinning:** Unchecked.
- Twin When Logged Out:** Unchecked.
- one-X Mobile Client:** Unchecked.
- Mobile Call Control:** Checked.
- Mobile Callback:** Unchecked.

**Figure 30 – Mobility Configuration for User**

## 5.10. Incoming Call Route

An Incoming Call Route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by service provider. To create an incoming call route, select **Incoming Call Route** in the left Navigation Pane, then right-click in the center Group Pane and select **New** (not shown). On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capability** to **Any Voice**
- Set the **Line Group ID** to the **Incoming Group 17** defined on the **Call Details** tab on the **SIP Line** in **Section 5.5.2**
- Set the **Incoming Number** to the incoming DID number on which this route should match
- Default values can be used for all other fields

Configuration	Incoming Call Route	Standard
BOOTP (4) Operator (3) Solution User (6) Group(0) Short Code(46) Directory(0) Time Profile(0) Account Code(0) User Rights(9) Location(0) IPO-SE System (1) Line (2) Control Unit (9) Extension (17) User (4) Group (0) Short Code (49) Service (7) Incoming Call Route Directory (0)	Line Group ID   Incoming Number 17   +473XXX4505 17   +473XXX4504 17   +473XXX4501 17   +473XXX4500	17 +473XXX4500 Standard   Voice Recording   Destinations Bearer Capability   Any Voice Line Group ID   17 Incoming Number   +473XXX4500 Incoming Sub Address   Incoming CLI   Locale   Norway (Norwegian) Priority   1 - Low Tag   Hold Music Source   System Source Ring Tone Override   None

Figure 31 – Incoming Call Route Configuration

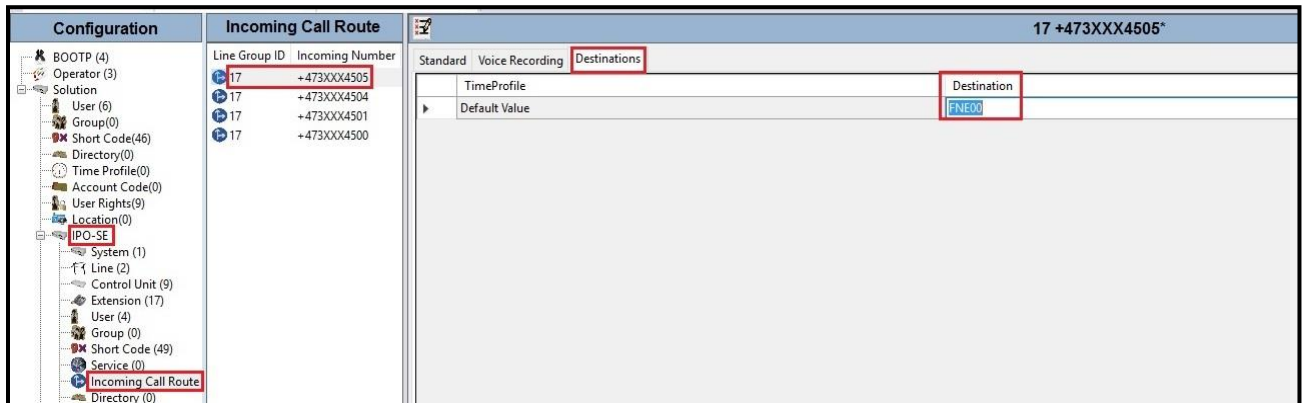
On the **Destination** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to **+473XXX4500** on line 17 are routed to **Destination 473XXX4500 473XXX4500** as below screenshot:

17 +473XXX4500*	
Standard   Voice Recording   Destinations	
TimeProfile	Destination
Default Value	473XXX4500 473XXX4500

Figure 32 – Incoming Call Route for Destination 473XXX4500

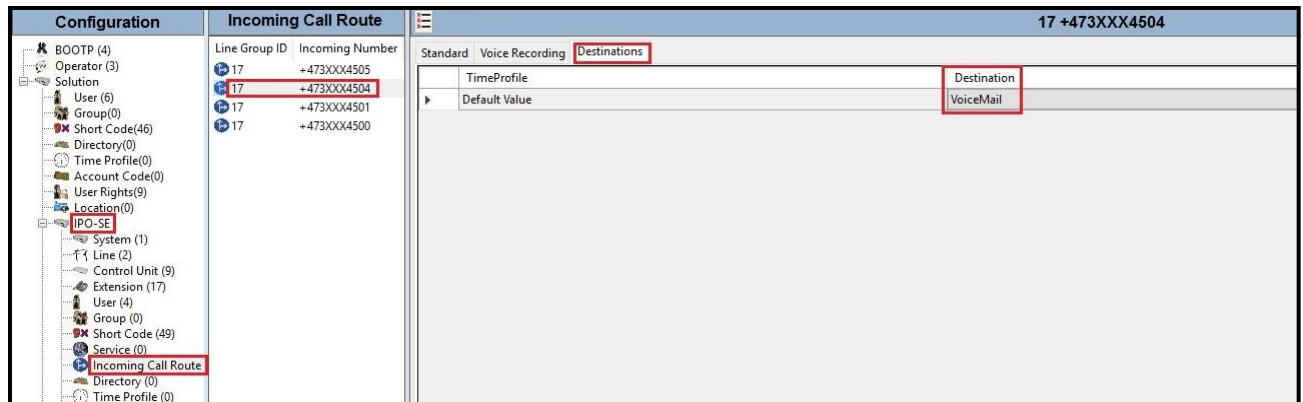


For Feature Name Extension Service testing purpose, the incoming calls to DID number **+473XXX4505** were configured to access **FNE00**. The **Destination** was appropriately defined as **FNE00** as below screenshot:



**Figure 33 – Incoming Call Route for Destination FNE**

For Voice Mail testing purpose, the incoming calls to DID number **+473XXX4504** were configured to access **VoiceMail**. The **Destination** was appropriately defined as **VoiceMail** as below screenshot:



**Figure 34 – Incoming Call Route for Destination VoiceMail**

## 5.11. Save Configuration

Navigate to **File** → **Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

## 6. Telia Norway SIP Trunk Configuration

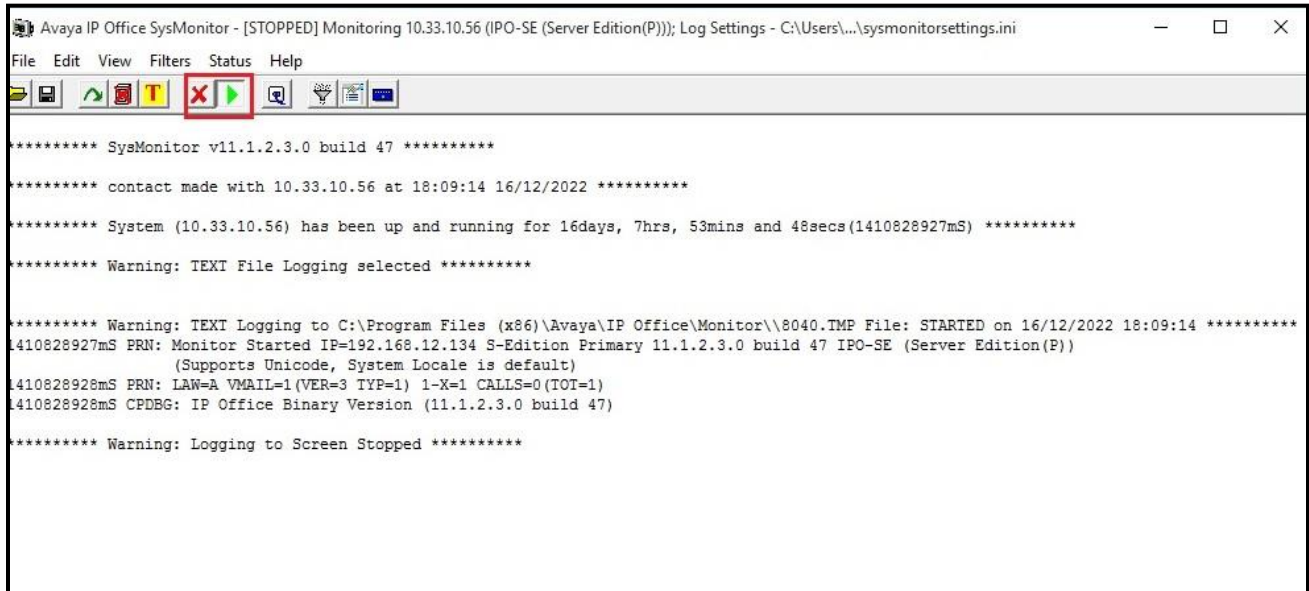
Telia Norway is responsible for the configuration of Telia Norway SIP Trunking Service. The customer must provide the IP address used to reach the Avaya IP Office Server Edition LAN2 port at the enterprise. Telia Norway will provide the customer necessary information to configure the SIP connection between Avaya IP Office Server Edition and Telia Norway. The provided information from Telia Norway includes:

- SIP Proxy IP address and port number used for signaling and media
- DID numbers
- Telia Norway root CA certificate
- Telia Norway SIP Trunk Specification

## 7. Verification Steps

The following steps may be used to verify the configuration:

- Use the Avaya IP Office Monitor application to monitor the active SIP call traces between the enterprise and Telia Norway. Launch the application from **Start → All apps → IP Office → Monitor** on the PC where Avaya IP Office Server Edition Manager was installed. Click start/ stop buttons to capture the SIP call traces.



**Figure 35 – SIP Trace Monitor**

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start → All apps → IP Office → System Status** on the PC where Avaya IP Office Server Edition Manager was installed. Select the SIP Line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** for each channel (The below screen shot showed 2 active calls at present time)

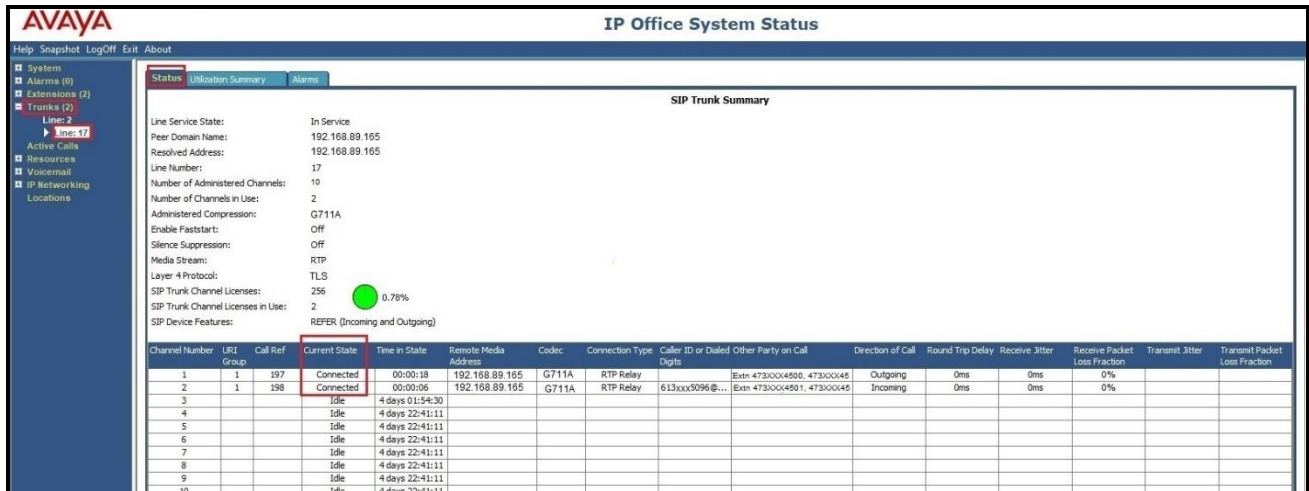
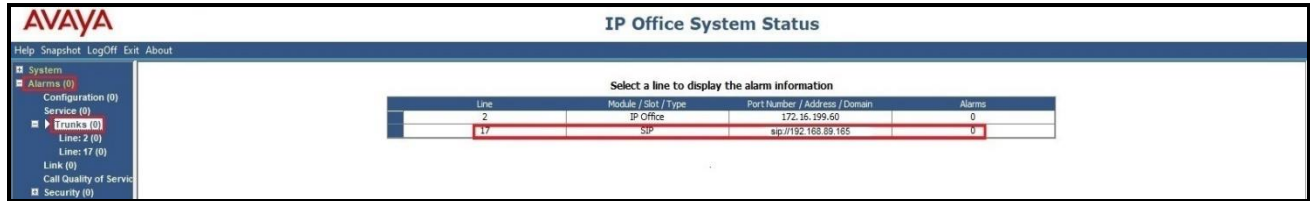


Figure 36 – SIP Trunk status

- Use the Avaya IP Office System Status application to verify that no alarms are active on the SIP line. Launch the application from **Start → All apps → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Select **Alarm → Trunks** to verify that no alarms are active on the SIP line



**Figure 37 – SIP Trunk alarm**

- Verify that a phone connected to the PSTN can successfully place a call to Avaya IP Office Server Edition with two-way audio
- Verify that a phone connected to Avaya IP Office Server Edition can successfully place a call to the PSTN with two-way audio
- Use a network sniffing tool e.g., Wireshark to monitor the SIP signaling between the enterprise and Telia Norway. The sniffer traces are captured at the LAN2 port interface of the Avaya IP Office Server Edition

## 8. Conclusion

Telia Norway passed compliance testing excepting the limitation in **Section 2.2**. These Application Notes describe the procedures required to configure the SIP connections between Avaya IP Office Server Edition and the Telia Norway system as shown in **Figure 1**.

## 9. Additional References

- [1] *Avaya IP Office Technical Bulletin 234 / General Availability (GA)- IP Office Release 11.1.2 Service Pack 2, Issue 1, 18<sup>th</sup> March 2022*
- [2] *Deploying IP Office Server Edition and Application Servers, Release 11.1 FP2, Issue 21, February 2022*
- [3] *Deploying Avaya IP Office Servers as Virtual Machines, Release 11.1 FP2, Issue 8, February 2022*
- [4] *IP Office Platform 11.1, Deploying an IP Office 500 V2/V2A in IP Office Basic Edition Mode, Issue 38e, Monday, February 28, 2022*
- [5] *Administering Avaya IP Office using Manager, Release 11.1.2, Issue 32, February 2022*

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

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