Yeastar

Swisscom Smart Business Connect Internet Trunk Configuration Guide

Effective on **P-Series PBX System**

From	Yeastar
Date	02-03-2026
Subject	How to configure Swisscom Smart Business Connect Internet Trunk registered trunk with Yeastar P-Series PBX System
Scope	Yeastar P-Series PBX homologation with Swisscom
PBX Firmware	Applicable firmware: ● P-Series Appliance Edition (PAE): Version 37.19.0.110 or later ● P-Series Software Edition (PSE): Version 83.20.0.23 or later ● P-Series Cloud Edition (PCE): Version 84.18.0.102 or later
Document Version	4.0
Status	Available
Valid from	Approval by Yeastar and Swisscom
Valid until	New version available
Document Name	Yeastar P-Series PBX with Swisscom Smart Business Connect Internet Trunk Configuration Guide

Contents

About Yeastar

Yeastar makes digital value easily accessible from ownership and adoption to daily usage and management by transforming how businesses connect and communicate.

Yeastar has established itself as a leading provider of UC solutions with a strong ecosystem, a global network of channel partners, and over 650,000 customers worldwide.

Committed to delivering the right technology to value-oriented businesses, Yeastar offers easy-first products and services for UC&C to enable them to win in the modern digital world. For more information about Yeastar or to become a Yeastar partner, please visit https://www.yeastar.com/.

Currently, Yeastar's main products are Yeastar's P-Series PBX and its supporting softphone Linkus and voice gateway.

|	
P-Series Appliance Edition (PAE)	Yeastar P-Series Hardware PBX including Yeastar P520/P550/P560/P570.
P-Series Software Edition (PSE)	Yeastar P-Series Software PBX can install on an on-premise Linux virtual machine or Cloud VPS.
P-Series Cloud Edition (PCE)	Yeastar P-Series Cloud PBX based on multi-tenant (multi-instance), provided Yeastar-hosted and self-hosted.
Linkus Client	Softphone with Web/Desktop/Mobile Client to work with P-Series PBX.
Yeastar Voice Gateway	Yeastar provides TA Gateway (FXS/FXO), TE Gateway (E1), TB Gateway (BRI) and TG Gateway (LTE). Among these, the TA gateway can be use to connect FAX machine for fax to email via Yeastar PBX.

Register the SIP Trunk

1. Create an Account with the Swisscom

To start with, you need to have a trunk account from Swisscom.

### 2. Add the SIP Trunk in P-Series PBX System

After you get the SIP trunk account, you need to add a SIP trunk in Yeastar P-Series PBX System. Go to **Extension and Trunk** > **Trunk**, click **Add**.
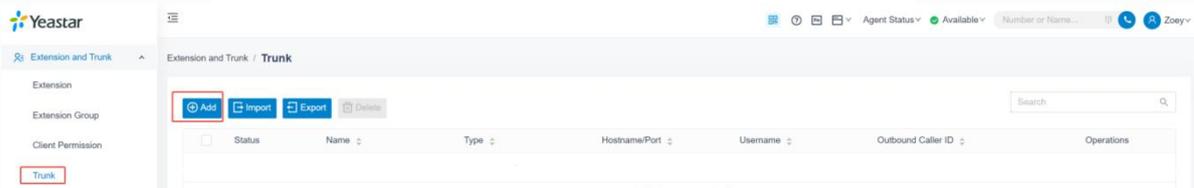
Then you need to configure the SIP Trunk registration parameter.

    Basic > Basic
- **Name:** Give this SIP trunk a name to help you identify it.
- **Select ITSP Template: Switzerland**
- **ITSP: Swisscom Smart Business Connect Internet Trunk**
- Make sure the trunk status is **"Enabled".**
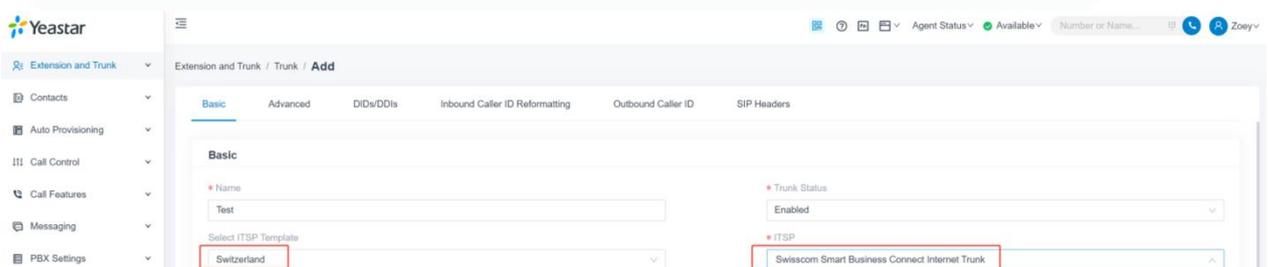
Basic > Detailed Configuration

All the parameters are embedded except the specific account registration information. You don't have to figure out Trunk Type, Transport, Hostname, Port, etc. Usually only the following account information needs to be filled in according to the content provided by Swisscom SBCon portal. (Please also check whether the other pre-filled information is consistent.)

Pre-filled information:
- **Trunk Type:** Register Trunk
- **Transport:** TLS
- **Hostname/IP:** strunkpub.join.swisscom.ch (Can be edited and modified according to the hostname provided by Swisscom SBCon portal, usually this is by default.)
- **Port:** 5061
- **Enable Outbound Proxy:** Disable

Specific account registration information needs to be filled by yourself:
- **Domain:** XXXXXX.join.swisscom.ch (Replace the six "X" with the customer's specific domain provided by Swisscom SBCon portal)
- **Username:** "SIP ID" provided by Swisscom SBCon portal. (Normally the number with +41XXXXXXXXX)
- **Password:** Your account password.
- **Authentication Name:** Fill in "SIP user" provided by Swisscom SBCon portal

Advanced > Codec Settings

In our certified templates, we will officially pre-fill a-law, G722 as we support and recommended codec. You can double check if your configuration is the same as the below example:
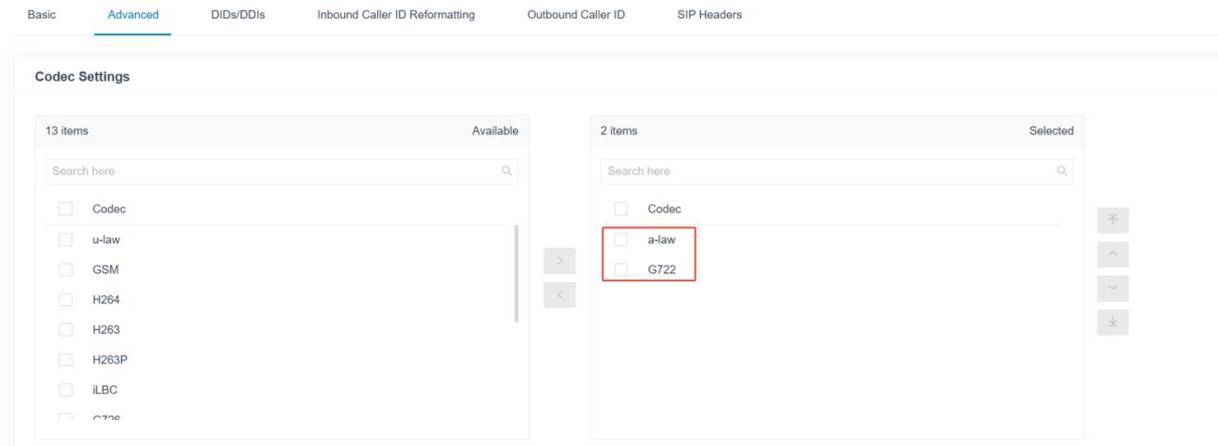
Advanced > VoIP Settings

In our certified templates, some parameters will be pre-filled to ensure compatibility, you can double-check if your configuration is the same as the below example:
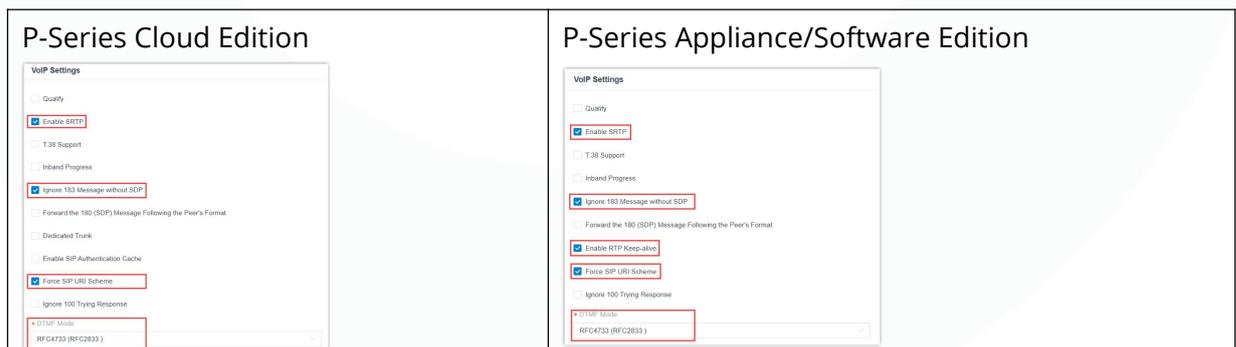
Enable: Enable SRTP

Enable: Ignore 183 Message without SDP

Enable: Enable RTP Keep-alive (This option is available and required only for P-Series Appliance and Software Edition)

Enable: Force SIP URI Scheme

DTMF Mode: RFC4733 (RFC2833)

P-Series Cloud Edition	P-Series Appliance/Software Edition

DIDs/DDIs

In P-Series Cloud PBX, our SBC servers are using DID to identify the PBXs. You need to add the available DIDs of this trunk, **otherwise, the call will fail**. (In the Software PBX and the hardware PBX you can omit this.) And please ensure that the DID number format entered here is consistent with the Trunk provider's number format.
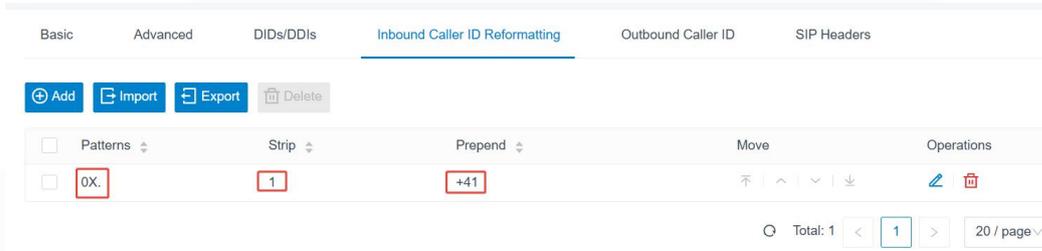
You can add a single DID or a range of DIDs.
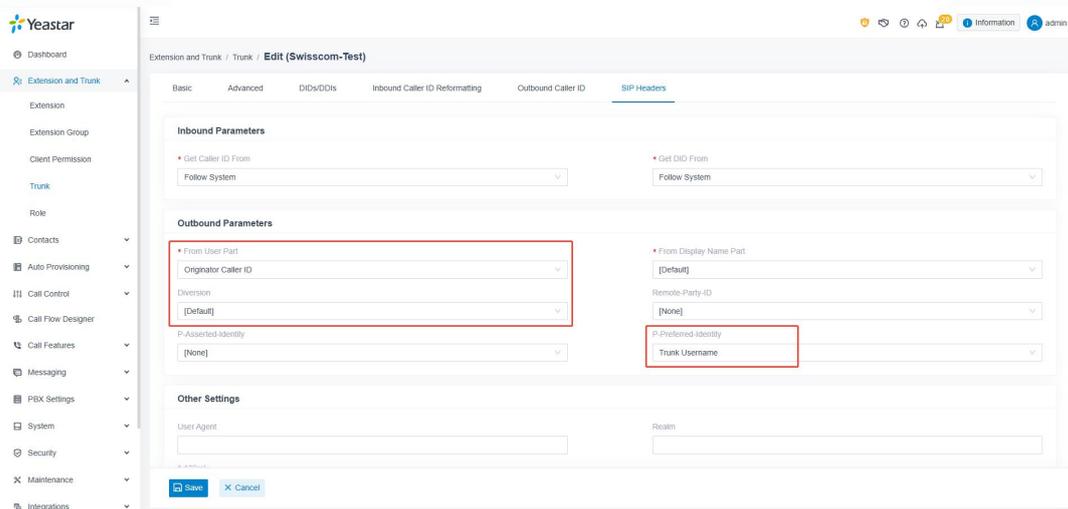
Inbound Caller ID Reformatting

In our certified templates, the parameters will be pre-filled to ensure compatibility, you can double check if your configuration is the same as shown below.

SIP Header > Outbound Parameters

In our certified templates, some parameters will pre-filled to ensure compatibility, you can double check if your configuration the same as the below example.
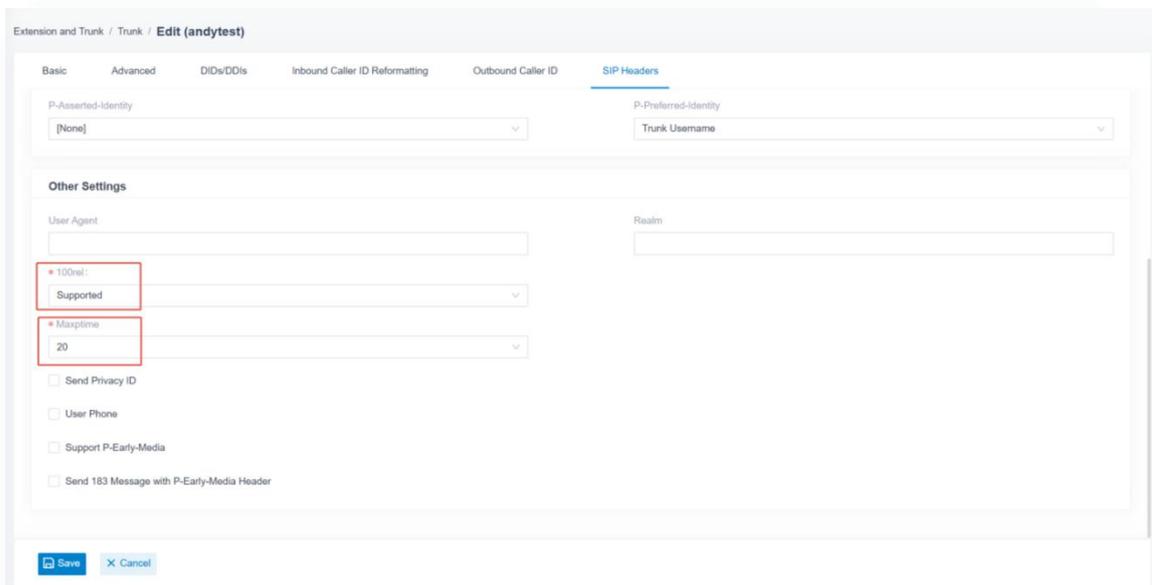
- **From User Part:** Originator Caller ID
- **Diversion:** [Default]
- **P-Preferred-Identity:** Trunk Username (This is the "SIP ID" you filled in Trunk > Basic > Detailed Configuration)

SIP Header > Other Settings

In our certified templates, some parameters will be pre-filled to ensure compatibility. You can double-check if your configuration is the same as the example below:

- **100rel:** Supported
- **Maxptime:** 20

After confirming the above information, you can click **Save** and **Apply** to complete the trunk configuration.

Note: The trunk status will show as ⊘ ('**Registered**') only after you have uploaded the TLS certificate to the PBX in Step 7.

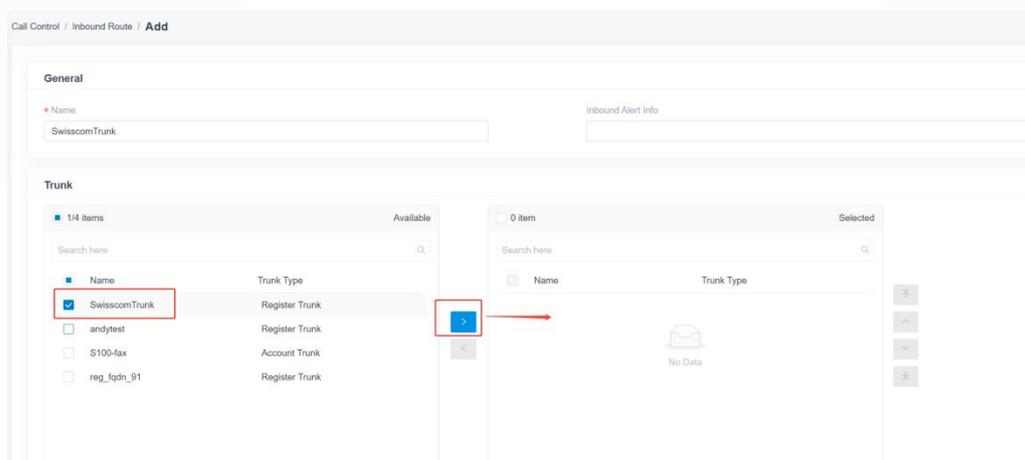## 3. Configure the Inbound Route with SIP Trunk

To specify how calls from the SIP trunk should be routed, you need to configure an inbound route for the SIP trunk.

Go to **Call Control** > **Inbound Route** and click **Add**.
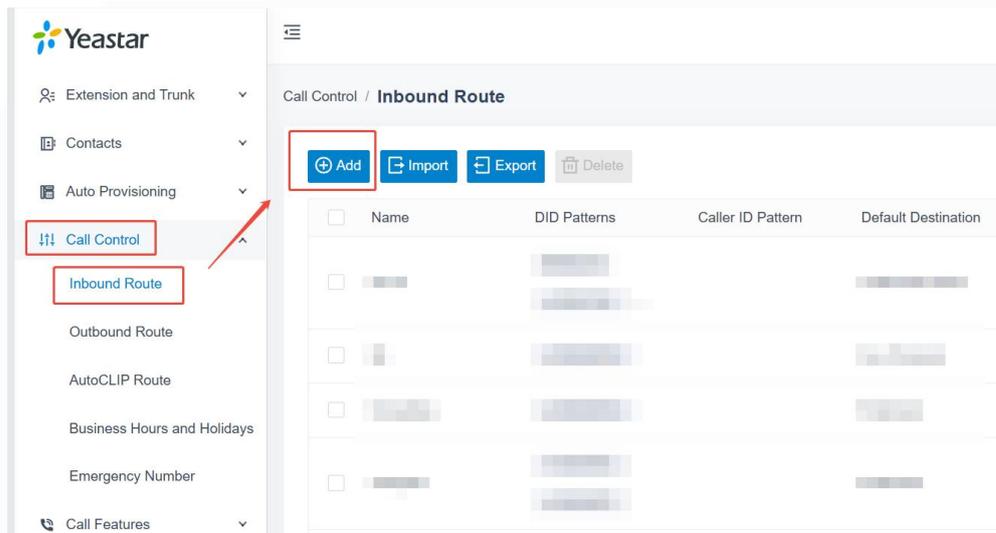
Then configure the necessary settings.

Select the trunk you created just now and move it on the right. And according to your own requirements to set the inbound routing conditions.

- **Name:** Give this inbound route a name to help you identify it.
- **Trunk:** Choose the Swisscom Smart Business Connect Internet SIP trunk you created.
- **Inbound Alert Info:** The Alert Info field is used to configure distinctive ring tones for incoming calls.
- **DID Pattern:** The DID pattern is used to match callee numbers for inbound call routing. In this example, only inbound calls with callee numbers matching the defined DID pattern will be routed to the specified extensions in a one-to-one correspondence.

  For more details on configuring different DID matching modes, see the guidance for P-Series Appliance Edition, Software Edition, and Cloud Edition.
- **Caller ID Pattern:** The pattern used to match caller ID. Only when the caller ID matches the pattern can users dials in through this route.
- **Default Destination:** Set the default destination or set with Time Condition.

After you configure all the conditions you want to, click **Save** and **Apply**. Then when customers call the DID number, the call will be routed to the destination configured on this inbound route.

4. Configure the Outbound Route with SIP Trunk

To make outbound calls via the newly created SIP trunk, you need to configure an outbound route for the trunk.

Go to **Call Control** > **Outbound Route** and click **Add**.
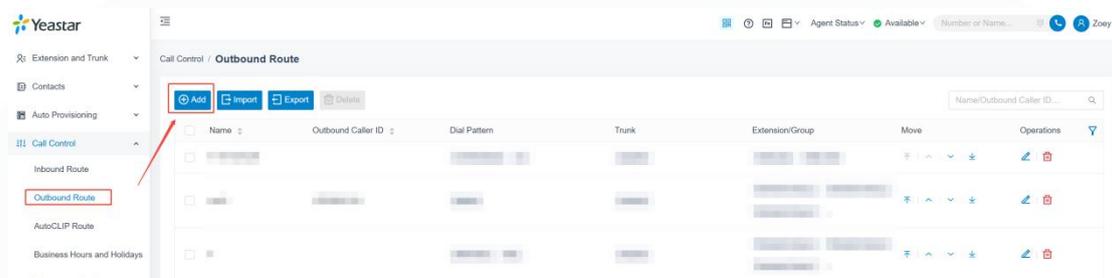
The system compares the number with the pattern that you have defined in your first route. If it matches, it will initiate the call using the selected trunks. If it does not, it will compare the number with the pattern you have defined in the second route and so on. The outbound route which is in a higher position will be matched first.

We suggest that you can follow the below example to set the Dial Pattern to meet the common outbound calling format requirement in Switzerland.

- **Name:** Give this outbound route a name to help you identify it.
- **Dial Pattern:** You can use a dial pattern to determine whether to use this trunk for outbound calls. For example, you can use different call prefixes before the number to distinguish between outbound routes.

  By default, it is necessary to add the following patterns **in order**. The system checks the dialed number against these patterns from top to bottom and applies the strip/prepend rule of the first pattern that matches, then sends the call.

  Note: The system processes emergency numbers with the highest priority, before evaluating these dial patterns. Please configure them in the Emergency Numbe section below.

Pattern	Strip	Prepend
00.	2	+
0.	1	+41
X.	Empty	Empty
+X.	Empty	Empty

- **Trunk:** Select the Swisscom Smart Business Connect Internet SIP trunk you created.

- **Outbound Route Password:** You can prompt users for a password before allowing calls to progress.

- **Extension/Extension Group:** Select the extensions or extension groups that are allowed to make calls through the outbound route.

- **Time condition:** You can select a time condition to specify when this outbound route can be used.


After you configure all the conditions you want, click **Save** and **Apply**.

You can now make outbound calls through the SIP trunk. If you have configured a special call prefix, you need to add a specific symbol or prefix number before making an outgoing call.

5. Emergency Number Settings

It is necessary to add **all Swiss emergency numbers** so that you can call them without any +41 prefix.

Note: Once the configuration is complete, emergency numbers must be tested.

Go to **Emergency Number** > **Emergency Number** and click **Add**.
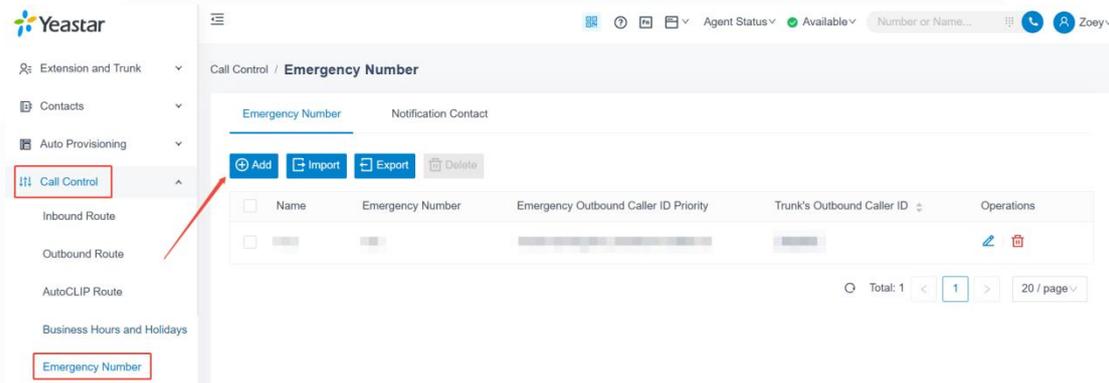

Configure the emergency numbers, for example for 112:

- **Name:** Give this emergency number a name to help you identify it.
- **Emergency Number:** Enter the emergency number 112.
- **Emergency Outbound Caller ID Priority:** Leave the Emergency Outbound Caller ID Priority field as the default setting. This setting is typically for enhanced emergency calling and will not affect basic emergency calling. For basic emergency calling, you don't need to set Emergency Outbound Caller ID for extensions and trunks.
- **Outbound Rules:** Select the trunk you created for emergency calls. Emergency calls have the highest priority. If the selected trunk is occupied, PBX will terminate the ongoing call, and place the emergency call.

6. ToS Settings

To ensure consistent quality for voice and video traffic, it is recommended to prioritize the traffic over other network traffic, and align the PBX's QoS settings with the QoS policy on the peer network.

Log in to PBX administrator portal, go to **PBX Settings** > **SIP Settings** > **QoS**, then set **ToS SIP** to **AF41** to prioritize SIP traffic.
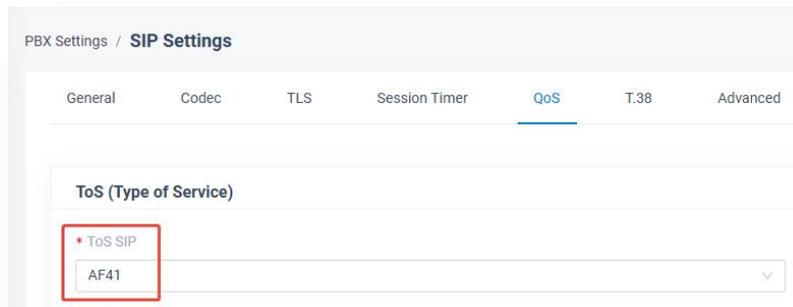
7. Upload the TLS certificates

This step is different for P-Series Appliance Edition, P-Series Software Edition and P-Series Cloud Edition.

P-Series Cloud Edition

You can skip this step because, for PCE, our mechanism does not verify the certificate. Instead, it verifies the IP through our proxy server. Each trunk is managed by the proxy server, and only the IPs of Swisscom can connect through TLS.

P-Series Appliance Edition & P-Series Software Edition

Yeastar P-Series Appliance Edition and Software Edition support the TLS protocol to secure SIP messaging. Before using the TLS protocol on the Swisscom Smart Business Connect Internet Trunk, Swisscom suggests uploading the TLS certificate. To successfully establish a TLS connection to Swisscom Smart Business Connect Internet trunk, the SwissSign Gold CA - G2 is required as a trusted certificate. Download: https://www.swisssign.com/support/ca-prod
The Gold G2 certificate in the **PEM** version is required.

Then Log in to PBX administrator portal. Go to **Security** > **Security Settings** > **Certificates**, click **Upload**.
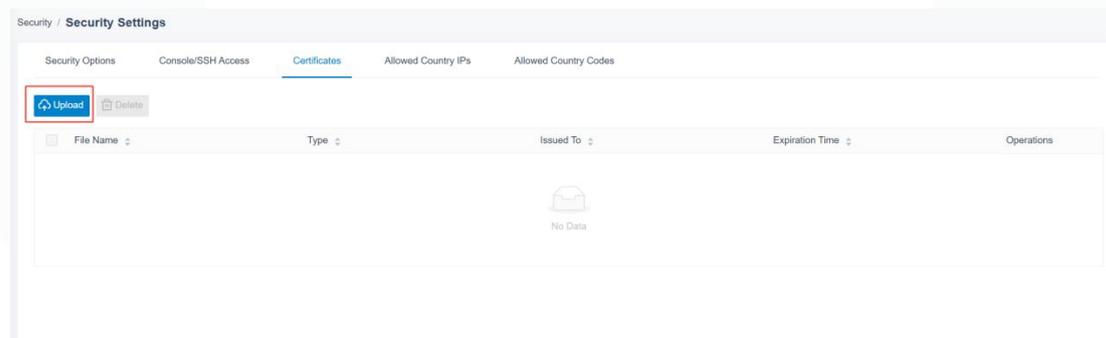
A window will pop up, requiring you to select the certificate type and upload the certificate.

After the certificate is uploaded successfully, it is displayed in the Certificates list. Next, go to **PBX Settings** > **SIP Settings** > **TLS** > **TLS Verify Server**. Enable the TLS option and the TLS Verify Server when the PBX is acting as a client, and then click **Save**.

8. Incompatible scenarios

Regarding the following special scenarios and user experience optimizations, we will continue to support them in future versions. If you require the implementation of these functions for regular use or special projects, please contact Yeastar. Thank you for your patience and trust.

- **Select DOD to call out with anonymous or normal (CLIR call-by-call)**
  Anonymous outgoing calls can only be configured for the entire trunk. If you want to use a special DID prefix pattern to control anonymity, we currently do not support it.

- **Press DTMF before the call is connected when making an outbound call**
  In our Yeastar Linkus Client, we cannot input DTMF before the call is connected when making an outbound call. For example, when navigating an IVR menu, the dial pad isn't accessible.

- **SIP 302**
  Currently not supported yet.

- **FAX transmissions limitation**
  Fax transmissions e.g. with Yeastar FXS Gateways need to be done through G.711a-law.

- **MOH situation**
  When the remote party initiates call hold signaling (SIP Re-INVITEs), Yeastar will not send the Re-INVITE. Due to different configurations, Yeastar will play the corresponding hold music according to the PBX configuration.

- **PANI**
  E112 Emergency Calling with PANI header is currently not implemented.

- **Billing with Special Arrangement**
  Billing with Special Arrangement: The billing will be done on the trunk main number instead of the user number.

SIP Trunk Configuration Guide

Support Contact

Yeastar
helpdesk@yeastar.com

Swisscom
Swisscom Support DE
Swisscom Support FR
Swisscom Support IT