

4.2 ENCRYPTED SIP Trunks ZRTP

Edit SIP Trunk

Name	<input type="text" value="new_pbx"/>
Failover Group	<input type="text" value="NONE"/>
Host	<input type="text" value="newpbx.hostname.com"/>
Outboundproxy	<input type="text"/>
Virtual Phone Number	<input type="text"/>
Username	<input type="text"/>
Password	<input type="text"/>
Register	<input type="checkbox"/>
Send keep alive	<input checked="" type="checkbox"/>
Port	<input type="text" value="5061"/>
Transport	<input type="text" value="TLS"/>
Secure	<input type="checkbox"/>
Insecure Call Policy	<input type="text"/>
Enable SIPS URIs	<input type="text"/>
Secure RTCP	<input type="text"/>
SDP Crypto Suite	<input type="text" value="DEFAULT"/>
Max Concurrent Calls	<input type="text" value="10"/>
Nat	<input type="text" value="NO"/>
Directmedia	<input checked="" type="checkbox"/>
Sendrpid	<input checked="" type="checkbox"/>
Audio Tones	<input type="text" value="No audio tones"/>
Dtmfmode	<input type="text" value="INFO"/>
Allow	<input type="text" value="amr:100"/>
Disallow	<input type="text" value="all"/>
Trusted	<input checked="" type="checkbox"/>

figure 1. "Edit Sip Trunk" form

In [figure 1. "Edit Sip Trunk" form](#) you can see an example configuration for creating a SECURE ZRTP SIP Trunk. The mandatory values are:

- **NAME:** a meaningful name for this trunk
- **HOST:** IP address/hostname of the SIP server provided by ITSP
- **PORT:** this is **5061** by RFC
- **TRANSPORT:** TLS
- **ENCRYPTION:** check it disabled

We do also suggest the following values to be set:

- **Audio Tones:** No audio tones
- **DTMFMODE:** choose your values considering the PBX on the other end of the Trunk. Usually we suggest to choose the value **INFO**
- **DIRECTMEDIA:** enabled (checked)
- **SENDRPID:** enabled (checked)
- **Allow:** amr:100
- **Disallow:** all
- **Trusted:** enabled (checked)

Other fields in the form depend by your network topology and by the features on the other end PBX.

When you are done with your changes, commit them by clicking on the **Update** icon.

4.2.1 Certificate Management

In order to validate a TLS peer for establishing an Encrypted SIP Trunk, you generally have to import the other party CA Root. This is important because the peer TLS certificate could not match PrivateServer actual Certificate Chain and thus the validation would fail.



PrivateServer comes with a bundle of the most known CA Roots certificates ready to be used. So this section is useful for minor certificate authorities and/or for self signed certificates.

Please read [2.3.3 Add certification authority](#) to understand how to import a new CA Root certificate.

[4.1 ENCRYPTED SIP Trunks](#)

[4.3 UNENCRYPTED SIP Trunks](#)