SIP Trunk Business

Technical Guide

Short summary of technical information

- The CPE ("OneAccess") used for the SIP Trunk connection is only used to provide the SIP Trunk service. If additional connectivity is required (e.g. Internet access), this must be ordered separately and is provided via separate lines and separate termination devices.
- The Customer’s IP-PBX must be connected directly to the Ethernet interface of the CPE without intermediate devices (e.g. Ethernet switch / firewall are not supported and are the responsibility of the customer. Automatic Failover is no longer possible, as the Ethernet switch or firewall do not show the status of private devices).
- The Tunnelling feature is not supported as POST’s SIP TRUNK solution uses its dedicated VOICE NET and is not connected to an open Internet access.
- Encryption is not supported on SIP TRUNK.

- The following IP address configuration is used as standard when a SIP trunk line is provided:
  - CPE (OneAccess) : 172.31.172.1
  - Customer PBX : 172.31.172.30
  - Netmask : 255.255.255.0
- An alternative user-specific IP configuration can be implemented at the customer's request. In this case, the IP configuration should be specified at the time of ordering or at least 5 days before installation.

- Standard SIP protocol according to RFC 3261, which is used for outgoing and incoming traffic
- Session transport options : UDP (TCP and TCP/TLS are not supported)
- UDP port number (destination port number for SIP INVITES) : 5060
- Voice codec : G.711 Alaw
- Fax protocol support : G.711 (Best effort)
- DTMF signaling mechanism : RFC2833 (Inband/SIP Info)
- No SDP information in initial SIP INVITE required
- SIP traffic from customer is only accepted if originates from a predefined IP address (i.e. 172.31.172.30 in the default configuration)

- No authentication between CPE and customer PBX required and no SIP Trunk registration required for each individual DID– SIP/RTP traffic must originate from predefined IP address (i.e. 172.31.172.30 for the default configuration) and use a phone number range assigned to the customer (i.e. root + DID)
- Digest Authentication is not supported
- Codec renegotiation during a call is not supported
- SDP port range : 10000 to 50000
- CLIP No Screening function supported (function is not supported by every operator national or international)
- No split technology (TDM and VOIP mixed)

Additional questions and support

- If you have further questions that are not described in this document, please open a request information ticket.

- to this e-mail address csc.telecom@post.lu with the following details:
  - Main number and company name involved
  - Explain or describe what information you need
  - Provide customer and integrator contact details
  - In case where traces have been made by you or your integrator, please include them

Thank you!

POST LUXEMBOURG