SIP TRUNK SPECIFICATIONS

Overview

Customers are able to connect their own PBX with the secure SIP trunk. The SIP trunk offers a redundant with two SIP registrations to facilitate failover situations. Besides a redundant registrations, it could be that the PBX is for some reason not reachable, malfunctioning or if all concurrent calls are used up, we offer a Trunk Failover feature. With this feature all incoming calls will automatically reroute to the 'failover' dialplan.

Features

- Limitation of concurrent calls
- Limit access to specific IP's or subnets
- Limit destinations that can be called by country or specific prefixes
- Limit amount of calls per hour, day and week to assist in fraud events
- One trunks means 2 SIP accounts to facilitate failover situations
- One trunk can receive multiple incoming numbers
- One trunk can use multiple numbers as Caller ID

Specifications and Considerations

Protocol	SIP/UDP	SIP over UDP, TCP is not supported.
Related Standards	RFC2833, RFC3261, RFC3264, RFC3325, RFC3326, RFC4566	Although we aim to be compliant with these standards, this might not be possible in all cases. For instance compatibility with certain non-compliant devices might be a reason to divert from the standards.
Supported Methods	ACK,BYE, INVITE, OPTIONS, REGISTER	Only these SIP Methods can be used, all others might be ignored.
Codecs	g711a, g711u	Codec g711a prefered. Others might be available on certain destinations but are not officially supported.
DTMF	RFC2833	We support only RFC 2833 for DTMF. SIP INFO or inband might work in certain cases but are not officially supported.
T. ₃ 8	Not supported	We do not support T.38 passthrough. Although it might work on certain destinations it is not officially supported.
Authentication	SIP DIGEST	SIP REGISTER with DIGEST authentication.
Call authorization	SIP DIGEST	SIP INVITE with DIGEST authorization.
Number Format	E164+	All SIP headers sent or received by Kwebbl, including Request URI, From and To, must use international E164+ format (example +3120202020).
Expiry Timer	Min SE	The value for the Session Expiry Timer is 1800 seconds.
Caller ID	P-Asserted- Identity	Besides the From header, the caller ID should always be set in the P-Asserted-Identity header. In some cases the Remote-Party-Id header might serve as a viable but unsupported alternative. Calls without correct Caller ID headers might be ignored.
Anonymous Caller ID	Privacy	Calling out anonymously can be done using the Privacy header, setting the header to "Privacy: id" will hide the caller id. The From and P-Asserted-Identity headers should always be set to a valid number.
Keepalive	OPTIONS	We might periodically send an OPTIONS packet to check if the trunk is alive. An OPTIONS packet should always be replied with a 200 OK. We will also reply 200 OK to received OPTIONS packets.
Failover	Dual Trunk	You will always receive 2 trunks. For outbound calls it is your responsibility to choose when to failover from one trunk to the other. We advise to use priority routing, first trying the primary trunk and then the secondary. We will automatically failover incoming calls if one of the trunks becomes unavailable.