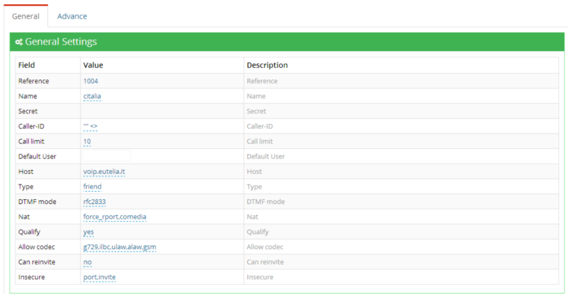
Edit a Trunk

The information about a Trunk can be viewed by clicking on Edit on the table row which contains the Trunk of interest.

To edit one parameter, simply click on its Value.

General Settings



Name: Trunk name

Secret: Trunk Secret

Caller-ID: defines the identifier, when there are no other information available

Call limit: number of simultaneous calls through this user/peer

Default User: your SIP username

Host: IP address

Type:

- User: used to authenticate incoming calls
- Peer: for outgoing calls
- Friend: covers both characteristics of the above

DTMF Mode: how DTMF (Dual-Tone Multi-Frequency) are sent:

- RFC2833: the default mode, the DTMF are sent with RTP but outside the audio stream.
- INBAND: The DTMF is sent in audio stream of the current conversation, becoming audible from the speakers. Requires a high CPU load.
- INFO: Although this method is very reliable, it is not supported by all PBX devices and many SIP Trunk.

Nat: this parameter specifies that the device is behind a NAT (Network Address Translator)

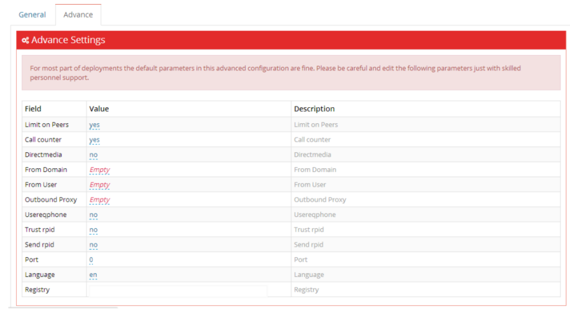
Qualify: defines the identifier, when there are no other information available

Allow Codec: with this parameter you can specify the codecs enabled for SIP; some of them are pre-selected by default

Can reinvite: thanks to this parameter two devices can establish directly the SIP RTP connection (Real Time Protocol). The result is to minimize the use of resources needed to establish the full-duplex communication.

Insecure: Specifies how to handle connections with peers.

Advanced Settings



Limit on Peers: yes|no, if set to yes use only the peer call counter for both incoming and outgoing calls

Call Counter: If enabled, this parameter allows Asterisk to provide useful information about the status of SIP devices.

From Domain: it sets default From: domain in SIP messages when acting as a SIP ua (client)

From User: specify user to put in “from” instead of $CALLERID(number) (overrides the callerid) when placing calls _to_ peer (another SIP proxy). Valid only for type=peer

Outbound Proxy: IP_address or DNS SRV name (excluding the _sip._udp prefix) : SRV name, hostname, or IP address of the outbound SIP Proxy. Valid only in [general] section and type=peer.

Usereqphone: yes|no, it indicates whether to add a “user=phone” to the URI. Default no.

Trust rpid: yes|no, if Remote-Party-ID SIP header should be trusted. Default no.

Send rpid: yes|no, if a Remote-Party-ID SIP header should be sent. Default no.

Port: Default SIP port of peer.

Language: A language code defined in indications.conf - defines language for prompts

Registry: callerID:secret@host/callerID

> **XCALLY - Twilio**
>
> Here you can find the XCALLY - Twilio guide: http://www.xcally.com/xcally-twilio/