

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

Field	Value	Description
Reference	1004	Reference
Name	cfata	Name
Secret		Secret
Caller-ID	** <>	Caller-ID
Call limit	10	Call limit
Default User		Default User
Host	voip.euclid.it	Host
Type	friend	Type
DTMF mode	rfc2833	DTMF mode
Nat	force_port_consmedia	Nat
Qualify	yes	Qualify
Allow codec	g729,libulaw,alaw,gsm	Allow codec
Can-reinvite	no	Can-reinvite
Insecure	port,mute	Insecure

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

Field	Value	Description
Limit on Peers	yes	Limit on Peers
Call counter	yes	Call counter
Directmedia	no	Directmedia
From Domain	Empty	From Domain
From User	Empty	From User
Outbound Proxy	Empty	Outbound Proxy
Usereqphone	no	Usereqphone
Trust rpaid	no	Trust rpaid
Send rpaid	no	Send rpaid
Port	0	Port
Language	en	Language
Registry		Registry

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

Send rpaid: 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/>