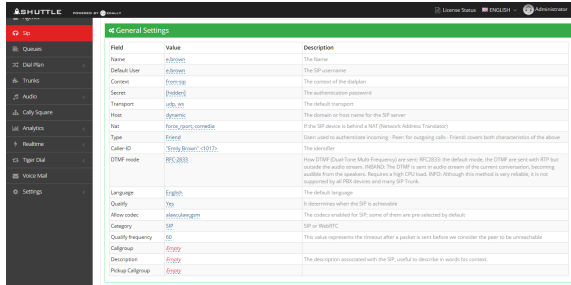


# Edit the SIP

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

To edit one parameter, simply click on its Value. You can edit **General** and **Advanced** Settings.

## General Settings



Field	Value	Description
Name	friend	The Name
Default User	friend	The SIP context
Context	friend	The context of the dialplan
Server	[friend]	The authentication password
Transport	udp, tcp	The default transport
Host	[friend]	The options to host name for the SIP server
Nat	force_rport, comedia	If the SIP device is behind a NAT (Network Address Translator) then enable nat support for the SIP device
Type	friend	When used in combination with nat, this covers both characteristics of the above
Caller ID	[SIPDevice]<1025>	The number
DTMF mode	RFC2833	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.
Language	English	The default language
Qualify	Yes	To determine when the SIP is achievable
Allow codec	ulaw, gsm	The codecs enabled for SIP; some of them are pre-selected by default
Category	SIP	SIP or WebRTC
Qualify frequency	50	The value represents the timeout after a packet is sent before we consider the peer to be unreachable
Callgroup	[group]	The SIP can be associated to one or several callgroups. The Callgroup settings are useful to letting someone else answer a call. You have to insert in this field the number of the callgroup. i.e. 1
Description	[group]	The description associated with the SIP, useful to describe in words his context.
Pickup Callgroup	[group]	The SIP can pickup the incoming calls of the Callgroups which are defined here, by calling *8 on the phone. You have to insert in this field the numbers relative to the Pickup Callgroups that you want to set for this SIP. i.e. 1-5,9

**Name:** the name of the SIP Context the context of the dialplan

**Secret:** authentication password of the SIP

**Transport:** set the default transports. The order determines the primary default transport.

**Host:** is the domain or host name for the SIP server.

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

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

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

**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.

**Language:** default language used by any Playback()/Background().

**Qualify:** to determine when the SIP is achievable

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

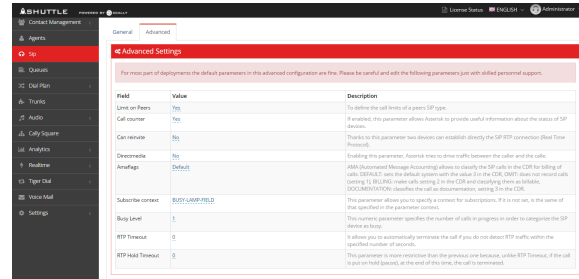
**Category:** WebRTC or SIP Qualify freq. It represents the timeout after a packet is sent before we consider the peer to be unreachable.

**Callgroup:** The SIP can be associated to one or several callgroups. The Callgroup settings are useful to letting someone else answer a call. You have to insert in this field the number of the callgroup. i.e. 1

**Pickup Callgroup:** The SIP can pickup the incoming calls of the Callgroups which are defined here, by calling \*8 on the phone. You have to insert in this field the numbers relative to the Pickup Callgroups that you want to set for this SIP. i.e. 1-5,9

**Description:** It's the description associated with the SIP, useful to describe in words his context.

## Advanced Settings



Field	Value	Description
Limit on Peers	Yes	To define the call limits of a "peers" SIP type.
Call counter	Yes	If enabled, this parameter allows Asterisk to provide useful information about the status of SIP devices.
Can Reinstate	No	On this parameter two devices can establish directly the SIP RTP connection (Real Time Protocol).
Direct Media	No	Enabling this parameter, Asterisk tries to drive traffic between the caller and the callee.
Amflags	Default	AMA (Automated Message Accounting) allows to classify the SIP calls in the CDR for billing of calls. Default: sets the default system with the value '3' in the CDR. Billing: make calls setting '2' in the CDR and classifying them as billable. Documentation: classifies the call as documentation, setting '3' in the CDR.
Subscribe context	BUSY_LEVEL	This parameter allows you to specify a context for subscriptions. If it is not set, is the same of that specified in the parameter "context".
Busy Level	1	This numeric parameter specifies the number of calls in progress in order to categorize the SIP device as busy.
RTP Timeout	0	It allows you to automatically terminate the call if you don't detect RTP traffic within the specified number of seconds.
RTP Hold Timeout	0	This parameter is more restrictive than the previous one because, unlike RTP Timeout, if the call is put on hold (pause), at the end of this time, the call is terminated.

**Limit on Peers:** to define the call limits of a "peers" SIP type. This can improve the status notification if you are using a "friend" SIP type for incoming calls. It also allows to evaluate incoming and outgoing calls; if it is set to yes value, Asterisk uses a counter for both incoming calls and outgoing.

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

**Can Reinstate:** 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.

**Direct Media:** enabling this parameter, Asterisk tries to drive traffic between the caller and the callee. Not all devices support this feature.

**Amflags:** AMA (Automated Message Accounting) allows to classify the SIP calls in the CDR for billing of calls: Default: sets the default system with the value '3' in the CDR. Billing: make calls setting '2' in the CDR and classifying them as billable. Documentation: classifies the call as documentation, setting '3' in the CDR.

**Subscribe context:** this parameter allows you to specify a context for subscriptions. If it is not set, is the same of that specified in the parameter "context".

**Busy Level:** this numeric parameter specifies the number of calls in progress in order to categorize the SIP device as busy.

**RTP Timeout:** it allows you to automatically terminate the call if you don't detect RTP traffic within the specified number of seconds.

**RTP Hold Timeout:** this parameter is more restrictive than the previous one because, unlike RTP Timeout, if the call is put on hold (pause), at the end of this time, the call is terminated.