

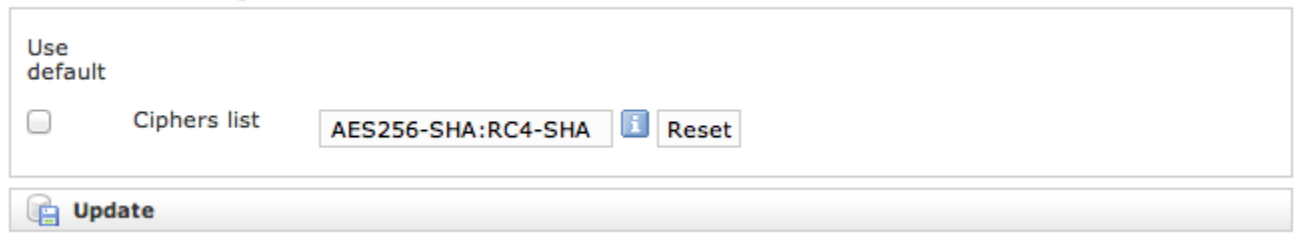
PSAM 2.4 Asterisk advanced configurations

Some advanced configuration settings about the PrivateServer behavior.

2.4.1 SIP/TLS

SIP/TLS is about configuring the encrypted communication channel among PrivateServer and its clients. The configuration form is reachable by the **SIP/TLS** main menu entry.

SIP/TLS Configuration



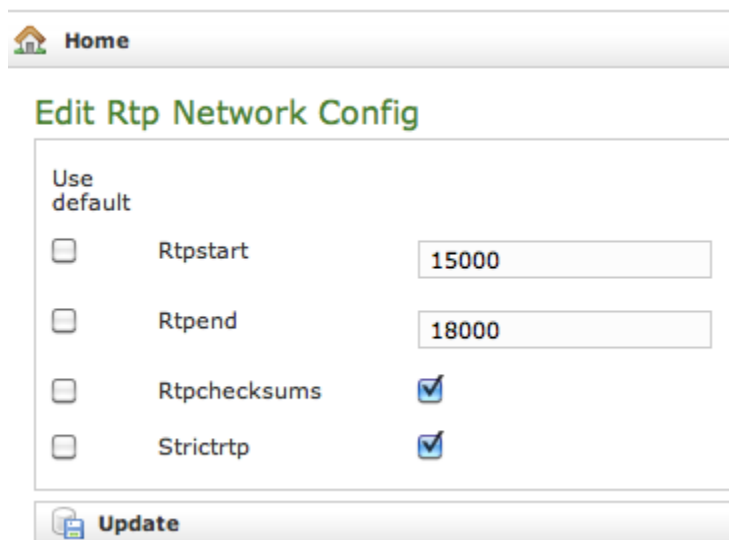
The SIP/TLS Configuration form contains a 'Use default' checkbox. Below it is a 'Ciphers list' section with a text input field containing 'AES256-SHA:RC4-SHA', an information icon, and a 'Reset' button. At the bottom of the form is an 'Update' button with a save icon.

figure 1. Cipher list configuration form

From the [figure 1. Cipher list configuration form](#) you can set up the cypher list of the PrivateServer. This is the list of accepted cipher suite, using OpenSSL format. Check at http://www.openssl.org/docs/apps/ciphers.html#CIPHER_LIST_FORMAT. Usually you can leave the default values.

2.4.2 RTP

"The **Real-time Transport Protocol (RTP)** defines a standardized packet format for delivering audio and video over IP networks" (quote from Wikipedia).



The 'Edit Rtp Network Config' form has a 'Use default' checkbox. Below it are four configuration items: 'Rtpstart' with a text input field set to '15000', 'Rtpend' with a text input field set to '18000', 'Rtpchecksums' with a checked checkbox, and 'Strictrtp' with a checked checkbox. At the bottom is an 'Update' button with a save icon.

figure 2. RTP configuration form

In this form that you get by the **RTP** main menu entry, you can set up the voice transport features. **Rtpstart** and **Rtpend** are the number of RTP ports available for the calls.

 Each call uses 4 ports, thus you can do your math on the RTP number necessary in your configuration multiplying the number of foreseen concurrent calls for 4.

In the example shown in [figure 2. RTP configuration form](#) you see:

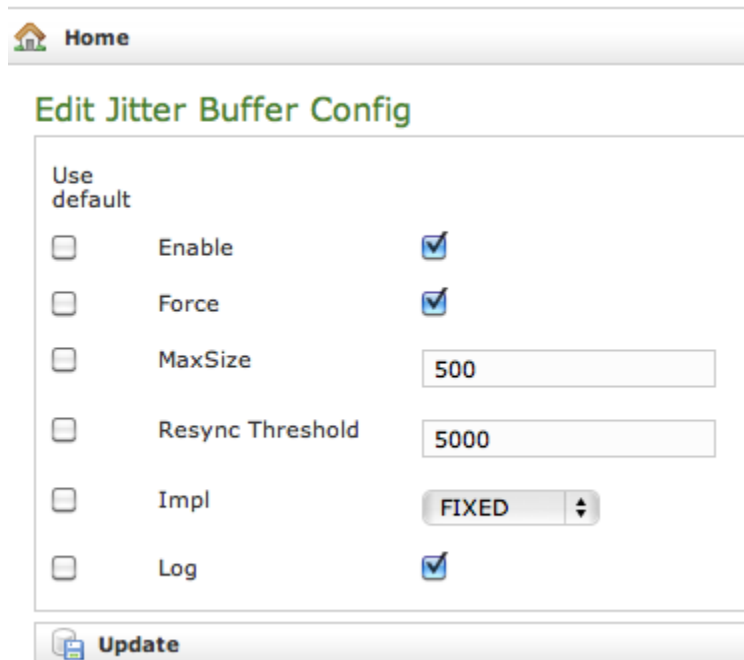
$18000 - 15000 = 3000$ ports available. This means $3000/4 = 750$ concurrent calls threshold.

The **Rtpchecksums** enable the application checksum over UDP encrypted voice transmission. This is an error detection commodity, which adds 16 bits per packets payload.

Strict RTP Enables the strict RTP protection. This will drop RTP packets that do not come from the source of the RTP stream. This option is disabled by default.

2.4.3 Jitter Buffer

Jitter is the undesired deviation from true periodicity of an assumed periodic voice streaming. The consequences of jitter, often called *jittering*, are a voice communication with holes in it or stirring metal voice effect. Mostly on a 3/4G network (and in general in a mobile network environment), the jitter is a sensible problem to face. To avoid jitter issues a **jitter buffer** is implemented in PrivateServer.



Home

Edit Jitter Buffer Config

Use default

☒	Enable	☒
☒	Force	☒
☐	MaxSize	500
☐	Resync Threshold	5000
☐	Impl	FIXED
☐	Log	☒

Update

figure 3. Jitter Buffer configuration form

The **jitter buffer** is an elastic buffer in which the signal is temporarily stored and then retransmitted at a rate based on the average rate of the incoming signal. Checking the **Enable** checkbox you allow PrivateServer to perform such jitter buffering in general, when needed.

The **Force** option obliges PrivateServer to create the jitter buffer over any new communication channel.

MaxSize is the max length of the jitterbuffer in milliseconds.

Resync Threshold is the buffer synchronization threshold. It's useful to improve the quality of the voice, with big jumps in/broken timestamps. Defaults to 1000.

Impl is the Jitterbuffer implementation, used on the receiving side of a SIP channel. Two implementations are currently available:

- FIXED (with size always equals to jbmmaxsize)
- ADAPTIVE (with variable size)

Defaults to FIXED.

Log enables jitterbuffer frame logging. Defaults to "no".

The following services need to be restarted to apply your changes: asterisk.

 [Home](#)

Edit Jitter Buffer Config

 Jitter Buffer Config updated

figure 4. The Management console asks to restart the service

To apply your changes just press the **Update** button and the management interface will ask you to restart the asterisk service in order to apply your new configuration, as shown in

2.4.4 Obfuscation

The **Obfuscation** is an internal VoIP communication stealth mode. It is useful to avoid QoS (Quality of Service) checks on VoIP as it masks the data.

 This practice is legal if you are not fooling your mobile provider or cheating your network administrator.

 [Home](#)

Edit Obfuscation Config

Mode

Key

 [Update](#)  [Update Accounts](#)

figure 5. Edit the obfuscation parameters

The configuration is quite simple. **Mode** enables/disables the obfuscation mode.

Key is a shared numeric key to be reported on the clients configuration as well.

 To avoid calls problems such as abruptly interrupted calls you make sure the obfuscation mode and key are equally set up on the server and the clients.

2.4.5 NAT Configuration

If you are using the appliance in an internal network then it's most possible that you need to configure the **NAT** option. NAT stands for **Network Address Translation** and it's commonly used to let services on a private IP address to be reachable by a public IP address.

NAT Configuration

Use default

☐

NAT

NO

☒

External media address

☒

External SIP address

☐

External port

143

☒

Keep-alive Frequency

Update

figure 6. NAT configuration form

Apart from your router/firewall configuration (please check [PrivateGSM installation pre-requisites](#)) and your network design/topology, from the PrivateServer point of view the only known thing is that the appliance is configured on a private IP address but the requests of the encrypted voice service are made to an external and public IP address. To avoid wrong replies the PrivateServer must know of this setup and be configured accordingly. Thus if you fall in the described scenario access to the "NAT Configuration" form (showed in [figure 6. NAT configuration form](#)) using the "NAT" link under "Server Configuration".

By default this option is disabled, so to enable it you first need to select "YES" in "NAT" option. If you have enabled the NAT then it's mandatory to configure the remaining options as well.



NEW FEATURE

The "Keep-alive Frequency" is part of a new feature that is not directly connected to the NAT setup. To better understand what a keep-alive is, please refer to [PSOM 1.0 Groups](#).

External media address

This is the **public IP** address used for the **RTP** delivery. It means that this is the **secured voice IP** you want to use.



Possible Misconfiguration

Unless you need to specify for some reason a specific IP address for RTP, you'd better leave this field empty and let Asterisk do the job for you!

External SIP address

This is the **public IP** address used for the **SIP** delivery. It means that this is the IP you want to use for **SIP signalling**.

External port

If you want to perform a **PAT** (Port Address Translation) in addition to the NAT, then please use this option to explain to the appliance which **port number** is used on the **external** interface for providing the **encrypted SIP service**.

2.4.5.1 Keep-alive Frequency

If you are using the keep-alive option (please refer to [PSOM 1.0 Groups](#)) then you may find this option handy. You can define here how many seconds should pass between each keep-alive request sent by the server to each client configured with the keep-alive option.



Please keep in mind that the **default** keep-alive **timeout** is **60 seconds** and thus it can lead to a quick battery drain since the radio system on the mobile device could never be idled.

If any mobile user has been configured with the keep-alive option on, then we **strongly suggest** you to set the **keep-alive Frequency** to **180 seconds** (i.e. 3 minutes) at least in order to save battery life.



This option is the same as the **qualifyfreq** one in the standard **Asterisk** configuration.

[PSAM 2.3 Certificates configuration](#)

[PSAM 2.5 Clock Configuration](#)