NAT Configuration FreePBX 12

NAT issues

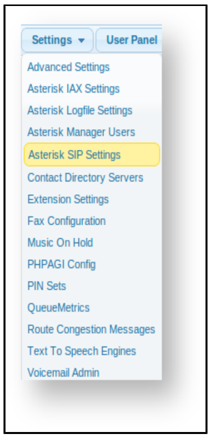
Some of the biggest problems that plague people such as "one way audio" or "Calls dropping after XX Seconds" are caused by NAT not being correctly setup.

Make sure you have a resolvable address on the Internet.

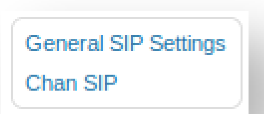
If you don't want to pay a few bucks to get a static IP address, and are served by an ISP that periodically changes your IP address, then get an account with a dynamic DNS service such as DynDNS . Your router may already have built-in support for one or more of these services, if so, use one that your router supports and then configure your router to automatically update your dynamic address when your ISP changes your IP address. Failing that, you can set up an updater program such as inadyn, there are instructions for doing that at this blog page

Adding NAT information in FreePBX

All of your settings will be under Settings > Asterisk SIP settings



Next Click Chan SIP in the right menu



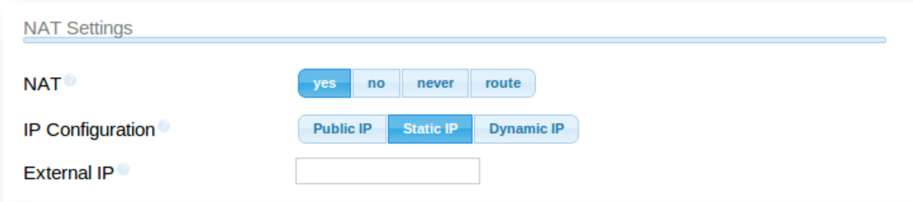
VERSION SPECIFIC

This right menu is specific to FreePBX 12. In 2.11 all settings are on the main page

Set NAT as yes

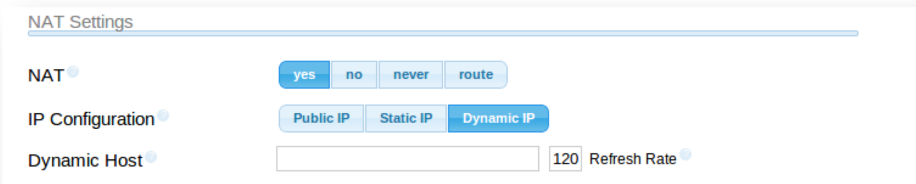
Static IP from your ISP

Select "Static IP" and enter your external IP



Dynamic IP Updated through dynamic IP service

Select "Dynamic IP" and put the Full host name in such as "foo.dyndns.net"



REMEMBER

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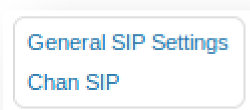
Whenever you make a change in the UI you need to "submit" the changes then click "APPLY" at the top

After clicking "submit changes" and the Red Apply click "General SIP Settings" on the right menu

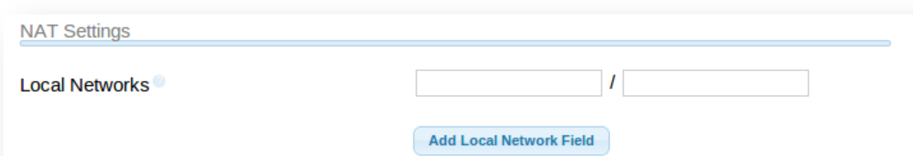


Local Networks

Under "NAT" you will see a box for "Local Networks"

In these boxes you will put your LAN information with the IP in the first box and the SUBNET in the second box

If your IP is 192.168.0.254 you would put 192.168.0.0 / 255.255.255.0



Click "Submit changes" And the red "APPLY" button.

RTP Port Range

Open the SIP and RTP ports to your Asterisk server

You must make sure that you open the correct UDP ports in your router's firewall and pointed at your Asterisk server. For SIP protocol, open UDP (NOT TCP) port 5060 (SIP) AND ports 10000-20000 (RTP, which must also be defined in /etc/asterisk/rtp.conf, see below). All these ports are UDP, opening the TCP ports will NOT help anything and may expose your system needlessly. While you are in your firewall configuration, you may as well also open UDP port 4569 (IAX), since sooner or later you'll probably want to accept IAX connections.

You can see the actual range under the "General SIP Settings" page.

If the port values are any different, change them. These MUST match what you opened in your firewall

Warning

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You may need to set this to start with 10001, as port 10000, conflicts with usage in Webmin. This only matters if you have installed Webmin

Some people feel the need to open fewer than 10,000 ports. I don't recommend this because six months from now when you start having audio problems you may not remember that you opened fewer than the recommended number of ports, and may spend hours troubleshooting the issue. But if you are simply obsessive about open ports, remember that each open SIP connection may require as many as FOUR concurrent ports, so don't cut it down to some ridiculously small number. For the non-paranoid, I suggest sticking with the recommendations above (and remember, if a hacker is looking at ports on your system, he's going to scan ALL of them, so having fewer UDP ports open really doesn't make you any more secure).