Asterisk SIP Settings User Guide

Overview

It was necessary in the past to hand edit files like "/etc/asterisk/sip_nat.conf" as part of the initial installation of any Asterisk based deployment. With the introduction of the Asterisk SIP Settings module, most SIP settings are made available in the GUI. This includes the all important NAT, External IP, Local Network, Enabled Codecs and Codec order.

Logging In

- From the top menu click **Settings**
- From the drop down click **Asterisk Sip Settings**

Settings

Allow Anonymous inbound SIP Calls

Allowing Inbound Anonymous SIP calls means that you will allow any call coming in from an unknown IP source to be directed to the 'from-pstn' side of your dialplan. This is where inbound calls come in. Although FreePBX severely restricts access to the internal dialplan, allowing Anonymous SIP calls does introduce additional security risks. If you allow SIP URI dialling to your PBX or use services like ENUM, you will be required to set this to Yes for Inbound traffic to work. This is NOT an Asterisk sip.conf setting, it is used in the dialplan in conjunction with the Default Context. If that context is changed to something custom, this setting may be rendered useless as well as if 'Allow SIP Guests' is set to no.

External Address

This should display your externally public facing IP address. You can select "Detect Network Settings" to have the PBX detect its External and Local networks, adjust accordingly if neccessary.

Local Networks

Local network settings in the form of "ip/mask" such as, "192.168.1.0/255.255.255.0." For networks with more than one LAN subnet, such as VPN network, us the "Add Local Network" button to add more fields. Blank fields will be removed upon submitting.

RTP Settings

RTP Ranges

The start and end ports for UDP RTP traffic. Default 10000-20000. You should have at least 4 ports per potential call.

RTP Checksums

Weather or not to enable UDP checksums for RDP traffic

Strict RTP

This will drop RTP packets that do not come from the source of the RTP stream. It is unusual to turn this off

Codecs

Check the desired codecs and drag to reorder. All others will be disabled unless explicitly enabled in a device or trunk configuration. Note that some codecs, such as g729, require commercial licensing.

Codecs

> **ℹ Helpful Information**
>
> This is the default Codec setting for new Trunks and Extensions.

- ☑ ulaw
- ☑ alaw
- ☑ gsm
- ☑ g726
- ☐ g723
- ☐ speex
- ☐ g722

Save

Click Submit to save