

# Asterisk SIP Trunk Settings – Vestalink

[Vestalink](#) is a new SIP trunk provider that has sprung up as a replacement for Google Voice trunking within Asterisk servers. They offer a very attractive pricing plan with 2000 mins/month going for \$39.99 per year, and unlimited plans at \$49.99 per year! This provides a single DID along with two SIP channels for the possibility of holding two simultaneous calls from one account (*call waiting*). The folks over to [Nerd Vittles](#) have written up a great guide to getting the service working with FreePBX, however I was hard pressed to find any documentation on how to get this working under a standard Asterisk server using only the CLI. Applying the Nerd Vittles FreePBX configuration to Asterisk CLI did not work out of the box. After some messing around I was able to successfully register my Asterisk server to my Vestalink account, and I have provided generic configurations that should help anyone looking to integrate a Vestalink SIP trunk with a vanilla Asterisk installation.

In `sip.conf` the following entries are necessary in order to register your Asterisk server to your Vestalink account. Note that the registry entry needs to be placed under the `[general]` section. Replace `YOURPHONENUMBER` with the 10 digit DID that you chose when signing up for an account [here](#).

Now in `extensions.conf` the following entries need to be made for inbound and outbound calling over your SIP trunk.