FreePBX Configuration Guide with Firewall

This document does not cover the installation of the FreePBX distribution itself and assumes knowledge of the system build and administration, to include administration access to FreePBX 2.210 running Asterisk 11.2.1 via web browser and secured shell or console.

It should be noted that these test results are applicable to FreePBX variants running Asterisk 11.2.1:

- Elastix
- PBX in a Flash
- AsteriskNOW
- Trixbox CE (END OF LIFE)

Configuration Notes for FreePBX 3.211.63-6 32-bit:

The FreePBX 3.211.63-6 32-bit ISO build ships with a deprecated maco coded in the “extensions_additonal.conf” file under /etc/asterisk. The deprecated macro is:

- cc_callback_macro

The correct/current callback macro for this release is:

- cc_callback_sub

The substitution must be placed in the “extensions_custom.conf” file so it will not be over-written during a system reload or restart:

From the command-line (or remote shell):

cd /etc/asterisk

grep cc_callback_macro extensions_additional.conf >> extensions_custom.conf

sed –i “s/cc_callback_macro/cc_callback_sub/g” extensions_custom.conf

REQUIRED INFORMATION (Provided by MegaPath)
Host: ___________________________________
Number of Trunks: ___________________________________

Pilot Number: ________________________________

Password: ________________________________

NOTE:

- The firewall used for voice-over-IP must support static port translation (or static NAT) meaning that the RTP port traverses the firewall unchanged. Static port translation is required to mitigate one-way audio issues.
- The SIP registration timeout interval should be set to 55 seconds. (See your FreePBX support documentation for details.)

The following screen capture is included as a reference.

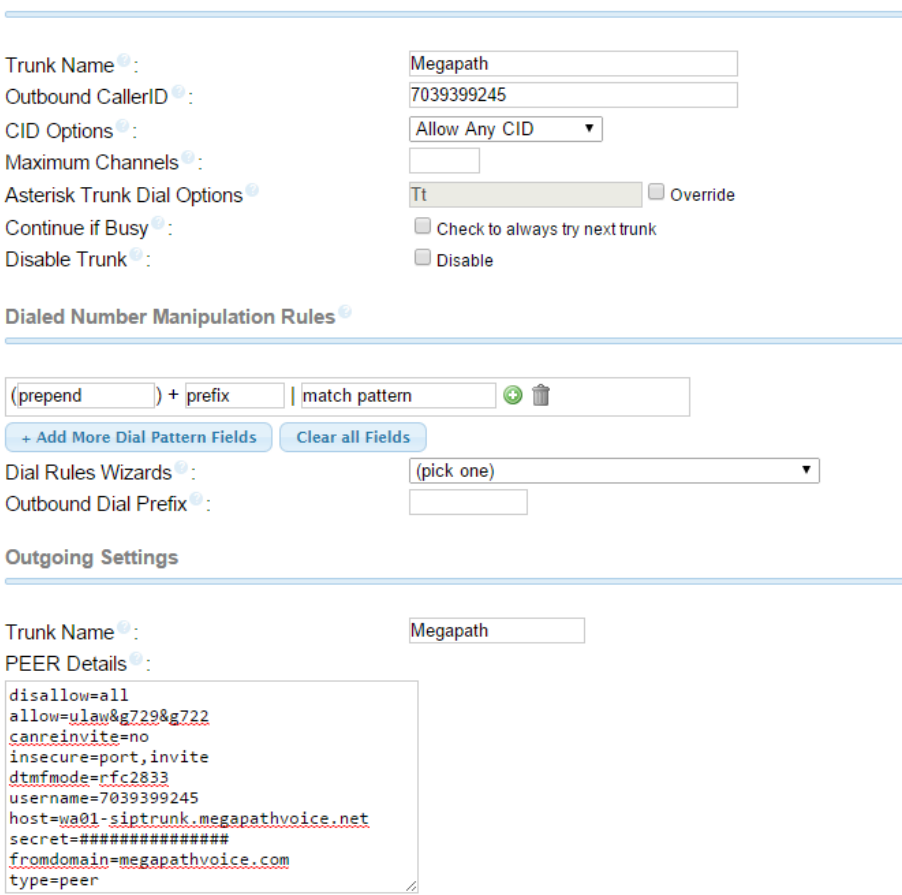

Figure: SIP trunk configuration with firewall

Trunk Name: Megapath
Outbound CallerID: <Pilot Number>
Maximum Channels: <Number of Trunks>

PEER Details:

```
disallow=all
allow=ulaw&g729&g722
canreinvite=no
insecure=port,invite
dtmfmode=rfc2833
username=<Authentication User Name>
host=<state-code>01-siptrunk.megapathvoice.net
secret=<Authentication User Name Password>
fromdomain=megapathvoice.com
type=peer
```

NOTES:

- To limit the allowed codec to G711 only, use the following “allow” statement:
  allow=ulaw

- To limit the allowd codec to G729 only, use the following “allow” statement:
  allow=g729

In the Registration String text box, enter the following registration string:

```
<Authentication User Name>:<secret>@<host>/<Authentication User Name>
```