



Asterisk-based (FreePBX) IP PBX Provisioning Guide

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DISCLAIMER

This document is provided as a basic guideline for setup and configuration of Asterisk systems with MegaPath's SIP Trunking service, based on MegaPath's testing and validation process. It does not include advanced configurations to enable features such as voicemail, Find-Me-Follow-Me, etc. MegaPath is not responsible for customer IP PBX configurations. For more information on advanced features, please refer to your IP PBX documentation.

OVERVIEW

Asterisk (www.asterisk.org) is an open source telephony solution providing PBX functionality such as call switching, route management, and advanced features including voicemail, conferencing, IVR, paging, and more. MegaPath has performed its Asterisk testing and functionality validation utilizing the FreePBX (www.freepbx.org) GUI (graphical user interface) to control and manage the Asterisk deployment. The steps outlined in this guide reflect the process used by MegaPath to successfully configure MegaPath's SIP trunks on the described software/hardware versions only.

SUPPORTED SOFTWARE DEPLOYMENTS

The following software deployments are supported with this setup guide:

- Asterisk 1.4.x and 1.6.x
- FreePBX 2.5.x

REQUIRED INFORMATION

The following information is required to complete the configuration. This information will be provided by your MegaPath Business Voice Implementation Engineer during the installation process.

Maximum Channels _____

Host _____

Username _____

Password _____

Registration String _____

CONFIGURATION

All configurations listed below are completed via the FreePBX web administration interface. For more information on accessing the administration interface, please refer to the FreePBX documentation.

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SIP Trunks

1. From the navigation menu on the left side of the screen, select "**Trunks**"
2. Select "**Add SIP Trunk**" from the "**Add a Trunk**" page
3. Enter the following information on the "**Edit SIP Trunk**" page
 - Set the "**Outbound Caller ID**" if all routes that use this trunk should show a specific caller ID (i.e. all outbound calls should show the company's main number)
 - Set the "**Maximum Channels**" to the number of SIP Trunks purchased from MegaPath.
 - **Dial Rules** will set how the calls will be sent from the trunk to MegaPath. All outbound calls must be sent as +1NPANXXXXXX. To allow 7 digit and 10 digit dialing, configure as below:
 - **Local Calling:** 1(NPA)+NXXXXXX (NPA is the local area code)
 - **10 Digit Dialing:** 1+NPANXXXXXX
 - **Outbound Dial Prefix:** This option will add the "+" into the outbound call to comply with MegaPath's E.164 requirement.
 - **Trunk Name:** MEGAPATH
 - **Peer Details:** Add the below lines verbatim, except for the username, secret, and host entries – for those entries, please use the information provided to you by MegaPath.

Version 1.4.x

```
context=from-pstn
dtmfmode=rfc2833
insecure=very
qualify=no
host=<Host provided by MegaPath>
username=<Username provided by MegaPath>
secret=<Password provided by MegaPath>
type=peer
```

Version 1.6.x

```
context=incoming
dtmfmode=rfc2833
insecure=invite,port
qualify=no
host=<Host provided by MegaPath>
username=<Username provided by MegaPath>
secret=<Password provided by MegaPath>
type=friend
```

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- **Registration String:** The details for the registration string will be provided by MegaPath.

Format:

<username>:<password>@<host>/<username>

Example:

9000000527:RsXHEE@wa1-siptrunk-a.voice.megapath.net/9000000527

4. Click the "**Submit Changes**" button at the bottom of the "**Edit SIP Trunk**" page and make sure to click the orange "**Apply Configuration Changes**" at the top of the screen.

Inbound/Outbound Routes

Please Note: The below information is only one example of how these routes may be configured. Your configuration may vary.

Outbound

1. From the navigation menu on the left side of the screen, select "**Outbound Routes**"
2. Select "**Add Route**"
3. Enter the following information on the "**Edit Route**" page
 - "Route Name" may be set to anything you'd like such as, "Outbound".
 - "Dial Patterns" identify what calls will be sent over the route.
 - 911 - will send 911 calls as dialed
 - 8|. - will send all calls that start with an 8, strip the 8 and pass to the trunk
 - "Trunk Sequence" specifies what trunk this outbound route will use.

Please Note: This example would require a user to dial an 8 for an outside line. This is not required and may be configured as you would like.

4. Click "**Submit Changes**" at the bottom of the "**Edit Route**" page.

Inbound

1. From the navigation menu on the left side of the screen, select "**Inbound Routes**"
2. Select "**Add Route**"
3. Enter the following information on the "**Edit Route**" page
 - Choose the person, group, or queue to route the call to under "Set Destination".

Please note: Each DID will need to have an inbound route built.

4. "**Submit Changes**" at the bottom of the "**Edit Route**" page.