FreePBX R14
SIP Trunk Provisioning Guide

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ABSTRACT
FreePBX 1.8 is a freely available software distribution sponsored by Bandwidth.com that offers a Linux-based (Centos 5.8, Linux kernel 2.6.18) IP PBX that uses Asterisk 1.8.13 as the telephony engine. The Asterisk 1.8.13 engine supports T38 FAX (pass-through mode), analog telephony, SIP telephony, presence management, voice call recording, call queue management, CDR tracking, and a host of third-party applications for added functionality. This document covers MegaPath R14 SIP trunk connectivity for FreePBX 1.8 in addition to the following open source, FreePBX-based distributions:

- PBX-in-a-Flash
- Elasix
- Trixbox CE
- AsteriskNOW

NOTE: The Broadsoft Identity/Device profile required for FreePBX and the FreePBX-based systems listed is the “Generic SIP Trunk Single Registration” Identity/Device profile.

For further information regarding FreePBX features, see http://www.freepbx.org/.
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INTRODUCTION
The following document covers MegaPath R14 SIP trunk configuration settings to enable SIP telephony services for FreePBX 1.8. FreePBX 1.8 is an open source distribution, comprised of Linux Centos 5.8, Apache 2.2, PHP 5.3, and MySQL 5.0; the telephony engine is Asterisk 1.8. The Asterisk engine features the Digium/Asterisk Hardware Device Interface DAHDI open source device interface binaries and libraries to control Digium and other legacy analog telephony interfaces.

This document does not cover the installation of the FreePBX 1.8 distribution itself and assumes knowledge of the system build and administration, to include administration access to the FreePBX 1.8 IP PBX via web browser and secured shell or console.

Other FreePBX-based IP PBX distributions covered by the guidelines in this document includes
- PBX-in-a-Flash
- Elastix
- Trixbox CE
- AsteriskNOW

This document covers MegaPath R14 SIP trunk configurations in conjunction with a third-party firewall and with an Edgemarc 4500-series SIP application-layer gateway.

The R14 Identity/Device profile required for FreePBX, as well as the FreePBX-based systems mentions is the “Generic SIP Trunk Single Registration” Identity/Device profile.

1. STANDARD FIREWALL
This configuration features a FreePBX build deployed behind a standard, third-party firewall. The firewall is configured the forward SIP and the Asterisk RTP range from the firewall WAN IP address to the internal IP address of the FreePBX server NATd behind the firewall.

![Figure 1.1 FreePBX server NATd behind a standard, third-party firewall](image)

IMPORTANT NOTES:
- For port-forwarding, the default Asterisk RTP/UDP port range is 10000:20000 and the SIP/UDP port is 5060.
- The third-party firewall SHOULD support static NAT for all outbound and inbound UDP traffic to mitigate one-way audio.
- If one-way audio is still an issue with a firewall known to support static NAT, then ensure that all SIP helper or SIP proxy services are disabled in the firewall.
1.1 Add SIP Trunk

To configure the R14 SIP trunk:
1. Log in to the FreePBX server.
2. From the main menu, go to the “Connectivity” menu option.
3. Select “Trunks”.
4. Select “Add a SIP Trunk”.
5. Input the options as shown in the following screens.

IMPORTANT NOTES:
- DO NOT USE the credentials and DID displayed in the screen captures that follow. USE the credentials and DID you have been provided with your service order of demo SIP trunk.
- Use one of the DID associated with the SIP trunk in the Outbound CallerID field or you WILL NOT be able to make outbound calls.

6. Under the “General Settings” section of the SIP trunk, give the SIP trunk a name. The name given will be used to identify the SIP trunk in inbound and outbound call routing configurations.
7. Enter the DID (or one of the DIDs) associated with the SIP trunk in the CallerID field.

Figure 1.1.1 SIP trunk General Settings.
8. Under “Outgoing Settings,” enter the Trunk Name.
9. Under “PEER Details” enter the following configuration parameters:

   disallow=all
   allow=all
   dtmfmode=rfc2833
   canreinvite=no
   insecure=very
   context=from-trunk
   host=wa1-siptrunk-srv.voice.speakeasy.net
   username=<Authentication username associated with SIP trunk>
   secret=<Authentication username password>
   type=peer

FIG 1.1.2 shows the “PEER Details” window in which these parameters are entered.

![Figure 1.1.2 SIP trunk PEER Details.](image-url)
10. Under "Registration" enter the registration string in the "Register String" field. The registration string format is:

\(<\text{username}>:\text{username password}@<\text{SIP gateway hostname}>/\text{username}\)

For example, for authentication username \textit{wa1siptrunk01} with password \textit{eEwd2wA90x1} and SIP gateway \textit{wa1-siptrunk-srv.voice.speakeasy.net}, the registration string will read:

\textit{wa1siptrunk01:eEwd2wA90dx1@wa1-siptrunk-srv.voice.speakeasy.net/wa1siptrunk01}

11. Submit and apply all changes.

1.2 Asterisk SIP Settings

1. From the main menu under "Settings" go to "Asterisk SIP Settings."
2. Under the "NAT" section, select the NAT option that reflects your local network.
3. Select "Auto Configure" to populate the "External IP" and "Local Networks" fields.

IMPORTANT NOTE: AsteriskNOW does not feature the "Asterisk SIP Settings" menu option; however, the lack of the "Asterisk SIP Settings" menu option does not impede its interoperability.
4. Under “Audio Codecs” ensure that at a minimum the “ulaw” codec is selected.

![Figure 1.2.2 Asterisk SIP Settings Audio Codecs.](image)

5. Submit and apply all changes.

### 1.3 Check the SIP Trunk Status

To check on the status of the SIP trunk registration:
1. From the main menu, go to “Reports”.
2. Select “FreePBX System Status”.

![Figure 1.3.1 Reports FreePBX System Status.](image)
The SIP trunk registration status can also be assessed in a secure shell or console session by issuing the following command at the command prompt to access the Asterisk command-line interface:

[freepbx]root # asterisk –vvvr

Once inside the Asterisk command line, issue the following command to check on the SIP trunk registration status:

freepbx*CLI> sip show registry

Figure 1.3.2 Asterisk CLI from a secure shell or console session.
2. EDGEMARC APPLICATION-LAYER GATEWAY

This configuration features a FreePBX build deployed behind an Edgemarc SIP application-layer gateway (SIP ALG). The SIP ALG is configured to forward SIP and the Asterisk RTP range from the SIP ALG WAN IP address to the internal IP address of the FreePBX server NATd behind the SIP ALG.

![Figure 2.1 FreePBX IP PBX NATd behind an Edgemarc SIP ALG.](image)

2.1 Add SIP Trunk

To configure the R14 SIP trunk:

1. Log in to the FreePBX server and from the main menu
2. Go to the “Connectivity” menu option and select “Trunks”.
3. Select “Add a SIP Trunk” and input the options as shown in the following screens.

IMPORTANT NOTES:

- DO NOT USE the credentials and DID displayed in the screen captures that follow. USE the credentials and DID you have been provided with your service order of demo SIP trunk.
- Use one of the DID associated with the SIP trunk in the “Outbound CallerID” field or you WILL NOT be able to make outbound calls.

4. Under the “General Settings” section of the SIP trunk, give the SIP trunk a name. The name given will be used to identify the SIP trunk in inbound and outbound call routing configurations.
5. Enter the DID (or one of the DIDs) associated with the SIP trunk in the “CallerID” field.
6. Under “Outgoing Settings,” enter the Trunk Name.
7. Under “PEER Details” enter the following configuration parameters:

NOTE: Instead of the IP address of the VOIP providers SIP gateway/session border controller, the LAN-SIDE IP address of the Edgemarc SIP ALG is used in the “host” parameter. In this example, the LAN-SIDE IP address of the Edgemarc SIP ALG is “172.16.248.1”

disallow=all
allow=all
dtmfmode=rfc2833
canreinvite=no
insecure=very
context=from-trunk
host=172.16.248.1
username=<Authentication username associated with SIP trunk>
secret=<Authentication username password>
type=peer
8. Under “Registration”, enter the registration string in the “Register String” field. The registration string format is:

\[
<\text{username}>: <\text{username password}>@<\text{LAN-SIDE IP address of Edgemarc SIP ALG}>/<\text{username}>
\]

For example, for authentication username \textit{j3rs3yr7} with password \textit{Wp3th0s4Z} and LAN-SIDE Edgemarc SIP ALP IP address \textbf{172.16.248.1}, the registration string will read:

\textit{j3rs3yr7:Wp3th0x4Z@172.16.248.1/j3rs3yr7}
2.2 Asterisk SIP Settings

1. From the main menu under "Settings" go to "Asterisk SIP Settings."
2. Under the "NAT" section, select the NAT option that reflects your local network.
3. Select "Auto Configure" to populate the "External IP" and "Local Networks" fields.

NOTE: AsteriskNOW does not feature the "Asterisk SIP Settings" menu option; however, the lack of the "Asterisk SIP Settings" menu option does not impede its interoperability.
5. Under "Audio Codecs" ensure that at a minimum the "ulaw" codec is selected.

![Audio Codecs]

Figure 2.2.2 Asterisk SIP Settings Audio Codecs.

6. Submit and apply all changes.

2.3 Edgemarc SIP ALG SIP Trunk Configuration

1. Log in to the Edgemarc SIP ALG. Under the "VoIP ALG" menu option, select the "SIP" option.
2. Under "SIP Settings," in the "SIP Server Address:" field, input the fully-qualified domain name or IP address of the SIP gateway or session border controller assigned to you by your VoIP service provider.
3. Input the SIP server port number in the "SIP Server Port:" field.
4. Enable "Use Custom Domain:"
5. Enter "speakeasy.net" in the "SIP Server Domain:" field.
### SIP Settings

SIP protocol settings.

The SIP Server settings specify the address and port that all client traffic shall be forwarded to.

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Server Address</td>
<td>lab-1-siptrunk-a voice spei</td>
</tr>
<tr>
<td>SIP Server Port</td>
<td>5060</td>
</tr>
<tr>
<td>Use Custom Domain</td>
<td>✔</td>
</tr>
<tr>
<td>SIP Server Domain</td>
<td>speakeasy.net</td>
</tr>
<tr>
<td>List of SIP Servers</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Setting</th>
<th>Options</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable Multi-homed Outbound Proxy Mode</td>
<td>□</td>
</tr>
<tr>
<td>Enable Transparent Proxy Mode</td>
<td>□</td>
</tr>
<tr>
<td>Limit Outbound to listed Proxies / SIP Servers</td>
<td>□</td>
</tr>
<tr>
<td>Limit Inbound to listed Proxies / SIP Servers</td>
<td>□</td>
</tr>
</tbody>
</table>

---

**Stale Timer**

The stale timer, if set, is used to automatically delete SIP clients that have not registered within the given time period.

**Stale client time (m):** 1440

Registration Rate-Pacing parameters are available on the Survivability page.

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10. Submit changes.
11. From the "SIP" sub-menu, select "Trunking."
12. In the "Add a trunking device" menu box:
   A. Set the "Action:" field to "Add new trunking device."
   B. Input a recognizable name for the trunking device in the "Name:" field.
   C. Input the IP address of theFreePBX server in the "Address:" field.
   D. Input the SIP port number of the FreePBX server in the "Port:" field.
   E. Select the "Commit" button to create the SIP trunking device.
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**SIP Trunking**

Configuration of SIP trunking devices.

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**SIP Trunking devices**

A SIP trunking device can be a PSTN gateway, or similar device, that does not issue REGISTER messages. Calls will be forwarded to the device based on the dial-plan rules below.

If VLANS are enabled, the SIP trunking device needs to be in the same VLAN as defined in the VoIP ALG page.

---

**SIP Trunking Devices**

<table>
<thead>
<tr>
<th>Select</th>
<th>Address</th>
<th>Port</th>
<th>Name</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>172.16.248.165</td>
<td>5060</td>
<td>IPPBX</td>
</tr>
</tbody>
</table>

---

**Add a trunking device**

- **Action:** Add new trunking device
- **Name:**
- **Address:**
- **Port:** 5060

---

**Figure 2.3.2 SIP Trunking, SIP Trunking devices**

6. Under “SIP Trunking” go to the “Rules” section to add an inbound rule.

**NOTE:** For a single SIP registration, we will be using a Default Rule to send all inbound call traffic to the LAN-SIDE FreePBX server.

7. In the “Add a rule” menu box:
   - **A.** Set the “Action:” field to “Add a new rule.”
   - **B.** Set the “Type:” field to “Inbound.”
   - **C.** Set the “Trunking device:” field to the recognizable name you created for the target trunking device in the last section.
   - **D.** Select “Commit” to create the Default Rule.
**Rules**

Rules are used to forward and/or modify incoming and outgoing calls. There are 3 types of rules:
- **Inbound**: from server to trunking device
- **Outbound**: from trunking device to server
- **Redirect**: from local phone to trunking device (w/o routing to server)

Outbound rules can match against and/or modify either the calling or called number. Inbound and redirect rules operate on the called number only. Stripped and added digits always apply to the left-most digits of the DID.

<table>
<thead>
<tr>
<th>Type</th>
<th>Party</th>
<th>PRIID</th>
<th>Pattern-match</th>
<th>Strip</th>
<th>Add</th>
<th>b2bua</th>
<th>az</th>
<th>Trunking device</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inbound</td>
<td>Default Rule</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>IPPBX (172.16.248.165:5060)</td>
</tr>
</tbody>
</table>

**Add a rule**

- **Action**: Add new rule
- **Type**: Inbound
- **Call Party**: Called
- **Default rule**: 
- **Priority (inbound & redirect only)**: 
- **Pattern-match (if not default)**: 
- **Strip digits**: 0
- **Add string**: 
- **Use B2BUA**: 
- **Use SIP proxy as secondary target**: 
- **Trunking device**: IPPBX (172.16.248.165:5060)

Figure 2.3.3 SIP Trunking, Rules, Add a rule.
2.4 Checking the SIP Trunk Status

To check on the status of the SIP trunk registration:
1. Log in to the FreePBX server via web browser
2. From the main menu, go to “Reports”.
3. Select “FreePBX System Status”.

Alternatively, log in to the FreePBX server in a secured shell session or in a console session, and issue the following command at the command prompt to access the Asterisk command-line interface:

[freepbx]root # asterisk –vvvr

Once inside the Asterisk command-line, issue the following command to check on the SIP trunk registration status:

freepbx*CLI> sip show registry

Figure 2.4.1 FreePBX System Status.

Figure 2.4.2 Asterisk CLI, SIP registration status.