Asterisk

Asterisk ??????????????? ?????? ???Asterisk ? Digium ??????·????1999?????????????????????????????Asterisk ??????????????????????????IP??????

- Installation
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Installation

Install Asterisk and FreePBX

Install FreePBX 15 with Asterisk 16 on Debian 10

Install Asterisk 16

Step 1: Update system

```
sudo apt update && sudo apt upgrade
sudo reboot
```

Step 2: Install Asterisk 16 LTS dependencies

```
sudo apt install git curl wget libnewt-dev libssl-dev libncurses5-dev subversion libsqlite3-dev build-essential
libjansson-dev libxml2-dev uuid-dev
```

Step 3: Download Asterisk 16 LTS tarball

```
cd /usr/src/
sudo curl -O http://downloads.asterisk.org/pub/telephony/asterisk/asterisk-16-current.tar.gz

sudo tar xvf asterisk-16-current.tar.gz
cd asterisk-16*/

# download the mp3 decoder library into the source tree
sudo contrib/scripts/get_mp3_source.sh

# Ensure all dependencies are resolved
sudo contrib/scripts/install_prereq install
```

Step 4: Build and Install Asterisk 16

```
sudo ./configure
sudo make menuselect
```

- Add-ons: chan_ooh323, format_mp3
- Core Sound Packages: CORE-SOUNDS-EN-*
- Music On Hold File Packages: MOH-OPSOUND-*
- Extra Sound Packages: EXTRA-SOUNDS-EN-*
- Applications: app_macro

```
sudo make
sudo make install
sudo make progdocs
sudo make samples
sudo make config
sudo ldconfig
```

Create Asterisk User

```
sudo groupadd asterisk
sudo useradd -r -d /var/lib/asterisk -g asterisk asterisk
sudo usermod -aG audio,dialout asterisk
sudo chown -R asterisk.asterisk /etc/asterisk
sudo chown -R asterisk.asterisk /var/{lib,log,spool}/asterisk
sudo chown -R asterisk.asterisk /usr/lib/asterisk
```

Set Asterisk default user to asterisk

```
$ sudo vim /etc/default/asterisk
AST_USER="asterisk"
AST_GROUP="asterisk"

$ sudo vim /etc/asterisk/asterisk.conf
runuser = asterisk ; The user to run as.
rungroup = asterisk ; The group to run as.
```

Restart asterisk service

```
sudo systemctl restart asterisk

# Enable asterisk service to start on system boot
sudo systemctl enable asterisk

# Test to see if you can connect to Asterisk CLI
sudo asterisk -rvv
```

Install FreePBX 15

Step 1: Install MariaDB Database server

```
sudo apt update
sudo apt install mariadb-server mariadb-client

# Initial DB setup and set root's password for DB
sudo /usr/bin/mysql_secure_installation
```

Step 2: Installing Node.js 10 LTS

```
sudo apt install curl dirmngr apt-transport-https lsb-release ca-certificates
curl -sL https://deb.nodesource.com/setup_10.x | sudo bash
sudo apt update
sudo apt install gcc g++ make
sudo apt install nodejs
```

Step 3: Install and configure Apache Web Server

```
sudo apt install apache2

# change Apache user to asterisk and turn on AllowOverride option
sudo cp /etc/apache2/apache2.conf /etc/apache2/apache2.conf_orig
sudo sed -i 's/^\(User\|Group\).*/\1 asterisk/' /etc/apache2/apache2.conf
sudo sed -i 's/AllowOverride None/AllowOverride All/' /etc/apache2/apache2.conf

# Remove default index.html page
sudo rm -f /var/www/html/index.html
```

Step 4: Install PHP and required extensions

```
sudo apt install wget php php-pear php-cgi php-common php-curl php-mbstring php-gd php-mysql \
php-gettext php-bcmath php-zip php-xml php-imap php-json php-snmp php-fpm libapache2-mod-php
```

Change php maximum file upload size

```
sudo sed -i 's/\(^upload_max_filesize = \).*/\120M/' /etc/php/7.3/apache2/php.ini
sudo sed -i 's/\(^upload_max_filesize = \).*/\120M/' /etc/php/7.3/cli/php.ini
```

Step 5: Install FreePBX 15

```
sudo apt install wget
cd /usr/src
wget http://mirror.freepbx.org/modules/packages/freepbx/freepbx-15.0-latest.tgz

tar xfz freepbx-15.0-latest.tgz
rm -f freepbx-15.0-latest.tgz

cd freepbx
sudo ./start_asterisk start
sudo ./install -n --dbuser root --dbpass "yourpassword"

# Enable Apache Rewrite engine
sudo a2enmod rewrite
sudo systemctl restart apache2
```

Step 6: Access FreePBX 15 Web Interface

Create the first admin account.

Q & A

Q: Online modules are not available.

> Error:
>
> Warning: Error retrieving updates from online repository(s) (https://mirror.freepbx.org 35). Online modules are not available.

A: Change the DNS to 8.8.8.8

```
vi /etc/resolv.conf

nameserver 8.8.8.8
#nameserver 67.207.67.3
#nameserver 67.207.67.2
```

Reference

- Install Asterisk 16 with FreePBX 15 on Ubuntu 20.04/18.04/16.04 & Debian 9
- Install Asterisk 16 LTS on Ubuntu 20.04/18.04/16.04 & Debian 10/9
- How To Install FreePBX 15 on Ubuntu 20.04/18.04/16.04 & Debian 10/9
- Installing FreePBX 15 on Debian 9.6

Installation

Install FreePBX 15 with Docker

Reference

- https://github.com/tiredofit/docker-freepbx#introduction

Incredible PBX

Post-Installation

Incredible PBX 2027 with Debian 11

Reset the hostname and password:

```
# Set the hostname
hostnamectl set-hostname <your-FQDN-name>


# Set the password
passwd           # for Root
admin-pw-change     # for FreePBX
apache-pw-change    # for Reminders and AsteriDex
```

Set Gmail as an SMTP Smarthost:

Create an App password for your Gmail account:
https://support.google.com/accounts/answer/185833?hl=en

```
/root/enable-gmail-smarthost-for-sendmail
```

Stop Webmin

```
systemctl stop webmin
systemctl disable webmin
```

Learning

Official URLs

Documentation

- PBX GUI Home - PBX GUI - Sangoma Documentation (atlassian.net)

Community

- Asterisk: https://community.asterisk.org/
- FreePBX: https://community.freepbx.org/

Github

- Asterisk: https://github.com/asterisk
- FreePBX: https://github.com/freepbx

Speech Recognition (Speech to Text)

Vosk-API

- Vosk speech recognition modules for Asterisk
- https://alphacephei.com/vosk/integrations
- Offline Speech to Text for Desktop Linux

Kamailio

- Handling SIP Flood Attacks Using Kamailio

Secure SIP Server

- APIBAN
    - Apiban Clients
    - Block unwanted SIP traffic efficiently

SIP Monitoring

- [sipexer - Modern and flexible SIP cli tool]

Auto Provisioning

- OSS End Point Manager - PBX GUI - Documentation (freepbx.org)
- https://www.voip-info.org/forum/threads/oss-epm-for-freepbx-16-ipbx-2027.26880/

WebRTC

- InnovateAsterisk/Browser-Phone
- [Video] WebRTC Browser Phone with Asterisk & Raspberry Pi

Billing

- MagnusBilling

HA with DRBD

- VitalPBX High Availability

Q & A

CDR Reports ??????

?? MySQL ???

```
# MySQL Credentials
cat /etc/freepbx.conf

# Check the mysql
mysql -u freepbxuser -p asteriskcdrdb -e 'SELECT * FROM cdr ORDER BY calldate DESC LIMIT 4'
```

?? asterisk module

```
asterisk -rx "module show like odbc"

Module                    Description                    Use Count  Status      Support Level
cdr_adaptive_odbc.so         Adaptive ODBC CDR backend            0         Running            core
cdr_odbc.so                 ODBC CDR Backend                      0         Running         extended
cel_odbc.so                 ODBC CEL backend                      0         Running            core
func_odbc.so                 ODBC lookups                        0         Running            core
res_config_odbc.so           Realtime ODBC configuration          0         Running            core
res_odbc.so                 ODBC resource                        6         Running            core
res_odbc_transaction.so       ODBC transaction resource             1         Running            core
```

??????????

```
fwconsole stop
fwconsole start
```

> [2022-06-03 10:38:42] WARNING[32144] res_odbc.c: res_odbc: Error SQLConnect=-1 errno=0 [unixODBC][Driver Manager]Can't open lib '/usr/lib/x86_64-linux-gnu/odbc/libmaodbc.so' : file not f

Solution:

```
#> locate libmaodbc.so
/usr/lib/i386-linux-gnu/odbc/libmaodbc.so


#> cp /etc/odbcinst.ini /etc/odbcinst.ini.orig
#> vi /etc/odbcinst.ini


# Change this line
Driver = /usr/lib/x86_64-linux-gnu/odbc/libmaodbc.so
```

????????????

```
fwconsole stop
fwconsole start
```

Can't send 10 type frames with SIP write

Frame type '10' is comfort noise (aka CNG) which Asterisk does not support.

However as of 13.18.0 this message will be silenced so you won’t see it anymore.

You can ignore it or disable CNG on all of your endpoints and ask the telecom providers as well to disable the CNG on your trunks.

FXO ??????? Answer ? IP ?

??? Gateway ????? (IP to PSTN)??? PSTN ????????Gateway ???? Answer ? IP ??

????????

Polarity Reversal (????)????????????????????????????????? ?????????? (??????)

The issue:

The VoIP gateway is sending an answer signal to the IP side, even when the call is not picked up on the PSTN side. This is not the expected behavior, as the gateway should only send an answer signal when the call is actually answered by the called party.

Possible causes:

1. **Improper FXO port configuration**: The FXO port on the gateway might not be configured correctly, leading to the gateway sending an answer signal prematurely.

2. **PSTN line issues**: There could be issues with the PSTN line, such as noise or electrical interference, that are causing the gateway to misinterpret the call status.
3. **Polarity Reversal Detection not functioning correctly**: The Polarity Reversal Detection feature on the gateway might not be working as expected, which could be contributing to the issue.

Polarity Reversal Detection:

Polarity Reversal Detection is a feature used to detect when a call is answered or hung up on the PSTN side. When a call is answered, the polarity of the PSTN line reverses, and the gateway can detect this change to determine the call status. If the Polarity Reversal Detection is not functioning correctly, the gateway may not be able to accurately determine the call status.

How Polarity Reversal Detection works:

When a call is made from the IP side to the PSTN side, the gateway monitors the PSTN line for a polarity reversal. When the called party answers, the PSTN line polarity reverses, and the gateway detects this change. The gateway then sends an answer signal to the IP side, indicating that the call has been answered.

Troubleshooting steps:

1. **Verify FXO port configuration**: Check the FXO port configuration on the gateway to ensure it is set up correctly. Consult the gateway's documentation or contact the manufacturer for guidance.
2. **Check PSTN line quality**: Verify that the PSTN line is clean and free of noise or electrical interference. You can use a line tester or consult with the PSTN provider to troubleshoot line issues.
3. **Verify Polarity Reversal Detection settings**: Ensure that the Polarity Reversal Detection feature is enabled and configured correctly on the gateway. Consult the gateway's documentation or contact the manufacturer for guidance.
4. **Monitor gateway logs**: Check the gateway logs to see if there are any errors or anomalies related to the Polarity Reversal Detection feature.
5. **Test with a different PSTN line**: If possible, test the VoIP gateway with a different PSTN line to isolate the issue.

By following these troubleshooting steps, you should be able to identify and resolve the issue causing the VoIP gateway to send an answer signal prematurely.

A2B ? FreePBX ???

A2B ?? Outbound Trunk ?

Call > FreePBX > A2B > SIP Carrier

? FreePBX ???????
?????????

```
username=51521171
fromuser=51521171
type=friend
secret=1234567
host=incoming.future-nine.com
insecure=port,invite
nat=yes
qualify=yes
context=from-trunk
allow=ulaw,g729,g726
trustrpid=yes
sendrpid=yes
canreinvite=no
```

FreePBX ?? Outbound Trunk ?

Call > A2B > FreePBX > SIP Carrier

? FreePBX ?)

1. ?? SIP extension: 9001

? A2B PBX)

1. ?? SIP Trunk: freepbx

```
[freepbx]
username=9001
type=peer
```

```
secret=<ext-secret>
insecure=very
host=<freepbx-ip-addr>
fromuser=9001
qualify=yes
```

2. ?? SIP Register String (**for incoming call only**)

```
9001:<ext-secret>@<freepbx-ip-addr>/from_freepbx
```

> NOTE: ???????? SIP number ??????(`/from_freepbx`)???????? SIP number 199 ??

3. ?? Outbound Route

A2Billing

URLs

- Home: http://www.asterisk2billing.org/
- Github: https://github.com/Star2Billing/a2billing/

Alternative to A2Billing

- MagnusBilling
  - Github: https://github.com/magnussolution/magnusbilling7
- ASTPP (based on FreeSWITCH)
  - Github: https://github.com/iNextrix/ASTPP

Voice Mail Transcription

IBM Watson STT

Creating IBM Watson Credentials

1. Login to IBM Cloud using your new credentials.
2. Once logged in, choose *IBM Cloud* from the Title Bar to display your Dashboard.
3. Choose *Create Resource*.
4. Click *Speech to Text* from the AI Section.
5. Name your STT service, choose the desired region, and choose Default resource group.
6. Select a *Pricing Plan*:
   - LITE provides 500 minutes/month free. Plan is deleted after 30 days of inactivity.
   - STANDARD is 2¢/minute with no free minutes.
7. When *Speech to Text* dialog opens, copy your *API Key* and *URL*.
8. Logout by clicking on image icon in upper right corner of dialog window.

Installing STT Engine

1. Unpack the file

```
wget http://incrediblepbx.com/sendmailibm-13.tar.gz
tar zxvf sendmailibm-13.tar.gz
cp sendmailmp3.ibm /usr/local/sbin/sendmailmp3
chmod 0755 /usr/local/sbin/sendmailmp3
```

2. Edit `sendmailmp3.ibm` and insert your IBM STT API_KEY and URL. Save file.

3. Edit `bluemix-test` and insert your IBM STT API_KEY and URL. Save the file.

4. Copy the updated `sendmailmp3.ibm` file to `sendmailmp3`:

```
cp sendmailmp3.ibm /usr/local/sbin/sendmailmp3
chmod 0755 /usr/local/sbin/sendmailmp3
```

5. Test your Bluemix STT setup: `bluemix-test`

Result should be: *we are now transferring you out of the company directory…*

FreePBX Setup

Settings > Voicemail Admin > Settings > Email Config > Mail Command: /usr/local/sbin/sendmailmp3

Set up voicemail for an extension and include your email address.

Tutorials

- Free IBM Voicemail Transcription with Incredible PBX 2020
- IBM's Speech Recognition Engine Comes to Asterisk
- Free Asterisk Voicemail Transcription with IBM Watson STT
- Creating IBM Watson Credentials
- Release notes for Speech to Text for IBM Cloud
- Getting started with Speech to Text
- Github: lgaetz/sendmail-bluemix
- Github: jtsage/sendmail.asterisk

Google STT

Tutorials

- Speech to text using Google Cloud (voicemails to text)
- FreePBX-VM-Transcription

Soft Phone

Open Source/Freeware

- Zoiper - Branding for your business, Freeware, Support mobile and desktop (Linux/Windows/macOS)
- Linphone - Open Source, Support mobile and desktop (Linux/Windows/macOS)
- MicroSIP - Open Source, based on PJSIP for Windows OS
- PhonerLite - Freeware, For Windows OS

OpenSIPS

Installation on Debian 10

- OpenSIPS v3 with GUI on Debian v10 MariaDB Apache install guide
- Quick Start to OpenSIPS Training 3.2

OpenSIPS 3.3

```
apt install gnupg2
apt-key adv --keyserver keyserver.ubuntu.com --recv-keys 049AD65B

# For Debian 10
echo "deb https://apt.opensips.org buster 3.3-releases" >/etc/apt/sources.list.d/opensips.list
echo "deb https://apt.opensips.org buster cli-nightly" >/etc/apt/sources.list.d/opensips-cli.list
# For Ubuntu 20
echo "deb https://apt.opensips.org focal 3.3-releases" >/etc/apt/sources.list.d/opensips.list
echo "deb https://apt.opensips.org focal cli-nightly" >/etc/apt/sources.list.d/opensips-cli.list

apt update
apt install opensips
apt install opensips-cli

# Install all other modules
apt install opensips-*

# Start opensips and check the status
systemctl start opensips
systemctl status opensips
```

OpenSIPS Database Support (MySQL)

```
# Install MySQL Server (MariaDB on Debian 10)
apt install mariadb-server

# Create the database opensips using the OpenSIPS command line interface
```

```
opensips-cli -x database create opensips

# Verify if the tables were created
mysql opensips -e "show tables"

# Set the root's password for MariaDB and complete a few secure steps.
MariaDB> alter user 'root'@'localhost' identified by 'newpassword';
MariaDB> flush privileges;
MariaDB> exit
```

OpenSIPS Control Panel 9.3.3

- [OpenSIPS Control Panel version class 9 (9.3.2, 9.3.3) documentation]

```
# Install Apache, PHP and other dependencies
apt-get install apache2 libapache2-mod-php php-curl php php-mysql php-gd php-pear php-cli php-apcu git

# Download the OCP 9.3.3
git clone -b 9.3.3 https://github.com/OpenSIPS/opensips-cp.git /var/www/opensips-cp
```

Configure Apache

```
# Remove the default configuration
rm /etc/apache2/sites-enabled/000-default.conf
```

Edit: /etc/apache2/sites-enabled/opensips.conf

```
<VirtualHost *:80>
        <Directory /var/www/opensips-cp/web>
                Options Indexes FollowSymLinks MultiViews
                AllowOverride None
                Require all granted
        </Directory>

        <Directory /var/www/opensips-cp>
                Options Indexes FollowSymLinks MultiViews
                AllowOverride None
                Require all denied
        </Directory>
```

```
    Alias /cp /var/www/opensips-cp/web

    <DirectoryMatch "/var/www/opensips-cp/web/tools/.*/.*/(template|custom_actions|lib)/">
        Require all denied
    </DirectoryMatch>

    ErrorLog ${APACHE_LOG_DIR}/error.log
    CustomLog ${APACHE_LOG_DIR}/access.log combined

</VirtualHost>
```

Set the permissions of directories

```
chown -R www-data:www-data /var/www/opensips-cp/
```

Creating the OCP tables

```
# This will create the OCP specific tables into the opensips database and add a first access user,
# the admin user with the opensips password.
mysql -uroot -p opensips < /var/www/opensips-cp/config/db_schema.mysql
```

set Cron jobs

```
cp /var/www/opensips-cp/config/tools/system/smonitor/opensips_stats_cron /etc/cron.d
sed -i 's/\/var\/www\/html\/opensips-cp/\/var\/www\/opensips-cp/g' /etc/cron.d/opensips_stats_cron
```

Restart Apache

```
systemctl restart apache2
```

Visit the OCP Web site: http://server-ip-address/cp , admin / opensips

RTPProxy

```
apt install build-essential
apt install libucl-dev
cd /usr/src
git clone -b master https://github.com/sippy/rtpproxy.git
git -C rtpproxy submodule update --init --recursive
cd rtpproxy
./configure
```

```
make clean all
make install
```

Configure the systemd

Edit: /etc/systemd/system/rtpproxy.service

```
[Unit]
Description=RTPProxy media server
After=network.target
Requires=network.target

[Service]
Type=simple
PIDFile=/var/run/rtpproxy/rtpproxy.pid
Environment='OPTIONS= -l 172.16.0.67 -A 154.19.187.227 -m 10000 -M 20000 -d INFO:LOG_LOCAL5'

Restart=always
RestartSec=5

ExecStartPre=-/bin/mkdir /var/run/rtpproxy
ExecStartPre=-/bin/chown opensips:opensips /var/run/rtpproxy

ExecStart=/usr/local/bin/rtpproxy -p /var/run/rtpproxy/rtpproxy.pid -s unix:/var/run/rtpproxy/rtpproxy.sock \
 -u opensips:opensips $OPTIONS
ExecStop=/usr/bin/pkill -F /var/run/rtpproxy/rtpproxy.pid

ExecStopPost=-/bin/rm -R /var/run/rtpproxy

StandardOutput=syslog
StandardError=syslog
SyslogIdentifier=rtpproxy
SyslogFacility=local5

TimeoutStartSec=10
TimeoutStopSec=10

[Install]
WantedBy=multi-user.target
```

Start the service

```
systemctl daemon-reload
systemctl start rtpproxy
systemctl enable rtpproxy
```

Configuration

OpenSIPS

Generate config file

```
# Install the package required
apt install m4

# -> Residential Script
# --> Configure Residential Script
# ---> Select all options except for TLS, VM_DIVERSION, PRESENCE
/usr/sbin/osipsconfig

mv /etc/opensips/opensips.cfg /etc/opensips/opensips.cfg.orig
mv /etc/opensips/opensips_residential_2023-3-19_6:6:6.cfg /etc/opensips/opensips.cfg
chmod 0644 /etc/opensips/opensips.cfg

# Restart OpenSIPS
systemctl restart opensips
```

opensips.cfg for server behind the firewall

```
/* For AWS and OpenStack Environment */
/* WAN IP: 123.123.123.123 */
/* LAN IP: 172.16.0.67
advertised_address="123.123.123.123"
alias="123.123.123.123"

socket=udp:172.16.0.67:5060
socket=tcp:172.16.0.67:5060
```

opensips.cfg for RTPProxy

```
### RTPProxy module ###
loadmodule "rtpproxy.so"
## Fixed for ERROR:rtpproxy:send_rtpp_command: proxy <udp:localhost:7890> does not respond, disable it
#modparam("rtpproxy", "rtpproxy_sock", "udp:localhost:7890")
modparam("rtpproxy", "rtpproxy_sock", "unix:/var/run/rtpproxy/rtpproxy.sock")
```

opensips.cfg for dispatcher

```
### Dispatcher modules ###
loadmodule "dispatcher.so"
modparam("dispatcher", "db_url", "mysql://opensips:opensipsrw@localhost/opensips")
modparam("dispatcher", "dst_avp", "$avp(271)")
modparam("dispatcher", "attrs_avp", "$avp(272)")
modparam("dispatcher", "grp_avp", "$avp(273)")
modparam("dispatcher", "cnt_avp", "$avp(274)")
modparam("dispatcher", "hash_pvar", "$avp(273)")
modparam("dispatcher", "ds_ping_method", "OPTIONS")
modparam("dispatcher", "ds_ping_from", "sip:sipcheck@outbound_IP:5060")
modparam("dispatcher", "ds_ping_interval", 10)
modparam("dispatcher", "ds_probing_threshhold", 3)
modparam("dispatcher", "ds_probing_mode", 1)
modparam("dispatcher", "options_reply_codes", "501,403,404,400,200")
```

OpenSIPS Control Panel (OCP)

OCP ?????????

??: config/modules.inc.php

???????

??: config/db.inc.php

Log file

Edit: /etc/rsyslog.d/opensips.conf

```
local0.*                        -/var/log/opensips.log
```

Restart rsyslog

```
touch /var/log/opensips.log
systemctl restart rsyslog
```

OpenSIPS CLI

```
# opensips-cli -x mi version
{
   "Server": "OpenSIPS (3.1.14 (x86_64/linux))"
}
```

FAQ

OCP ? dispatcher ??????

Solution: ?? dispatcher ? mi_http ???????????????? OCP ? MI Commands ?? `ds_list` ?????????????????

Links

- https://opensips.org/
- https://www.rtpproxy.org/
- Quick Start to OpenSIPS Training 3.2
- opensips ?? 1/2

Dispatcher

- OpenSIPS + FreeSWITCH ???????
- OpenSIPS????-dispatcher??????-?????????????

CGRateS

- Installation
- CGRateS Usage
- https://fossies.org/linux/opensips/modules/cgrates/README
- https://nickvsnetworking.com/category/voip/cgrates/

FreePBX

FreePBX is a web-based open-source graphical user interface (GUI) that manages Asterisk, a voice over IP and telephony server.

FreePBX

fwconsole

Tutorials

- CLI Commands

Help

```
fwconsole help

# lists all commands
php /usr/sbin/fwconsole list
```

Service Start/Stop

```
# Start Asterisk and run other needed FreePBX commands
fwconsole start

# Stop Asterisk and run other needed FreePBX commands
fwconsole stop
```

Module Admin

```
# Check Online Repository
fwconsole ma listonline

# Install a module
fwconsole ma download ivr
fwconsole ma install ivr

# Installing specific module versions with multiple modules
fwconsole ma install foomodule:15.1.3 barmodule:15.0.9

# Upgrade all modules
fwconsole ma listonline | grep "upgrade"
fwconsole ma upgradeall
```

```
# Apply the settings changed
fwconsole reload
```

Database

????? asterisk (??? /etc/freepbx.conf ??????)

```
fwconsole m
```

FreePBX

Post-Installation

Set root's password for MySQL

```
mysql_secure_installation
```

Log File Rotation

If this is not done the log files will keep growing indefinitely.
Edit `/etc/logrotate.d/asterisk`

```
/var/spool/mail/asterisk
/var/log/asterisk/*log
/var/log/asterisk/full
/var/log/asterisk/dtmf
/var/log/asterisk/freepbx_dbug
/var/log/asterisk/fail2ban {
 weekly
 missingok
 rotate 4
 #compress
 notifempty
 sharedscripts
 create 0640 asterisk asterisk
 postrotate
 /usr/sbin/asterisk -rx 'logger reload' > /dev/null 2> /dev/null || true
 endscript
 su root root
}
```

TFTP

If you plan to use hardware SIP phones you will probably want to set up TFTP.

```
yum -y install tftp-server
nano /etc/xinetd.d/tftp
```

```
change server_args = -s /var/lib/tftpboot
to server_args = -s /tftpboot
change disable=yes
to disable=no
```

```
mkdir /tftpboot
chmod 777 /tftpboot
systemctl restart xinetd
firewall-cmd --permanent --zone=public --add-port=69/udp
firewall-cmd --reload
```

MPG123

This is used in combination with sox to convert uploaded mp3 files to Asterisk compatible wav files.

```
cd /usr/src
wget http://ufpr.dl.sourceforge.net/project/mpg123/mpg123/1.22.4/mpg123-1.22.4.tar.bz2
tar -xjvf mpg123*
cd mpg123*/
./configure --prefix=/usr --libdir=/usr/lib64 && make && make install && ldconfig
```

Digum addons

To register digium® licenses.

```
cd /usr/src
wget http://downloads.digium.com/pub/register/linux/register
chmod +x register
./register
```

To install the individual addons refer to the README files and ignore the register instructions.

- http://downloads.digium.com/pub/telephony/codec_g729/README
- http://downloads.digium.com/pub/telephony/res_digium_phone/README
- http://downloads.digium.com/pub/telephony/fax/README
- http://downloads.digium.com/pub/telephony/hpec/README

Password protect http access

A simple way to block scanners looking for exploits on apache web servers.

```
mkdir -p /usr/local/apache/passwd
htpasswd -c /usr/local/apache/passwd/wwwpasswd someusername
htpasswd -c /usr/local/apache/passwd/wwwpasswd someotherusername
nano /var/www/html/.htaccess
```

```
# .htaccess files require AllowOverride On in /etc/httpd/conf/httpd.conf
AuthType Basic
AuthName "Restricted Access"
AuthUserFile /usr/local/apache/passwd/wwwpasswd
Require valid-user
```

Alternatively, the above .htaccess config can be added to /etc/httpd/conf/httpd.conf or as a separate file in /etc/httpd/conf.d/ as follows.

```
<Directory /var/www/html>
AuthType Basic
AuthName "Restricted Area"
AuthUserFile /usr/local/apache/passwd/wwwpasswd
Require valid-user
</Directory>
```

Whitelist protect http access

If http access is only required from certain IP addresses.
NOTE: Apache 2.4 ????????
Edit `/etc/httpd/conf.d/whitelist.conf`

```
<Location />
 <RequireAny>
 ## Uncomment the following line to disable the whitelist
 #Require all granted
 Require ip x.x.x.x
 Require ip x.x.x.x x.x.x.x x.x.x.x
 Require ip x.x
 Require ip x.x.x.0/255.255.255.0
 Require host somedomain.com
 #
 ## See http://httpd.apache.org/docs/2.4/mod/mod_authz_host.html for more examples
```

```
 #
 </RequireAny>
</Location>
```

?? Apache ??
NOTE???????? AllowOverride All ??
.htaccess?

```
order deny,allow
deny from all
# Alang's IPs
allow from 123.123.123.1
allow from 111.222.222.2
allow from 192.168.99.
```

G.729 Codec

- https://www.asterisk.org/products/add-ons/g729-codec/
- http://asterisk.hosting.lv/

FreePBX

OSS Endpoint Manager

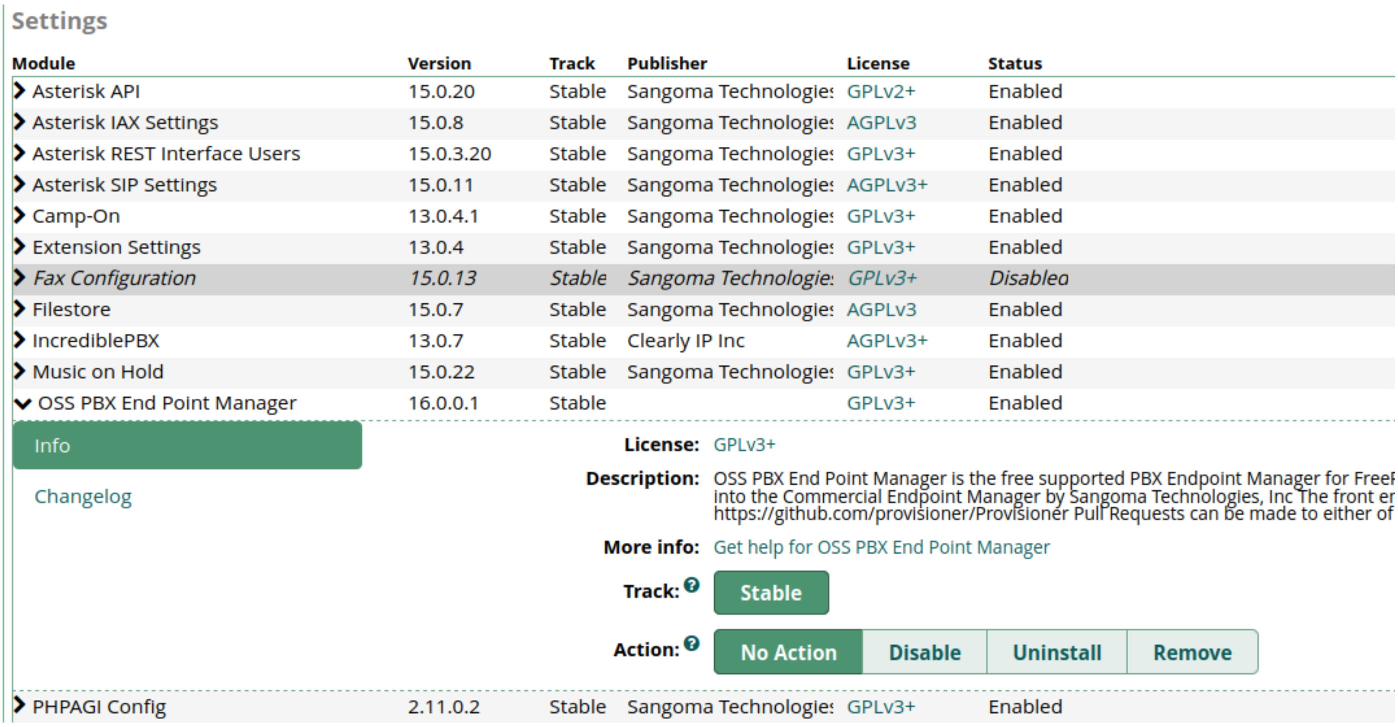
Links

- Github: https://github.com/billsimon/endpointman
- Doc: https://wiki.freepbx.org/display/FPG/OSS+End+Point+Manager
- EPM-Supported Devices
- Introducing OSS Endpoint Manager for FreePBX 16 & Incredible PBX 2027 – Nerd Vittles

Installation

Incredible PBX 2027

```
cd /var/www/html/admin/modules
wget http://incrediblepbx.com/ossepm16.tgz
tar zxvf ossepm16.tgz
rm -f ossepm16.tgz
rm -f /tmp/*
fwconsole ma install endpointman
fwconsole reload
```



Configuration

Package Server

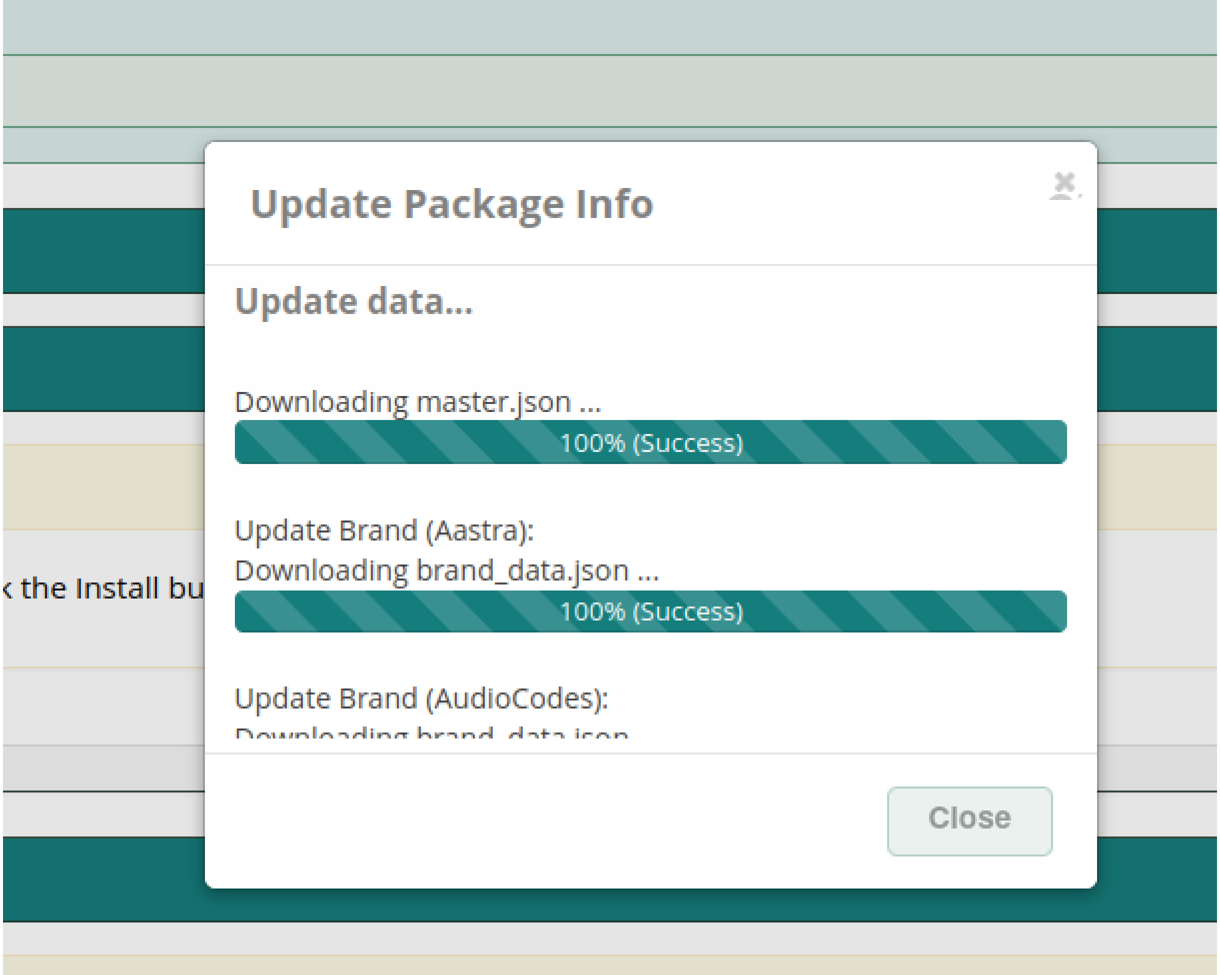
FreePBX GUI > Settings > OSS Endpoint Manager > Settings

- Package Server: `http://provision.lol/`

FreePBX GUI > Settings > OSS Endpoint Manager > Package Manager

- Click the Check for Update

> ??????????????????????

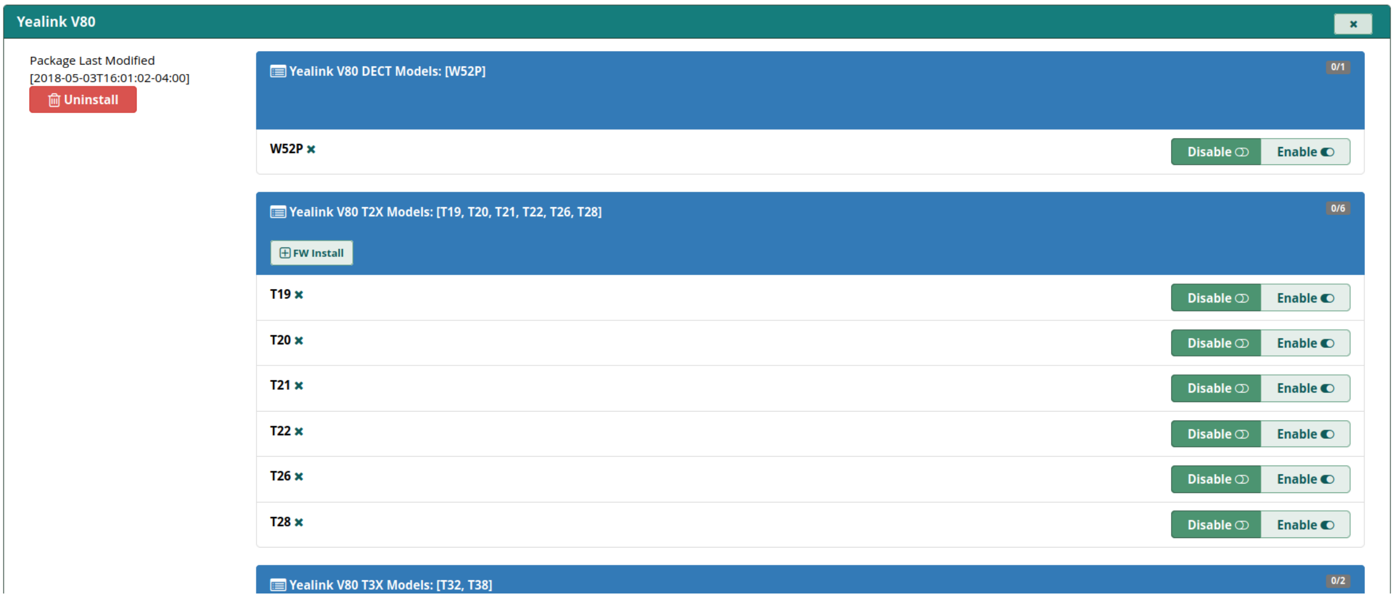


Additional brands (Grandstream & Yeallink V80)

- Download: [SourceForge]

FreePBX GUI > Settings > OSS Endpoint Manager > Settings > Package Import/Export

- Brand Package: `grandstream.tgz` `yealinkv80.tgz`



IP & NTP & Type

FreePBX GUI > Settings > OSS Endpoint Manager > Settings

- IP address of phone server: <server-ip-addr>
- Internal IP address of phone server: <server-ip-addr>
- Configuration Type: Web (HTTP)
- Time Zone: Asia/Taipei
- Time Server: `tw.pool.ntp.org`

> ??????? Settings ?????? Template Editor?????????? Extension Mapping??? Selected Phone Options ?? Global Phone Options??? Rebuild???????????????????

Extension Provisioning

Add Device: Linksys PAP2T

FreePBX GUI > Settings > OSS Endpoint Manager > Package Manager

- Cisco/Linksys - PAP2T : Enable

Create Template: my-pap2t

> ??????????? PAP2T ??????????????????? my-pap2t?

> ????? template ???????????? Extension Mapping ???????????? Save?????????????

FreePBX GUI > Settings > OSS Endpoint Manager > Template Manager

- Template Name: my-pap2t
- Product Select: Linksys/Cisco
- Clone Template From: PAP2T

Edit the template: my-pap2t

- Profile Resync: 3600 (?????? `Resync_Periodic` ??????????????????)
- Enable Webserver: Yes
- Enable Webserver Admin: Yes
- Administrator Password: <set-your-password>
- User Password: <set-your-password>

> ????? template ?????? Edit Global Setting Overrides???????? template?????? ?? template???? template ????????

> ???template ??????? (`spa$mac.xml`) ????????????????????? HTTP ????????? `http://freepbx-ip-addr/provisioning/p.php/spaxxxxxxxx.xml`?xxxxxxx ???? MAC address ??????????????????

Extension Mapping

FreePBX GUI > Settings > OSS Endpoint Manager > Extension Mapping

- MAC Address: <pap2t-mac-addr>
- IPEI: <blank>
- Brand: Cisco/Linksys
- Model: PAP2T
- Line: 1
- Extension Number: <select-your-extension>
- Template: my-pap2t

PAP2T ??

?? PAP2T ???? (advanced view) > Provisioning

- Provision Enable: yes
- Profile Rule: `http://<freepbx-ip-addr>/provisioning/p.php/spa$MA.xml`

???????????????

Provisioning Template Files

SPA-3102/PAP2T

File: spa$mac.xml

???????? `/var/www/html/admin/modules/_ep_phone_modules/endpoint/cisco/linksysata/`

?????

1. ???????????
2. ?? LAN ?? DHCP?SPA3102 ?????
3. ?????????????????? tftp ???

SIP Response Codes

180 v.s. 183

Code 180 and 183 are both SIP response codes used to indicate the progress of a call. While they may seem similar, they have distinct differences in their meanings and usage.

Code 180: Ringing

- The 180 Ringing response code indicates that the called party's device is being alerted or ringing.
- It is typically sent by the called party's User Agent (UA) to indicate that the call is being presented to the user.
- The 180 response code is often sent in response to an INVITE request, indicating that the called party's phone is ringing.
- The calling party may receive multiple 180 responses if the call is forwarded to multiple destinations.

Code 183: Session Progress

- The 183 Session Progress response code indicates that the call is in progress, but the called party has not yet answered.
- It is typically sent by the called party's UA to indicate that the call is being connected or processed, but the called party has not yet accepted the call.
- The 183 response code is often used to indicate that the call is being connected to a voicemail system, an IVR, or a queue.
- Unlike 180, the 183 response code does not necessarily imply that the called party's phone is ringing.

Key differences:

- **Ringing vs. Call Progress**: 180 specifically indicates that the called party's phone is ringing, while 183 indicates that the call is in progress, but the called party has not yet answered.
- **Device Alerting**: 180 implies that the called party's device is being alerted, whereas 183 does not necessarily imply device alerting.
- **Call State**: 180 indicates that the call is in the "ringing" state, while 183 indicates that the call is in the "session progress" state.

In summary, while both 180 and 183 response codes indicate call progress, 180 specifically indicates that the called party's phone is ringing, whereas 183 indicates that the call is in progress, but the called party has not yet answered.

Predictive Dialer

????????

?????

- (??)????????
- ????????
- ??????
- CRM ????????

????

- ICTBroadcast
  - ?????????? 50 channels ????? *(?????????????)*
  - ?????????? Asterisk ?????????
- WombatDialer
  - ????????? 2 channel ????? *(?????????????)*
  - ??????????? Asterisk ?????????
- SIP Caller (Cloud Hosted)
  - ????? Channel ???????
  - ???????????????????
- XactDialer
  - ??????
  - ?????FreePBX ?????

WombatDialer Setup

- Doc: docs.loway.ch :: Loway Documentation Center
- FAQ: Frequently Asked Questions | WombatDialer
- Manual Install: https://docs.loway.ch/WombatDialer/050_Sysadmin.html
- GitHub: https://github.com/Loway/OpenWombatDialerAddOns

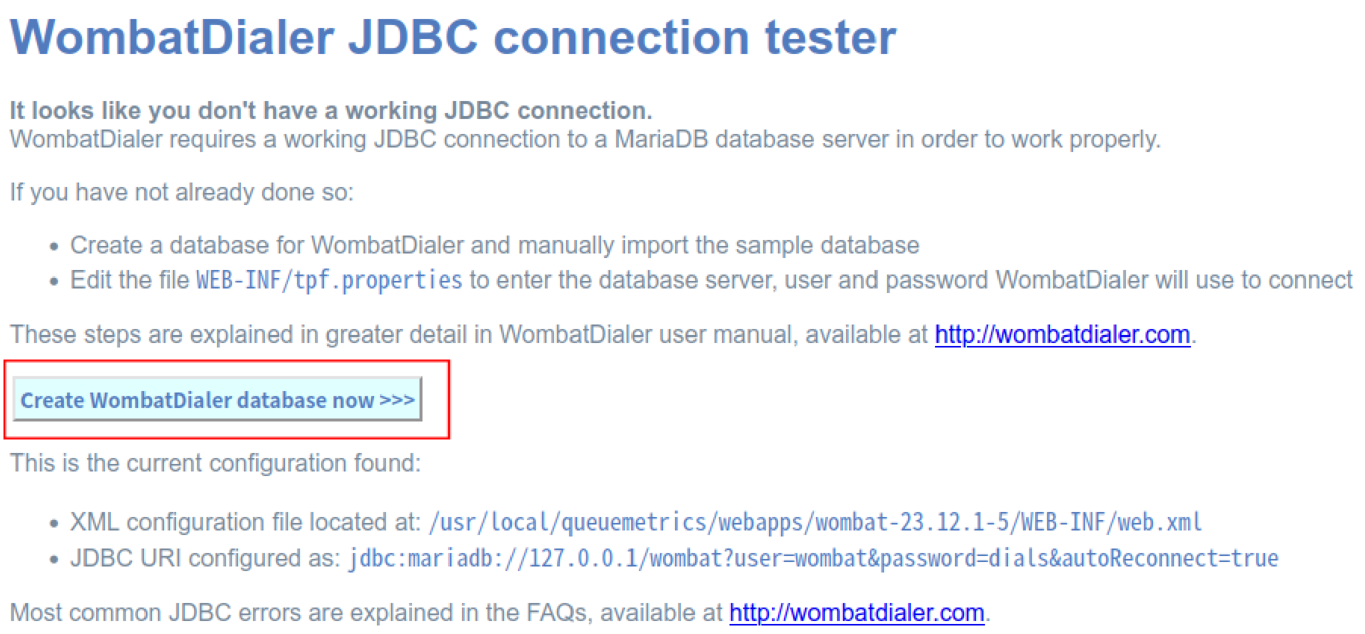
> It looks like you don't have a working JDBC connection.
> WombatDialer requires a working JDBC connection to a MariaDB database server in order to work properly.

> If you have not already done so:
>
>   - Create a database for WombatDialer and manually import the sample database
>   - Edit the file WEB-INF/tpf.properties to enter the database server, user and password WombatDialer will use to connect

Solution:

??? Maridb ?????????????????????????????

- ?????
  - ????? `JDBC_URL=jdbc:mariadb://127.0.0.1/wombat?user=wombat&password=dials&autoReconnect=true`
  - ???? `/usr/local/queuemetrics/webapps/wombat-23.12.1-5/WEB-INF/tpf.properties`
- ??????? `/usr/local/queuemetrics/webapps/wombat-23.12.1-5/WEB-INF/web.xml`
- ?????demoadmin / demo , demouser / demo



chan_mobile

Set Up Bluetooth on Linux

? Gempro Wireless VoIP Gateway ??

Install packages

```
# For RedHat
yum install bluez bluez-tools

# For Debian/Ubuntu
apt-get install bluetooth bluez
```

Service `bluetooth`

```
# For RedHat
systemctl restart bluetooth
systemctl status bluetooth
```

Pair the BT devices

> ?????????????? Gempro Gateway ?????

```
bluetoothctl

[bluetooth]#
Controller 00:1A:7D:DA:71:13
	Name: freepbx.sangoma.local
	Alias: freepbx.sangoma.local
	Class: 0x000104
	Powered: no
	Discoverable: no
	Pairable: no
```

```
	UUID: Generic Attribute Profile (00001801-0000-1000-8000-00805f9b34fb)
	UUID: A/V Remote Control        (0000110e-0000-1000-8000-00805f9b34fb)
	UUID: PnP Information           (00001200-0000-1000-8000-00805f9b34fb)
	UUID: Generic Access Profile    (00001800-0000-1000-8000-00805f9b34fb)
	UUID: A/V Remote Control Target (0000110c-0000-1000-8000-00805f9b34fb)
	Modalias: usb:v1D6Bp0246d052C
	Discovering: no

[bluetooth]# power on
[bluetooth]# agent on
[bluetooth]# scan on
[bluetooth]# devices
Device 00:19:5D:3E:01:77 GP-712-1
Device 00:19:5D:24:C6:98 GP-712-2

[bluetooth]# pair 00:19:5D:3E:01:77
...
...
[agent] Enter PIN code: 0003
[CHG] Device 00:19:5D:3E:01:77 UUIDs: 00001108-0000-1000-8000-00805f9b34fb
[CHG] Device 00:19:5D:3E:01:77 UUIDs: 0000111e-0000-1000-8000-00805f9b34fb
[CHG] Device 00:19:5D:3E:01:77 ServicesResolved: yes
[CHG] Device 00:19:5D:3E:01:77 Paired: yes
Pairing successful

[bluetooth]#
[bluetooth]# paired-devices
Device 00:19:5D:3E:01:77 GP-712-1
```

Set Up Asterisk

```
[root@freepbx ~]# asterisk -rx "module show like chan_mobile"


Module                         Description                              Use Count  Status      Support Level
chan_mobile.so                 Bluetooth Mobile Device Channel Driver   0          Not Running      extended
1 modules loaded
```

Create `/etc/asterisk/chan_mobile.conf` :

```
[general]
interval=30             ; Number of seconds between trying to connect to devices.

; The following is a list of adapters we use.
; id must be unique and address is the bdaddr of the adapter from hciconfig.
; Each adapter may only have one device (headset or phone) connected at a time.
; Add an [adapter] entry for each adapter you have.

[adapter]
id=usbbt1
address=00:1A:7D:DA:71:13
;forcemaster=yes	; attempt to force adapter into master mode. default is no.
;alignmentdetection=yes ; enable this if you sometimes get 'white noise' on asterisk side of the call
		       ; its a bug in the bluetooth adapter firmware, enabling this will compensate for it.
		       ; default is no.

; The following is a list of the devices we deal with.
; Every device listed below will be available for calls in and out of Asterisk.
; Each device needs an adapter=xxxx entry which determines which bluetooth adapter is used.
; Use the CLI command 'mobile search' to discover devices.
; Use the CLI command 'mobile show devices' to see device status.
;
; To place a call out through a mobile phone use Dial(Mobile/[device]/NNN.....) or Dial(Mobile/gn/NNN......) in your dialplan.
; To call a headset use Dial(Mobile/[device]).

[GP712P1]
address=00:19:5D:3E:01:77       ; the address of the phone
port=1                          ; the rfcomm port number (from mobile search)
context=incoming-mobile         ; dialplan context for incoming calls
adapter=usbbt1                  ; adapter to use
group=1                         ; this phone is in channel group 1
;sms=no                         ; support SMS, defaults to yes
;nocallsetup=yes                ; set this only if your phone reports that it supports call progress notification, but does not do it. Motorola L6 for example.
```

bluetoothctl (optional)

```
Available commands:
    list
    show [ctrl]
    select <ctrl>
    devices
    paired-devices
    system-alias
    connect <address>
    disconnect
    remove <address>
    power <on/off>
    mode <mode>
    agent <on/off/capability/timeout>
    default-agent
    scan <on/off>
    pairable <on/off>
    pair <address>
    discoverable <on/off>
    info <address>
    menu <command>
    quit
```

Other commands

```
# Listing all known devices
[bluetooth]# devices

# Powering the Bluetooth controller on or off
[bluetooth]# power on

```

```
# Pairing with a device
[bluetooth]# pair <mac_addredd>

# Remove a device
[bluetooth]# remove <mac_address>

# Connecting to a paired device
[bluetooth]# connect <mac_address>

# Disconnecting from a paired device
[bluetooth]# disconnect <mac_address>
```

Ring Strategies

Tutorials

- Ring Group and Follow-Me Ring Strategies (1 of 2) | FreePBX - Let Freedom Ring
- Ring Group and Follow-Me Ring Strategies (2 of 2) | FreePBX - Let Freedom Ring
- Ring Groups Module User Guide - PBX GUI - Sangoma Documentation (atlassian.net)

VitalPBX

VitalPBX is a unified communications PBX system based on Asterisk and Linux (Debian 11). VitalPBX provides a robust and scalable platform, which will allow you to manage your PBX in an easy and intuitive way.

> ???? Starter License ?????????????????????????

- VitalPBX - Advanced PBX System
- Doc: VitalPBX – VitalPBX Wiki
- Forum: VitalPBX Community
- Video: https://www.youtube.com/@VitalPBX/videos