Installing Asterisk 13 from source (Ubuntu 15.04)

Prerequisites

Install Ubuntu 15.04 LTS

If unstalling on a virtual machine, make sure the network adapter is configured in “Bridged” mode.

Enable SSH remote access into your Ubuntu system:

```
sudo apt-get install openssh-server
```

Build and Install iksemel (if you want to support Google Voice)

Prerequisites for iksemel

iksemel requires the gnutls-dev package.

```
sudo apt-get install libgnutls-dev
```

gnutls-config is no longer distributed with Ubuntu, but you can make one as follows (reference: https://code.google.com/p/iksemel/issues/detail?id=29):

```
sudo cat > /usr/bin/libgnutls-config
#!/bin/bash
if [ "$1" == "--version" ]; then
pkg-config --modversion gnutls
else
pkg-config $1 gnutls
fi
^D
sudo chmod 755 /usr/bin/libgnutls-config
```

Download and install iksemel

```
sudo wget https://iksemel.googlecode.com/files/iksemel-1.4.tar.gz
sudo tar xfvz iksemel-1.4.tar.gz
cd iksemel-1.4/
./configure
make
sudo make install
```

Build and Install pjsip

Download pjsip v2.4:
http://www.pjsip.org/download.htm

./configure --prefix=/usr --enable-shared --disable-sound --disable-resample --disable-video --disable-opencore-amr CFLAGS='-O2 -DNDEBUG'
make dep
make
sudo make install
sudo ldconfig
sudo ldconfig -p|grep pj

Build and Install jansson

git clone https://github.com/akheron/jansson.git
autoreconf -i
cd jansson/
autoreconf -i
./configure
make
sudo make install

Build and Install Asterisk 13:

Download Asterisk 13
http://www.asterisk.org/downloads/asterisk/all-asterisk-versions

```
tar xfvz ~/Downloads/asterisk-13-current.tar.gz
cd asterisk-13.3.2/
sudo apt-get install uuid-dev
sudo apt-get install libjansson-dev
sudo apt-get install libncurses-dev
sudo apt-get install libxml2-dev
sudo apt-get install libsqlite3-dev
```

```
sudo apt-get install subversion
./configure
```
`make menuselect` (Note: If you want MP3 support, you need to manually turn on 'format_mp3' on the first page)

```
contrib/scripts/get_mp3_source.sh
make
sudo make install
sudo make samples
sudo make config
sudo make install-logrotate
```

Install Extra Sounds

cd /var/lib/asterisk/sounds
sudo wget http://downloads.asterisk.org/pub/telephony/sounds/asterisk-extra-sounds-en-wav-current.tar.gz
sudo tar xfz asterisk-extra-sounds-en-wav-current.tar.gz
sudo rm -f asterisk-extra-sounds-en-wav-current.tar.gz
# Wideband Audio download
sudo wget http://downloads.asterisk.org/pub/telephony/sounds/asterisk-extra-sounds-en-g722-current.tar.gz
sudo tar xfz asterisk-extra-sounds-en-g722-current.tar.gz
sudo rm -f asterisk-extra-sounds-en-g722-current.tar.gz

Configure
http://<server_ip>:<bindport>/static/docs/index.html
sudo vi /etc/asterisk/http.conf
sudo /etc/init.d/asterisk status
sudo asterisk -r

Build and Install FreePBX:
http://wiki.freepbx.org/display/HTGS/Installing+FreePBX+12+on+Ubuntu+Server+14.04+LTS

```
sudo apt-get install -y build-essential linux-headers-`uname -r` openssh-server apache2
mysql-server\
  mysql-client bison flex php5 php5-curl php5-cli php5-mysql php-pear php-db php5-gd curl sox\
  libncurses5-dev libssl-dev libmysqlclient-dev mpg123 libxml2-dev libnewt-dev sqlite3\
  libsqlite3-dev pkg-config automake libtool autoconf git subversion unixodbc-dev uuid
uuid-dev\
  libasound2-dev libogg-dev libvorbis-dev libcurl4-openssl-dev libical-dev libneon27-dev
libsrtp0-dev\
  libspandsp-dev libiksemel-dev libiksemel-utils libiksemel3
```

Set password for mysql server root user.

```
sudo wget http://mirror.freepbx.org/freepbx-12.0.43.tgz
sudo tar vxfz freepbx-12.0.43.tgz
```

```
sudo -i
useradd -m asterisk
chown asterisk. /var/run/asterisk
chown -R asterisk. /etc/asterisk
chown -R asterisk. /var/{lib,log,spool}/asterisk
chown -R asterisk. /usr/lib/asterisk
rm -rf /var/www/html

sed -i 's/\(^upload_max_filesize = \).*/\120M/' /etc/php5/apache2/php.ini
cp /etc/apache2/apache2.conf /etc/apache2/apache2.conf_orig
sed -i 's/^\(User\|Group\).*/\1 asterisk/' /etc/apache2/apache2.conf
service apache2 restart

export ASTERISK_DB_PW=`dd if=/dev/urandom bs=1 count=32 2>/dev/null | base64 - | cut
-c2-18`
mysqladmin -p -u root create asterisk
mysqladmin -p -u root create asteriskcdrdb
mysql -p -u root -e "GRANT ALL PRIVILEGES ON asterisk.* TO asteriskuser@localhost
IDENTIFIED BY '${ASTERISK_DB_PW}';"
mysql -p -u root -e "GRANT ALL PRIVILEGES ON asteriskcdrdb.* TO asteriskuser@localhost
IDENTIFIED BY '${ASTERISK_DB_PW}';"
mysql -p -u root -e "flush privileges;"

./start_asterisk start
./install_amp --installdb --username=asteriskuser --password=${ASTERISK_DB_PW}
amportal chown
amportal a ma installall
amportal a reload
amportal a ma refreshsignatures
amportal chown
```

Ubuntu Firewall Configuration

```
sudo ufw allow 5060/tcp
sudo ufw allow 5060/udp
sudo ufw allow 5061/tcp
sudo ufw allow 5061/udp
```

Custom Extensions

Map 555-xxxx, etc to PBX extensions 6xxx

```
:/etc/asterisk# cat extensions_custom.conf
[voicemenu-anac-1]
exten = s,1,NoOp(ANI)
exten = s,2,Answer()
exten = s,3,Wait(1)
exten = s,4,SayDigits(555)
exten = s,5,SayDigits(${CALLERID(num)})
exten = s,6,Wait(5)
exten = s,7,SayDigits(555)
exten = s,8,SayDigits(${CALLERID(num)})
exten = s,9,Wait(5)
exten = s,10,Busy()

[from-internal-custom]
exten = _92555599XX,1,Goto(default,60${EXTEN:8},1)
exten = _192555599XX,1,Goto(default,60${EXTEN:9},1)
exten = _55599XX,1,Goto(default,60${EXTEN:5},1)
exten = _925555XXXX,1,Goto(default,${EXTEN:6},1)
exten = _1925555XXXX,1,Goto(default,${EXTEN:7},1)
exten = _555XXXX,1,Goto(default,${EXTEN:3},1)
exten = 9581111,1,Goto(voicemenu-anac-1,s,1)
exten = 7958,1,Goto(voicemenu-anac-1,s,1)
exten = 958,1,Goto(voicemenu-anac-1,s,1)
```

Securing Asterisk and FreePBX

1. Enable .htaccess: http://wiki.freepbx.org/display/F2/Webserver+Overrides
2. Change Asterisk Manager password (FreePBX->Settings->Advanced)

Protect your System

Install fail2ban

```
sudo apt-get install fail2ban
```

https://www.digitalocean.com/community/tutorials/how-to-install-and-use-fail2ban-on-ubuntu-14-04

Install email server and client for Fail2Ban:

```
sudo apt-get install sendmail
sudo apt-get install mailutils
```

Check to make sure fail2ban is working properly:
In one terminal: `sudo tail -f /var/log/fail2ban.log`
In another: sudo `fail2ban-client reload`

Sendmail through gmail

http://linuxconfig.org/configuring-gmail-as-sendmail-email-relay

Google Voice Setup

Install Motif module from FreePBX Admin page:
Module Admin, select the “Unsupported” repository and scroll down to install “Google Voice/Chan Motif” from the “Connectivity” section.

Enable Less Secure Apps

https://www.google.com/settings/u/1/security/lesssecureapps

Make sure you have a Google Voice number associated with your Google Voice account. If you don’t, the caller ID will come up as “GOOGLE,GUY”

Setting up Asterisk to Work with ACTS Payphones

Common Bell System payphones are of the Automated Coin Toll Service (ACTS) type, and require a special coin line in order for the phone to collect money. Joshua Stein created an interesting project on GitHub by jcs that allows Asterisk to recognize the coin tones. https://github.com/jcs/payphone. I started my own project to add full coin control using a hardware interface between Asterisk and the payphone: https://github.com/hharte/1dcoinctrl

Install Asterisk-perl

Asterisk-perl 1.03 is needed by the AGI scripts in order to support Perl. Information about Asterisk-perl can be found here: http://asterisk.gnuinter.net/

Install as follows:

```
wget http://asterisk.gnuinter.net/files/asterisk-perl-1.03.tar.gz
tar xfvz asterisk-perl-1.03.tar.gz
cd asterisk-perl-1.03/
perl Makefile.PL
make all
sudo make install
```

Modifying Asterisk to Recognize Coin Tones

Patch per:
https://github.com/hharte/1dcoinctrl/blob/master/asterisk/main/dsp.c-patch

Then rebuild Asterisk:

```
make
sudo make install
sudo service asterisk restart
```

Installing Payphone AGI Script

```
git clone https://github.com/hharte/1dcoinctrl
cd 1dcoinctrl
sudo cp asterisk/agi-bin/payphone.agi /var/lib/asterisk/agi-bin
sudo cp asterisk/agi-bin/invalid_number.sln /var/lib/asterisk/agi-bin
sudo cp asterisk/agi-bin/not_deposited.sln /var/lib/asterisk/agi-bin
sudo cp asterisk/sounds/* /var/lib/asterisk/agi-bin/
sudo chown asterisk.asterisk /var/lib/asterisk/agi-bin/payphone.agi
```

```
sudo chown asterisk.asterisk
/var/lib/asterisk/agi-bin/invalid_number.sln
sudo chown asterisk.asterisk
/var/lib/asterisk/agi-bin/not_deposited.sln
sudo chmod 744 /var/lib/asterisk/agi-bin/payphone.agi
sudo chown asterisk.asterisk /var/lib/asterisk/agi-bin/*.wav
```

in /etc/asterisk/extensions_custom.conf:
[payphone-totalizer]
exten => 6200,1,Answer
exten => 6200,2,AGI(payphone.agi)
exten => 6200,3,Hangup

#include pjsip.endpoint_custom.conf

[6001]
type=endpoint
aors=6001
auth=6001-auth
disallow=all
allow=ulaw,alaw
dtmf_mode=inband
context=payphone-totalizer
callerid=device <6001>
mailboxes=6001@device
use_avpf=no
ice_support=no
media_use_received_transport=no
trust_id_inbound=yes
rtp_symmetric=yes
rewrite_contact=yes

Coin Collect / Refund Controller Interface

In order to actually collect and refund coins deposited into the coin hopper, a hardware interface is required to supply +130VDC and -130VDC to the coin relay at the appropriate times.

1D Coin Controller Project

https://github.com/hharte/1dcoinctrl

Coin Control details:

+/-130VDC minimum 41mA (Collect/Refund)
+/-48VDC maximum 20mA (Stuck Coin/Initial Rate Tests)

+48VDC also for 5c overtime test (same as stuck coin test.)

Coin Relay operate pulse is 600ms.
After coin control function is complete, the system will make one recycle attempt if coin ground (stuck coin test) is still present.