Installing Asterisk 13 from source (Ubuntu 14.04 LTS)

Build and Install pjsip

Download pjsip v2.3:
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 Asterisk 13:

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

```
tar xfvz ~/Downloads/asterisk-13-current.tar.gz
cd asterisk-13.1.0/
sudo apt-get install uuid-dev
sudo apt-get install libjansson-dev
./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
```

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

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)

Sendmail through gmail

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

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
cp -a asterisk-perl-1.03/lib/Asterisk /usr/lib/perl5/
cp -a asterisk-perl-1.03/lib/Asterisk.pm /usr/lib/perl5
```

Modifying Asterisk to Recognize Coin Tones

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

Then rebuild Asterisk:

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

Installing Payphone AGI Script

```
wget https://github.com/jcs/payphone/archive/master.zip
unzip master.zip
cp payphone-master/payphone.agi /var/lib/asterisk/agi-bin
cp payphone-master/invalid_number.sln /var/lib/asterisk/agi-bin
cp payphone-master/not_deposited.sln /var/lib/asterisk/agi-bin
chown asterisk.asterisk /var/lib/asterisk/agi-bin/payphone.agi
chown asterisk.asterisk /var/lib/asterisk/agi-bin/invalid_number.sln
chown asterisk.asterisk /var/lib/asterisk/agi-bin/not_deposited.sln
```

```
chmod 744 /var/lib/asterisk/agi-bin/payphone.agi
```

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.