Voice over IP using Asterisk (*)
What is Asterisk?

- Asterisk (http://www.asterisk.org) is an Open Source Private Branch Exchange (PBX) and Interactive Voice Response (IVR) system.
- Asterisk was written in C originally by Mark Spencer.
- Asterisk requires Linux for full support, BSD (lacks some (T1/E1, ISDN)) and Windows (lacks all) hardware support.
Asterisk Features

- Voicemail
- Interactive Voice Response (Press 1 for…)
- Auto Attendant
- Call Forwarding
- Call Queues
- Call Detail Reports
- Agents (Support, sales, etc)
- Directories
- Scalability
- Many others!
Asterisk Hardware for VoIP

- Server (600 MHz +) *
- Internet Connection (Cable/DSL or faster)
- E1/T1 cards (only if you are a VoIP provider)
  - Single or Quad port

Server speed requirement depends on amount of calls and codecs.
More Hardware

- **ATA (Analog Telephone Adapter)**
  - “External FXS”
  - Cisco, IAXy, Sipura, etc.
  - Linksys PAP2-NA (Limited Production Currently)
  - Use a normal analog phone

- **IP Phones**
  - Cisco 7960 / 7940
  - Grandstream
  - Software phones
  - Many others
Service Provider

- VoIP Termination Provider (connection to the public phone system).
  - Nufone
  - Voicepulse
  - Broadvoice
  - VoIPJet
  - Many others

Vonage is a VoIP provider, but they no longer fully support * and restrict setup and configuration.
Asterisk Protocols

- IAX (Inter-Asterisk eXchange) *
- SIP (Session Initiation Protocol)
- H.323 **
- MGCP (Media Gateway Control Protocol)

* Most popular
**Not fully supported (AKA you support it yourself), evil...
Asterisk Codecs

- GSM
- ILBC (Needs a lot of CPU)
- **G.729** (available through purchase of commercial license(s))
- **G.711** (ALaw/ULaw)
- G.723.1 (pass through)
- Linear
- ADPCM
- G.726
- LPC-10
- Speex
- MP3 (decode only)
Asterisk IAX Configuration

[general]
bandwidth=low
register => USER:PASSWORD@switch-1.nufone.net
register => USER:PASSWORD@switch-2.nufone.net
register => USER:PASSWORD@iaxtel.com
allow=gsm

[NuFone]
type=peer
host=switch-1.nufone.net
context=from-nufone
secret=PASSWORD
Asterisk SIP Configuration

[general]

port = 5060 ; Port to bind to (SIP is 5060)
bindaddr = 192.217.247.64 ; Address to bind to

disallow=all ; Disallow all codecs
allow=g729
allow=gsm
allow=ulaw
allow=alaw
[1100]

type=friend
username=1100
secret=PASSWORD
host=dynamic
canreinvite=no
context=administration
mailbox=1100@local
callerid="Ryan Brown" <1100>
Asterisk Extensions Config

[globals]
PHONENUM=8776825483
PHONE1=1100
PHONENAME=The Burgh Live, LLC
IAXINFO=USER:PASSWORD

[from-nufone]
exten => 1899,1,Goto(default,s,1)
exten => 8776825483,1,Goto(default,s,1)
Extensions Config Cont.

[default]
include => autoattend
include => internal-ext
[internal-ext]
include => default

; The Burgh Live
exten => 1100,1,Answer
exten => 1100,2,Dial(SIP/1100,20|m)
exten => 1100,3,Voicemail(u1100)
exten => 1100,104,Voicemail(b1100)
exten => 1100,105,Hangup
Extensions Config Cont.

[tollfree-out]
exten => _9.,1,SetCallerID(${PHONENUM})
exten => _9.,2,SetCIDName(${PHONENAME})
exten => _91866NXXXXXXX,3,Dial,IAX2/tblive@NuFone/${EXTEN:1}
exten => _91877NXXXXXXX,3,Dial,IAX2/tblive@NuFone/${EXTEN:1}
exten => _91888NXXXXXXX,3,Dial,IAX2/tblive@NuFone/${EXTEN:1}
exten => _91800NXXXXXXX,3,Dial,IAX2/tblive@NuFone/${EXTEN:1}

[nufone-out]
exten => _9.,1,SetCallerID(${PHONENUM})
exten => _9.,2,SetCIDName(${PHONENAME})
exten => _91NXXNXXXXXXX,3,Dial,IAX2/tblive@NuFone/${EXTEN:1}
exten => _9011N.,3,Dial,IAX2/tblive@NuFone/${EXTEN:1}
Asterisk Voicemail Config

[general]
format=gsm|wav49|wav ; Default formats for writing Voicemail

; Who the e-mail notification should appear to come from
serveremail=voicemail@sage.pit.tblive.com

; Should the email contain the voicemail as an attachment
attach=yes

[local]
; format: exten, password, name, email address voicemail msgs
1100 => PASSWORD,Ryan Brown,rbrown@tblive.com
Many Other Config Files

- Music on Hold
- Call Detail Reports
- Festival
- Meet Me
- Parking
- Queues
- Etc…
## Cost Savings vs. POTS

<table>
<thead>
<tr>
<th>Feature</th>
<th>VoIP (Nufone)</th>
<th>POTS (Verizon)</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Incoming Calls</strong></td>
<td>$7.50 (unlimited, MI #) 2 cents / min 800 #</td>
<td>$15 / month (unlimited)</td>
</tr>
<tr>
<td><strong>Local Calling</strong></td>
<td>2 cents / min (6 second)</td>
<td>Included in Plan</td>
</tr>
<tr>
<td><strong>Long Distance</strong></td>
<td>2 cents / min (6 second)</td>
<td>5 cents / min (1 min)</td>
</tr>
<tr>
<td><strong>Monthly Fee</strong></td>
<td>FREE</td>
<td>$15</td>
</tr>
<tr>
<td><strong>Taxes</strong></td>
<td>NO</td>
<td>YES</td>
</tr>
<tr>
<td><strong>Tariffs</strong></td>
<td>NO</td>
<td>YES</td>
</tr>
<tr>
<td><strong>911</strong></td>
<td>NO</td>
<td>YES</td>
</tr>
</tbody>
</table>
VoIP / Asterisk Disadvantages

- NAT (IAX works, SIP hates it, but can be made to work)
- Can’t control Internet lag
- Server must be up and Asterisk running
- Power reliability (POTS is up when power is out)
- 911
Other Things * Can Do

- Interface with analog POTS via PCI cards
- Caller ID spoofing (Asterisk makes it very easy as shown in extensions.conf, your provider must support this)
- Expandability, if Asterisk can't do it now - you can probably make it do it.
- Traditional phone system, FXO (Phone line) / FXS (Phone) PCI cards
Asterisk Resources

- [http://www.asterisk.org](http://www.asterisk.org)
- [http://www.voip-info.org](http://www.voip-info.org)
- [http://www.automated.it/guidetoasterisk.htm](http://www.automated.it/guidetoasterisk.htm)
Asterisk Questions / Comments?