Ryan Brown
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The Burgh Live, LLC

Voice over IP using Asterisk (*)

What is Asterisk?

- * (http://www.asterisk.org) is an Open Source Private Branch Exchange (PBX) and Interactive Voice Response (IVR) system.
- * was written in C originally by Mark Spencer.
- * 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
```

SIP Configuration Cont.

```
[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 => _91866NXXXXXX,3,Dial,IAX2/tblive@NuFone/${EXTEN:1}
exten => _91877NXXXXXX,3,Dial,IAX2/tblive@NuFone/${EXTEN:1}
exten => _91888NXXXXXX,3,Dial,IAX2/tblive@NuFone/${EXTEN:1}
exten => _91800NXXXXXX,3,Dial,IAX2/tblive@NuFone/${EXTEN:1}

[nufone-out]
exten => _9.,1,SetCallerID(${PHONENUM})
exten => _9.,2,SetCIDName(${PHONENAME})
exten => _91NXXNXXXXXX,3,Dial,IAX2/tblive@NuFone/${EXTEN:1}
exten => _9011N.,3,Dial,IAX2/tblive@NuFone/${EXTEN:1}
```

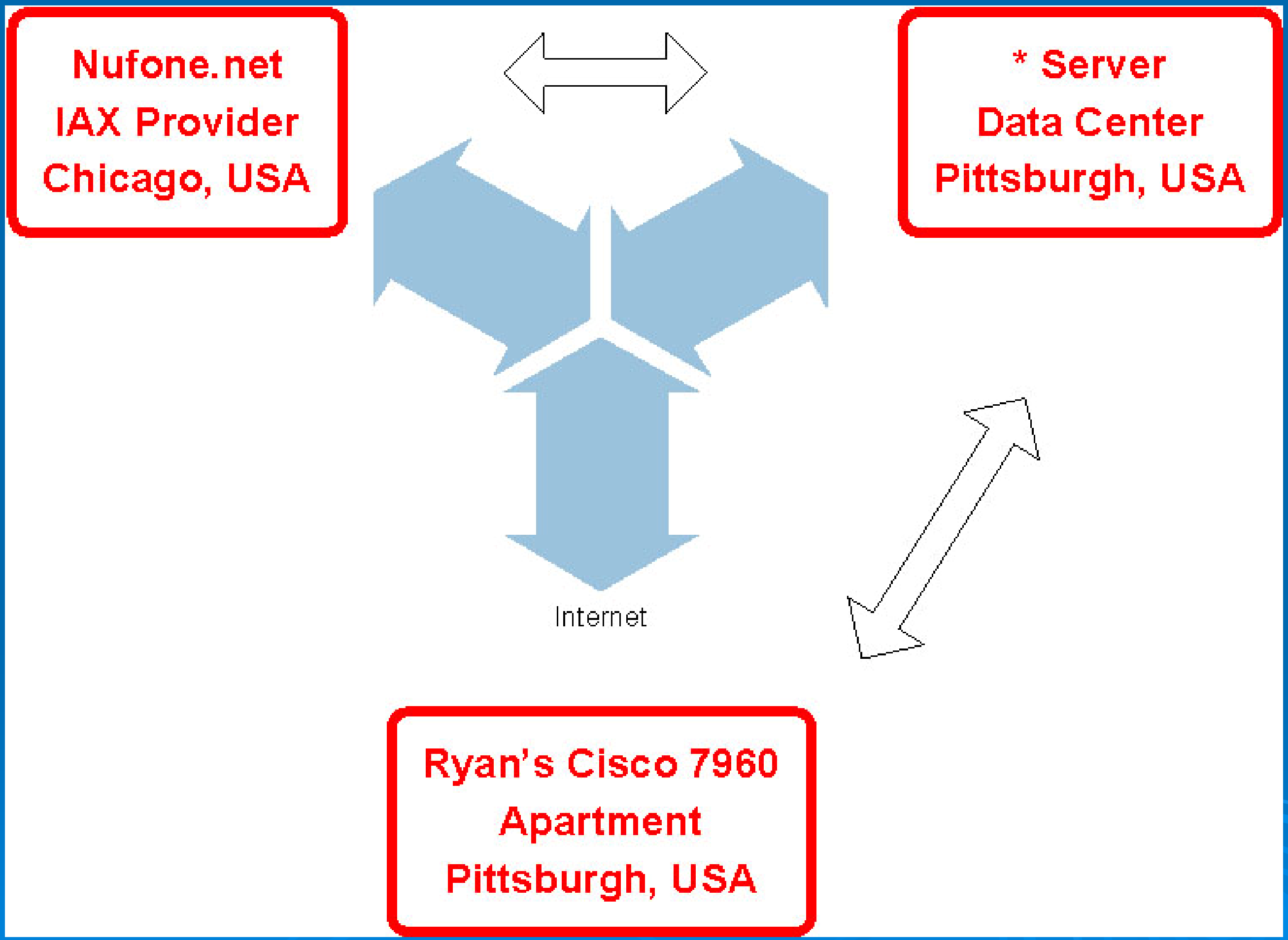
Asterisk Voicemail Config

```
[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…

Asterisk in Production



Cost Savings vs. POTS

Feature	VoIP (Nufone)	POTS (Verizon)
Incoming Calls	$7.50 (unlimited, MI #) 2 cents / min 800 #	$15 / month (unlimited)
Local Calling	2 cents / min (6 second)	Included in Plan
Long Distance	**2 cents / min (6 second)**	5 cents / min (1 min)
Monthly Fee	**FREE**	$15
Taxes	**NO**	YES
Tariffs	**NO**	YES
911	NO	**YES**

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.voip-info.org
- http://www.automated.it/guidetoasterisk.htm
- http://www.voip-news.com/

Asterisk Questions / Comments?