VoIP
What it can do for you

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Apology

- LCA Payment gateway
- http://justblamepia.com
- Pay Now!!
- We need volunteers!

Background

- Using VoIP for 5 years
- Basic Setup at home
- Mobile Least Cost Routing
- IVR for Beagle
- Beagle Distributed Call Centre
- Throw away the copper

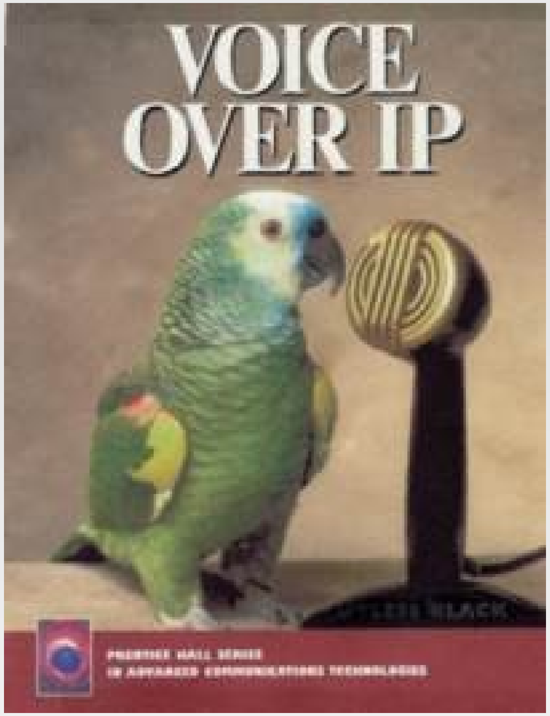
FXO and FXS

- What do they mean?
  - who cares?
- Some ports are for phones
- Some ports are for PSTN/PRI/BRI
- They are usually well labeled
- http://www.patton.com/technotes/fxs_fxo.pdf

What we will cover

- VoIP Basics
- VoIP Hardware
- VoIP Software
- VoIP Providers
- Beagle Case Study
- Demo

VoIP Basics



VoIP over IPoAC

Survey

- How many of you have used VoIP?
  - without knowing it (Calling cards)
- Use a VoIP provider?
- Have an Analogue Telephone Adapter?
- Use something like Asterisk?

What is VoIP

- Voice over Internet Protocol
- Digitise the sound put it in an IP Packet
- Usually UDP
- Discrete packets rather than circuits
- Latency sensitive
- Jitter Sensitive

Standards

- SIP
  - Session Initiation Protocol
  - Similar to HTTP
  - Used for signaling
  - Voice goes in UDP RTP packets
- IAX
  - Inter Asterisk Exchange
  - Asterisk Specific
  - NAT Safe

Standards...

- Others
  - H.323
  - Skinny
- NAT unfriendly
  - STUN

SIP Packet

INVITE sip:7038@10.38.38.9 SIP/2.0
Via: SIP/2.0/UDP 10.38.38.61:5060;branch=z9hG4bK-8e2d7763
From: John Ferlito Cordless <sip:jf_portable@10.38.38.9>; tag=78cb213254d4f38bo0
To: <sip:7038@10.38.38.9>
Call-ID: e5cee37e-b3c535af@10.38.38.61
CSeq: 101 INVITE
Max-Forwards: 70
Contact: John Ferlito Cordless <sip:jf_portable@10.38.38.61:5060>
Expires: 240
User-Agent: Sipura/SPA2000-2.0.10(e)
Content-Length: 424
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura
Content-Type: application/sdp

v=0 o=- 92586134 92586134 IN IP4 10.38.38.61 s=- c=IN IP4 10.38.38.61
t=0 0 m=audio 16444 RTP/AVP 0 2 4 8 18 96 97 98 100 101
a=rtpmap:0 PCMU/8000 a=rtpmap:2 G726-32/8000 a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000 a=rtpmap:18 G729a/8000 a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000 a=rtpmap:98 G726-16/8000 a=rtpmap:10NSE/8000
a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-15 a=ptime:30 a=sendrecv.

VoIP Packets

- Small packets
  - Massive overhead
  - 8kbps codec turns into 31.2kbps
- Latency (300ms Max)
- Jitter (50ms)
- http://www.connect802.com/voip_bandwidth.php

```
11:58:25.041186 IP 10.38.38.61.16448 > 10.38.38.9.15034: UDP, length: 252
11:58:25.073074 IP 10.38.38.61.16448 > 10.38.38.9.15034: UDP, length: 252
11:58:25.102809 IP 10.38.38.61.16448 > 10.38.38.9.15034: UDP, length: 252
11:58:25.132563 IP 10.38.38.61.16448 > 10.38.38.9.15034: UDP, length: 252
11:58:25.162312 IP 10.38.38.61.16448 > 10.38.38.9.15034: UDP, length: 252
11:58:25.192064 IP 10.38.38.61.16448 > 10.38.38.9.15034: UDP, length: 252
```

Codecs - G.711

- uLaw or aLaw ie PCM/WAV like
- 64kbps Raw
- 87.2kbps with overhead
- Full Quality
- Low CPU
- High Bandwidth (need >=128kbps)

Codecs - G.729

- Patented
- License US$10 per channel
- Built in to phone/ATAs
- Industry Standard
- 8kbps Raw -> 31.2kbps
- Good Quality

Codecs - GSM

- Proprietary
- Royalty Free
- Built in to phone/ATAs
- Industry Standard
- 8kbps Raw

Codecs - Others

- iLBC
  - Good for modems
- Speex
  - Variable Bit Rate

Anecdote

- Talking to Jeff on the phone
- “So you know what would be a really good demo...”
- VoIP call drops out
- Seriously this happened this morning.
- I choose to blame the PSTN connection at Jeff's end :)

VoIP Hardware

ATA's

- Analogue Telephone Adapters
- Turn an Analogue handset into a VoIP phone
- More expensive ones turn a PSTN line into something a VoIP phone can talk to as well

Sipura/Linksys [23]000

- SPA2000 – 2 Phones
- SPA3000 – 1 Phone, 1 PSTN
- $100-$150
- Web Interface
- Mass deployment via DHCP/BOOTP

VoIP Handset

- Gives you more functionality
- More buttons and features at touch of a button
- Better quality as 100% digital end to end

Snom 300 series

- Great VoIP handset
- $300 - $500 depending on model
- Multiple buttons
- Flashing Lights
- No backlit LCD
- Mass deployable

Line Cards

- Plug into POTS
- PSTN
- BRI (Onramp 2, OnRamp Home)
- PRI (10+ Channels)

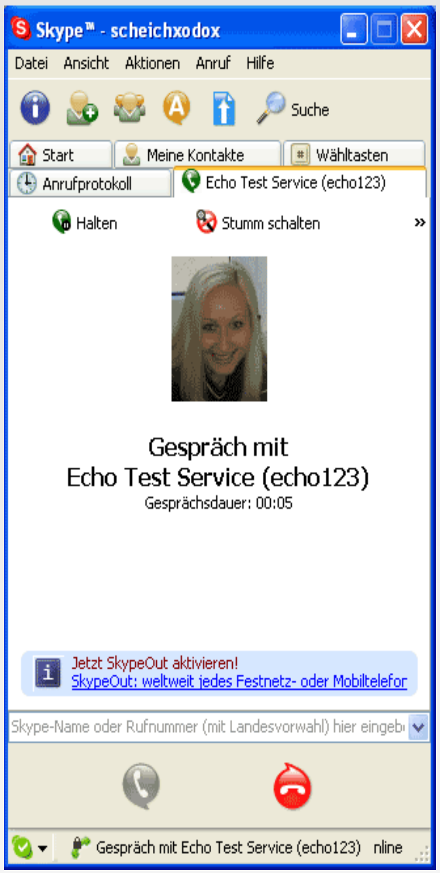
Line Cards Examples

- PSTN
  - Digium x100P clones
- ISDN
  - Fritz
    - Ebay == Cheap
    - No AusTick
  - HiSax/Tiger/NetJet (Traverse)
- PRI
  - Digium
  - Eicon

Mobile PODs

- Used to plug a mobile into a VoIP system to use cheap mobile Caps
- Some Providers do this
- Bluetooth and asterisk sort of work
- Use an ATA type method

VoIP Software



Asterisk.
The Open Source PBX

Softphones

- Run as an application
- Use you standard speakers and mic
- Get a headset
- echo can be a problem
- Hard phones have echo cancellation in hardware

Soft Phones

- xten
- sjphone
- gnophone
- iaxcomm
- linphone
- SNOM Softphone

PBX

- Perform all the functions of a real PBX
- Extension
- Voicemail
- Transfer calls
- Conferencing
- Call Monitoring
- Call Parking
- On Hold Music
- etc

PBXs

- Asterisk
- Asterisk@Home
- FreeSwitch
- sipX
- SER – SIP Express Router

VoIP Providers

What they do?

- Provide Many things
- Outbound calls
- Direct in Dial (DID)
- Partial to Complete PBX solutions

What to look for?

- Supported Codecs
- Latency
- Price, Flagfall
- Protocols
- Where they have DIDs
- Number concurrent calls

My preferences

- SIPMe for outbound
  - Cheapest mobile rates
- OzTell for inbound
  - Cheapest per number DID
- Other places to look
  - whirlpool
  - your ISP
  - Bundling

Beagle Case Study

Keith

"VoIP you say, maybe we should sell that sucker"

Phase 1

- Second ISDN number rang my desk phone
- All out bound calls went via ISDN
- Could have just used an analogue phone

Phase 2

- Setup a 3 level menu IVR
- Menus gave some basic help
- Phone on my desk rang
- Voicemail
- Multiple mailboxes for Sales, Support, etc
- Later voicemail emailed direct to Request Tracker

Phase 3

- Distributed Call Centre
  - My Desk
  - My Sisters
  - Mum's kitchen
  - 2 Suburbs away
  - Ireland
  - Brazil

Phase 4

- Mobile calls where costing too much
- Use a mobile POD with Optus SIM card
- Didn't work to well due to reception

Phase 5

- Switch to outbound provider SIPMe
- Saved on STD and mobile calls
- No drop in quality
- Can now accept more than 2 calls at once

Phase 6 (Future)

- Remove reliance on copper and Telstra
- Move Asterisk into an IDC
- Point 1300 number at VoIP provider indials
- Indial per state to reduce 1300 number costs

Benefits

- Everyone has a phone
- Calls between offices are free
- Conferencing
  - Hardware phones make this easy
- LCA emergency number uses this setup and only rings during AU business hours
- Hands off to mobile phones

Demo

Hopefully time for this.
Although demos never work anyway.
See I predicted it so I have an excuse :)

Resources

- http://asterisk.org
- http://voip-info.org
- http://www.connect802.com/voip_bandwidth.php
- http://www.whirlpool.net.au
- http://ozvoip.com

Questions?

Thanks for listening!