VOIP, Linux, and Asterisk
Making Beautiful Voice Together

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POTS World - Today

Incumbent Local Exchange Carrier
Competitive Local Exchange Carrier
Inter-exchange Carrier (Long Distance)
Connections to the Telephone Company

- Analog phone lines

- ISDN – Digital phone lines. Two **B Channels** for voice and one **D Channel** for control

- Primary Rate Interface – Digital phone lines. 23 **B Channels** and one **D Channel**.
Networked World

Central Office

ISP

Internet

Sever

NID

Home Wiring

ADSL Router

Ethernet

Cable Head End

Coaxial Cable

Cable Modem

Ethernet
Crossover Into Voice Over Internet Protocol

- VOIP crosses over between the Internet and the PSTN at several possible locations
- Intraoffice – VOIP phones on the desktop
- Direct Inward Dial – A phone number people can call
- Termination – Calling local or long distance numbers.
VOIP Gear

• Foreign eXchange Station – analog telephone

• Foreign eXchange Office – Device that to phones

• Analog Telephone Adapter – An interface with ethernet and an FXS port. Examples include Motorola VT1000 or Sipura 1000
VOIP Gear

- **Portable Branch eXchange** – A local telephone switch
- **Interactive Voice Response** – A voice menu
- **Key System** – A type of PBX that tightly tracks phone lines in and out of the system.
VOIP Protocols

- **Session Initiation Protocol** – Manages a phone connection
- **Realtime Transport Protocol** – Carries the voice data
- **Inter Asterisk eXchange** – Voice and control information between two PBXs.
- **H323** – An older voice/video teleconferencing protocol
VOIP Encoding

• Voice is digitized and compressed for transmission.

• Each voice channel requires some bandwidth.

• Converting between encodings is called **transcoding**

• **ulaw** and **alaw** (aka g711) are highest quality lowest compression. Essentially equivalent to analog voice.

• **g729a** is very good, but proprietary.

• Other formats include **gsm, ilbc, adpcm** (aka **g726**)

• 56kbps down to about 10kbps, but you lose quality as you drop.
Network Protocols

- **Network Address Translation** – Allow multiple machines to share on network address

- **Quality of Service** – A protocol for prioritizing network traffic
Starting to VOIP

- Headset is highly recommended for better voice quality
- VOIP Providers – Free World Dialup, Sipphone, Earthlink, or Skype (non standard)
- Free calls to other VOIP users
- **Peering numbers** to call from one VOIP provider to another
- Uses SIP/RTP between your computer and VOIP provider
- **Soft phone** – is a software phone that allows one to make VOIP calls
- SIP Address – Resembles an email address for SIP calls

![Diagram of VOIP setup]
Making a SIP Call

- Register your SIP device. Let a **proxy server** know you're there so that it can ring you.

- Dial a SIP URL (or a number)

- SIP connects to the destination and tells them what RTP ports to use and what encodings are supported

- RTP stream starts sending voice packets.

- If the call is forwarded to another SIP device, the client may be told to **reinvite** and reconnect directly to that host.

- Call completes SIP says goodbye
Some providers will route PSTN calls to your SIP phone number for free

No choice of phone numbers. Usually a long distance call.

ipkall.com is one such service

They make money from **settlements**

People with standard phones can call you, but you can't call out

Good for testing incoming setup before attaching it to a live number.
There are many residential VOIP providers. (Vonage, Broadvoice, packet8, VoicePulse, Sipphone, etc)

You connect a standard phone via an ATA. Some let you bring your own device

They provide a DID (phone number) people can call

Many choices of services such as voice mail, many calling features, 800 numbers, etc.

Many give unlimited calling locally, nationally, or even to some international destinations.
Replace a Phone (cont)

- If possible calls are sent entirely via the internet.
- If not, then they are routed via the Internet to the closest **Point Of Presence** before going to the PSTN.
Connecting Your PSTN and VOIP

- Add a device that supports an FXO port and it can be connected to the local exchange carrier.
- Sipura 3000 is an example of this that supports a single line.
- Calls can be routed out either port.
- A **dial plan** is used specify which calls are sent out which port.
Asterisk

• Asterisk can speak SIP, IAX, and H323 over an ethernet port

• Asterisk supports cards that talk to analog lines via FXO or FXS

• Asterisk allows multiple lines to be shared by multiple devices

• Asterisk can play prerecorded sounds

• Asterisk can detect Dual Tone Modulation Frequency (touch tones)

• Asterisk can run programs to control various actions
First Tests With Asterisk

- Configure Asterisk to register with FWD using IAX
- Configure Asterisk to play a sound when it receives a call
- Use a soft phone with FWD to call Asterisk

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- Configure IPKall to point at your FWD SIP address
- Call your IPKall number
### IAX.conf

```plaintext
[general]
bandwidth=low
disallow=lpc10 ; Icky sound quality... Mr. Roboto.
allow=ulaw
allow=gsm
allow=alaw
allow=ilbc
allow=adpcm
jitterbuffer=no
register=>123456:PASSWORD@iax2.fwdnet.net
tos=lowdelay
;mailboxdetail=yes

; Guest must exist to avoid unauthorized users from connecting
[guest]
type=user
callerid="Guest IAX User"

; ; Trust Caller*ID Coming from iax.fwdnet.net
; [iaxfwd]
type=user
context=from-fwd
auth=rsa
inkeys=freeworlddialup
```

### extensions.conf

```plaintext
[from-fwd]
exten => 123456,1,Answer
exten => 123456,2,Playback(monkeys)
```
IVR and Voicemail With Asterisk

extensions.conf

[macro-mainmenu]
exten => s,1,Answer
exten => s,2,DigitTimeout,5
exten => s,3,ResponseTimeout,10
exten => s,4,SetMusicOnHold,random
exten => s,5,Background(greeting)

[incoming]
include => extensions
; IVR
exten => 1,1,VoiceMail2(u201)
exten => 2,1,VoiceMail2(u202)
exten => 8,1,VoiceMailMain2
exten => 8,2,Hangup
exten => 9,1,Directory(default)
; Invalid
exten => i,1,Playback(invalid)
exten => i,2,Background(greeting)
; Timeout default mailbox
exten => t,1,VoiceMail2(u201)

[from-fwd]
include => incoming
exten => ${FWDUSERID},1,Macro(mainmenu)

voicemail.conf

[general]
format=wav49|gsm|wav
servermail=asterisk
attach=yes
maxsilence=10
silencethreshld=128
maxlogins=3
fromstring=Digital Ordnance Voicemail
pagerfromstring=DO VMail
emailsubject=New VM (${VM_MSGNUM}) for
${VM_MAILBOX} from ${VM_CALLERID}
emailbody=Dear ${VM_NAME}:

You have a
${VM_DUR} long message (#${VM_MSGNUM})
in mailbox ${VM_MAILBOX} from ${VM_CALLERID}
on
${VM_DATE}
The Digital Ordnance Voicemail
 tz=pacific

[default]
; Each mailbox is listed in the form
; <mailbox>=<password>,<name>,<email>,
; <pager_email>,<options>
201=>1234,Daryll Strauss,daryll@nospam.com
202=>1234,Daryll Strauss,daryll@nospam.net
Interfacing With Asterisk

- Soft phones
- ATA's with analog phones
- SIP phones
- Analog phones into cards
- VOIP Providers over ethernet
- PSTN connection via cards
- PSTN via gateway
Interfacing With Asterisk

extensions.conf

[global]
MYNAME=Digital Ordnance
MYPHONE=1234567890

FWDUSERID=12356
FWDPASSWD=PASSWORD
FWDSERVER=iax2.fwdnet.net

[macro-dialfwd]
exten => s,1,SetCallerID(${MYPHONE})
exten => s,2,SetCIDName(${MYNAME})
exten => s,3,Dial(IAX2/${FWDUSERID}: ${FWDPASSWD}@${FWDSERVER}/${ARG1})
exten => s,4,Congestion

[macro-makecall]
exten => s,1,Dial(${ARG1},32,m)

[macro-stdexten]
exten => s,1,Playback(pleasewait)
exten => s,2,Macro(makecall,SIP/{ARG1})
exten => s,3,Goto(s-${DIALSTATUS},1)
exten => s-NOANSWER,1,Macro(vmessage,u${ARG1})
exten => s-NOANSWER,2,Goto(incoming,s,1)
exten => s-BUSY,1,Macro(vmessage,b{ARG1})
exten => s-BUSY,2,Goto(incoming,s,1)
exten => s-,1,Goto(s-NOANSWER,1)
exten => a,1,Macro(vmessage,${ARG1})

[extensions]
exten => 201,1,Macro(stdexten,201)
exten => 202,1,Macro(stdexten,202)
exten => 444,1,Meetme(1234)

[fwd-forced]
exten => _7.,1,Macro(dialfwd,${EXTEN:1})

[incoming]
include => extensions
; IVR
exten => 1,1,Macro(stdexten,201)
exten => 2,1,Macro(stdexten,202)
exten => 8,1,VoiceMailMain2
exten => 8,2,Hangup
exten => 9,1,Directory(default)
; Invalid
exten => i,1,Playback(invalid)
exten => i,2,Background(greeting)
; Timeout default mailbox
exten => t,1,Macro(stdexten,201)

[from-fwd]
include => incoming
exten => ${FWDUSERID},1,Macro(mainmenu)

[default]
include => incoming
exten => s,1,Macro(mainmenu)

[home]
include => fwd-forced
include => extensions
Interfacing With Asterisk

sip.conf

[general]
disallow=all ; Disallow all codecs
allow=gsm
allow=ilbc
allow=adpcm
allow=ulaw
allow=alaw
dtmfmode=rfc2833
srvlookup=yes

register => <NUMBER>:<PASSWORD>@sip.voiprovider.com/<NUMBER>

[201]
; Sipura ATA Phone line
type=friend
host=dynamic
context=home
secret=PASSWORD
callerid=Daryll
mailbox=201
nat=no

[202]
; Soft phone
type=friend
host=dynamic
callerid=Daryll
mailbox=201
nat=no

[voipprovider]
type=friend
username=1234567890
fromuser=1234567890
secret=PASSWORD
host=sip.voipprovider.com
callerid=Daryll
mailbox=201
nat=yes
canreinvite=no
dtmfmode=inband
qualify=yes
Additional Features

• Asterisk can monitor and record calls

• Asterisk can provide features, like putting calls on hold, even if the phone doesn't support it.

• Asterisk can have dial plans that select among many VOIP providers

• Pickup groups can be defined

• Call queues can be created

• Asterisk can have time sensitive rules.
Going Beyond Your Father's PBX

• Asterisk can read/write values from/to a database

• Asterisk can send data to/read data from from an application

• Asterisk can be controlled by an external manager application

• Festival can be used for speech generation

• Speech recognition is harder, but also possible
Example Applications

• Credit card/Prepaid calling
• Dating service
• Live chat
• Follow me
• Call center (Asterisk agents)
• Games (Lost Vault, Taboo)
• Training
• Virtual Office
• Web calling/Presence
Gotchas

• SIP behind NAT is hard, because SIP encodes RTP port numbers in packets. Use IAX or a Virtual Private Network to tunnel behind a NAT. Simple Tunneling of UDP through NAT helps a lot with the problem, but isn't perfect.

• Echo can be a problem when transitioning between digital and analog network

• Asterisk doesn't support all features (like key system features) It's still very young and a lot of development is still being done.

• Encryption is not widely support for SIP (Evesdropping on SIP calls)
Gotchas (cont)

- Asterisk doesn't support SIP URLs well.

- Learning curve is steep – read the docs, take small steps and test changes.

- Overloading the Asterisk box will degrade call quality. Asterisk should have a dedicated box. Transcoding (converting between formats) takes lots of cycles.

- 911 is problematic. Where are you? With VOIP you can be calling from anywhere. VOIP also requires power unlike analog phones.
Gotchas (cont)

• Network traffic can cause you to loose quality. QoS can prioritize voice traffic over data. Consider private/VLAN voice ethernet.

• Fax and Data calls can be a problem. Fax works well with some encodings or T.38. Data doesn't work (Tivo/DirecTV calls)

• Devices from VOIP providers may be locked.

• VOIP providers may not support IAX, Asterisk, or soft phones.
Asterisk Add Ons

• **ASTMan** is a manager that lets you manipulate Asterisk while it is running via a network connection.

• **AMP** is a GUI for configuring Asterisk and some of its features. Using a GUI makes the setup easier at the cost of some of the scripting flexibility.

• **Flash Operator Panel** is a program that allows the user to control Asterisk (monitor, transfer, hangup, etc. calls).

• **Asterisk@Home** is a GUI based on AMP and other tools for using Asterisk in a home environment.
Other Open Source VOIP Systems

- **SIP Express Router** – A SIP processor that does not handle the media stream. Scales to very large numbers of users. SER and Asterisk work well together.

- **SIP Foundry** – A PBX that focuses on SIP. Has a nice web interface for configuration.
Q: Why do we use phone numbers?  
A: SIP URLs are easier to remember. **SRV** records allow you to do that.

Q: How do I know if a phone number is VOIP?  
A: **E164** allows users to register phone numbers that redirect to SIP URLs.

Q: How do I route my call?  
A: With the wide variety of VOIP service providers you can select on a call by call basis whichever one best meets your needs (functions, cost, quality).
Conclusions

• My goal was to introduce you to telephony and VOIP. Teach you the basic terminology.

• Give you examples you can do yourself for very little cost

• Get you thinking of Asterisk not only as a PBX but as a voice application platform
Don't forget the VOIP panel at 3:00 today.
Resources

Websites:
http://www.voxilla.com
http://www.asterisk.org
http://www.voip-info.org
http://www.asteriskdocs.org

Mailing Lists:
asterisk-users mailing list (HIGH volume)