VOIP, Linux, and Asterisk
Making Beautiful Voice Together

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SCALE 3x
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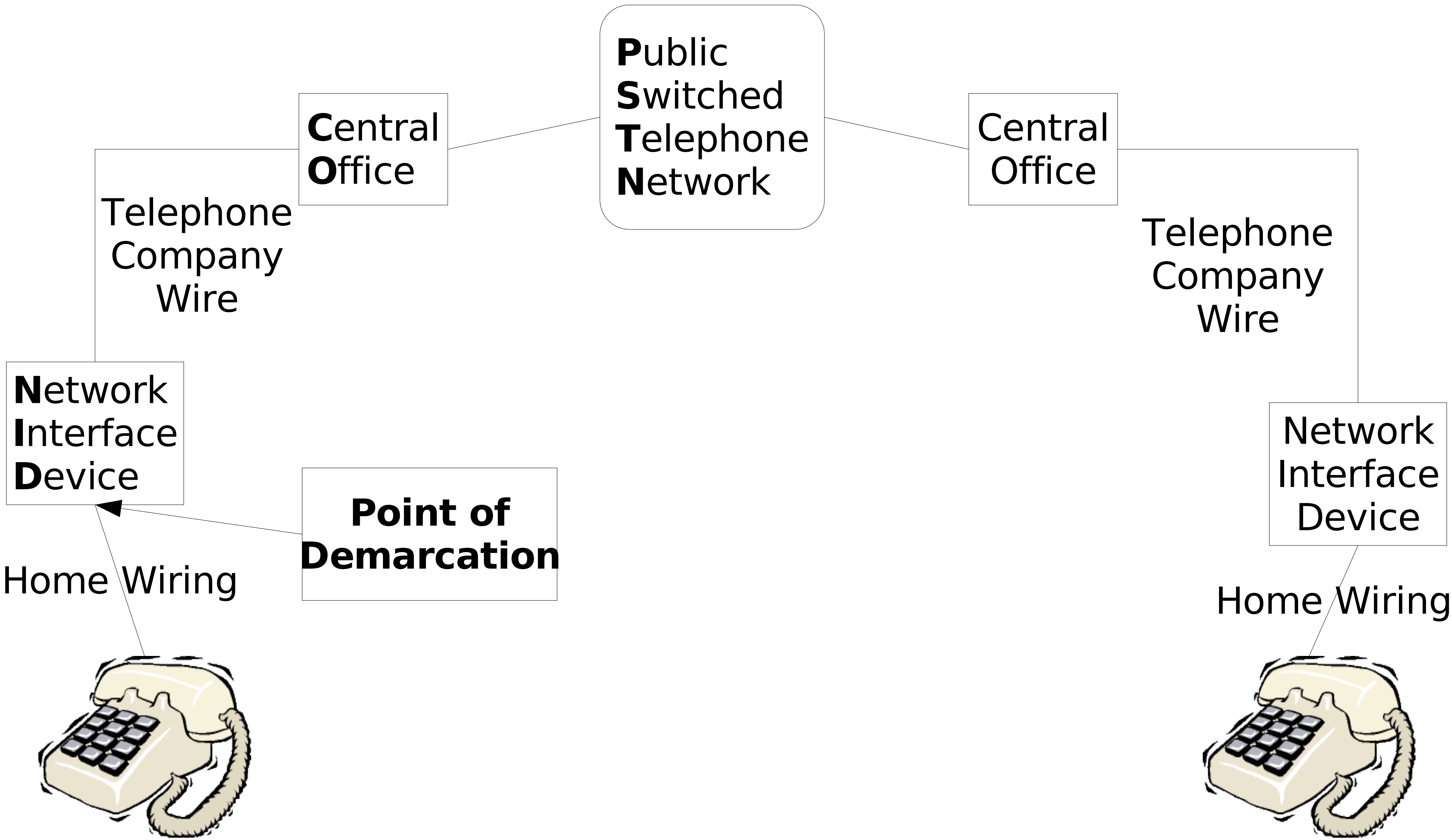
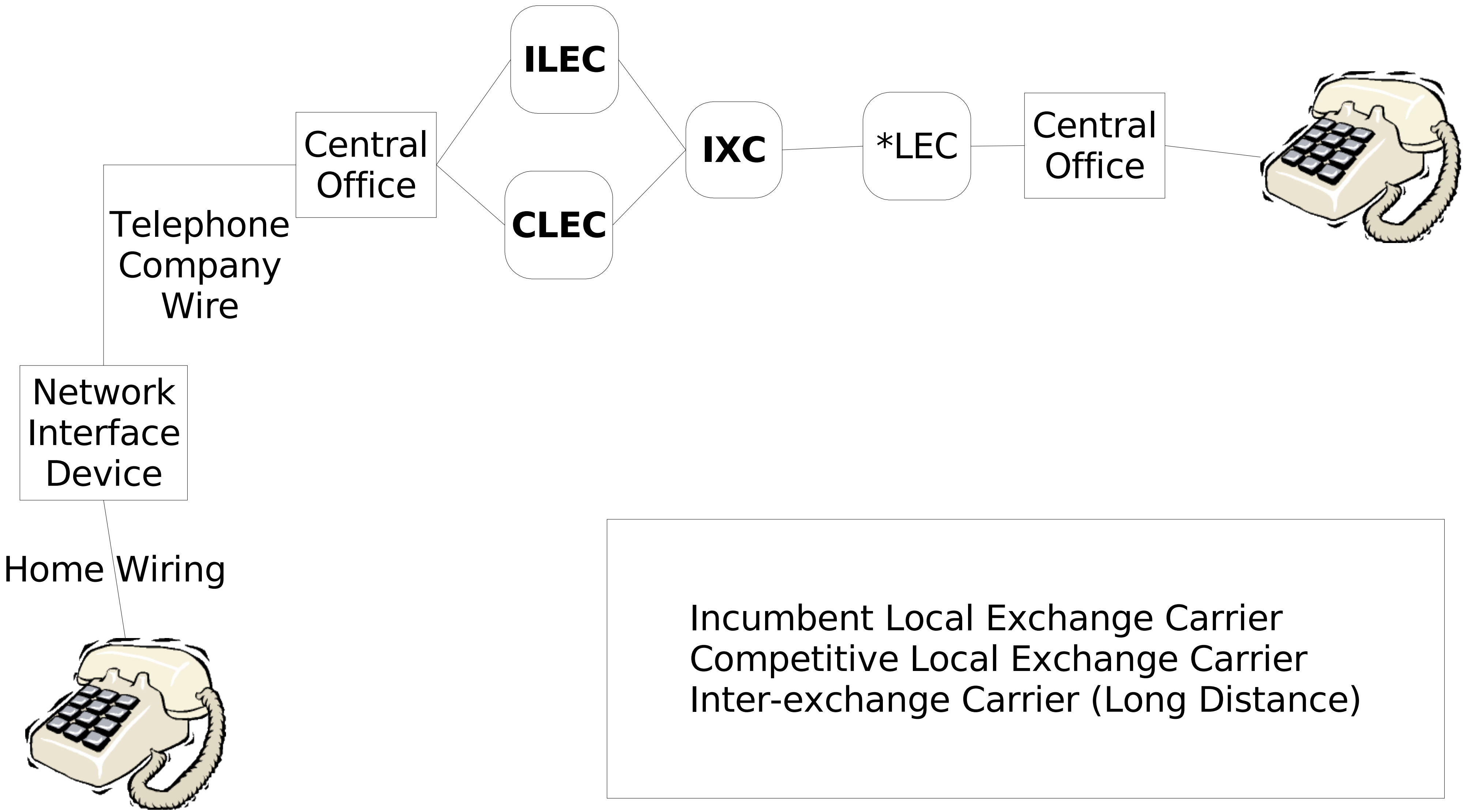
POTS World – Ma Bell



POTS World - Today

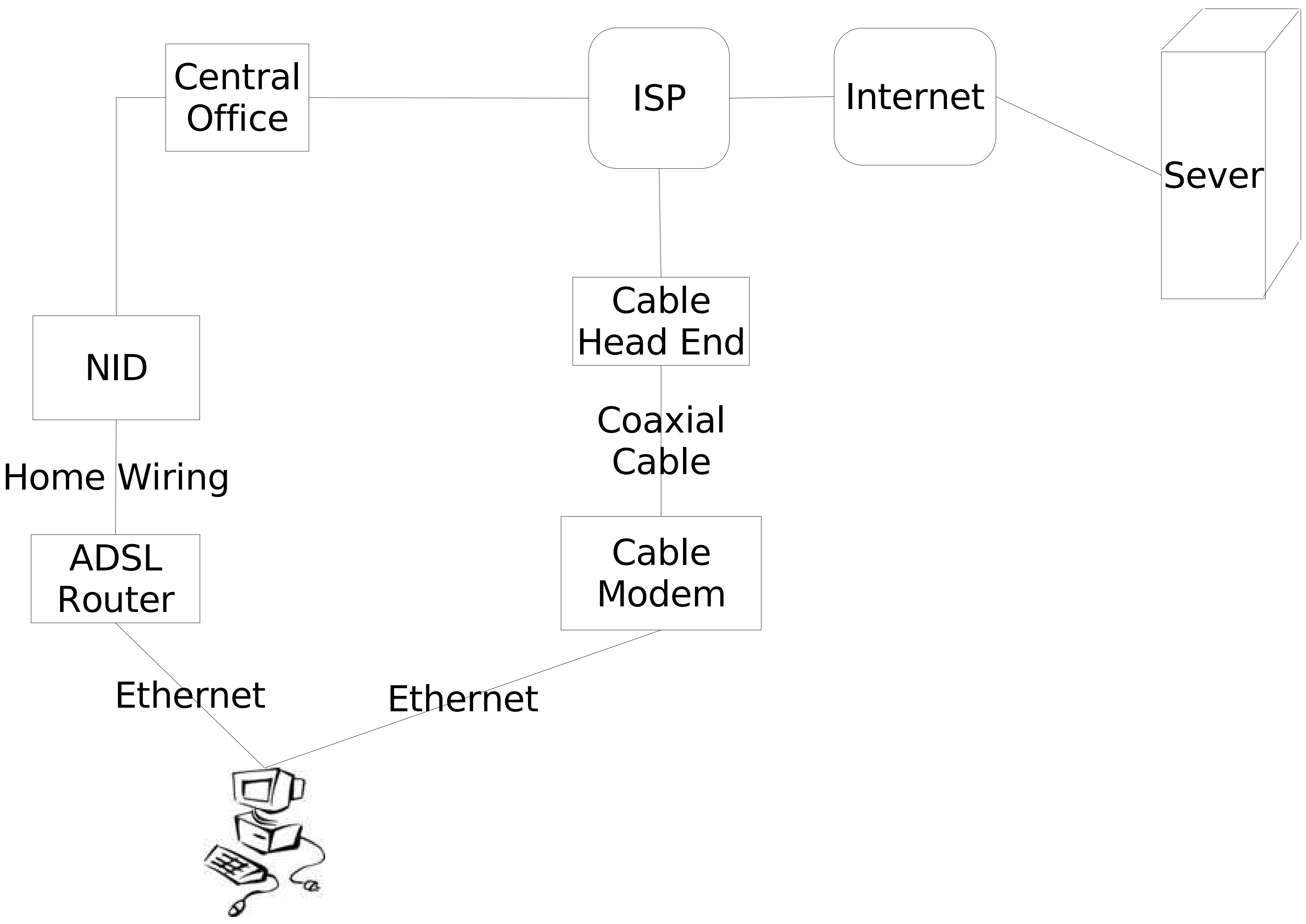


Incumbent Local Exchange Carrier
Competitive Local Exchange Carrier
Inter-exchange Carrier (Long Distance)

Connections to the Telephone Company

- Analog phone lines
- ISDN – Digital phone lines. Two **B Channels** for voice and one **D Channel** for control
- **P**rimary **R**ate **I**nterface – Digital phone lines. 23 **B Channels** and one **D Channel**.

Networked World



Crossover Into **V**oice **O**ver **I**nternet **P**rotocol

- VOIP crosses over between the Internet and the PSTN at several possible locations
- Intraoffice – VOIP phones on the desktop
- **D**irect **I**nward **D**ial – A phone number people can call
- **Termination** – Calling local or long distance numbers.

VOIP Gear

- **F**oreign e**X**change **S**tation – analog telephone
- **F**oreign e**X**change **O**ffice – Device that to phones
- **A**nalog **T**elephone **A**dapter – An interface with ethernet and an FXS port. Examples include Motorola VT1000 or Sipura 1000

VOIP Gear

- **P**ortable **B**ranch e**Xchange** – A local telephone switch
- **I**nteractive **V**oice **R**esponse – A voice menu
- **Key System** – A type of PBX that tightly tracks phone lines in and out of the system.

VOIP Protocols

- **S**ession **I**nitiation **P**rotocol – Manages a phone connection
- **R**ealtime **T**ransport **P**rotocol – Carries the voice data
- **I**nter **A**sterisk e**X**change – Voice and control information between two PBXs.
- **H323** – An older voice/video teleconferencing protocol

VOIP Encoding

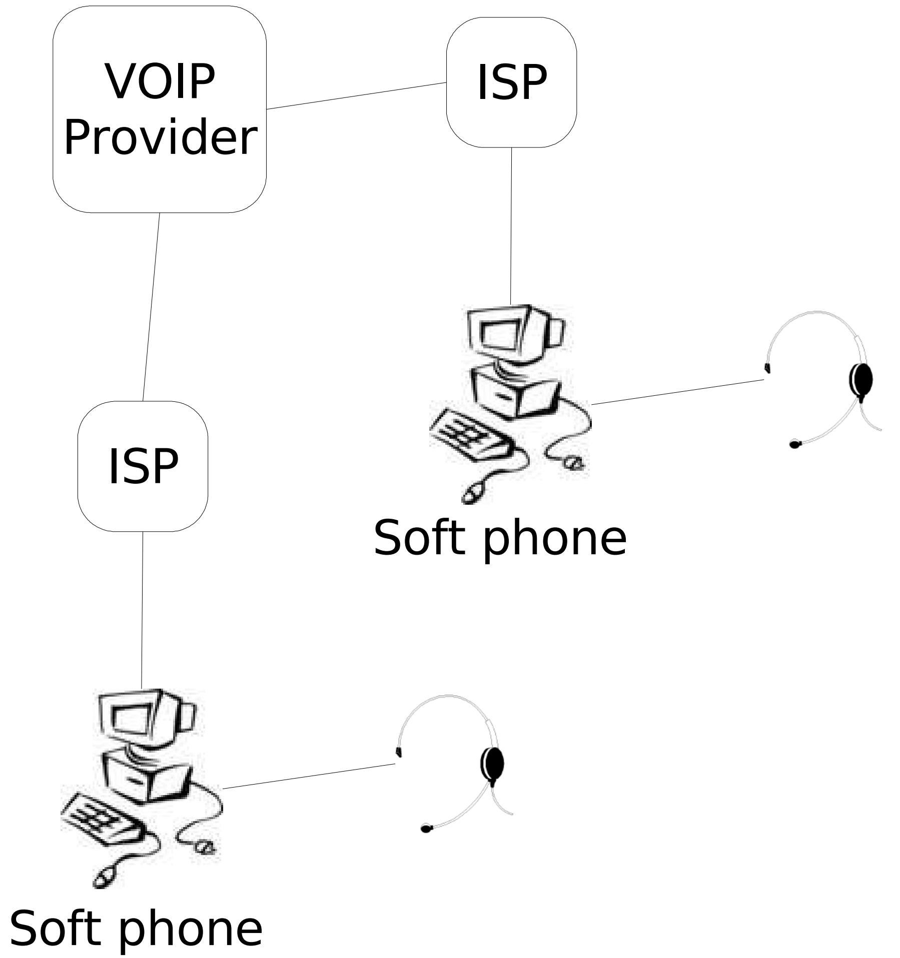
- Voice is digitized and compressed for transmission.
- Each voice channel requires some bandwidth.
- Converting between encodings is called **transcoding**
- **ulaw** and **alaw** (aka g711) are highest quality lowest compression. Essentially equivalent to analog voice.
- **g729a** is very good, but proprietary.
- Other formats include **gsm**, **ilbc**, **adpcm** (aka **g726**)
- 56kbps down to about 10kbps, but you lose quality as you drop.

Network Protocols

- **N**etwork **A**dress **T**ranslation – Allow multiple machines to share on network address

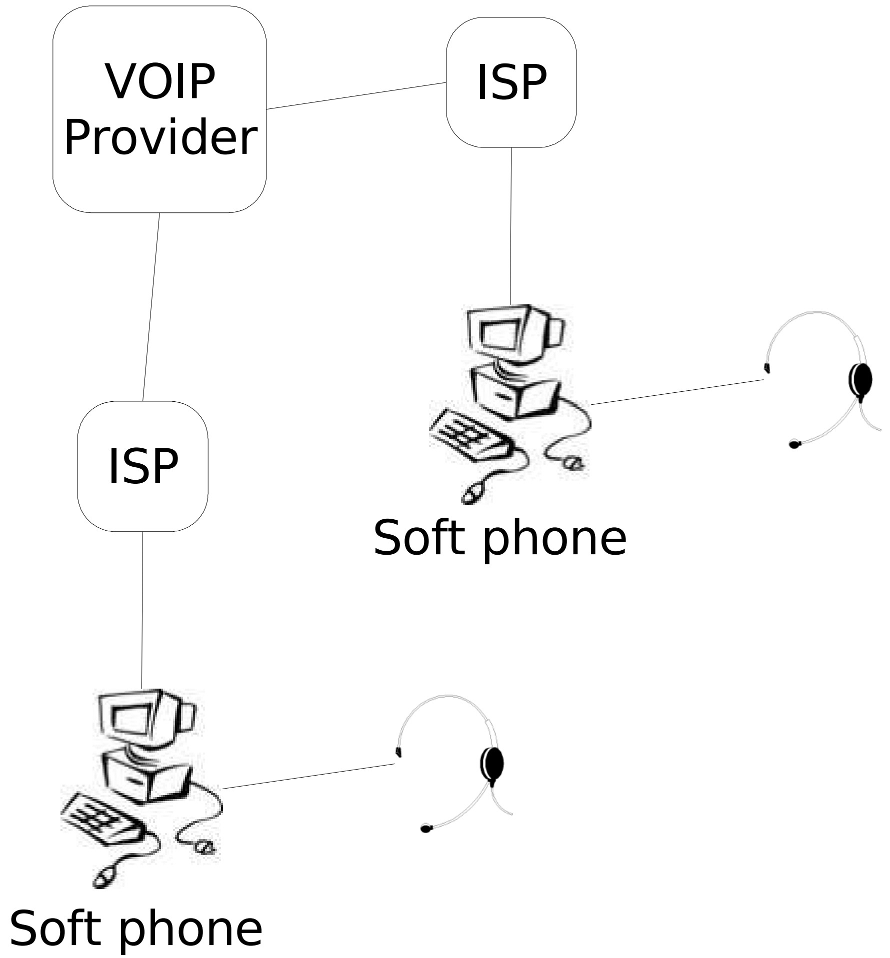
- **Q**uality **o**f **S**ervice – A protocol for prioritizing network traffic

Starting to VOIP



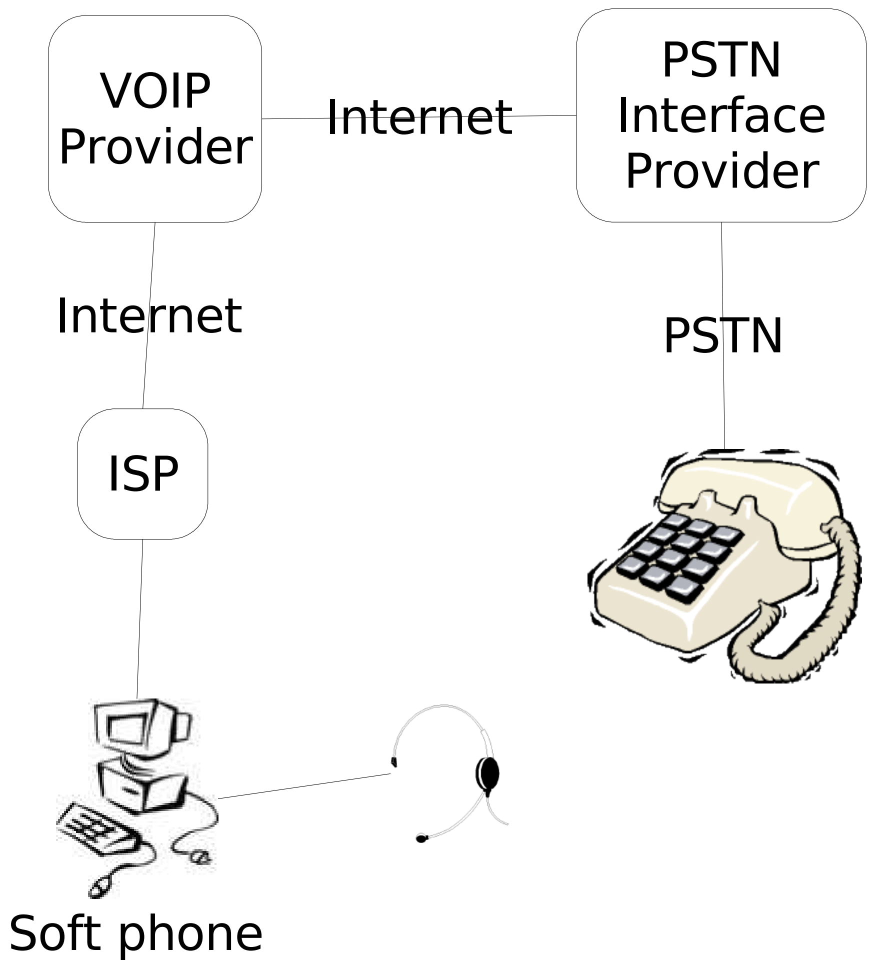
- Headset is highly recommended for better voice quality
- VOIP Providers – **F**ree **W**orld **D**ialup, Sipphone, Earthlink, or Skype(non standard)
- Free calls to other VOIP users
- **Peering numbers** to call from one VOIP provider to another
- Uses SIP/RTP between your computer and VOIP provider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

Making a SIP Call



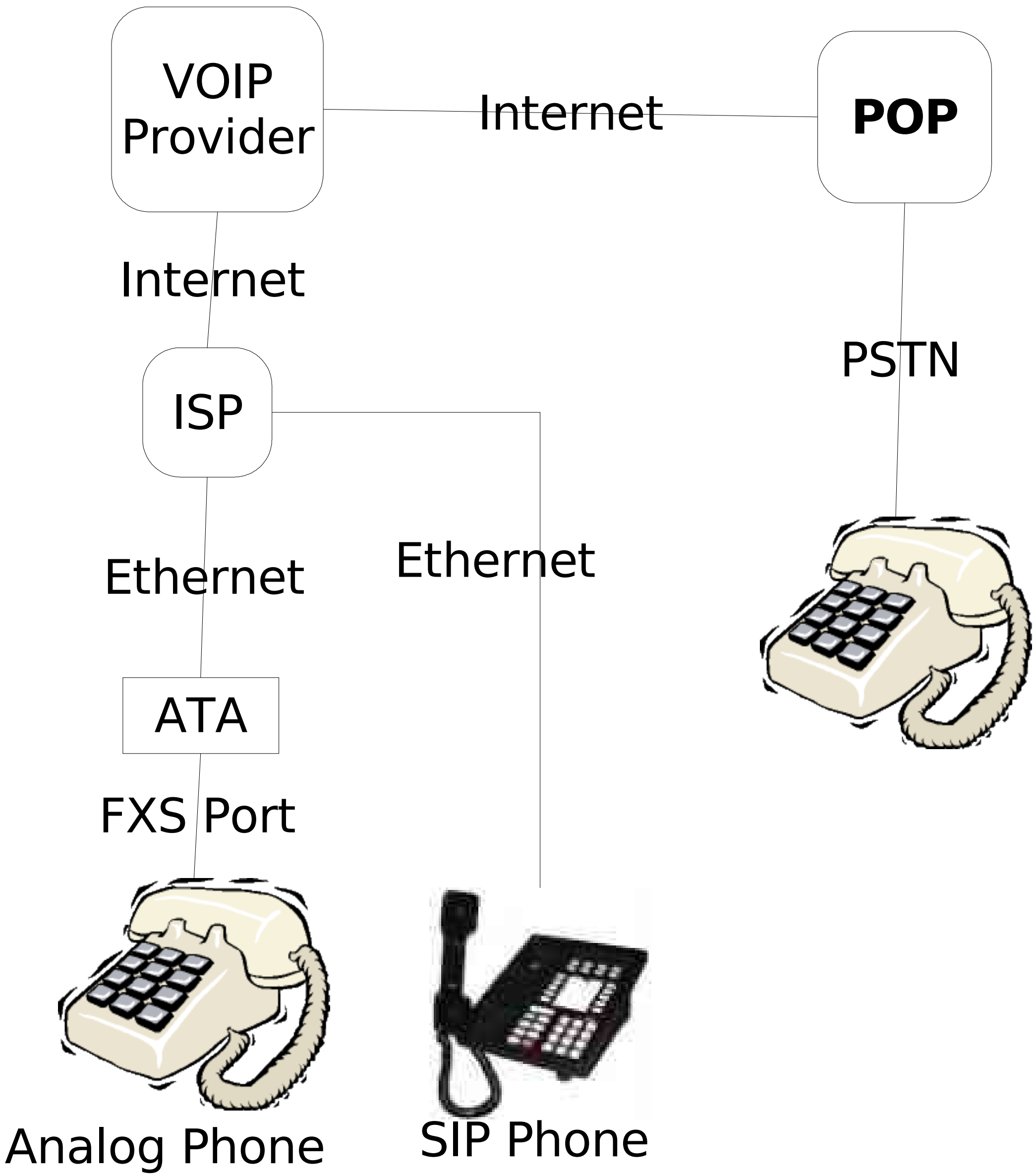
- Register your SIP device. Let a **proxy server** know you're there so that it can ring you.
- Dial a SIP URL (or a number)
- SIP connects to the destination and tells them what RTP ports to use and what encodings are supported
- RTP stream starts sending voice packets.
- If the call is forwarded to another SIP device, the client may be told to **reinvite** and reconnect directly to that host.
- Call completes SIP says goodbye

PSTN to VOIP



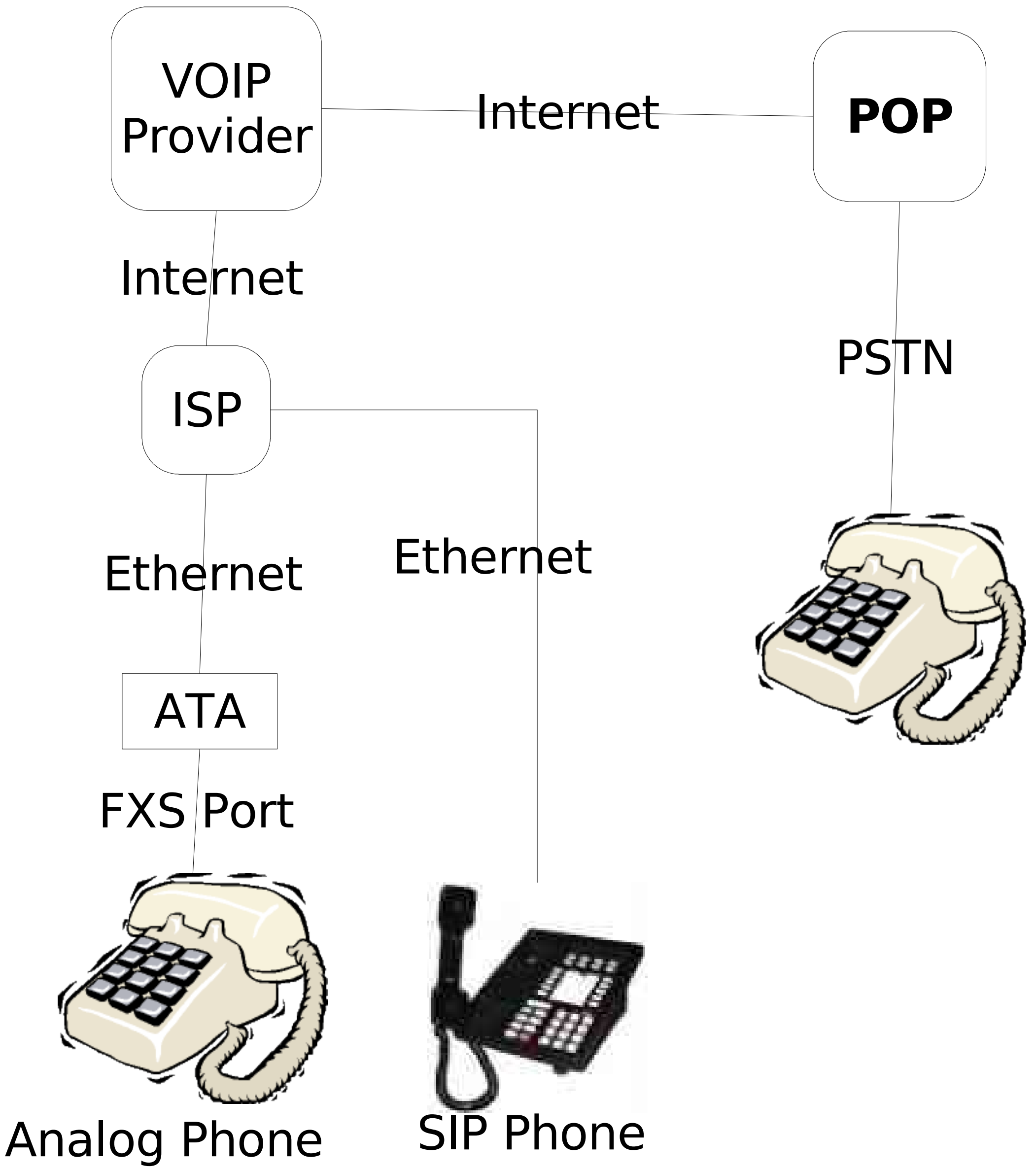
- Some providers will route PSTN calls to your SIP phone number for free
- No choice of phone numbers. Usually a long distance call.
- ipkall.com is one such service
- They make money from **settlements**
- People with standard phones can call you, but you can't call out
- Good for testing incoming setup before attaching it to a live number.

Replace a Phone



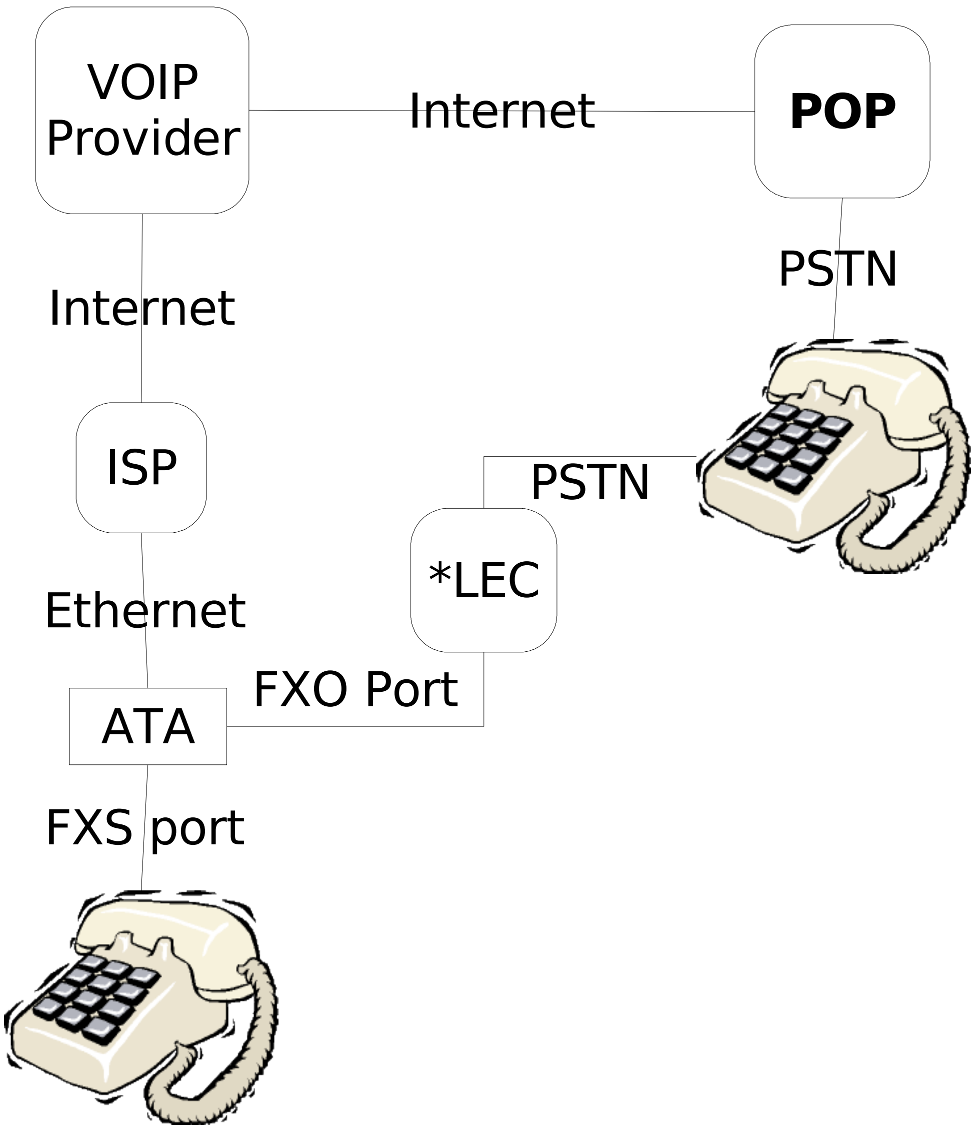
- There are many residential VOIP providers. (Vonage, Broadvoice, packet8, VoicePulse, Sipphone, etc)
- You connect a standard phone via an ATA. Some let you bring your own device
- They provide a DID (phone number) people can call
- Many choices of services such as voice mail, many calling features, 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 **P**oint **O**f **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

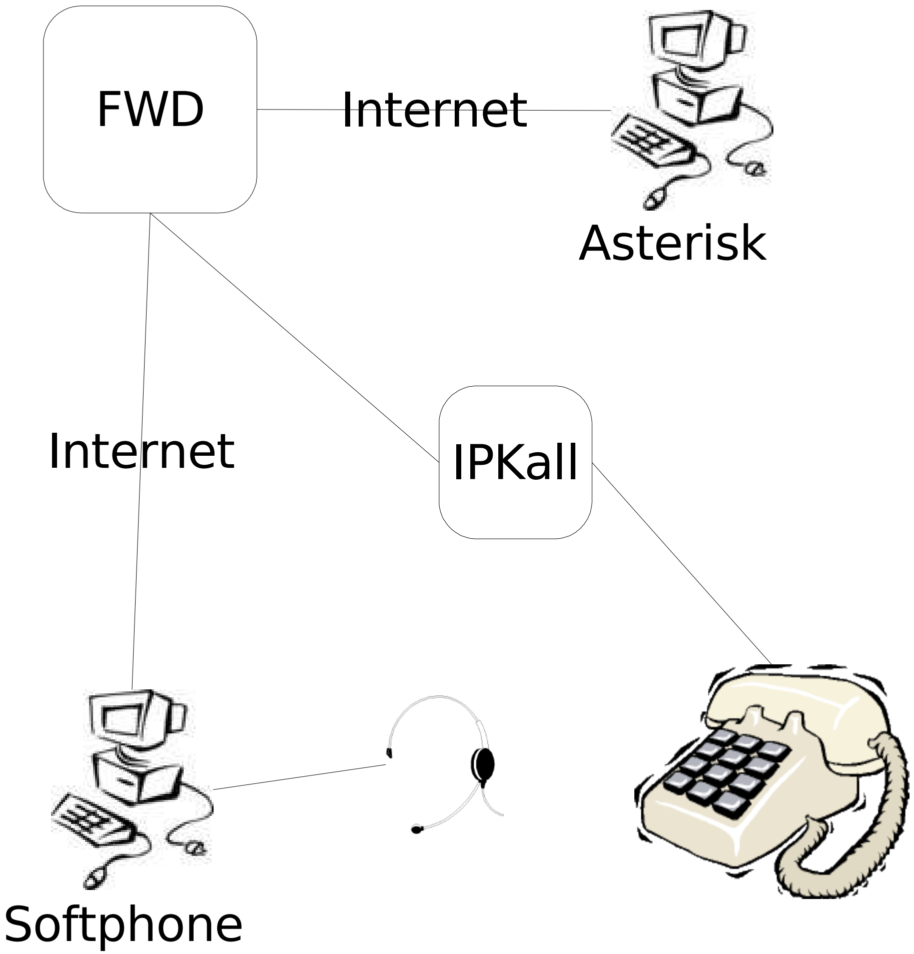
- Asterisk allows multiple lines to be shared by multiple devices

- Asterisk can play prerecorded sounds

- Asterisk can detect **D**ual **T**one **M**odulation **F**requency (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

- Configure IPKall to point at your FWD SIP address

- Call your IPKall number

Config Files

IAX.conf

```
[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
context=default
callerid="Guest IAX User"

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

extensions.conf

```
[from-fwd]
exten => 123456,1,Answer
exten => 123456,2,Playback(monkeys)
```

IVR and Voicemail With Asterisk

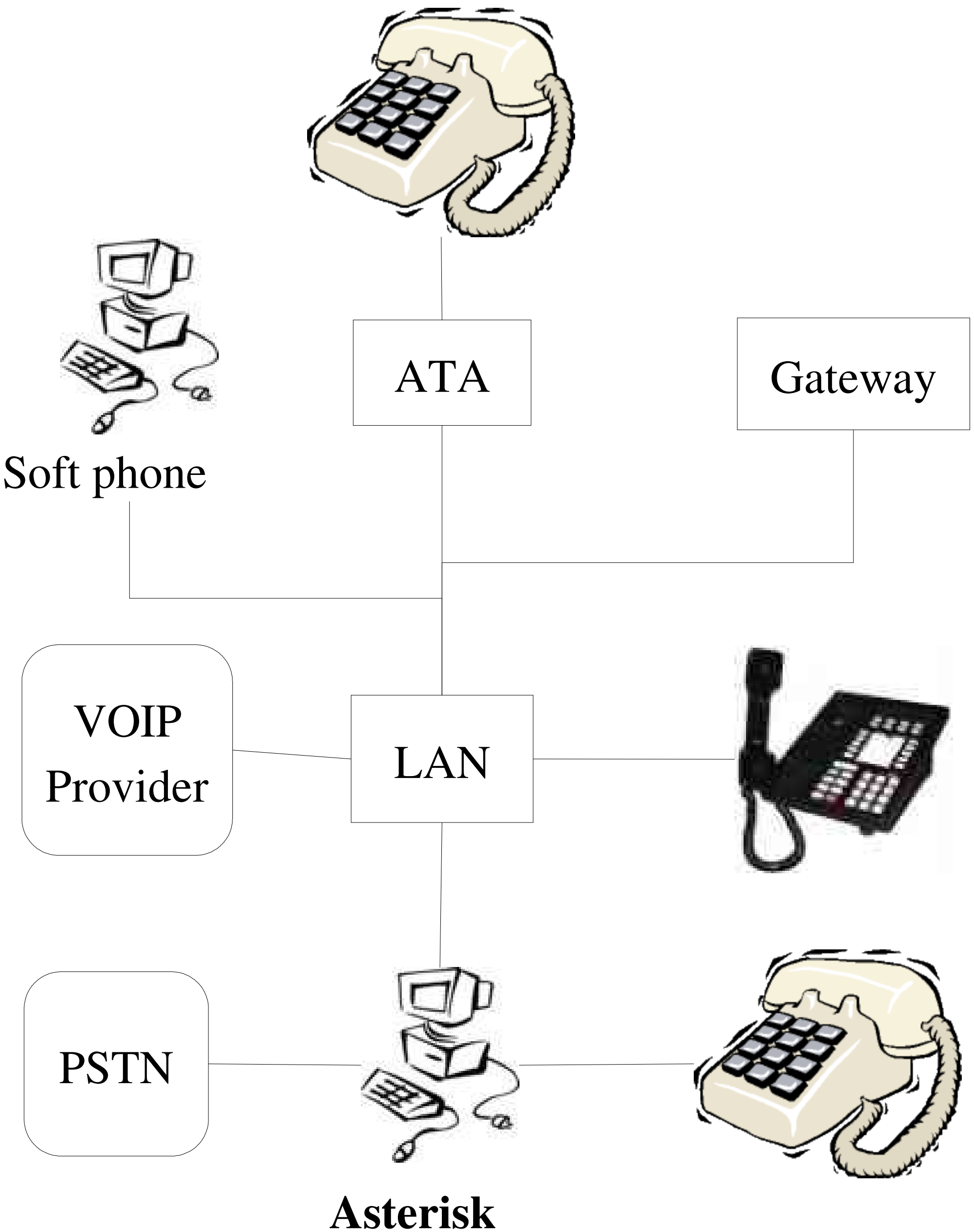
extensions.conf

```
[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}:\n\nYou have a
${VM_DUR} long message (#${VM_MSGNUM})
in mailbox ${VM_MAILBOX} from ${VM_CALLERID} on
${VM_DATE}\nThe Digital Ordnance Voicemail\n
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
context=home
secret=PASSWORD
callerid=Daryll
mailbox=201
nat=no

[voipprovider]
type=friend
username=1234567890
fromuser=1234567890
secret=PASSWORD
host=sip.voipprovider.com
context=from-voiprovider
fromdomain=sip.voipprovider.com
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 **V**irtual **P**rivate **N**etwork to tunnel behind a NAT. **S**imple **T**unneling of **U**DP through **N**AT 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 manager that lets you manipulate Asterisk while it is running via a network connection.

- AMP is GUI for configuring Asterisk and some of it's 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

- **S**IP **E**xpress **R**outer – 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.

A Brave New World

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

Q&A

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)