

VOIP

POTS SIP AND ALL THAT JAZZ

Welcome to CIALUG



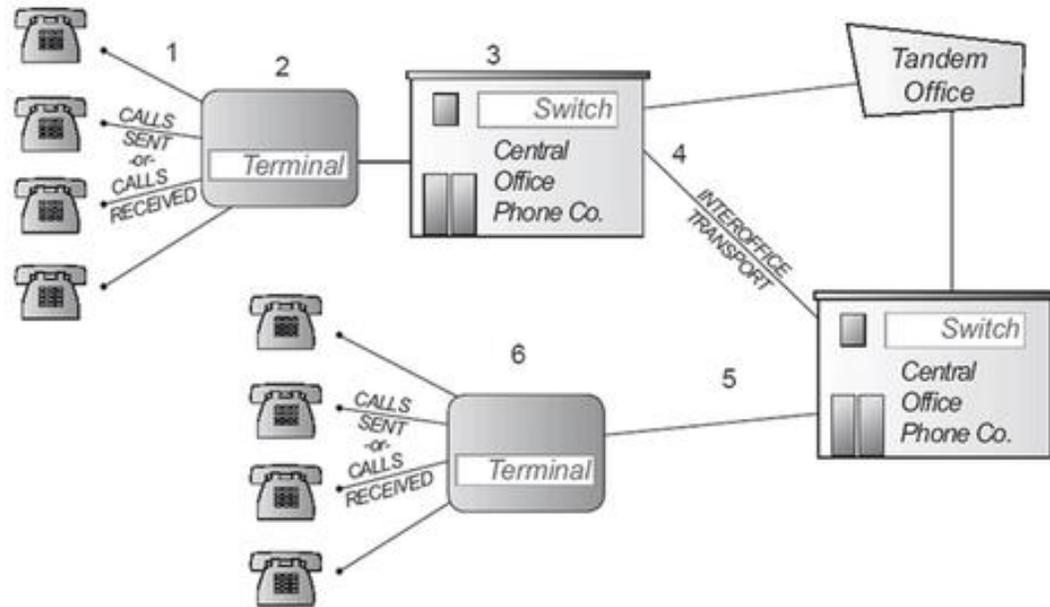
First a little housekeeping

- ▶ A little about me:
 - ▶ Andrew Denner (denner@gmail.com) (@adenner)
 - ▶ Slides will be posted tonight at <http://denner.co>
 - ▶ Not a telecom guy or expert in this so please don't hurt me!
- ▶ We have a mailing list and website (<http://cialug.org>)
- ▶ IRC
- ▶ Do you want to present on a topic kinda associated with Linux? This is your chance!
- ▶ Tentative date for Joint Meeting with Des Moines Azure group in April... details to follow



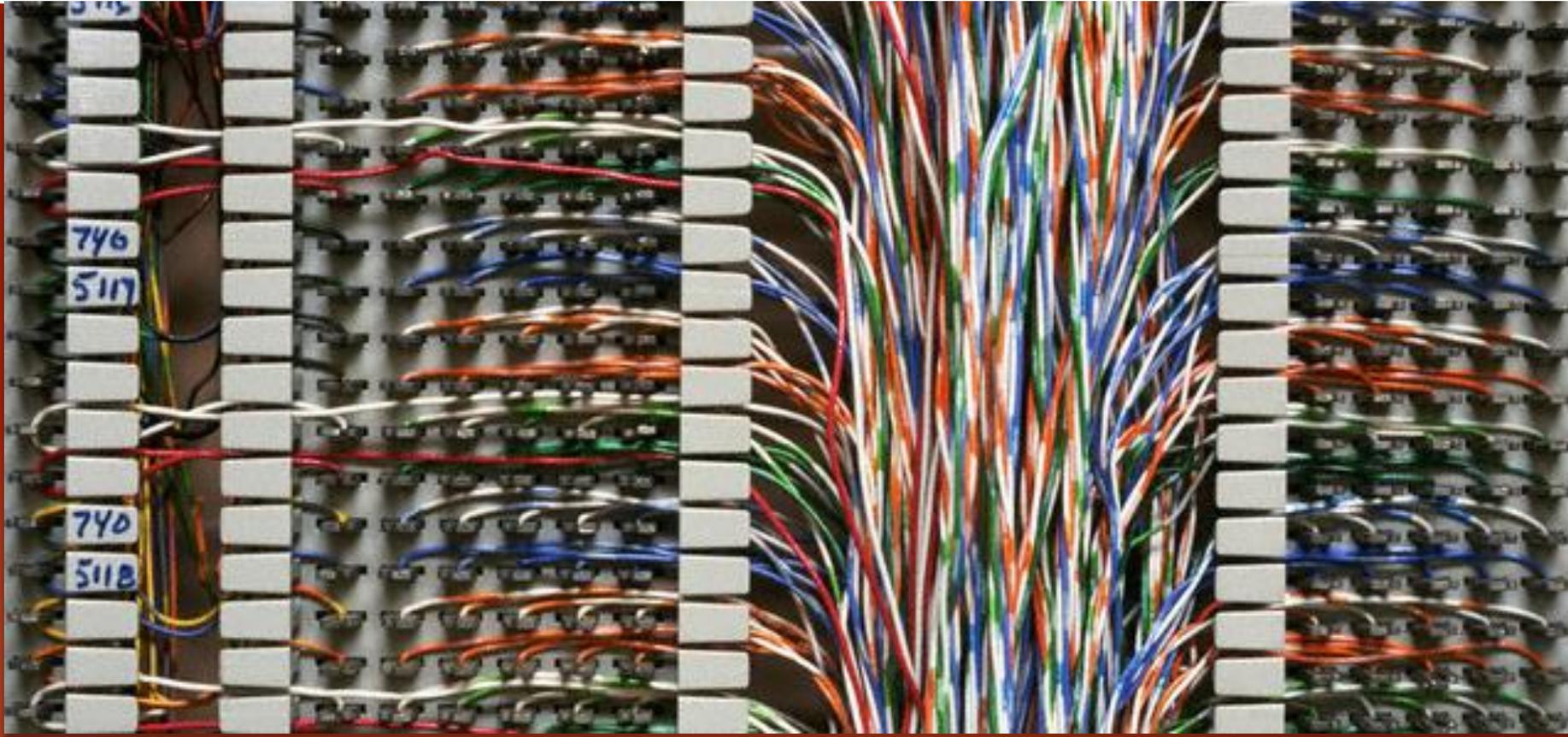
Now, on
with the
show

Plain Old Telephone Service (POTS) Networking



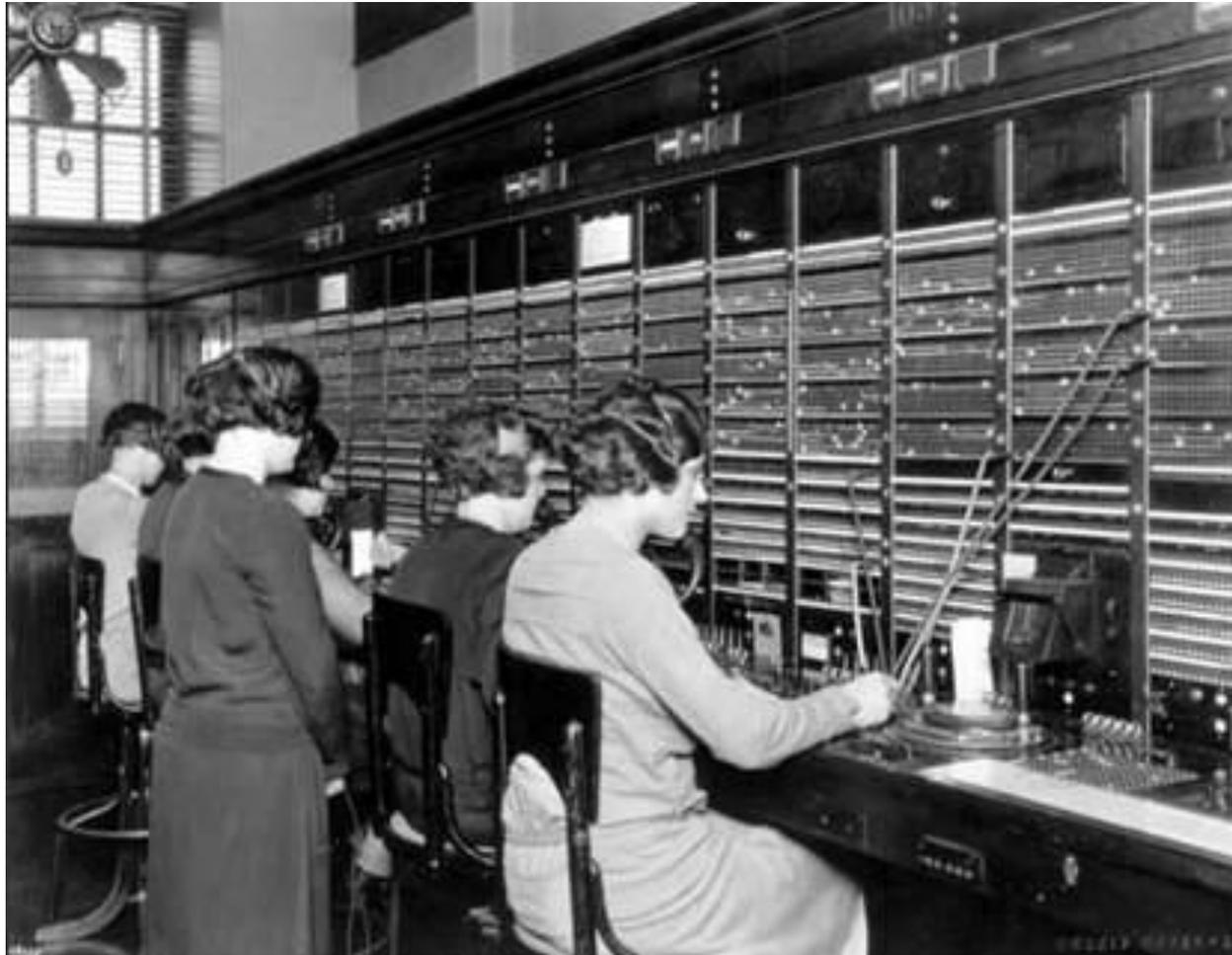
POTS

PLAIN OLD TELEPHONE SERVICE AKA PUBLIC SWITCHED TELEPHONE NETWORK (PSTN)



One wire, one path from point to point

OK IT'S NOT REALLY LIKE THAT ANYMORE... NOW EVERYTHING IS SWITCHED
AGGREGATED AND FIBER...



PBX

PRIVATE BRANCH
EXCHANGE



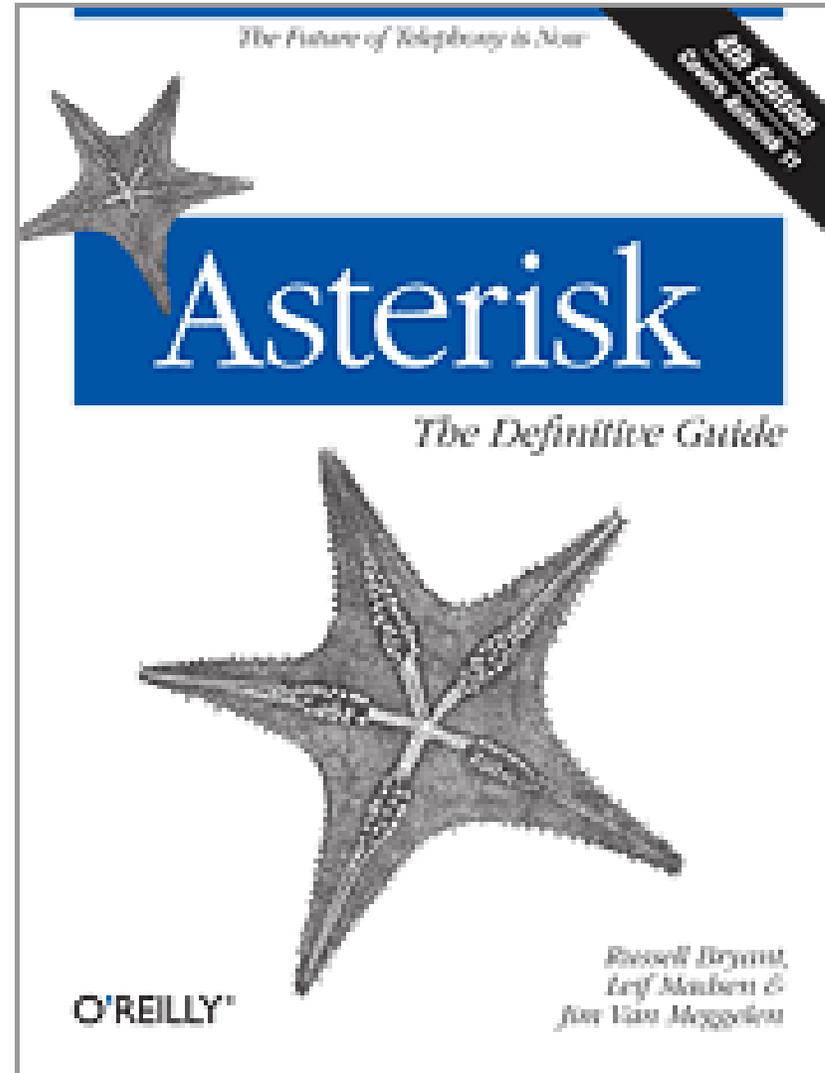
Slightly newer hardware... still expensive and inflexible! There's got to be a better way!

The Asterisk Project

- ▶ 1999 Mark Spencer was Comp E student at Auburn
- ▶ During the .com bubble
- ▶ Linux was big but there was no support
- ▶ Mark decided to fill the niche and opened the company "Linux Support Services"
- ▶ It became popular and they needed a "real phone system" \$\$\$\$\$\$\$\$\$\$
- ▶ So Mark wrote his own... and open sourced it
- ▶ In 2001 LSS became [Digium](#)

If you really want
to go down the
rabbit hole...

The sky is the limit...



VOIP Codecs

- ▶ [G.711](#) dates back to 1972 Low processor, high bandwidth at least 128k
- ▶ [G.722](#) twice range of frequency bandwidth adaptive can be better than PSTN
- ▶ [iLBC](#) Low Bitrate Codec Acquired by google popular with open source
- ▶ [Speex](#)-Low bandwidth
- ▶ [SILK/OPUS](#)-Designed by skype used by others like [whatsapp](#)

Best advice is to figure out how much [bandwidth](#) you have to use, what your upstream provider can do, and what sounds the best but YMMV

So you want to talk to the world...

- ▶ You can do direct connections between voip servers
 - ▶ Firewalls and NAT are hard...
- ▶ For PSTN you need a provider
 - ▶ I have had good luck with vitelity.net costs ~\$0.014 a minute dropped \$20 in an account ages ago and don't use it that much so it lasts almost for ever...
 - ▶ You can also get a DID number so that others can call you

FreePBX

▶ Tonight we are using a pre-packaged system called Free PBX

▶ Why?

▶ Easy

▶ Quick

▶ GUI-Web page based configs

▶ <http://freepbx.org>

The screenshot shows the FreePBX website homepage. At the top, there is a navigation menu with links for Home, Support, Community, Store, Development, Downloads, About Us, Portal, and Contact Sales. The main content area features a green background with a large image of a green frog. On the left, there are three green buttons: "Community Forums", "Bug & Feature Request", and "Why choose FreePBX?". On the right, there is a section titled "The 'Free' Stands for Freedom" with a paragraph of text and a red "DOWNLOAD FreePBX" button. Below this, there is a banner for "Elastix Usuarios" with a "CONVERTIR AHORA" button and a "Herramienta de conversión disponible!" message. To the right of the banner is a "Video Library" button. At the bottom, there is a "Welcome to FreePBX!" message and a "Recent News!" section.

FreePBX
let freedom ring™

Register Forgot Password

Home Support Community Store Development Downloads About Us Portal Contact Sales

Community Forums

Bug & Feature Request

Why choose FreePBX?

The "Free" Stands for Freedom

FreePBX is a web-based open source GUI (graphical user interface) that controls and manages Asterisk (PBX), an open source communication server. FreePBX is licensed under the GNU General Public License (GPL), an open source license. FreePBX can be installed manually or as part of the pre-configured FreePBX Distro that includes the system OS, Asterisk, FreePBX GUI and assorted dependencies.

DOWNLOAD
FreePBX

Elastix Usuarios
Vuelve a casa a FreePBX
Herramienta de conversión disponible! CONVERTIR AHORA

Video Library

Welcome to FreePBX!

Recent News!

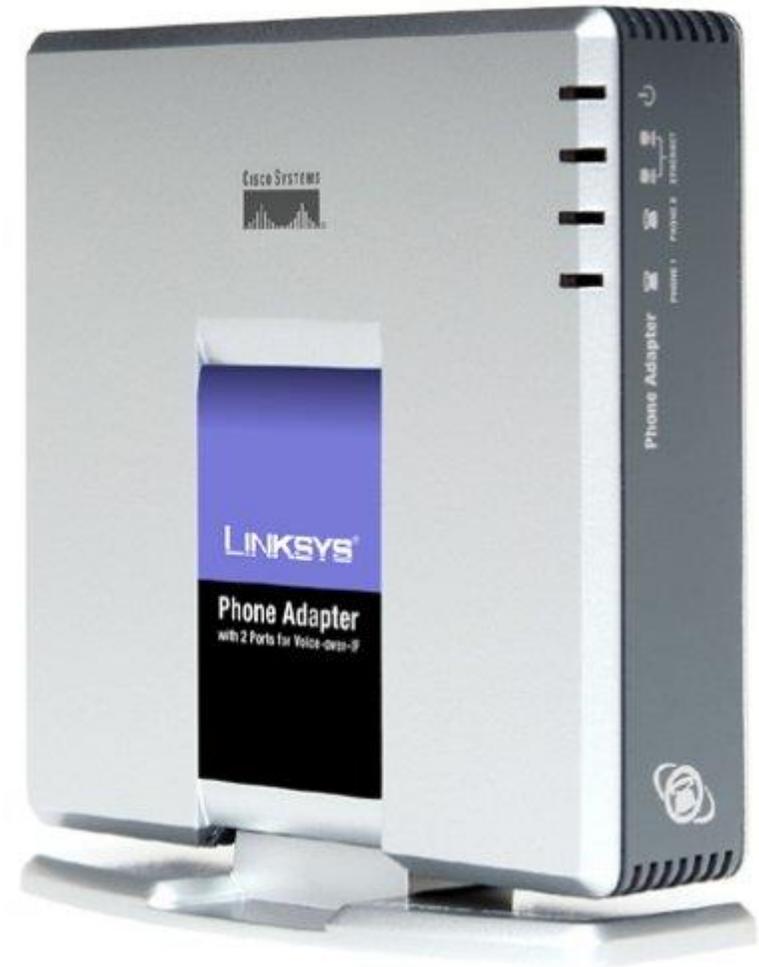
PAP2

Currently what I have running at home

Mostly used as a "Where is my cellphone"

No DID

It's worked for 5+ years directly connected to vtelity and I don't want to break it



Cisco 7960

- ▶ Was originally going to use, but ran out of time
- ▶ TFTP is hard...



Ekiga soft phone

- ▶ Apt-get install ekiga
- ▶ Free, cheep, easy...

Ekiga

Home Screenshots Features Downloads Community Documentation

Welcome to Ekiga.org

Submitted by Yannick Defais on Tue, 10/27/2009 - 21:49

Ekiga (formerly known as GnomeMeeting) is an open source **SoftPhone, Video Conferencing** and **Instant Messenger** application over the Internet.

It supports **HD** sound quality and video up to DVD size and quality.

It is **interoperable** with many other standard compliant softwares, hardwares and service providers as it uses both the major **telephony standards** (SIP and H.323).

Would you like to know [more](#)?

Free Your Speech!

 **Ekiga 4.0**
Get The Sources 
Sources (4.0.1)

Other [downloads](#).

[Read more](#)

Get Ekiga!

Download Ekiga!

Free SIP Address

Get your personal Free SIP address at ekiga.net.

Demo time