IP Telephony with Asterisk

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There lived the PSTN ….

- A few years ago, everyone struggled to convert data (IP) into sound, and move it over the Public Switched Telephone Network (PSTN) infrastructure [using MODEMs]

Enter VoIP ….

The packetisation and transport of classic public switched telephone system audio over an IP network.

The analog audio stream is encoded in a digital format, with possible compression and filtering, before encapsulating it in IP for transport over LAN/WAN or the public internet Infrastructure

Convergence or Extinction?

- Now … everyone is struggling to convert PSTN sound into data, and move it over well established IP links. [using CODECs]

Technology has just reversed the process

IP vs VoIP

VoIP introduces a collection of protocols and devices that allow for the encoding, transport and routing of audio calls over IP networks.

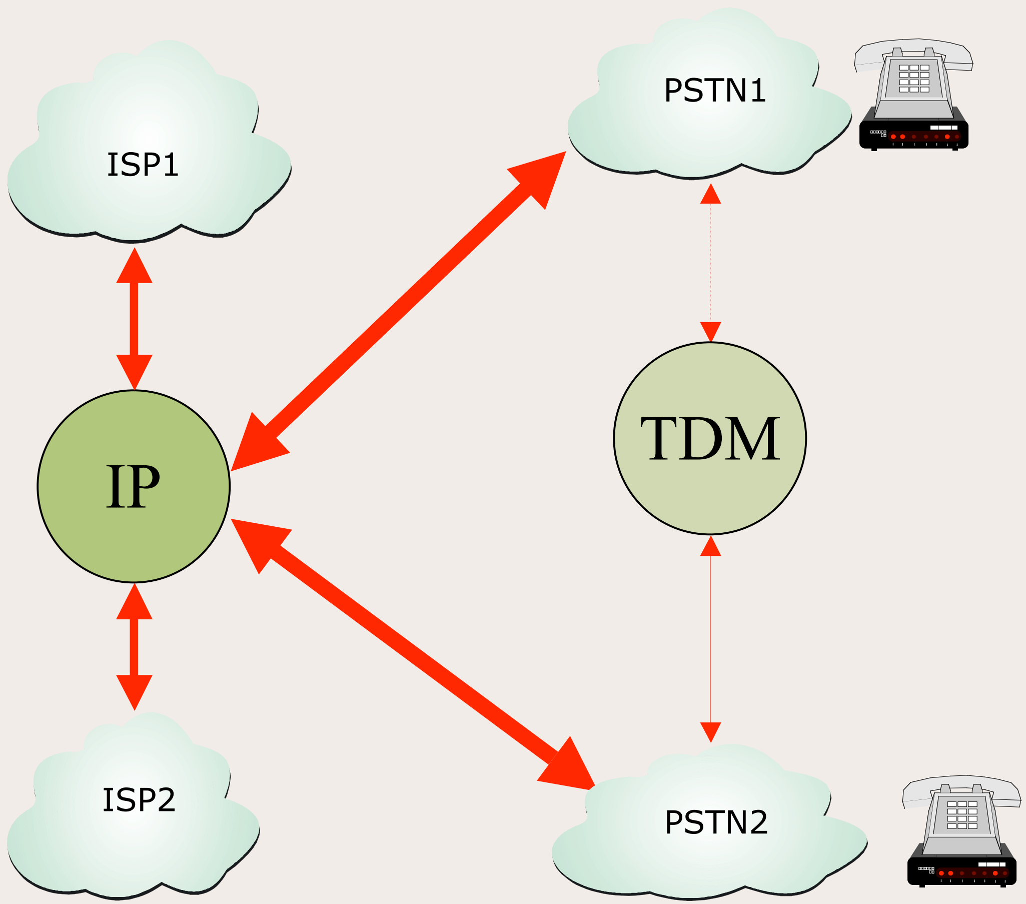
Voice → IP → Voice [P2P, Skype, Messenger]

Voice → IP → PSTN [Net2Phone, Deltathree]

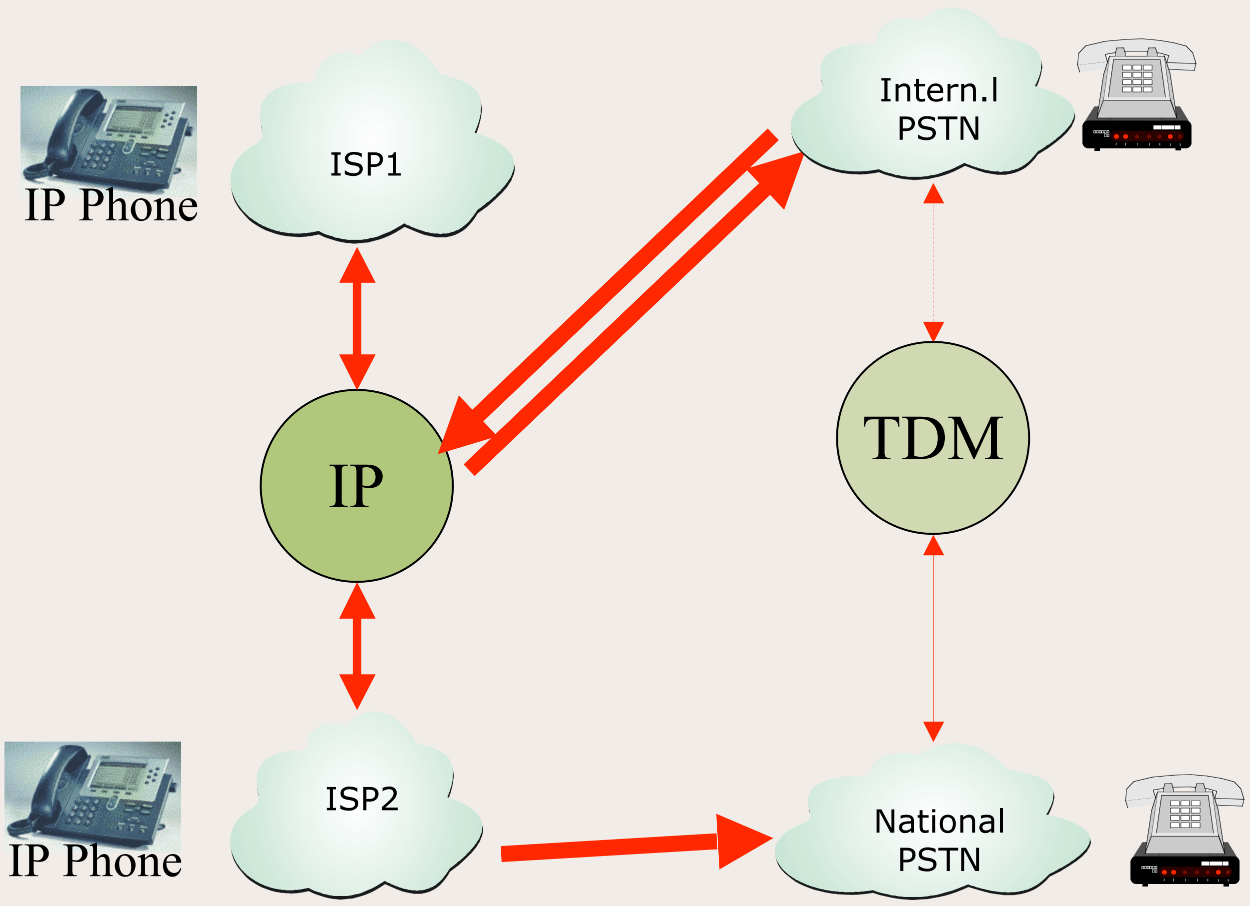
Voice [PSTN] → IP → PSTN [iBasis, ITXC]

Voice [GSM] → IP → GSM/PSTN [???]

Games the big boys play …



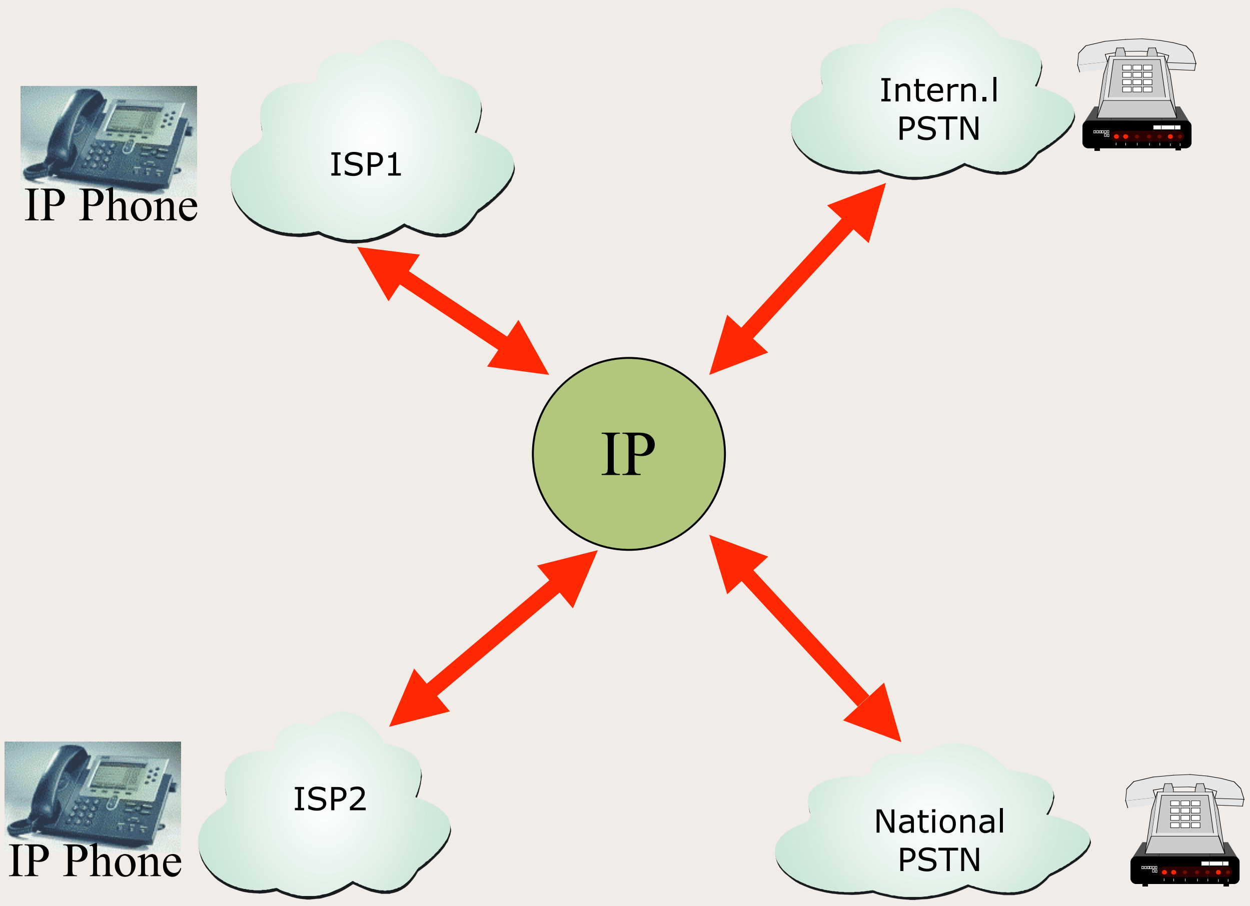
Little kids also play …



The VoIP edge

- IP is Scaleable
- IP conserves capacity
- IP simplifies charging and billing
- A turf for ISPs to play on …
  - Softphones for PC to Phone and PC to PC calls
  - Web-based applications for web to phone services
  - Move phones into the IT department and away from the expensive PBX consulting firm
  - Interconnecting office PBXs at zero network cost
  - Give ubiquitous access to the PBX for home/traveling employees
  - PBX features such as Voicemail, Call blocking, Call forwarding, Call Conferencing, Follow me etc as added services

Universal Access



VoIP Building block

VoIP is not built on TCP, but RTP

- RTP (Real-Time Transport Protocol)
- RTCP (Real-Time Control Protocol)
  - RTP is a UDP stream with no intelligence for QOS or resource reservation
  - Contains a packet number for detection of packet loss and re-sequencing of out of order packets.
  - Unidirectional : two streams in any call

VoIP Building block

- Calls are CODed to IP or DECoded from IP.
- CODECS vary in sample size, usually Kbits per second
- Decoding can include echo cancellation
- Decoding can compensate for jitter
- IP routers do not need to decode voice passing through them

VoIP Building block

Sample CODEC Sizes

– G711alaw	64k
– G711ulaw	64k
– ILBC	15k
– Speex	2.15 – 44.2k
– Gsm	13k
– G729	8k
– G723	5.3 - 6.3k
– Iax2 (trunked)	4k

Codecs that compress to lower bandwidth are CPU intensive, unless the codec is implemented in hardware. Strike a balance!

Control Protocols

- H323 – Complex, multiple flow, ancient
  - Has a large install base
- Session Initiation Protocol (SIP)
  - New, simple, only sets up RTP streams
- Cisco Skinny (Proprietary)
  - Allows complete phone customization
- MGCP (media Gateway Control Protocol)
  - Good but Not widely deployed as SIP
- IAX (Inter-Asterisk eXchange)
  - Simple, transverses NAT, Compressed

SIP

- SIP messages are HTTP-like and readable
- Supports Video
- There's lots of hardware SIP units available
  - Grandstream BT-101/2
  - Cisco 79xx)
- Not suited for Trunking (pbx to pbx)
- SIP is responsible for the increased use of VoIP

IAX(2)

- Inter Asterisk Exchange
- Not many Hardware phones support IAX.
- Soft Clients available for *unix/Windows
- Works behind NAT
- Has Trunking support built in
- Very low bandwidth requirement
- Built for asterisk

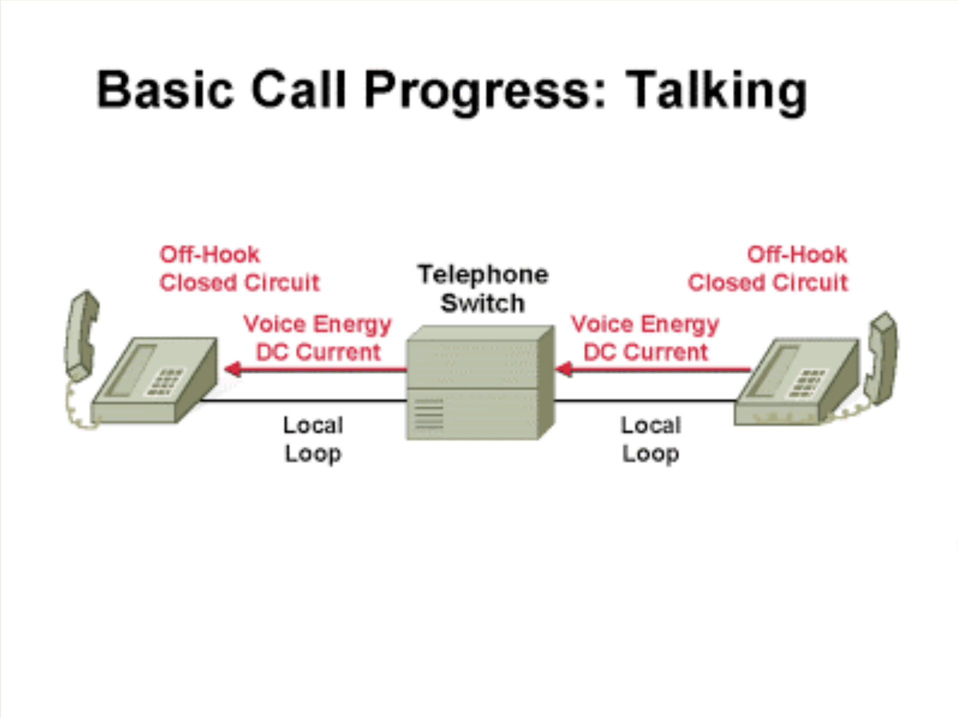
Phones

- Soft phones
  - X-lite - www.xten.com (Windows)
  - Lipz - www.lipz4.com (Linux)
  - DIAX - http://www.laser.com/dante/diax/diax.html (Windows)
  - PhoneGaim www.phonegaim.com(Linux)
  - Linphone - www.linphone.org (FreeBSD)
  - Sjphone - http://www.sjlabs.com/sjp.html (Windows, WinCE, Mac)
  - Lots of others

Phones

- Hard phones
  - Cisco 79XX's
  - Grandstream BT 10X's
  - Snom 100/200's
  - LOTS of h.323 phones from .tw ;-)
  - Many other phones

Most IP phones can work Peer to Peer



It is the Ability to use a PC as switch or PBX that really makes VoIP rock!! Simply loading a software PBX on a PC offers new possibilities …

PBX Software

Call Manager

- Closed Source
- 13 → 16 CD's
- Web Interface
- Requires CCNA to setup
- Needs extremely powerful Server
- Leaves PRI/FXO/FXS to other devices

Asterisk

- Open Source
- A large array of tools and add-ons
- Uses industry-wide devices and equipment
- Can be setup in one night

What is in VoIP for operators?

Some uncharted colonies …

- WiFi/WiMax Phones for universal access
- True Global roaming ;-)
- Enum adoption
- Numbering plan, being able to really “Play”
- Receivership for Long Distance companies

Asterisk Open-Source IP PBX

Asterisk is not …

- A billing system
- A CRM system
- A web server or XML server (re: Cisco 79xx)
- A configuration tool for VoIP devices
- A voice recognition system
- A USENET or email client

Asterisk is a ….

- Telephony gateway (TDM - PRI,POTS)
- VoIP Gateway (IP channels)
- IVR system (Interactive Voice Response)
- Voicemail system
- Meet-me Conference system
- Scriptable telephony-to-anything (Perl, C, etc.)
- Automatic Call distribution (ACD) system

Practical Uses (office)

- Ditch your LD company
- Interconnect office PBXs at zero network cost
- Get “Unified Messaging”
- Give ubiquitous access to the PBX for home/traveling employees
- Disaster recovery scenarios
- Move phones into your IT department and away from your expensive PBX consulting firm
- Eliminate adds/moves/changes as physical chores

System Requirements

- No clear rule of thumb on processor size; at least 400mhz PIII recommended
- Works on almost all Linux Distributions and FreeBSD
- Source + binaries (including sounds) are ~35Mb
- Using complex codecs (i.e.: G.729, speex, etc.) will increase processor load dramatically

Estimated CPU Sizing

Purpose	Simultaneous calls	Minimum Recommendation
Hobby System	<5	X86 400Mhz 256MB
SoHo System	5 - 10	X86 1Ghz 512Mb
SMB System	10 - 15	X86 3Ghz 1GB
Large	>15	Dual CPU, Clusters

Compatible Interfaces

Many interfaces for converting between Voice/IP/TDM are compatible with Asterisk. These include

- POTS cards (Digium, Zapata, Voicetronix, etc.)
- TDM Digital (AdTran VoFR, Digium E1/T1, etc.)
- CAPI (ISDN card support for Linux ISDN driver)
- USB dongle for FXS
- Modem drivers for certain modems
- Speaker/headphones via soundcard

Basic Installation Steps

1. Setup CPU and operating System
2. Install desired hardware based on application intended
3. Download asterisk from www.asteriskpbx.org
4. Compile and install with “Make”
5. Load Appropriate drivers [None is needed for IP or soft phone]
    1. Configure modules.conf
    2. Configure either sip.conf or iax.conf
    3. Configure extensions.conf
6. Start Asterisk
7. Make calls!

Extensions.conf (Call Flow)

- Calls come in on channels and are then handed to the “extensions.conf” file, which is the dialplan
- Dialplan contains logical sections of matches called ‘Contexts,’ and each channel sends a call into the dialplan with a context name and a dialed number
- The dialplan then matches (with modified regexp’s) the number being **dialed**, and runs applications accordingly
- Each match on the dialed number has an order of steps called ‘Priorities’, and are indicated with an integral incrementing number (BASIC-like)

Other use ….

- Call queues - you can build a call center with Asterisk, with various call weightings and agent logins/hot seating
- Multi-ring, cascading ring with different technologies (inbound calls forward to your desk line and your cell phone - first answer gets it)
- Multi-language support with same dialplan
- Festival integration for voice synthesis

References ….

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