VoIP for Business

A discussion of issues relating to deploying Asterisk based VoIP systems in a business setting

Tim Frichtel
TimFrichtel at yahoo dot com

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Disclaimer

- Views expressed are **my personal opinions**, not those of my employer
- Not affiliated with any vendor mentioned
- I'm not an Asterisk developer

Presentation Goals

- Overview of VoIP
- Business implementation of Asterisk
- Lessons learned
- Costs
- Resources

- *Cheap long-distance not covered*
- *Configuring Asterisk not covered*

VoIP vs PSTN

- PSTN
  - Public Switched Telephone Network
  - Calls go through physical circuit paths
  - 100+ years of maturation/optimization
- VoIP
  - Calls are packetized, carried over IP
  - Can be peer-to-peer or through a PBX
  - Systems can be open or proprietary

Why install VoIP?

- New calling features
- Better integrate voice & data (CTI)
- Easier/cheaper administration
- Increased redundancy

- Maybe to reduce long distance costs

- Not to get “better” voice
- Not necessarily to get better support

Open Source VoIP Pitfalls

- Compared to PSTN, VoIP is immature
- No single vendor “Turn-Key” solutions
- More features = greater complexity
- Total control = Total responsibility
  - For configuration
  - For testing
  - For integration
  - For troubleshooting
- *You are the integrator!*

Why Open Source VoIP?

- To eliminate vendor lock-in
- To take total control of system
- To avoid license costs
- Choice in:
  - Phones
  - PBX Hardware
  - PSTN Gateways
  - SIP Service providers

Why we chose Open Source

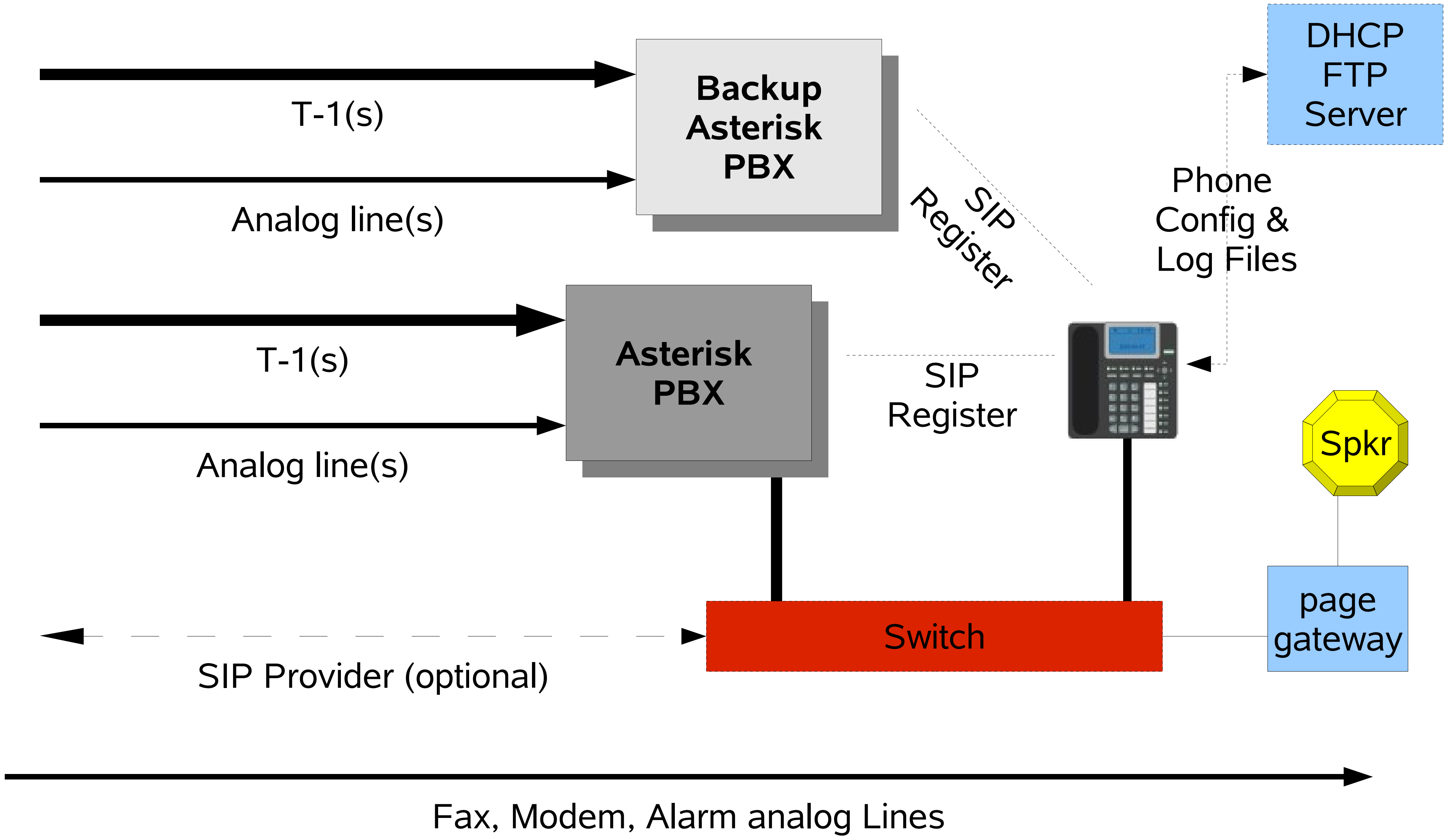
- Cost
- Speed of deployment
- Desire to avoid vendor lock-in
- Wanted to really “own” system
- Comfort with Linux and other OSS

- *No big vendor would do a pilot*

Our Requirements

- Single Building
- 100 employees + conference rooms etc.
- Receptionist
- Overhead Paging
- Voicemail
- Conferencing
- A few Fax & Modems
- NOT a call center

Business VoIP System



Choosing a Prototype

- Implement a small system for test group
- Researched OSS & Commercial
  - Cisco, Nortel, Mitel, Zultys, Siemens
  - Asterisk, Pingtel
  - Full system quotes - from $85,000 to $110,000
- Asterisk looked (and *IS*) hard
- Hybrid OSS/Commercial chosen
  - Switchvox (based on Asterisk)

Switchvox selected

- Switchvox targets smaller deployments
- Vendor support valuable
- Low cost, Low Risk
  - Standard Dell Server
  - Polycom Telephones
  - Pre-configured to our needs

Switchvox Pilot Solution

- Dell 1800 Server
- Switchvox software
  - Fedore Core
  - Asterisk
  - Proprietary Switchvox Admin Interface
- Digium TDM 400P Analog Card
- Polycom 300 and 600 telephones
- PBX & phones came pre-configured
- It worked perfectly when we plugged it in!

Test successful but...

- Analog PSTN interface prone to echo – eventually mostly resolved
- Didn't replicate every feature of existing Nortel Meridian
- Network and power challenges became obvious

- Pilot demonstrated how involved replacing a **10-year old stable system** would be...

Pilot Phase II

- Upgraded to 4 port Digium T-1 Card
- Excellent tech support from Switchvox
- T-1 cable was a guess
- Using phones from other vendors was a steep learning curve – but worked
  - Cisco 7940
  - Grandstream GX2000

Pilot Lessons Learned

- Features are split between phone & PBX
- Phones are “partners” with PBX
- Configuring phones manually is tedious
- Lots of data in a phone system
- “Closed” Switchvox system:
  - Great at what it does
  - No direct access to Asterisk
  - Out of luck if you want to do something it doesn't

Prep to Deployment

- Designed PoE Network
- Chose Telephones
- Created database to track names, extensions, telephone MAC address and PBX passwords for telephones
- Created app to generate XML files to configure Phones
- FTP server for phone configuration files & phone log files

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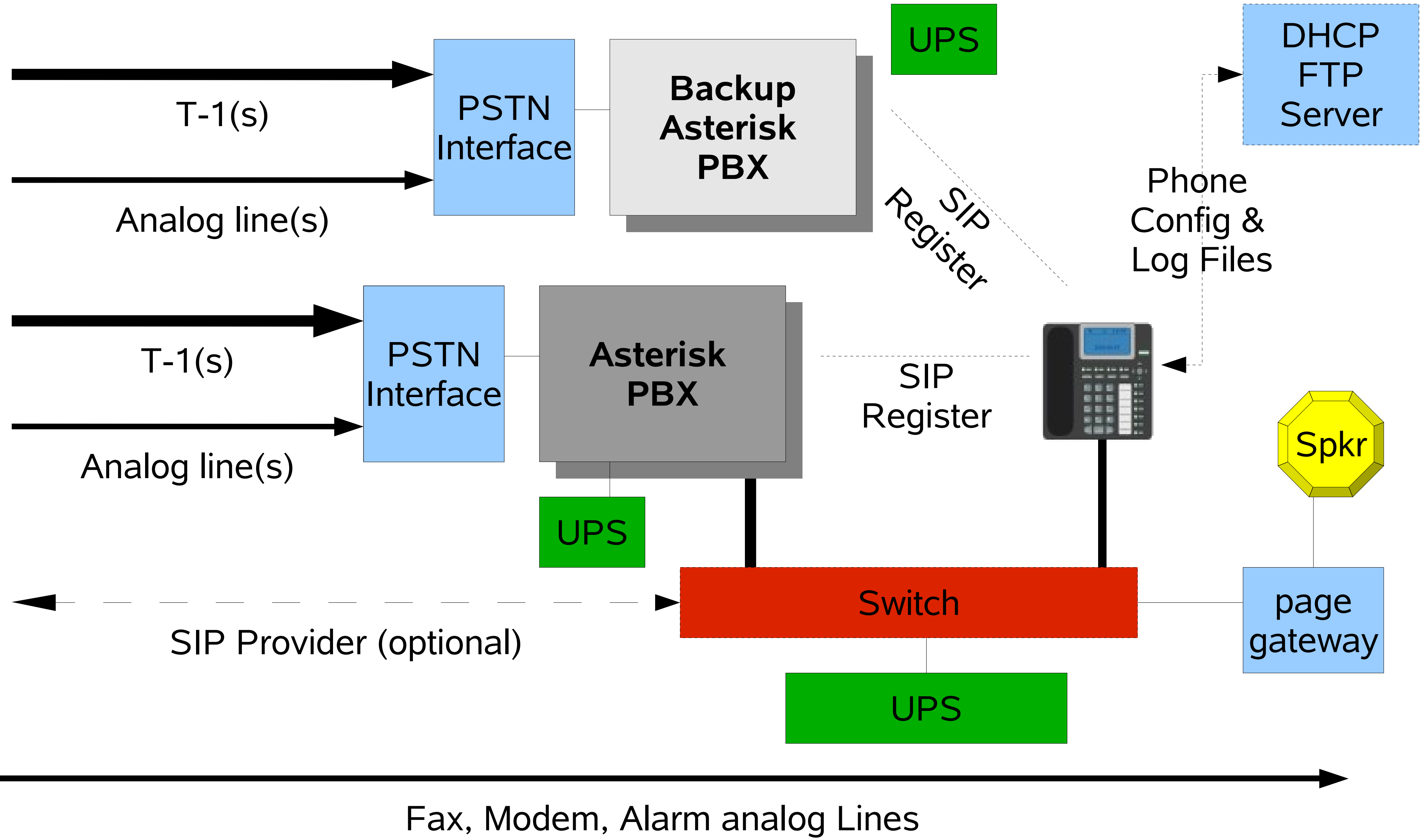

“Component” Costs

- Linux - $0
- Asterisk - $0
- PBX Hardware - $1,000 ~ $3,000
- Telephones - $80 ~ $500 each
- PoE Switch - $800 ~ $3,500
- UPS for Server and PoE switch
- PSTN Interface $200 ~ $5,000

PSTN to VoIP Integration

- Telcom people don't know VoIP and IT people don't know Telecom
- Be prepared to learn Telecom
- You are the integrator
- Good Telecom support is critical

PSTN Interfaces

- Analog
  - Analog Telephone Adapter (ATA) – Sipura
  - Analog Interface card* (Digium)

- Digital (PRI – T1/E1)
  - Internal Interface Card* (Digium, Sangoma)
  - External Router (Cisco)

  Caution! - Internal PSTN cards aren't always compatible with, and may not fit in all PCs/servers

PSTN Interface - Analog

- Analog Lines
  - Good for low external call volume
  - ATAs are inexpensive
  - Echo is a bigger issue
  - Analog lines cost tens of dollars/month
  - “Built in” fall-back – plug a telephone in!
  - Your phone number is tied to Central Office
  - Expensive for multiple DID (direct inward dial) lines

PSTN Interface - Digital

- Digital Lines
  - T-1 PRI provides 23 circuits
  - Can be less expensive than same number of analog
  - Echo rarely an issue
  - PRI has more features than analog
  - Need expensive, specialized test equipment to troubleshoot -- (or a vendor)
  - T-1 Interface card more expensive, but only real way to go for many DID lines
  - Channel banks can be used to split analog lines out of T-1 for fax/modem

PBX Server Selection

- Buy reliable hardware
  - Redundant power supplies
  - RAID
  - ECC memory
- Moderate speed is OK
- RAM > 512 (more for conferencing)
- Disk – voicemail is biggest consumer
- *Check compatibility with PSTN cards!*
- UPS recommended

Asterisk PBX Provides

- PBX functions to control calls
- Voicemail
- Call Conferencing
- Music On Hold
- Call Queues
- Integrated Voice Response (IVR)
- Integration with Data System
- Call Reporting/Logging

Asterisk Configuration

- Create SIP channels
- Create PSTN channels
- Create **dialplan**
- Configure voicemail, email integration
- Create IVR/Call Queues

- “User Friendly” interfaces are important
  - Switchvox (can't see Asterisk files)
  - Sigman (can see Asterisk files)
  - Asterisk@Home (can see Asterisk files)

Telephones

- Choose SIP
- Features, Cost & Quality vary widely
- “Feel” and “Sound” of phones important
- Message lights, buttons matter
- Headsets support generally integrated
- Check CODEC support
- Power choices
  - AC Adapter
  - Power Over Ethernet (may require special cable)

Voice Quality Factors

- Handset versus softphone
  - Quality of sound card
  - Quality of microphone/earphones
  - PC stability and availability
- Codec - G711
  - free, high quality & bandwidth – for LANs
- Codec - G729
  - licensed, good quality, low bandwidth – for WANs
- Others

Voice Quality Factors -2

- Echo
  - Echo is part of telephone life
  - Is less with digital (T-1/E-1) PSTN connections
  - Some echo is good (*side tone*)
- Latency
  - Delay due to equipment, routing, distance etc
- Jitter
  - Due to dropped, out of sequence packets
- Dropped Packets
  - A few lost packets won't be noticed

Telephones – PoE

- No need for power outlet by phone
- Reduce cabling at telephone
- Phones may use bulky, expensive cable
- Only reliable if supported with UPS
- Not available on all phones or ATAs

Polycom PoE Cable

- PoE cable is 9' long
- PoE block is 1.5' from connector
- Must be connected to standard cat-5

Telephones - Configuration

- Telephones require configuration
- Most phones support:
  - Keypad entry (limited and painful)
  - In-phone web server
  - XML file – *best way to go*
- Typically XML file:
  - Features
  - Phone extension
  - PBX passwords
  - Multiple PBX registrations

Network - Dedicated/Shared?

- VoIP needs good networking
- Shared VoIP & Data
  - Less cost (only 1 set of switches)
  - Less wire / power / space
  - Quality of Service (QoS) require network skills
- Dedicated VoIP Network
  - Expensive
  - “Throwing money at the problem”
  - Avoids contention with data network

Power over Ethernet (PoE):

- Two PoE Options:
  - PoE Switch (Power and Switch together)
  - Power Injector (between phone and switch)
- PoE switches
  - Are more expensive
  - Draw more power
  - Generate more heat
  - Have louder fans due to heat
  - May require external power supplies for redundancy

VoIP "Sidekicks"

- *Optional but recommended*
- User & telephone database
- Backup system for PBX config files and Voicemail
- TFTP, FTP or HTTP server to deliver telphone config files
- DHCP server to provide IP addresses to phones

Username/Password Mania!

- Phone
  - password to access Asterisk
  - password to change in-phone settings
  - password to access FTP site for config files
  - password for telephone web config pages
- User
  - Voicemail password
  - Web interface password

More Username/Password Mania!

- Admin
  - PBX server root
  - PBX administration interface
  - Telephone FTP upload password
  - Vendor support web site login
  - Network Switches
  - Backup server (if separate)

Bringing it All Together

- Build the network
- Install & configure the server
- Build the user database
- Provision the telephones
- Connect to the PSTN
- Train the users

Our System

- Live since October 2005
- 2,000 calls/day
- $39,000
  - 2 Dell PBX (1 switchvox, 1 just asterisk)
  - 2 T-1 cards (1 per server)
  - 120 telephones
  - Dedicated voice PoE network
  - Overhead speakers for 30,000' office
- Employees happy

Good Luck!

Miscellaneous

"Selling" Management on Asterisk

- Hire a vendor
- Positive reviews in PC Mag, E-week etc
- Demonstrate with low-cost pilot system
  - Switchvox, Signate, Asterisk@Home
- Demonstrate using multi-vendor phones
- Showcase cool new features
  - Email a voicemail
  - Use softphone

- Don't Over Sell!

VoIP Resources

- www.voip-info.org (wiki)
- Asterisk mailing list (not for the meek!)
- Books:
  - Switching to VoIP (O'Reilly)
  - Asterisk (O'Reilly)
  - VoIP Telephony with Asterisk (Signate)
- White Paper
  - Voice over IP 101 (Juniper Networks)
- Asterisk overview video – revision3.com/systm/ episode #5

Resources – Asterisk Training

- Digium – 3 days, $1,500
- Signate – 4 days, $1,795
- Astricon – 5 days, $3,000

Resources – End User Training

- Download telephone manuals
- Create “help sheets”
- Telephone demonstration classes

Softphone Issues

- Softphone success depends on
  - Quality of sound card
  - Quality of microphone/earphones
  - PC stability and availability
  - PC network
- User reaction
  - Headsets can be uncomfortable
  - Not as convenient as physical phone
  - OK for travel and occasional use

VoIP – Protocols

- Signaling to <u>control</u> calls
  - **SIP** = Session Initiation Protocol
  - SCCP = Skinny Client Control Protocol (Cisco)
  - H.323 – older, more complex
  - MGCP = Media Gateway Control Protocol
- Voice Data ("payload")
  - **RTP** = Real Time Protocol

VoIP – Common Terms

- PSTN = Public Switched Telephone Network
- Central Office = phone company center
- ATA = Analog Telephone Adapter
  - connects analog phones to VoIP
- Foreign Exchange (Station, Office)
  - FXS = Interface to connect analog telephone device to VoIP
  - FXO = Interface to connect analog line to VoIP

Overhead Paging

- Another Integration Challenge
  - Speaker System Choice & Design
  - Cable Installation
  - VoIP to Speaker integration
- Cobbled solution works!
  - Grandstream 2000 phone cut open to feed amplified speaker system
- Better solution coming
  - Sipura ATA and Bogen TAM-B interface