Production Quality Open Source VoIP: Next Generation Telephony at Penn

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Agenda

- **Penn background and goals**
- **Review of Penn's open source project**
- **Open source applications and architectures that have worked best**
- **Major obstacles in the use of open source IP Telephony**
- **Lessons learned from open source experiences**

Penn Background

- An Ivy League university in Philadelphia, PA founded in 1740 by Ben Franklin

- 40,000 faculty, staff and students
- 25,000 phone lines/12,000 voicemail users
- 50,000 IP addresses in use
- 9,000 analog video connections
- Over 1.5 Gbps external IP capacity

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Voice Services at Penn Today

- Most telephony via LEC-provided Centrex service
- LEC brings copper loop infrastructure to campus Rate Demarcation Points (RDPs), Penn maintains in-ground and in-building copper to the wallplate
- This infrastructure is old and failing resulting in service outages and requiring expensive repairs
- Replacement of these cables would be $3-5M

Business Drivers

- Aging phone-cabling infrastructure
  - In-ground cabling is 25-75 years old
  - Cable faults are increasingly frequent
  - Cable repairs are expensive and time consuming
- Operational inefficiencies
  - Separate cabling infrastructure
  - Moves, adds and changes are time consuming and costly (3000/yr @ $125= $375k)
- Legacy Voice Mail and ACD are business critical, but fragile and costly to support

Voice Services Goals

- **Generally**:
  - Cost–effective, reliable voice communications with flexibility to meet evolving demands
  - Vendor independence
  - Single integrated network infrastructure
- **Specifically**: Convert 25,000 analog voice customers to VoIP on a converged IP network with added functionality and lower costs in 5 years or less

History of VoIP at Penn

- Worked with H.323-based VoIP in late 1990s, later moved to Session Initiation Protocol (SIP)
- Worked with a range of applications:
  - Cisco Call Manager, an enterprise solution (2001-2002)
  - BroadSoft's Broadworks, a carrier-class VoIP software solution, piloted on campus (2001-2003)
  - Externally hosted IP Centrex solutions (2003-2004)
  - Open-source SIP Express Router with Asterisk Comedian Voice Mail (2003-Present)

Whether to use Open Source

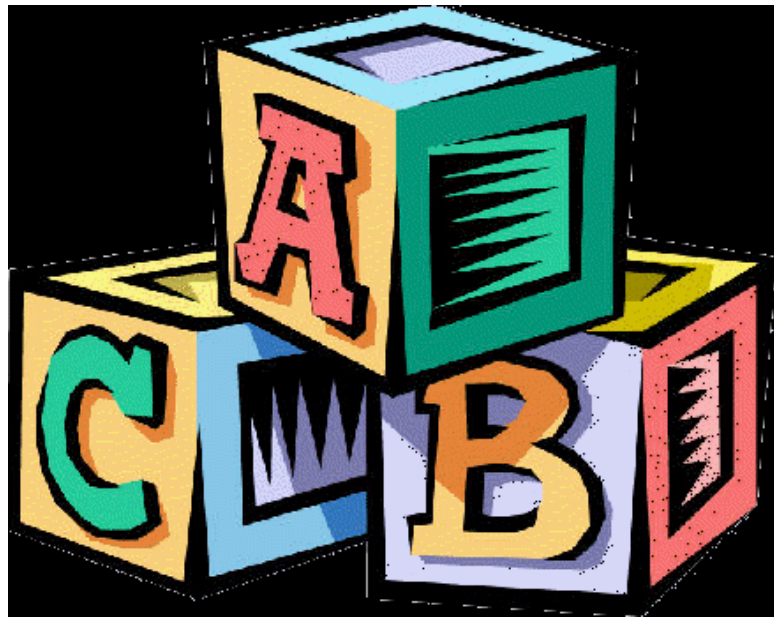
- Does IP Telephony lend itself to open source server software?
  - Can it be reliable enough?
  - Will we have the feature set we need?
  - Will we be able to maintain it well?
- We have an IT staff experienced in the 24x7 operation of key services based on open standards and open source
- Positive experience with open source server software encouraged us to take this as a serious option

Open Source Strategy

- Select open source tools written in programming languages in which we have expertise
- Assure that multiple staff members are familiar with each open source tool
- Avoid the over-customization trap
  - Use existing functionality where possible
  - Maintain active relationship with developers
  - Try to get any essential changes of general interest built back into mainline code
- Notable Asterisk customization
  - **IMAP access** to voice mail message store for true **unified messaging** from **phones** and **email clients**

VoIP Building Blocks

- SIP Clients or User Agents
- SIP Proxy Servers
- Media & Feature Servers
- Gateways and ITSP services



SIP User Agents (UAs)

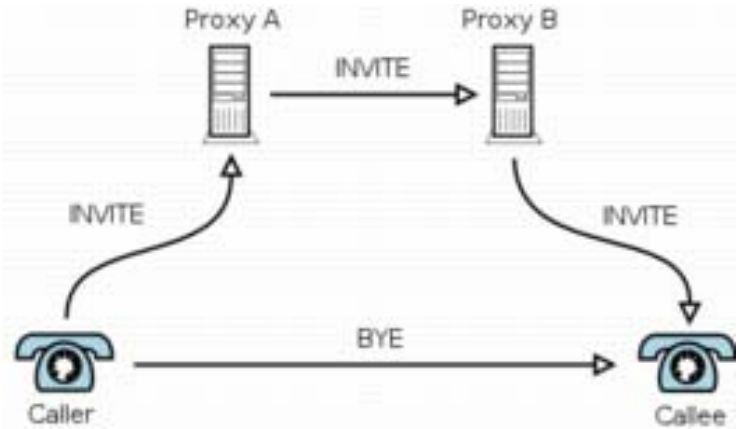
- Hardware SIP phone, in the familiar desk set form factor. Penn currently uses the Cisco 7940 phone.

- Software that acts as a SIP client or user agent and runs on a desktop, laptop or handheld. Shown here is eyeBeam from Counterpath.

SIP Proxy Servers

- Server that accepts “registration” from valid users
- Allows client to signal their call information
- Handles call set up and gets out of the way
- Penn uses SER (iptel.org)

and OpenSER (openser.org)

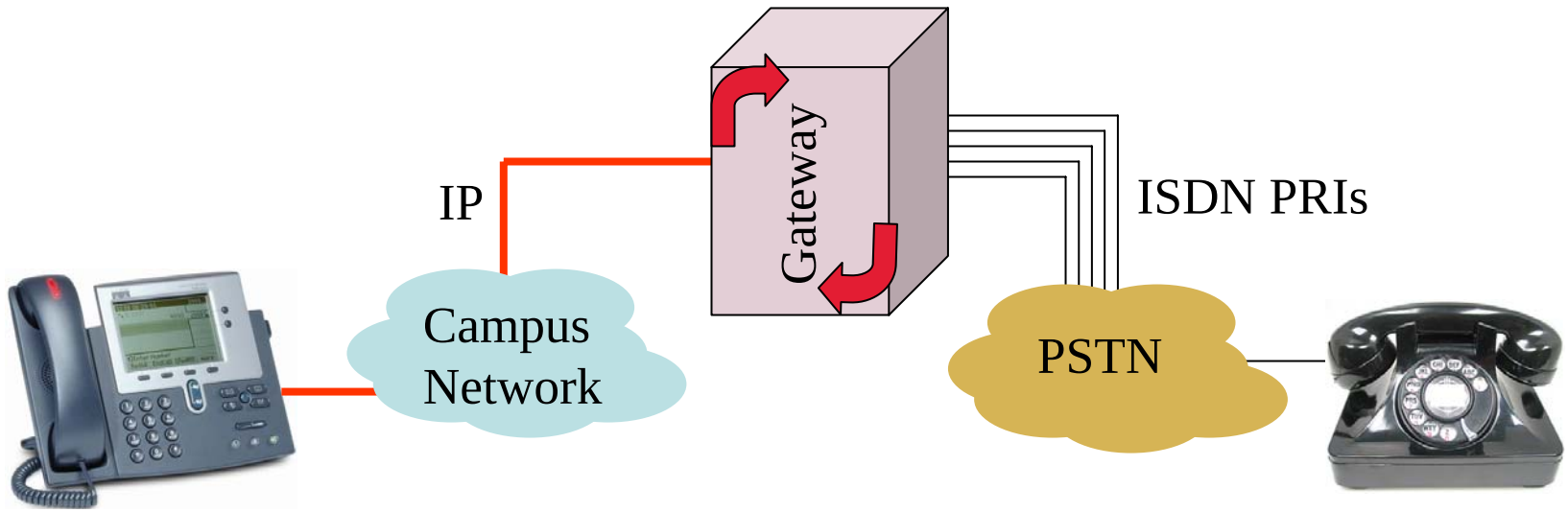


Media & Feature Servers

- Server that handles “media” such as voice mail and music on hold
- Provides for some additional call features
- Co-exists with provisioning services
- Penn currently uses Asterisk (www.digium.com)

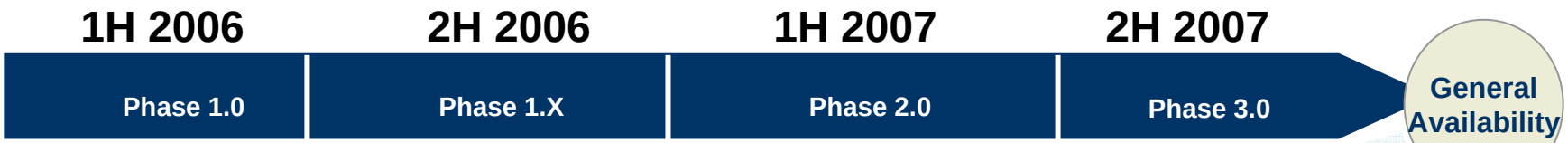
Gateway (and ITSP)

- Interconnect between campus network and PSTN
- Penn currently uses Cisco 36xx routers



- Penn also uses ITSP services from major carriers for off campus calls

Penn iPhone Roadmap



Phase 1.0

- VoIP Campus Pilot Begins
- Penn iPhone web portal
- Directory lookups

Phase 1.X

- IMAP Message Store
- Multi-line Pilots
- Network security stage 1
- Handset Evals
- 600+ phones

Phase 2.0

- Feature Rel. 2.0
- Softphone Pilots
- Centrex-Asterisk Pilot(s)
- ITSP/LNP in Production
- New Handset Testing

Phase 3.0

- Octel voice mail migration
- New Handset Production
- Softphone Pilots
- SRTP deployments
- Multi-line Pilots
- 2000 phones deployed

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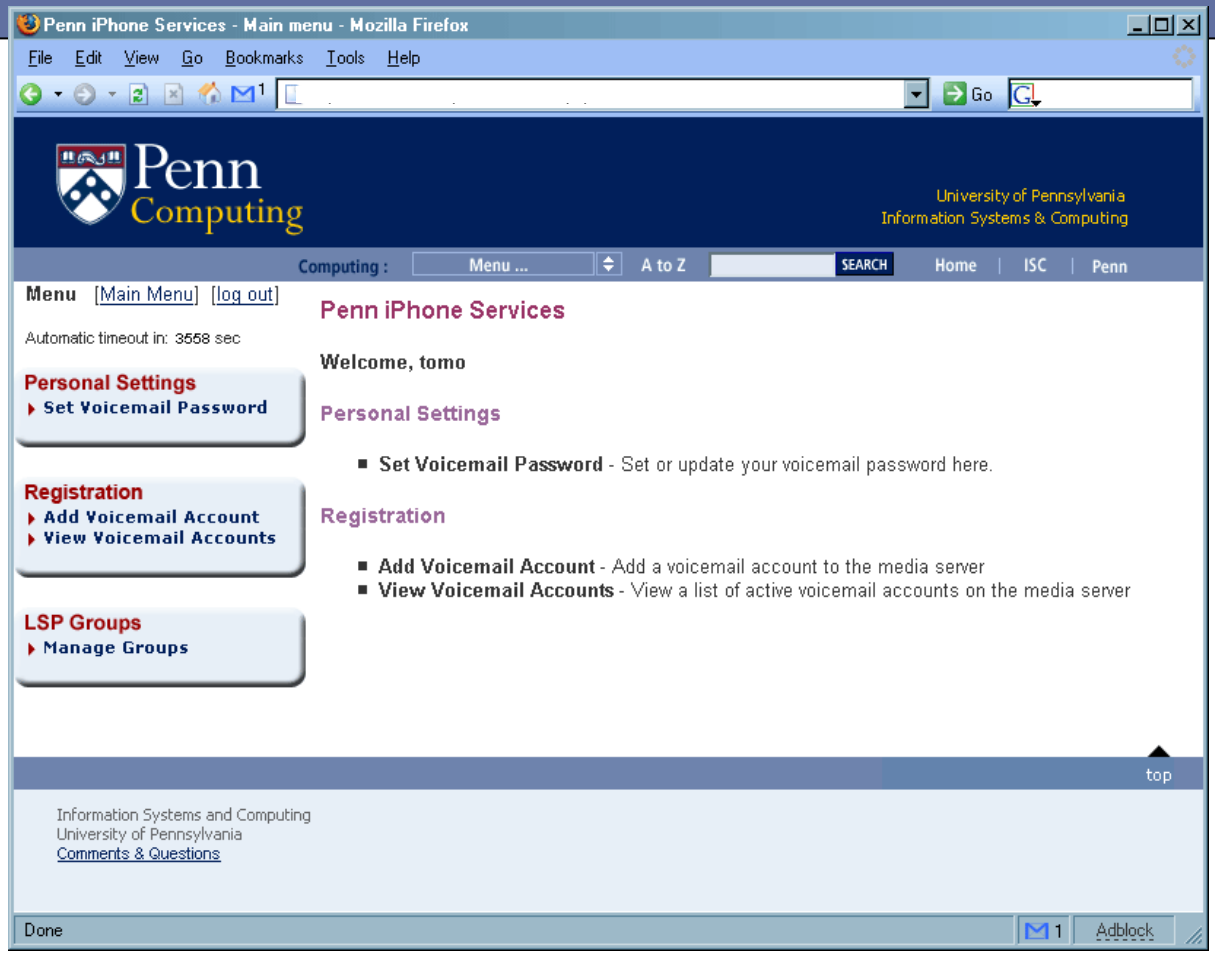
Current State of Service

- Production-grade, enterprise VoIP Services
  - Redundant servers, gateways and PRIs
  - 24x7 monitoring and support
  - Single-line features, email/voicemail integration
  - Basic multi-line services
- 911 support equal to legacy system
- IMAP-based unified messaging
- Layer 2 QoS, separate VLANs & subnets
- 1,000 VoIP phones in full production

Managing Penn iPhone

- Locally developed web provisioning tool to allow
  - Installation staff to configure phones for deployment
  - Support staff to provide local support
  - End-users to select and maintain configuration options

Penn iPhone Web Services



Self-service
Voice mail
Change Email Password
Do Not Disturb
Call Forward
Reject Anon.
Blocked Caller ID
More . . .

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Unified Messaging

- **By phone or by email?** Users want to decide how to access their voice messages at listening time, not at delivery time.
- We wanted to support this choice ***without*** message replication
- Penn-developed extension to Asterisk **for IMAP access** to voice mail for true **unified messaging** from **phones** and **email clients,** included in Asterisk version 1.4

Emergency Services & Location

- Ability to route 511/911 calls, including support for location information
- Three phases towards improvement
  - Phase 1 - Fixed location phones
  - Phase 2 - VoIP phones movable by IT Staff, with web form location reporting
  - Phase 3 - Dynamic location updates

Current Development Projects

- Advanced multi-line support
- Using ITSPs (Internet Telephony Service Providers) for bi-directional off campus calls
- Migrating Centrex users to our *next generation* voice mail
- Power Over Ethernet (PoE) network designs
- Evaluating additional VoIP handsets

Primary Challenges

- Implementing required phone features for which open SIP standards are not yet in place, such as Bridged Line Appearance
- Implementing standardized features not yet widely implemented in phones, such as SRTP and associated key management
- Finding cost-effective, fully featured phones with strong SIP software loads

Collaboration within Higher-Ed

- Several major universities are using a very similar approach
- Sharing plans, source code and best practices through monthly collaboration calls
- Open to participation by other groups seriously engaged in similar efforts

Lessons & Recommendations

- Cross-train staff, and allow time for adjustment. Augment team with consultants in key areas
- Create and maintain a dedicated voice services development environment
- Deploy new voice services in your own department first, as a pre-pilot
- *Replace* phones with VoIP (don’t simply add)
- Make project a priority across the organization

Looking To The Future

- With the elimination of legacy services and networks, we anticipate:
  - Better customer service
  - Increased functionality and reliability
  - Improved operational efficiencies
  - Telecommunications cost reductions of 10-30%