Open Source Voice over IP (VoIP) at Penn

Shumon Huque
University of Pennsylvania

Winter 2011 Joint Techs Conference
January 31st 2011, Clemson, SC, U.S.A.
The University of Pennsylvania is in the midst of a multi-year deployment of a campus-wide Voice over IP system based on open source components on the server side of the infrastructure and open protocols (SIP). This talk will review the architectural details, progress to date, future plans, and touch on some of the specific technical challenges we've faced.
Brief background

• Analog Telephone system:
  • Verizon Centrex, over 20K lines
  • Old copper infra, outages, long prov time
• Protocol research & testing in late 90’s
  • H.323 initially, later SIP (Session Initiation Protocol)
• Formal VoIP project began 2005/2006
• 6,500 VoIP lines so far (production)
Server Infrastructure

- SIP Registrar & Proxy servers (iptel SER)
- Voicemail servers (Asterisk)
- SIP Presence servers (OpenSIPS)
- PSTN gateways (cisco 3845 routers + voice cards)
- In-trial: SIP Trunking (Verizon ITSP)
Clients

- Handsets from Polycom (Soundpoint IP 321/550/650, Soundstation 6000)
- Have previously used Cisco handsets (7940 and 7960)
- Soft Clients: experimental, small number of users; not supported in production
Sampling of Features

- Basic Single Line
- Ring Groups
- Call Hold & Transfer
- Call Forward All
- Call Forward Busy
- Call Forward No-Ans
- Call Hunt
- Music on hold

- Staged/timed services
- Do Not Disturb
- Per extension VM dest
- Caller ID block
- Anonymous Rejection
- Out-call notification
- Distribution messages
- Advanced Caller Menus
Web Feature Management

Features and Voice Mail Settings - 215-898-2623

Your current PennNet Phone services are listed below. You can change your selections at any time. Changes will take place immediately after clicking the submit button unless the setting is marked as "Handset restart required".

Phone Number to View or Update Information
215-898-2623 - tomc

PennNet Phone Settings

Advance One
Advance One: on/off
* When on, Call Forward on Busy & No Answer will not work
Destination Number: [37471]
* 5 digit PennNet Phone Number

Call Forward All
Call Forward All: on/off
* Call Forward All will be off unless enabled by an entry below.
* Each entry overrides all entries above it for overlapping time spans. [Scheduler Help]
* The call forward all destination number may be displayed to other PennNet Phone subscribers when this feature is enabled.

<table>
<thead>
<tr>
<th>Action</th>
<th>Destination</th>
<th>Date</th>
<th>Start Time</th>
<th>End Time</th>
</tr>
</thead>
<tbody>
<tr>
<td>1:</td>
<td>215-833-7586</td>
<td>everyday</td>
<td>All Day</td>
<td>All Day</td>
</tr>
<tr>
<td>2:</td>
<td>9-1-0000-0000</td>
<td>on weekdays</td>
<td>09:00 AM</td>
<td>05:00 PM</td>
</tr>
</tbody>
</table>

[+] Add a new schedule entry

Call Forward on No Answer
Call Forward on No Answer: on/off
* Incoming calls are forwarded if your number doesn't answer.
Call Forward on No Answer Destination: [215-122-4587] [see valid formats...]

Caller ID
Blocked Caller ID: on/off
* When on, your caller ID is not sent when you place calls.

Voice Mail Settings

Voice Mail Delivery Method

1: Telephone only.
* Notification of messages via indicator light on handset an icon on telephone display and stutter dial tone is heard when the
handset is lifted. Messages are managed by telephone.

Submit
Challenges/Issues

• Many bugs and interoperability issues
• Timer issues, call loops, call transfer, forward, phone crashes
• System tuning and scaling issues
• IMAP storage of voicemail messages (for UC)
• Keeping up with SER community development
• BLA/SLA (Bridged/Shared Line Appearance)
BLA Issues

- **Bridged Line Appearance**: multiple sets share a number; call can be picked up at one set; held; transferred to another set etc

- Bugs and Interoperability issues with presence server (OpenSIPS) and handset (Polycom)

- Unclear (and unfinished) technical specifications for BLA (expired Internet-drafts etc; new BLA “requirements” draft)

- Deployed; backed out; debugging & repairing work going on for past 2 years

- Early Jan: working reliably in our lab
BLA Issues

- Dialogs stuck in various states (early, confirmed) -- stuck or incorrect lights on UI
- Stability issues with OpenSER
- Subtle interaction issues with other features (eg. call transfer, call forward, etc)
- Many rounds of fixes by various involved parties (us, opensips, polycom, etc)
Future Enhancements

- ITSP (SIP Trunking)
- Security Enhancements
  - Secure Signalling (SIP over TLS, etc)
  - Secure Media (SRTP, ZRTP, etc)
- Production support of Soft Clients
- Automatic location tracking (public safety)
- Proxy server update: “SIP Router” 3.x
Assessment

• OpenSource VoIP works and at large scale

• But, implementing certain advanced business class telephony features is challenging

• Need to be closely involved in open source development community and participate

• State of maturity of protocol specs is lacking

• Need strong relationships with other vendors
Assessment

- Cost savings: no purchase or license fees
- Vendor neutrality
- Locally customizable, locally fixable
- Ability to troubleshoot and debug better
- Shared community of knowledge
- Developers interested in open-standards and compatibility
Questions?

Shumon Huque
shuque -@- upenn.edu
Didn’t address

- Organizational/Staffing issues
- Project management structure
- Local IT and user support issues
- etc