VoIP Review

IT Partners

Wednesday, July 23, 2007
Agenda

- Project objectives
- Financial overview
- Current status
- New VoIP architecture
- Enterprise services demo
- Personal sip accounts
- Timeline
- Strategic Issues
- Support and provisioning model
- Communication and training
VoIP Project Objectives - 1

- Provide a flexible infrastructure for MIT faculty, students, and staff.
  - managed enterprise service for most users, and
  - open source environment allowing members of the community to experiment

**Basic Functionality**
- IP Phone Users
- Operates like ISDN phone
- new VMail ≠ old VMail
- No / little training required
- Full Support

**Advanced Functionality**
- Call control using Sylantro web pages
- Visual voicemail
- Remote office features
- Full Support

**Personal SIP Accounts**
- SIP account only
- BYO Phone, Client or Application
- Telephone number assigned
- Limited (5 digit) outbound dialing
- Limited IS&T Support, Peer Support
VoIP Project Objectives - 2

Smoothly migrate MIT to Voice over IP (VoIP) with minimal impact to the end user community; determine optimal timing for the transition.

- Highly robust, carrier class infrastructure.
- Roll-out plan developed in coordination with departments, labs, and centers:
  - site surveys of existing telephony services
  - scheduled migration
- Deployment support from intensive & local hands-on by IS&T to cooperative support for DLC IT staff.
VoIP Project Objectives - 3

- Deliver advanced end-user feature/functionality!

- Work remotely:
  - Remote IP phone
  - Bridge work line with a remote telephone line
  - Laptop client

- Call flow tailoring:
  - Simultaneous ring, find-me, follow-me features
  - Call screening by calling party, time of day, etc.

- Fax inboxes

- Converged WiFi/Cell phone:
  - Dual telephone numbers today, eventually migrating to a single telephone number with multiple personalities

- Real-time, presence based multimedia negotiation (future capability)
  - Video, Voice, IM
VoIP Project Objectives - 4

Embrace productivity gains from running a single network infrastructure, and supporting voice as an IP application.

- Single data network vs. separate data and telephony networks:
  - MIT-net carries both voice and data (Fiber/Ethernet/IP) vs.
  - MIT-net for data and telephony network (twisted copper/ISDN & POTS) for voice.
- Converged support: VoIP is another IP application.
- IP application model: VoIP is code running on Linux or Sun servers.
- Ability to move handsets rather than => service order and dispatch.
- Easily integrates with other voice service platforms: audio conferencing, IVRs, ACDs, etc.
VoIP Financial Plan

- Total cost to transition to VoIP over 7 years is ~$35.5M
  - One time implementation cost is ~$7.1M of total (>35% of this is for the telephone devices)
- Cost over same period if we “do nothing” is ~$31.8M
- Additional investment = ~$3.7M
- Partially offset by ~$1.5M in MIT savings from not wiring new buildings for traditional telephone service
- Savings in annual operating costs begin in FY12

CAN WE AFFORD THIS?
VoIP Status Update

MIT Open Source VoIP Pilot
- 1025 telephony accounts, ~ 975 VoIP devices registered (including ~65 WiFi phones)
- ~ 25 on campus, non IS&T subnet phones
- ~10 off campus phones
- Call routing (SIP proxy) service delivery platform (openSER)
- Voicemail platform (Asterisk)

Network and service architecture evolution based on:
- Feedback from Pilot re: features and functionality,
- Pilot support experience.

New architecture includes:
- Session Border Controllers that provide security and robustness.
- Voicemail/Unified Messaging service spanning VoIP and 5E environments (Iperia)
- IP Centrex VoIP cluster to support telephony features (Sylantro)
Pilot VoIP Architecture

5/9/2007

Internal Access (Line Side)

External Access (Trunk Side)

Shared Services

~ 1075 SIP accounts
~ 975 IP Phones
(inc. ~65 WiFi phones)

Outbound Proxy
Pilot: openSER

Location dB

Internal (routing) Proxy
Pilot: openSER

DMS Proxy
Pilot: openSER

Voicemail
Pilot: Asterisk

VoIP Gateways
Pilot: Cisco 5850

ENUM (DNS)

PSTN

Fax

Emergency Phones

Pilot VoIP Architecture

• ~ 1075 SIP accounts
• ~ 975 IP Phones
  (inc. ~65 WiFi phones)

5E

Octel Voicemail

Modern, point of sale, alarms, etc.

Public IP

Modem, point of sale, alarms, etc.
Intermediate VoIP Architecture

5/9/2007

Open Source

IP-Centrex

Session Border Controller (SBC)

Outbound Proxy
Pilot: openSER

Location dB

Internal Access (Line Side)

ENUM (DNS)

DMZ SBC

Shared Services

Internal (routing) Proxy
Pilot: openSER

SYL Presence Server

External Access (Trunk Side)

ENUM (DNS)

Voicemail (Iperia)

DMZ SBC

VoIP Gateways
Pilot: Cisco 5650

New VoIP Platform

Pilot Platform

New Analog Platform

Public IP

5E

Emergency Phones

Modem, point of sale, alarms, etc.

Fax

Public IP

PSTN

Emergency Phones

Fax

Public IP

PSTN

Emergency Phones

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Emergency Phones

Fax
Long Term VoIP Architecture

5/9/2007

~ 1-2,000 Open Source?
~ 13,000 IP-Centrex
~ 2,000 Analog Telephone Lines

- Internal Access (Line Side)
  - Location dB
  - Outbound Proxy
    - Pilot: openSER
  - SYL Presence Server
  - SYL Control Server (feature server, registration dB)

- Shared Services
  - ENUM (DNS)
  - Internal (routing) Proxy
    - Pilot: openSER
  - SYL Route Server

- External Access (Trunk Side)
  - DMZ SBC
  - Carrier SBC
  - Public IP
  - PSTN
  - VoIP Gateways
    - Pilot: Cisco 5850

- Messaging (Iperia)

- Emergency Phones
- Fax
- Modem, point of sale, alarms, etc.

- Other Shared Services (conferencing, presence, etc.)

New VoIP Platform
Pilot Platform
New Analog Platform

IST Information Services & Technology
VoIP Services Demo

End User Demo

- Bridge line appearance
- Simultaneous ring
- Find-me ↔ follow-me features
- Click to conference
- Call screening
- Remote office
Personal SIP Accounts

- Based on pilot/open source infrastructure
- Bring your own device
  - Soft clients (applications) that run on Windows, Macs, Linux, mobile devices
  - Phones & soft clients that support video
- Outgoing calls are restricted to Internet and campus phones
- Modified support model
  - Limited IS&T support
  - Peer support: wiki and user forums
  - Leverages Mobile Partners
- Allows for experimentation
  - Greater flexibility – lesser reliability
  - Features may evolve over time
  - Encourages innovation
VoIP Project Timeline

- **VoIP Pilot** (Feb 07)
- **Transition Plan** (Feb 07)
- **Infrastructure Development** (Feb 07)
- **SBC Deploy** (Feb 07)
- **Personal SIP Accounts** (Feb 07)
- **Migrate VMail inc 5E OPTIN** (Feb 07)
- **Pilot Server Migration** (Feb 07)
- **Full VoIP Deployment** (Feb 07)

**FY 3Q08**
- 1500 phones

**FY 4Q08**
- 1500 phones

**FY 3Q10**
- 1500 phones

**FY 4Q10**
- 1500 phones

**Jan-07 Jul-07 Oct-07 Jan-08 Apr-08 July-08 Jan-10 Jul-10**
Strategic Issues

- Limited support for VoIP over WiFi
  - Not part of the generally available enterprise model, supported by exception.
  - Currently, no guarantee for quality of service or ability to centrally configure and manage dual-mode handsets.
  - Expected to change over the life of the service.

- Service delivery model for VoIP provided over non-IS&T campus networks. Issues include:
  - Service level agreements, phone configuration and initialization, emergency services/endpoint location, and quality of service.

- Evolving ability to support emergency services (“100/911”)
  - Using network data to self-report ip phone location.
  - Need for additional analog phones (key offices and hallway placement) in the event of power and/or network outages.

- Whether to provide a student VoIP service offering
  - Parity with existing telephony service offering
Support and Provisioning Model

Pilot Model

- voip-pilots-support@mit.edu for support and provisioning

VoIP Deployment Model

- Roll-out:
  - Consulting and collaborating with designated DLC contacts
  - Site surveys of existing telephony services
  - Scheduling migration
  - Training demos and documentation
  - Deployment support from intensive hands-on to supporting DLC IT staff
  - Post-deployment support availability as necessary

- On-going support:
  - Integrated with existing telephony and computing help desk
  - Customer Service Representatives (CSR) as point of contact
  - Self-Service options
  - Clear escalation paths
  - Stock answer database (FAQs)
  - Support documentation
Communication and Training Strategy

- Develop a coordinated plan
- Target audience and key stakeholders
- Engage the community and develop a feedback loop
- Leverage learning
  - Project showcases
  - VoIP demos
  - Online demos and documentation
- VoIP Advisory Board
VoIP Core Team

- Dennis Baron
- Elliot Eichen
- Lu Keohane
- Angie Milonas
- Kathy Pagones O’Neill
- Jeff Schiller
- Mark Silis
- Jana Tarasenko
- Oliver Thomas
- Irina Vainstock
Enhanced Shared Call Appearance (a.k.a. Bridged Call Appearance)

SIP Feature Server (Sylantro)

Call Placed

Jerry’s Phone

Joanne’s Phone. Has Joanne’s Telephone Number but also shares (subscribes to) Jerry’s calls.
Enhanced Shared Call Appearance (a.k.a. Bridged Call Appearance)

Joanne answers call
Jerry’s phone blinks red

SIP Feature Server (Sylantro)

Joanne’s Phone. Has Joanne’s Telephone Number but also shares (subscribes to) Jerry’s calls.
Enhanced Shared Call Appearance (a.k.a. Bridged Call Appearance)

Jerry answers call, media is bridged

SIP Feature Server (Sylantro)

Jerry’s Phone

Joanne’s Phone. Has Joanne’s Telephone Number but also shares (subscribes to) Jerry’s calls.
Enhanced Shared Call Appearance (a.k.a. Bridged Call Appearance)

Joanne hangs up, call indicator flashes

SIP Feature Server (Sylantro)

Jerry’s Phone

Joanne’s Phone. Has Joanne’s Telephone Number but also shares (subscribes to) Jerry’s calls.