SIP: Standardization, Interoperability, New Horizons

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VON Fall 1999 (Atlanta)
September 28, 1999

Joint work with Jonathan Lennox, Jonathan Rosenberg, Elin Wedlund
Overview

- overview/review
- standardization status
- interoperability bake-offs
- SIP futures: event notification, mobility
Architecture

- SIP3
- Architecture
- Megaco/MGCP/MDCP
- SIP
- RTP
- Circuit-switched voice (POTS, ISDN)
- Internet
- PSTN
- Gateway
- Proxy
- H.323
- Megaco/MGCP/MDCP

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SIP 101

1. SIP = signaling protocol for establishing sessions/calls/conferences/…

2. session = audio, video, game, chat, …

3. called server may map name to user@host

4. callee accepts, rejects, forward (→ new address)

5. if new address, go to step 2

6. if accept, caller confirms

7. …conversation …

8. caller or callee sends BYE
SIP Standardization Status

- Feb. 2, 1999: IETF Proposed Standard
- March 17, 1999: IETF RFC 2543
- working towards Draft Standard (mainly clarifications)
- new SIP working group (move from mmusic)
SIP Bake-Offs

1st April 1999 Columbia University, New York
2nd August 1999 Pulver, Long Island
3rd December 1999 Ericsson, Dallas

- roughly 12-15 groups
- tested
  - hardware
  - PSTN gateways
  - proxy/redirect servers
  - clients
  - test instrument, ...
- interoperability and “torture test”
### Participants at SIP Bake-Offs

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<th>Company</th>
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<tr>
<td>3Com</td>
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<td>University of Tampere, Waterloo</td>
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SIP Bake-Off Results

- almost all implementations could establish basic calls – either on arrival or after minor on-site fixes
- tested redirection, proxying, security, registration, …
- generated interoperability test cases and tools
- will fold clarifications into Draft revision of RFC and web page at http://www.cs.columbia.edu/sip
- public test servers:
  - sip:sip.pcs.ellemtel.net
  - sip:siphappens.com (3Com)
  - sip:sip.pulver.com (Columbia sipd)
SIP Work Items

- PINT (control of PSTN)
- sip-cgi
- call processing language
- SIP servlet APIs
- reliable provisional (1xx) responses
- caller preferences
- third-party call control
- SIP for subscribe/notify
- SIP–ISUP interworking (BCP)
- SIP–H.323 interworking
- billing
- reverse channel setup for call progress tones
- pre-ringing resource reservation
- SIP for mobility

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Integrating Signaling and Instant Messaging: Some Ideas

- “reverse” signaling: callee indicates availability
- buddy lists = special case of *event notification*
- other events: “sensor 17 smells smoke”, “Beanie Babies are on sale”, “(voice) mail has arrived”, …
- subscribe – notify – set up call
- useful for call parking
- many SIP mechanisms apply: security, redirection, proxying, content negotiation, …
SIP for Event Notification

- add two methods: SUBSCRIBE and NOTIFY
- proxy server may intercept SUBSCRIBE
- use message body for event description
- default: presence, indicated by REGISTER
- one of many proposals for presence (IETF WG!)
SIP for Event Notification

Alice (subscriber) → proxy

Bob (subscriber) → proxy

proxy → Carol (publisher)

SUBSCRIBE

NOTIFY

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Mobility

- move to new network ➔ IP address changes (DHCP)
- mobile IP hides address changes
- but: little deployment
- encapsulation overhead
- dog-legged routing
- may not work with IP address filtering
SIP mobility overview

- pre-call mobility ➔ SIP proxy, redirect
- mid-call mobility ➔ SIP re-INVITE, RTP
- recovery from disconnection
SIP mobility: pre-call

- MH acquires IP address via DHCP
- optional: MH finds SIP server via multicast REGISTER
- MH updates home SIP server
- optimization: hierarchical LR (later)
SIP mobility: mid-call

- MH→CH: new INVITE, with Contact and updated SDP
- re-registers with home registrar
SIP mobility: multi-stage registration

Don’t want to bother home registrar with each move

Contact: alice@CA
From: alice@NY
Contact: 193.1.1.1

REGISTER
INVITE

Los Angeles
San Francisco

From: alice@NY
Contact: alice@CA
From: alice@NY
Contact: 192.1.2.3
Conclusion

- SIP basic standard stable
- multiple interoperating implementations
- backward-compatible features:
  - interoperation with legacy signaling systems
  - mobility
  - caller preferences
  - call transfer
  - ...
- programming of services:/cgi, CPL, applets
For more information...

**SIP:**  http://www.cs.columbia.edu/sip

**RTP:**  http://www.cs.columbia.edu/~hgs/rtp

**Papers:**  http://www.cs.columbia.edu/IRT