SIP: Status and Directions

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Nextone

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Overview

- SIP overview/review
- SIP services
- SIP standardization status
- SIP bake-off
- SIP for notification
- SIP for mobility
Architecture

- SIP
- Megaco/MGCP/MDCP
- RTP
- circuit-switched voice (POTS, ISDN)
- Internet
- PSTN
- proxy
- GK
- gateway
- MGC

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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 Operation in Proxy Mode

1. INVITE henning@columbia.edu
2. 200 OK
3. INVITE hgs@play
4. 200 OK
5. PLAY
6. ACK hgs@play
7. 200 OK
8. ACK henning@columbia.edu
9. media stream

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SIP Operation in Redirect Mode

1. INVITE henning@ieee.org
   302 Moved temporarily
   Contact: hgs@columbia.edu
2. hemming
   columbia.edu
3. columbia.edu
4. INVITE henning@ieee.org
5. ACK henning@ieee.org
6. INVITE hgs@columbia.edu
7. 200 OK
8. ACK hgs@columbia.edu
SIP Advanced Features

- operation over UDP or TCP
- multicast invitations ➤ basic ACD
- “interactive web response” (IWR)
- UA ↔ proxy = proxy/redirect ↔ proxy/redirect
- stateless proxies: self-routing responses
- forking proxies: call several in sequence and/or parallel
- security: basic (password), digest (challenge/response), PGP
More SIP Internet Telephony Services

- camp-on without holding a line
- short message service ("instant messaging")
- schedule call into the future
- call with expiration date
- add/remove parties to/from call ➔ mesh
- “buddy lists”
Internet Telephony – as Part of Internet

- email address = SIP address
- SIP URLs in web pages
- forward to email, web page, chat session, …
- include web page in invitation response (“web IVR”)
- RTSP: choose your own music-on-hold
- include vCard, photo URL in invitation
SIP Extensibility

- headers that receiver may ignore, e.g., Photo
- new methods and inquire about those supported (OPTIONS)
- features that receivers needs to understand: Required → Unsupported
  - e.g., Required: com.nextone.feature
- proposed: features supported via Supported header
SIP Standardization Status

- Feb. 2, 1999: IETF Proposed Standard
- March 17, 1999: IETF RFC 2543
- eligible for Draft Standard: 6 months, 2 implementations √
- new SIP working group (move from mmusic)
- working on updated draft based on implementation experience
- mostly clarifications + optional headers, no new version
SIP Work Items

- sip-cgi
- call processing language (CPL)
- reliable provisional (1xx) responses
- caller preferences
- third-party call control
- SIP for subscribe/notify
- SIP–ISUP interworking
- SIP–H.323 interworking
- billing
- reverse channel setup for call progress tones
- pre-ringing resource reservation
SIP Bake-Off

- 3 bake-offs: April, August, December
- from 15 to 33 groups
- hardware, PSTN gateways, proxy/redirect servers, clients, test instrument, ...
SIP Bake-Off Participants

3Com
dynamicsoft
Mitel
8x8
Ellemtel
Netspeak
Agilent
Ericsson
Nortel
Alcatel
Facet
Nuera
Broadsoft
Helsinki Univ.
OZ.com
British Telecom
Hewlett-Packard
Pingtel
Catapult
Indigo
Radcom
Cisco
IPcell
Telygo
Columbia University
Lucent
Vovida
Dialogic
MCI Worldcom
VTEL
Mediatrix
SIP Bake-Off Goals

- basic call set-up
- registration, user location
- proxies and redirect server operation
- advanced features: security
- identify implementation bugs and robustness issues
- identify spec ambiguities
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/~hgs/sip
- install public testing mechanisms (Pulver OpenTestNet, www.siphappens.com)
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

subscribe

Alice

Bob

proxy

CAROL

publisher

1999-12-13
Mobility

- new network $\rightarrow$ new IP address (DHCP)
- mobile IP hides addr. changes
- but: little deployment
- $-\rightarrow$: encapsulation overhead
- $-\rightarrow$: dog-legged routing
- $-\rightarrow$: 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

LaTeX code:

\begin{center}
\begin{tikzpicture}
  \node (ca) at (0,0) {CA};
  \node (ny) at (4,0) {NY};
  \node (losangeles) at (-2,-2) {Los Angeles};
  \node (sanfrancisco) at (2,-2) {San Francisco};

  \draw[blue,->] (ca) -- node[fill=white] {From: alice@NY\newline Contact: 193.1.1.1} (sanfrancisco);
  \draw[blue,->] (sanfrancisco) -- node[fill=white] {From: alice@NY\newline Contact: alice@CA} (ca);
  \draw[blue,->] (ca) -- node[fill=white] {From: alice@NY\newline Contact: 192.1.2.3} (losangeles);
  \draw[blue,->] (losangeles) -- (ca);

  \draw[green,->] (ca) -- (ny);
  \draw[green,<-] (ny) -- (ca);

  \draw[blue,->, dashed] (ca) -- (losangeles);
  \draw[blue,->, dashed] (losangeles) -- (ca);

  \draw[blue,->, dashed] (ca) -- (sanfrancisco);
  \draw[blue,->, dashed] (sanfrancisco) -- (ca);

  \node at (-4,-4) {Los Angeles};
  \node at (0,-4) {San Francisco};

  \node at (0,-4.5) {REGISTER};
  \node at (0,-4.8) {INVITE};
\end{tikzpicture}
\end{center}
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