SIP: Status and Directions

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Sylantro

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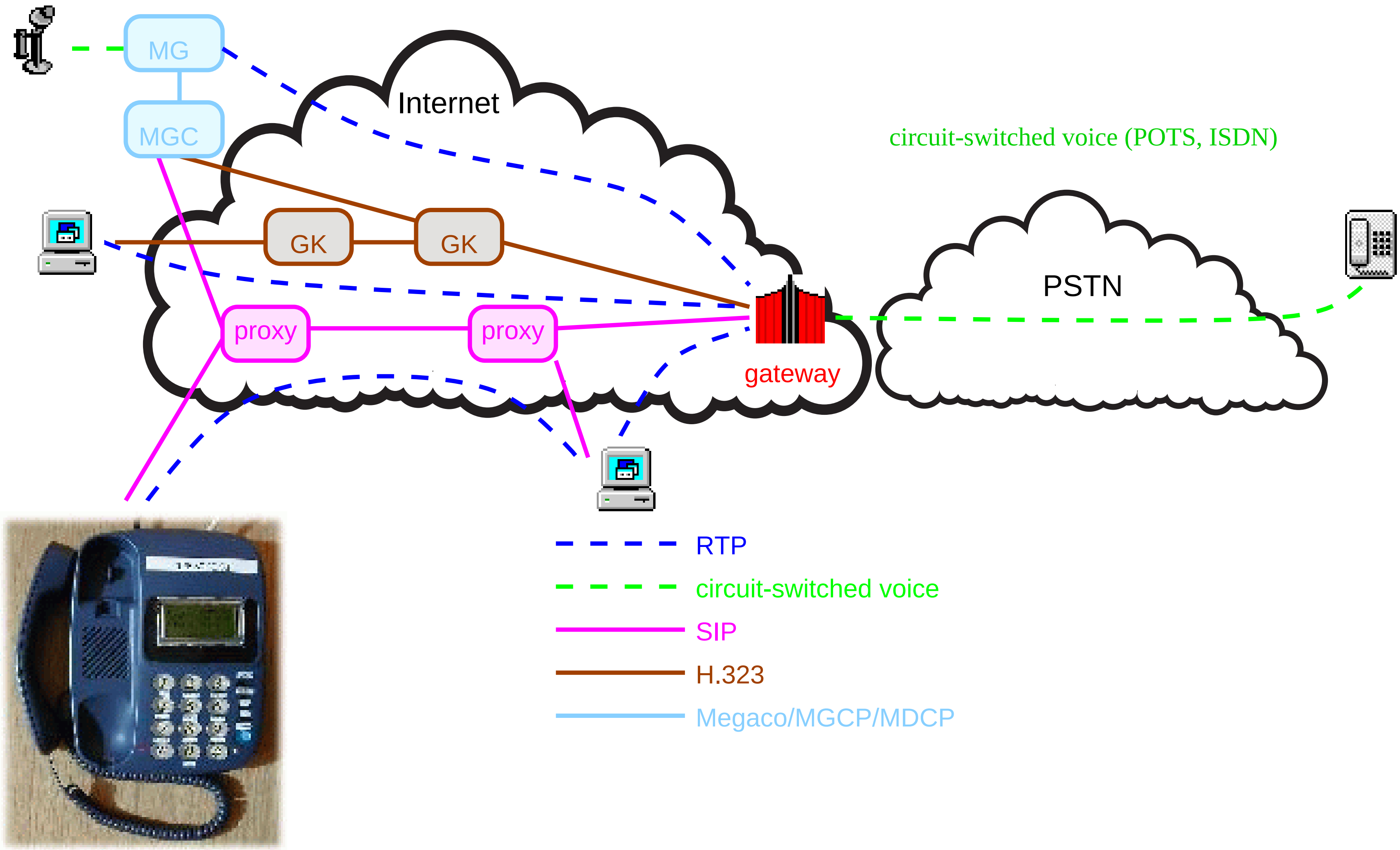
Overview

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

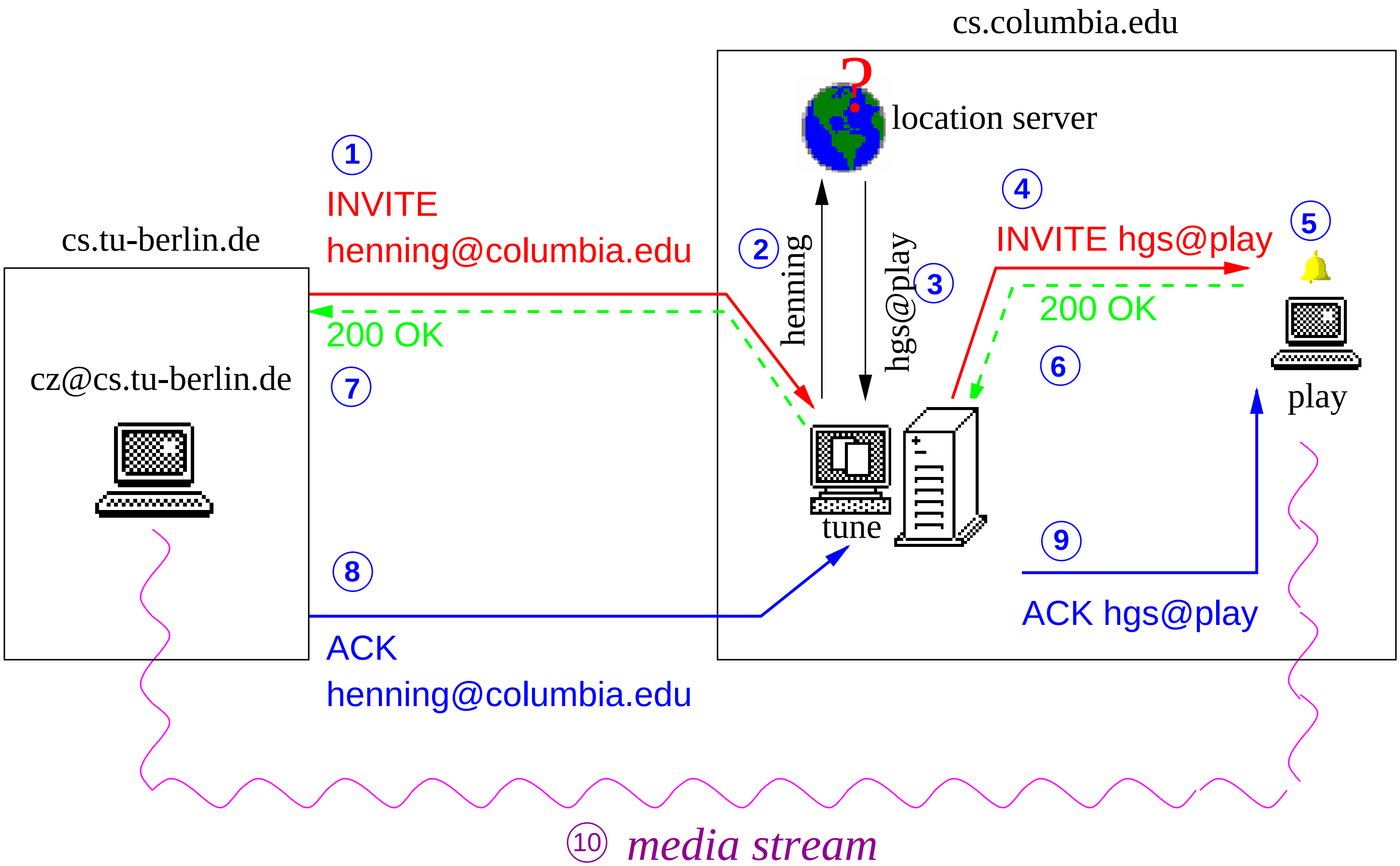
Architecture



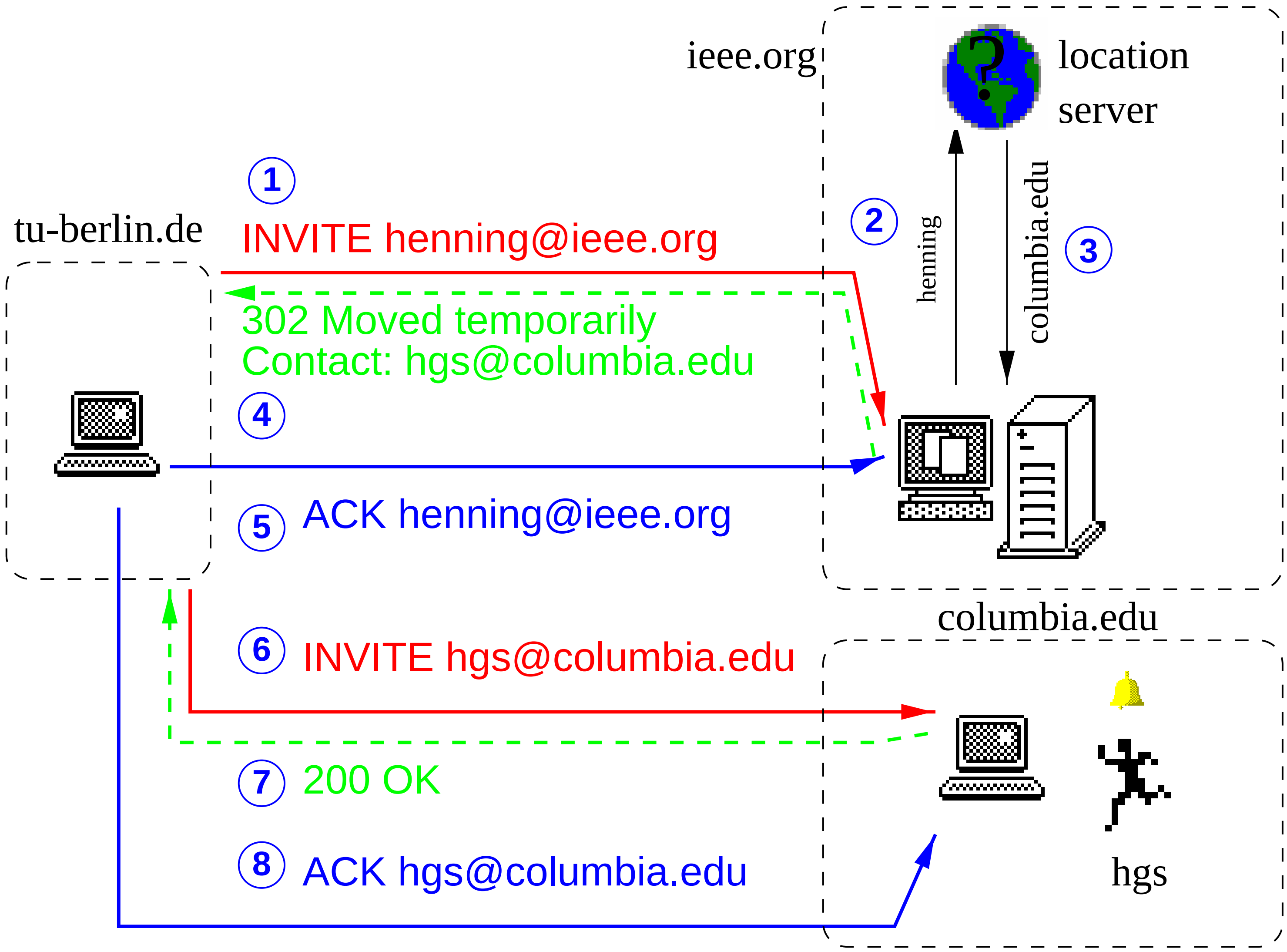
SIP 101

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 ($\rightarrow$ 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



SIP Operation in Redirect Mode



SIP Advanced Features

- operation over UDP or TCP
- multicast invitations ➠ basic ACD
- “interactive web response” (IWR)
- UA $\leftrightarrow$ proxy = proxy/redirect $\leftrightarrow$ 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 $\longrightarrow$ Unsupported
- e.g., `Required: com.sylantro.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 $\surd$
- 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
8x8
Agilent
Alcatel
Broadsoft
British Telecom
Catapult
Cisco
Columbia University
Dialogic
dynamicsoft
Ellemtel
Ericsson
Facet
Helsinki Univ.
Hewlett-Packard
Indigo
IPcell
Lucent
MCI Worldcom
Mediatrix
Mitel
Netspeak
Nortel
Nuera
OZ.com
Pingtel
Radcom
Telogy
Vovida
VTEL

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)

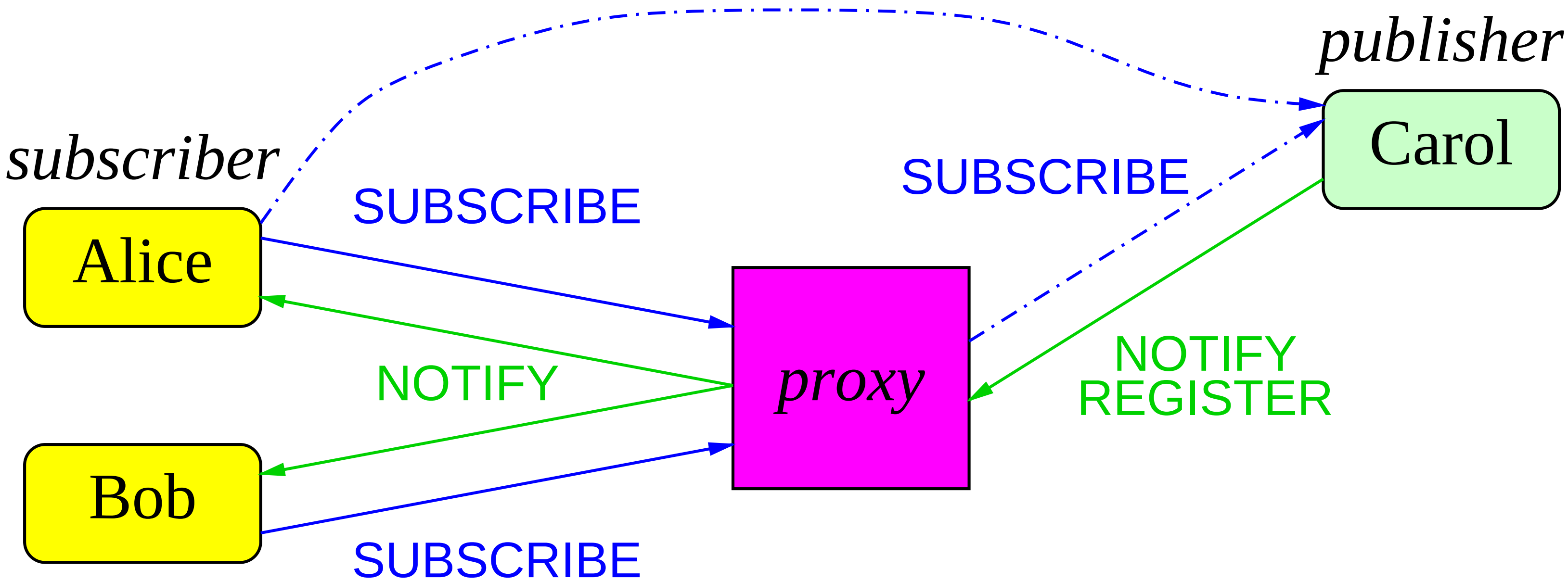
Integrating Signaling and Instant Messaging: Some Ideas

- “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

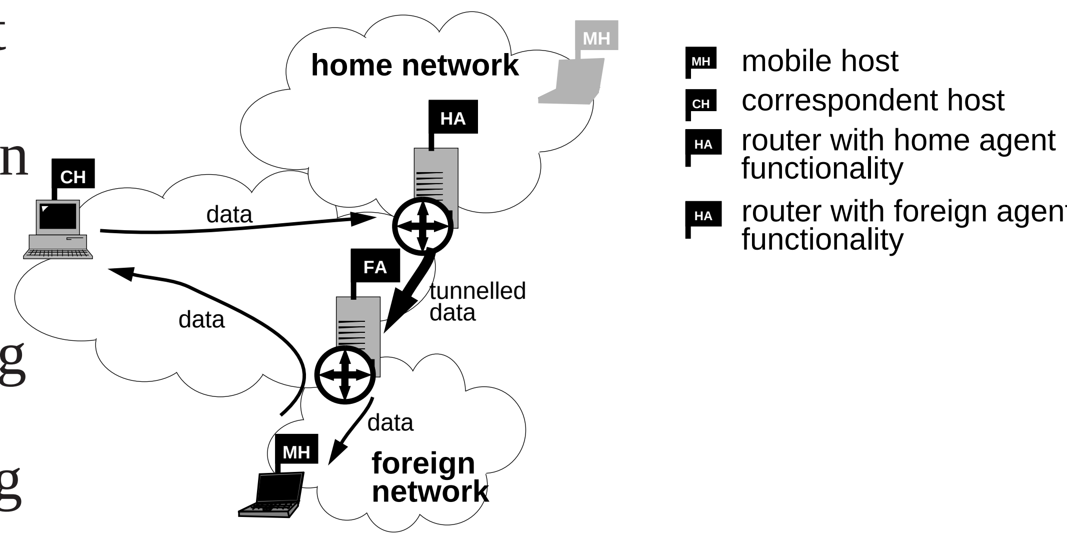
- add two methods: SUBSCRIBE and NOTIFY
- proxy server may intercept SUBSCRIBE
- use message body for event description
- default: presence, indicated by REGISTER
- one of *many* proposals for presence (IETF WG!)

SIP for Event Notification



Mobility

- new network ➠ new IP address (DHCP)
- mobile IP hides addr. changes
- but: little deployment
- –: encapsulation overhead
- –: dog-legged routing
- –: IP address filtering

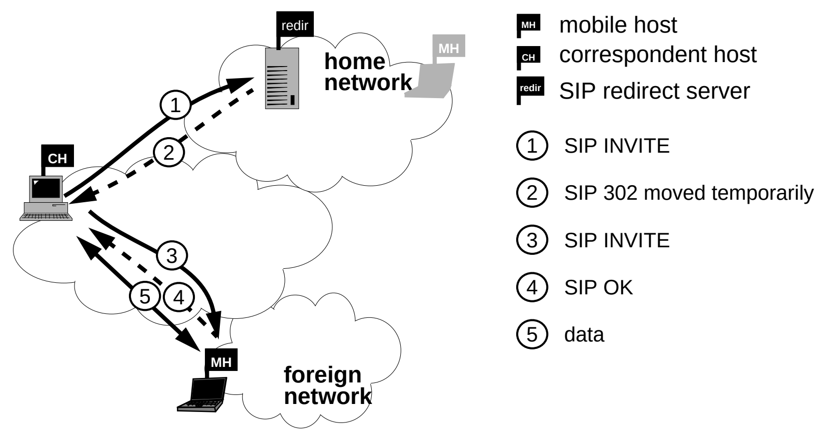


SIP Mobility Overview

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

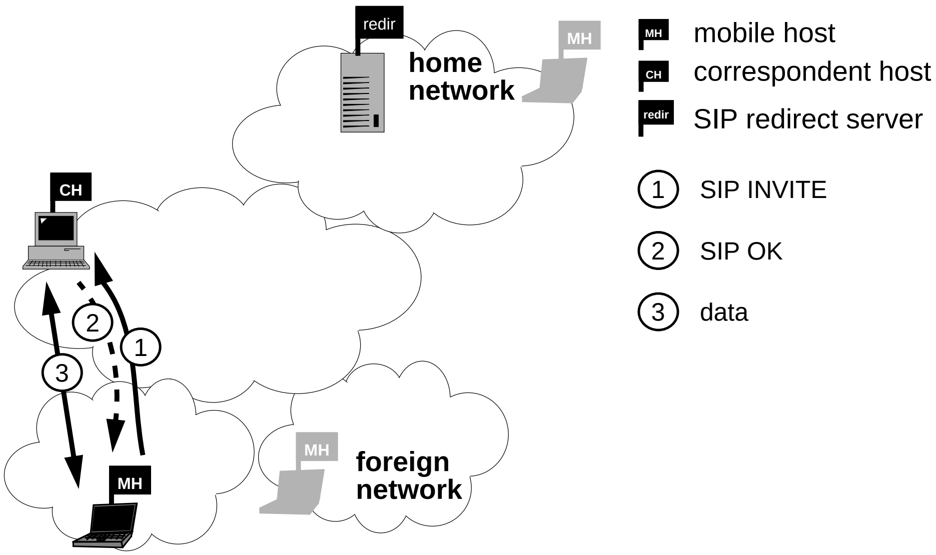
SIP Mobility: Pre-call

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



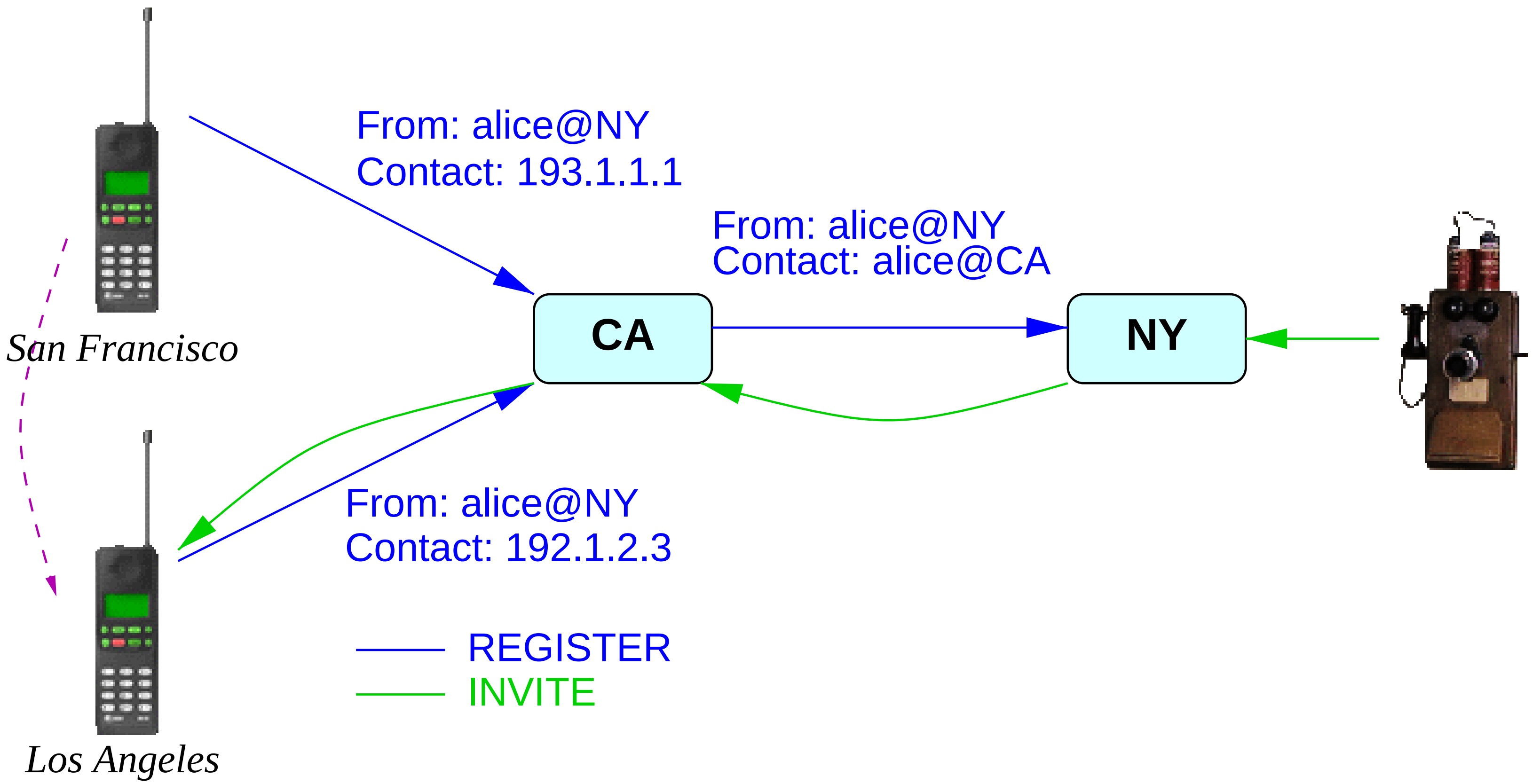
SIP mobility: mid-call

- 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



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