SIP: Standardization, Interoperability, New Horizons

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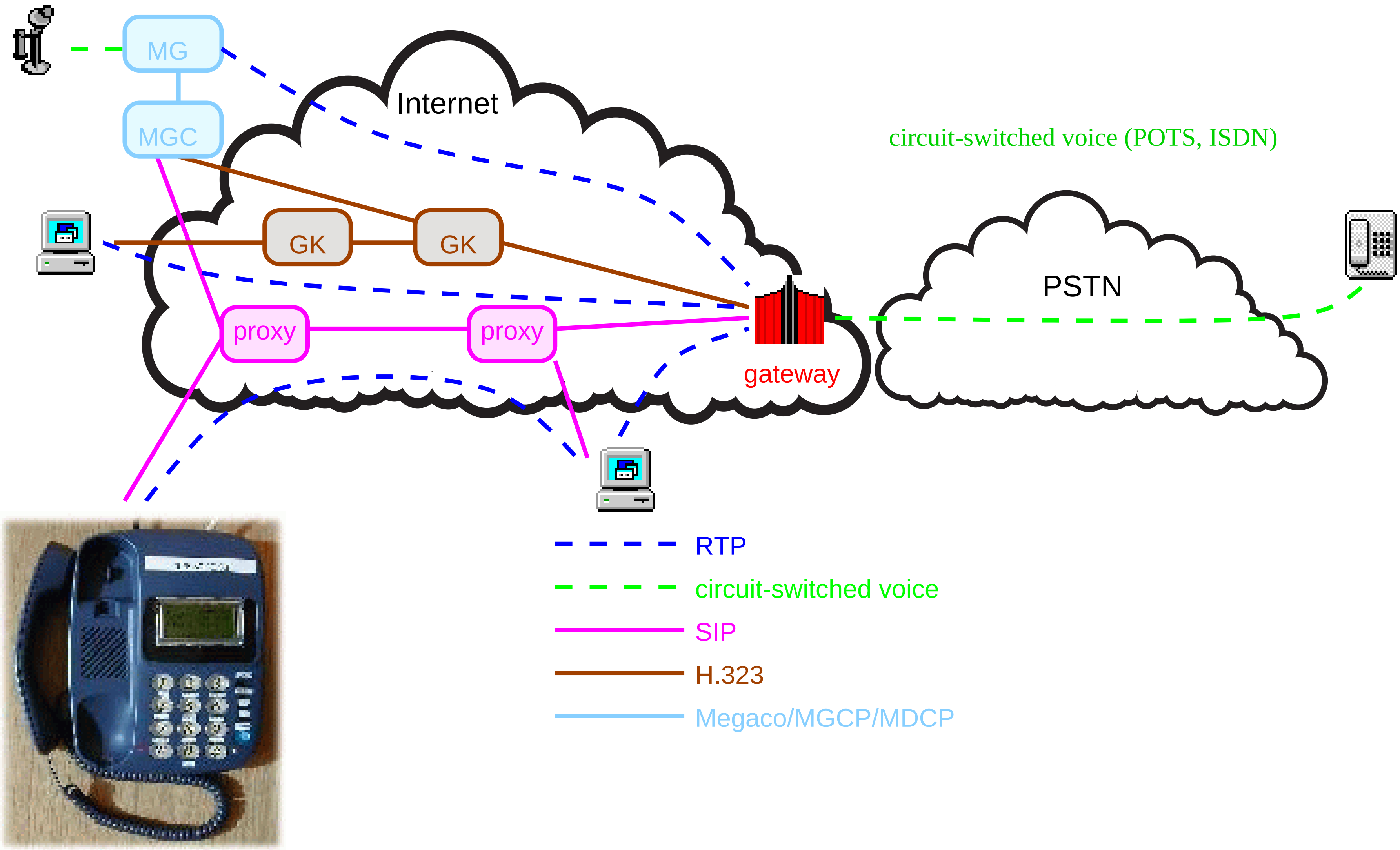
Overview

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

3Com	Ericsson (2)
8x8	Helsinki Univ. of Technology
Alcatel	Hewlett-Packard (2)
Broadsoft	Lucent
British Telecom	MCI Worldcom
Cisco	Mitel
Columbia University	Mediatrix
Dialogic	Nortel
dynamicsoft	Pingtel
Ellemtel	University of Tampere, Waterloo

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

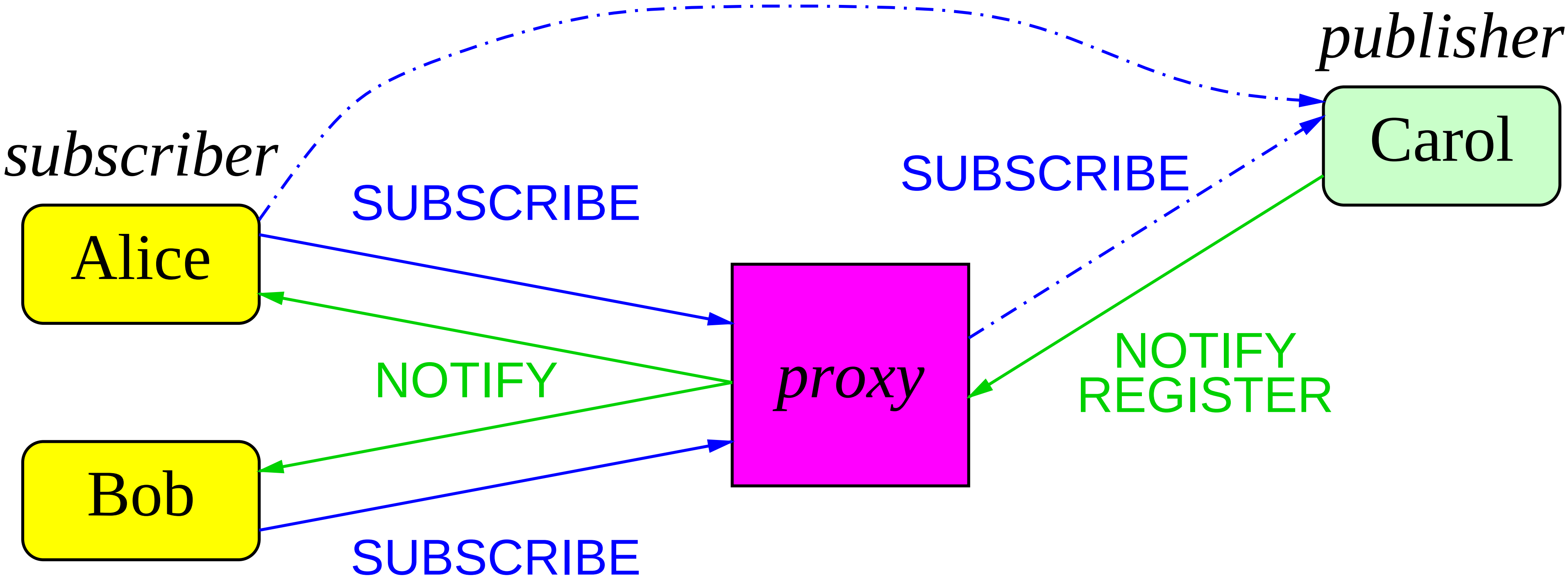
Integrating Signaling and Instant Messaging: Some Ideas

- “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

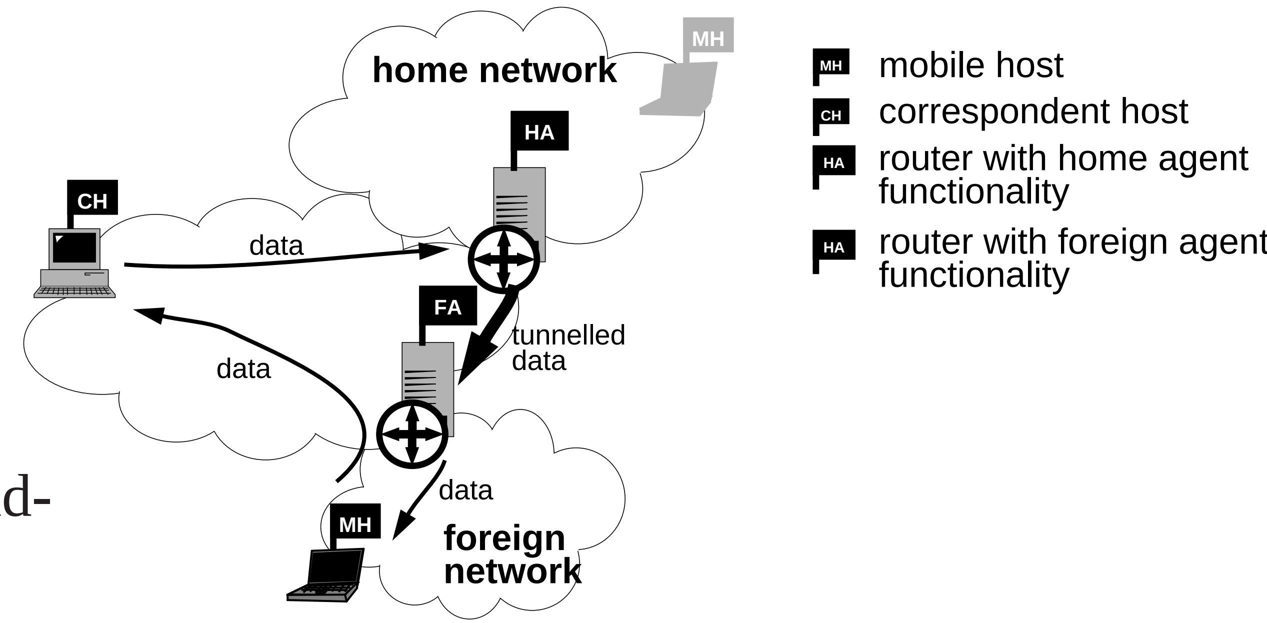
- add two methods: SUBSCRIBE and NOTIFY
- proxy server may intercept SUBSCRIBE
- use message body for event description
- default: presence, indicated by REGISTER
- one of *many* proposals for presence (IETF WG!)

SIP for Event Notification



Mobility

- 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

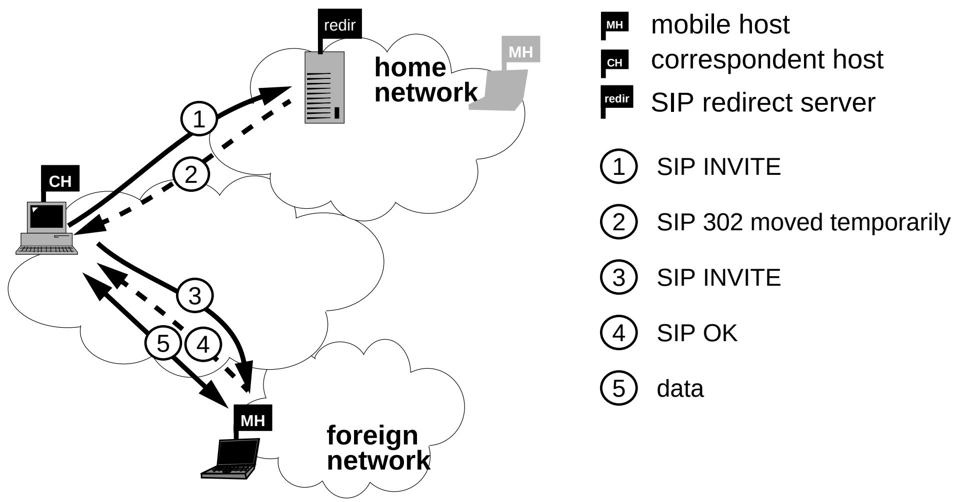


SIP mobility overview

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

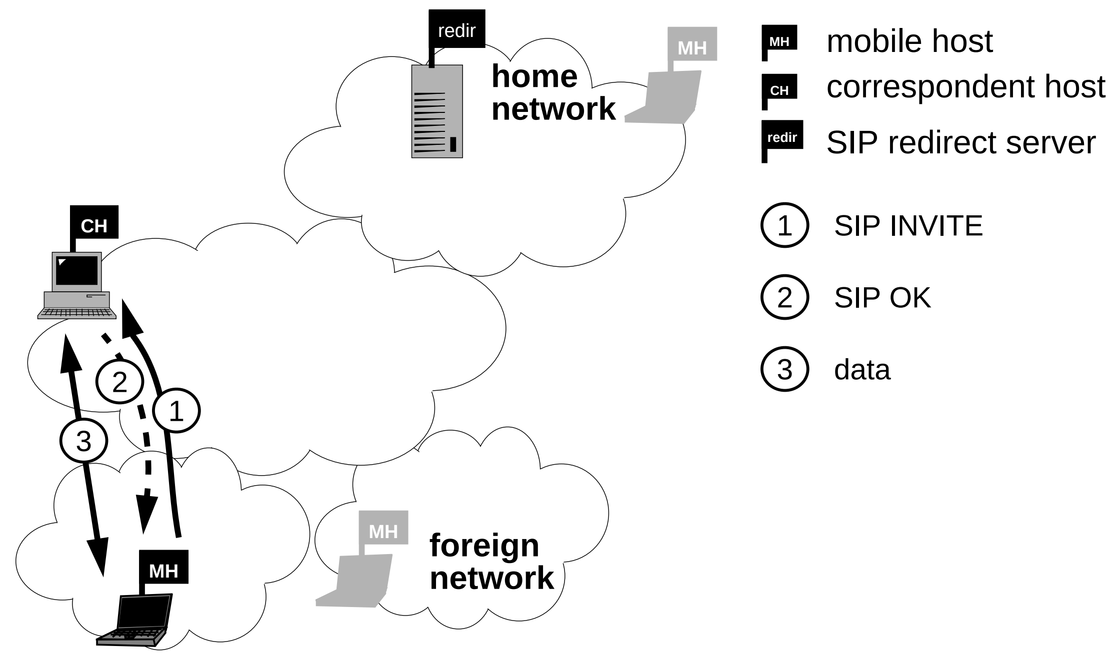
SIP mobility: pre-call

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



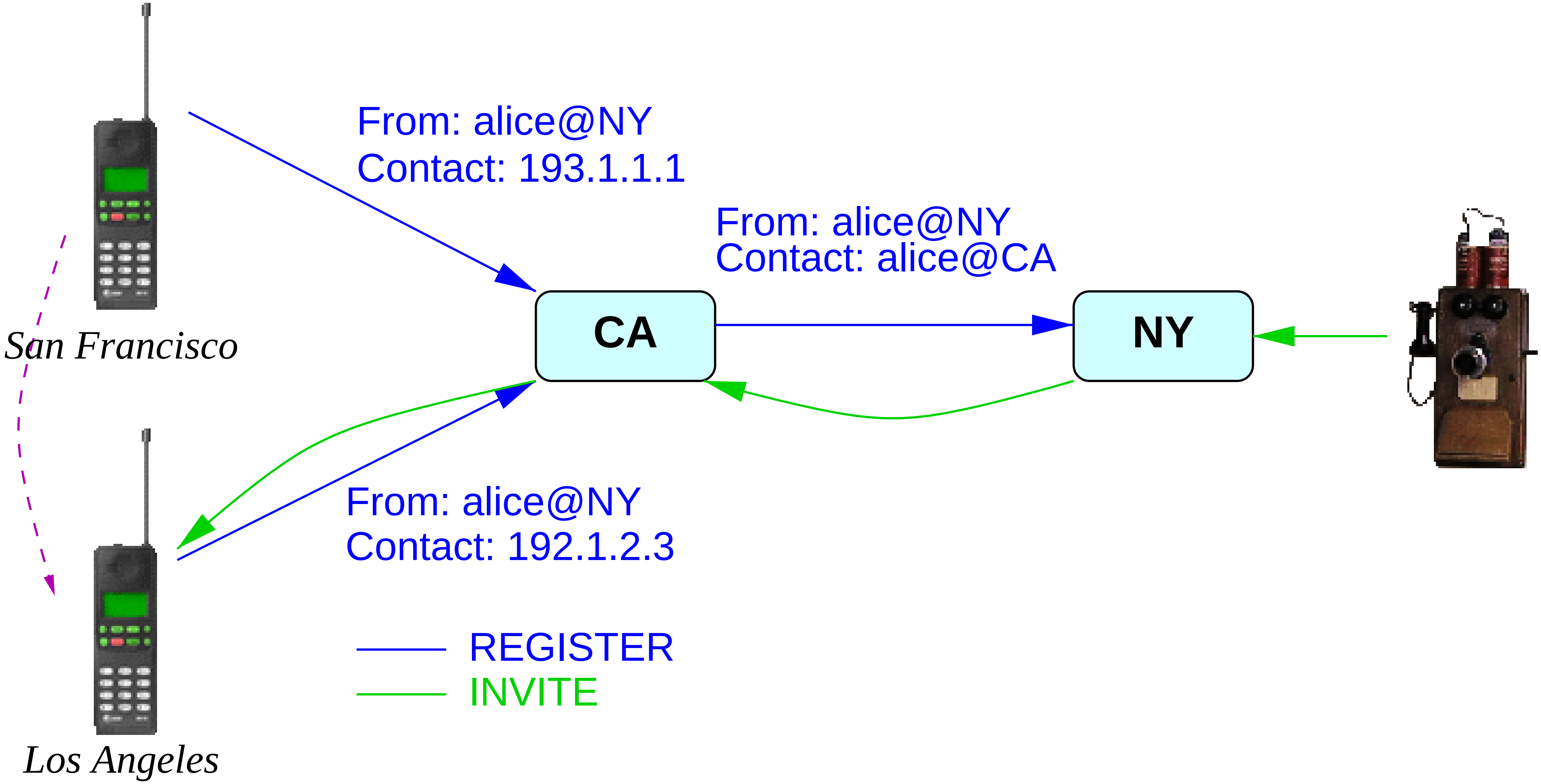
SIP mobility: mid-call

- 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



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