SIP: Standardization, Interoperability, New Horizons

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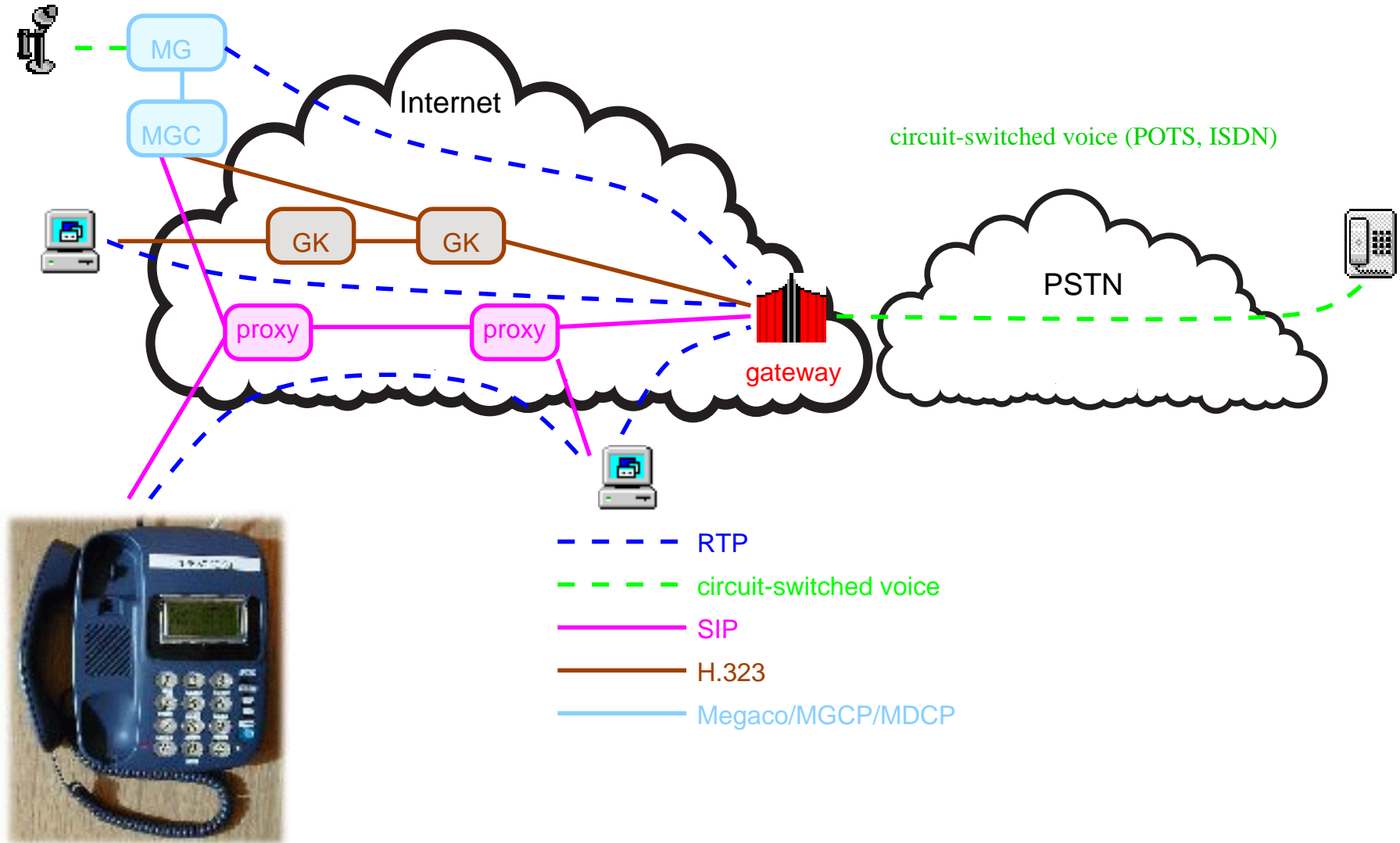schulzrinne@cs.columbia.edu

VON Fall 1999 (Atlanta)

September 28, 1999

Overview

- overview/review

- standardization status

- interoperability bake-offs

- SIP futures: event notification, mobility

Architecture

Internet

circuit-switched voice (POTS, ISDN)

MG

MGC

GK   GK

proxy   proxy

gateway

PSTN

- - - - RTP
- - - - circuit-switched voice
——— SIP
——— 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 ($\rightarrow$ 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

- <mark>SIP for subscribe/notify</mark>

- SIP–ISUP interworking (BCP)

- SIP–H.323 interworking

- billing

- reverse channel setup for call progress tones

- pre-ringing resource reservation

- <mark>SIP for mobility</mark>

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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
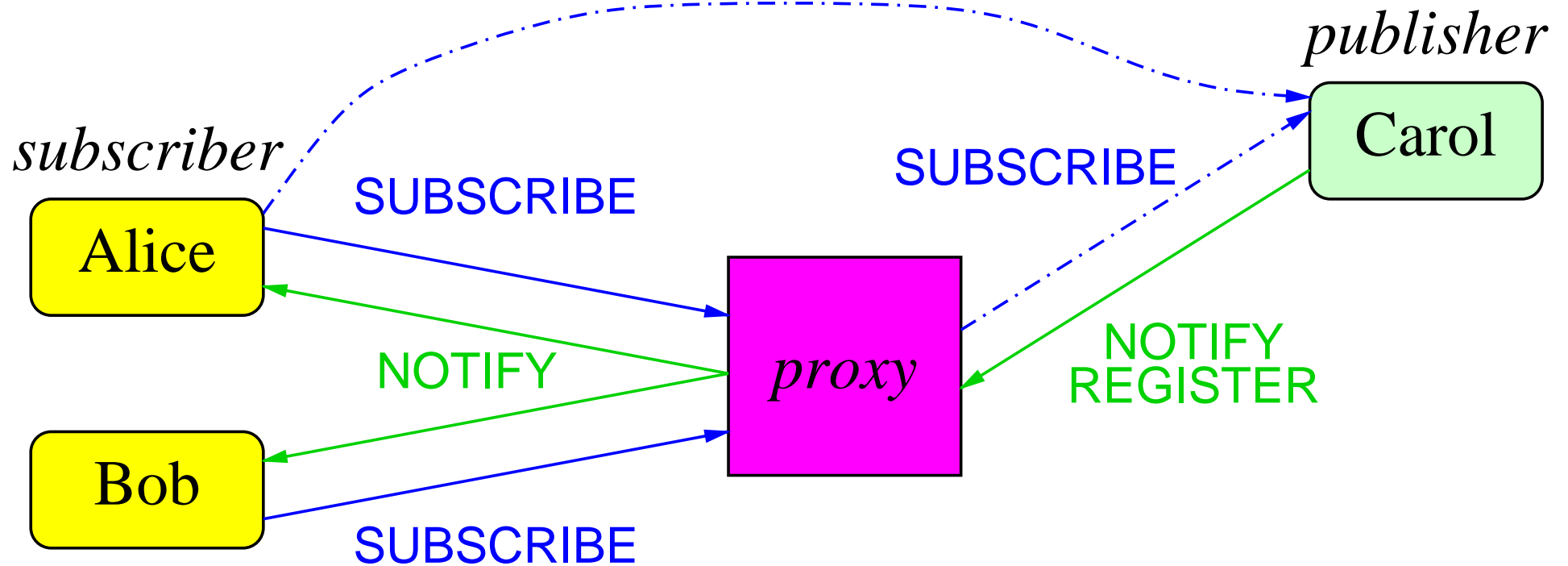
Mobility

- move to new network ⇒ IP address changes (DHCP)

- mobile IP hides address changes

- but: little deployment

- encapsulation overhead

- dog-legged routing
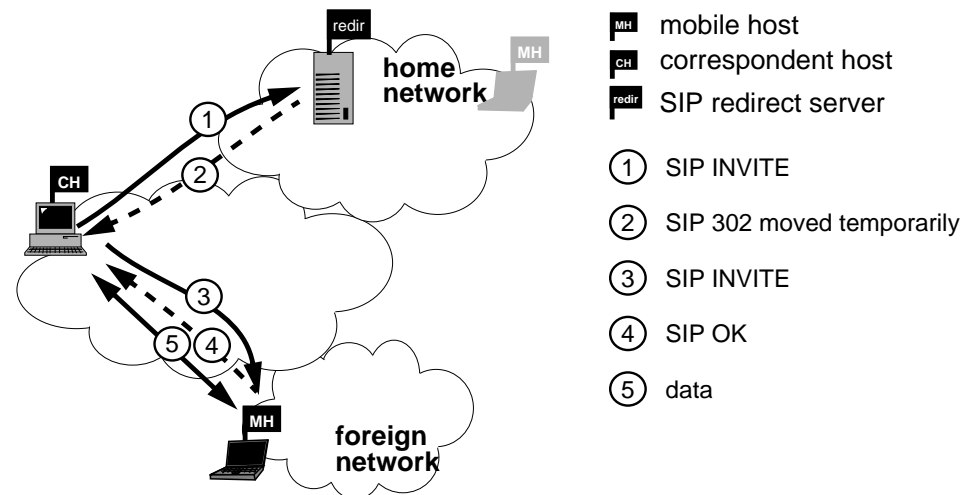
- may not work with IP address filtering

home network

MH

HA

CH

data

FA

tunnelled data

data

MH

foreign network
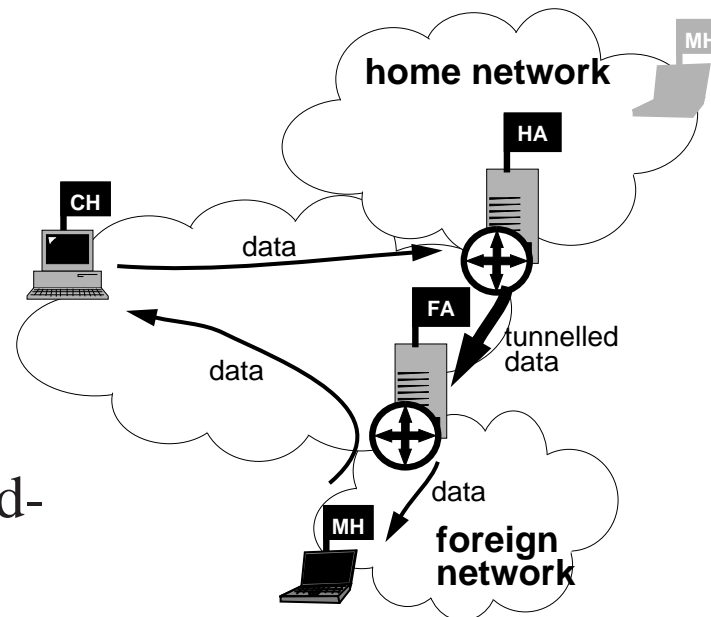
MH mobile host

CH correspondent host

HA router with home agent functionality

HA router with foreign agent functionality

SIP mobility overview

- pre-call mobility ➠ SIP proxy, redirect

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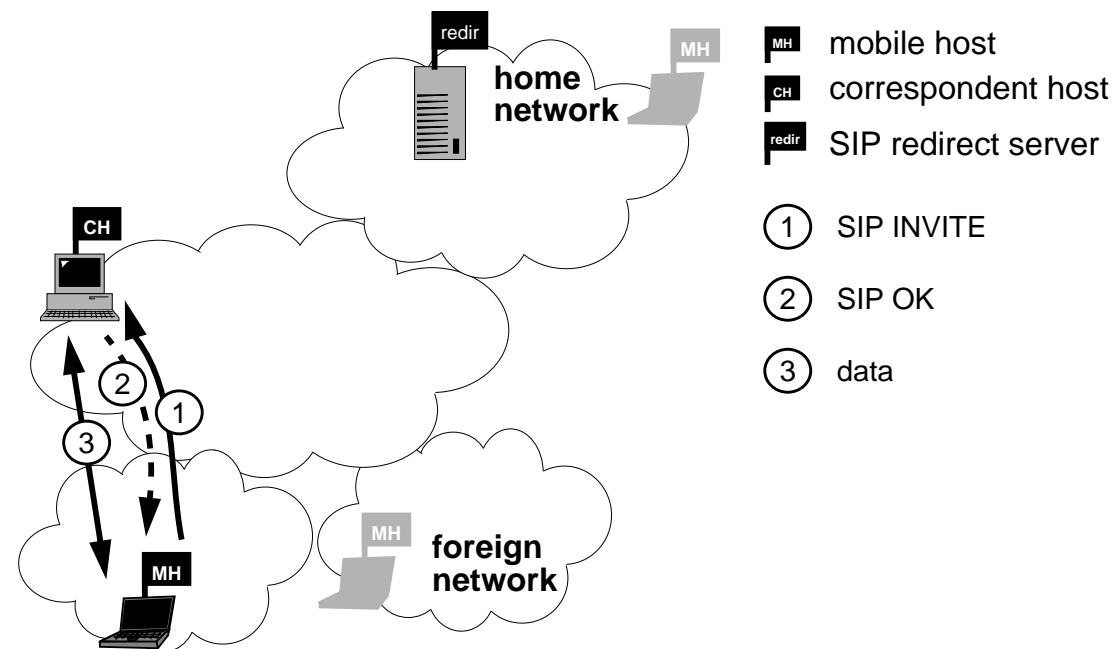
- recovery from disconnection

SIP mobility: pre-call

- MH acquires IP address via DHCP

- optional: MH finds SIP server via multicast REG-ISTER

- MH updates home SIP server
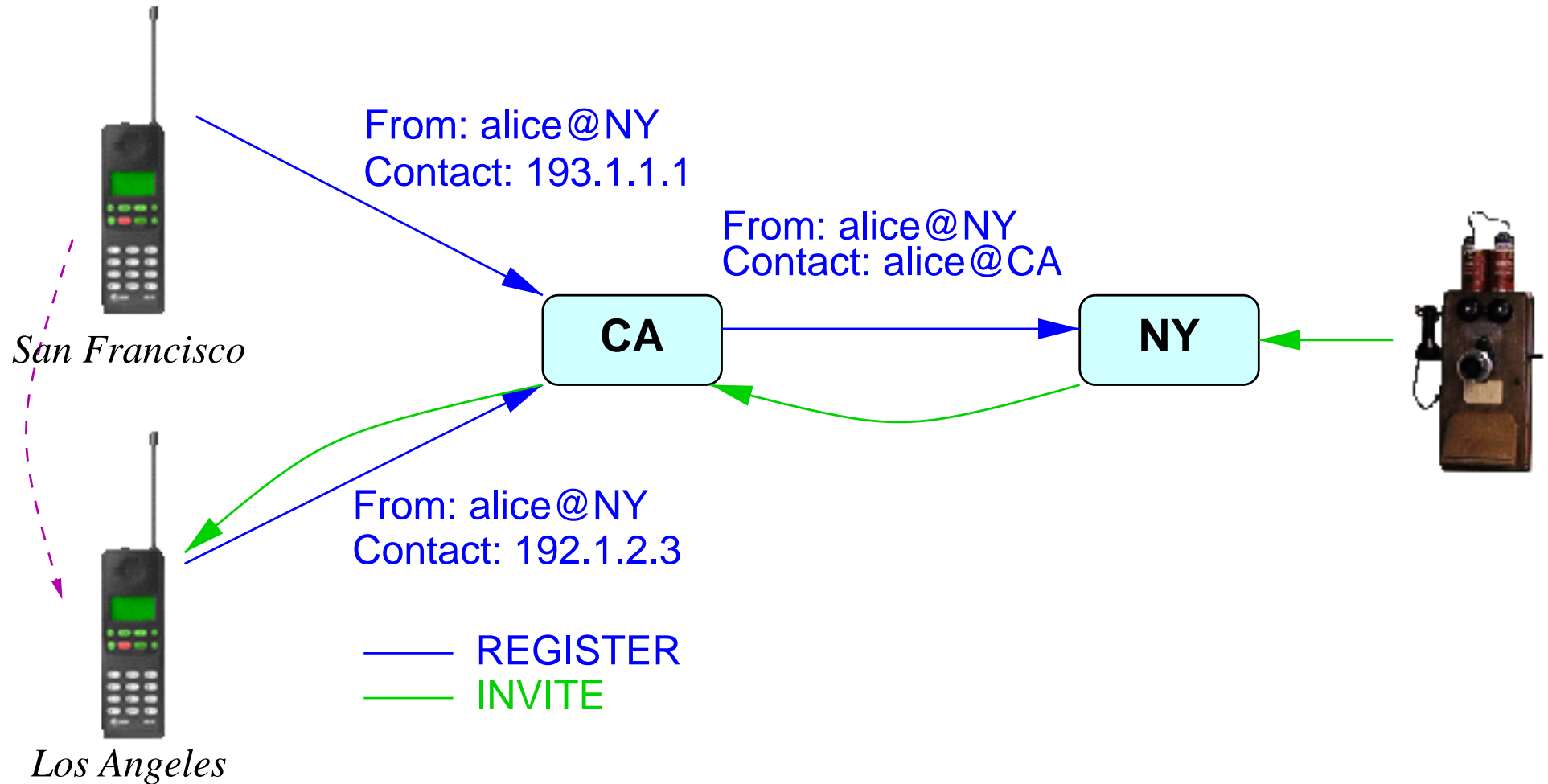
- optimization: hierarchical LR (later)

MH   mobile host
CH   correspondent host
redir SIP redirect server

① SIP INVITE
② SIP 302 moved temporarily
③ SIP INVITE
④ SIP OK
⑤ data

home network
foreign network

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

From: alice@NY
Contact: 193.1.1.1

From: alice@NY
Contact: alice@CA

San Francisco

CA   **NY**

From: alice@NY
Contact: 192.1.2.3

―――― REGISTER
―――― INVITE

Los Angeles

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