

Mobility Support Using SIP

Elin Wedlund and *Henning Schulzrinne*

Dept. of Computer Science

Columbia University

New York, New York

schulzrinne@cs.columbia.edu

WoWMoM, Seattle

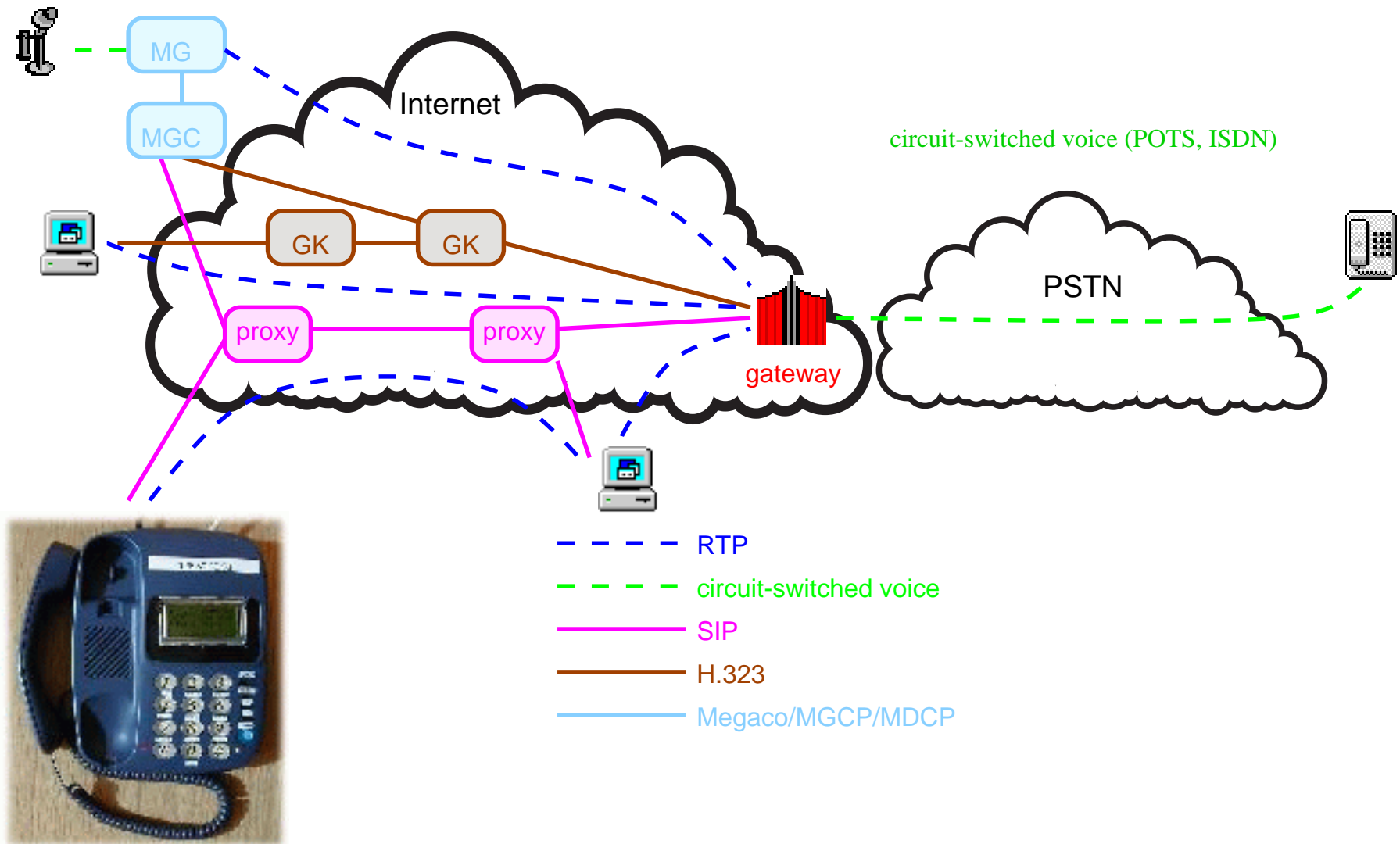
August 20, 1999

Overview

pure-IP mobility \leftrightarrow IP over GSM, 3G, ...

- SIP
- mobile applications
- mobile IP issues for Internet telephony
- mobility support using SIP
- performance
- future work

Internet Telephony Architecture



SIP (Session Initiation Protocol)

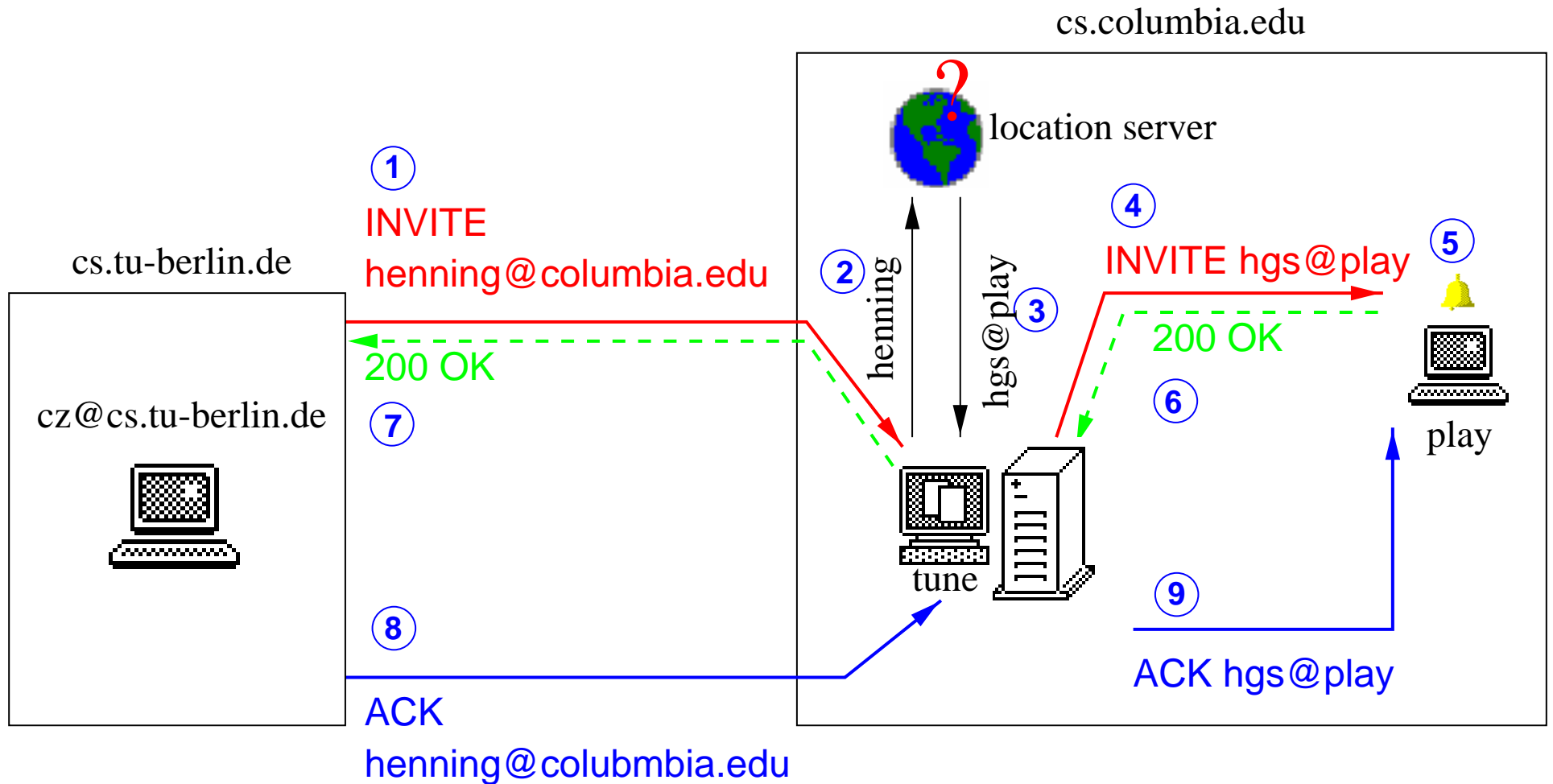
- SIP = “out-of-band” *signaling* protocol for establishing sessions/calls/conferences/...
- multimedia data typically uses RTP
- may travel completely different path than data
- session = audio, video, shared application, game, chat, ...
- session description: SDP, ...
- “personal mobility” = single address for multiple end systems ||, \rightsquigarrow

SIP Operation

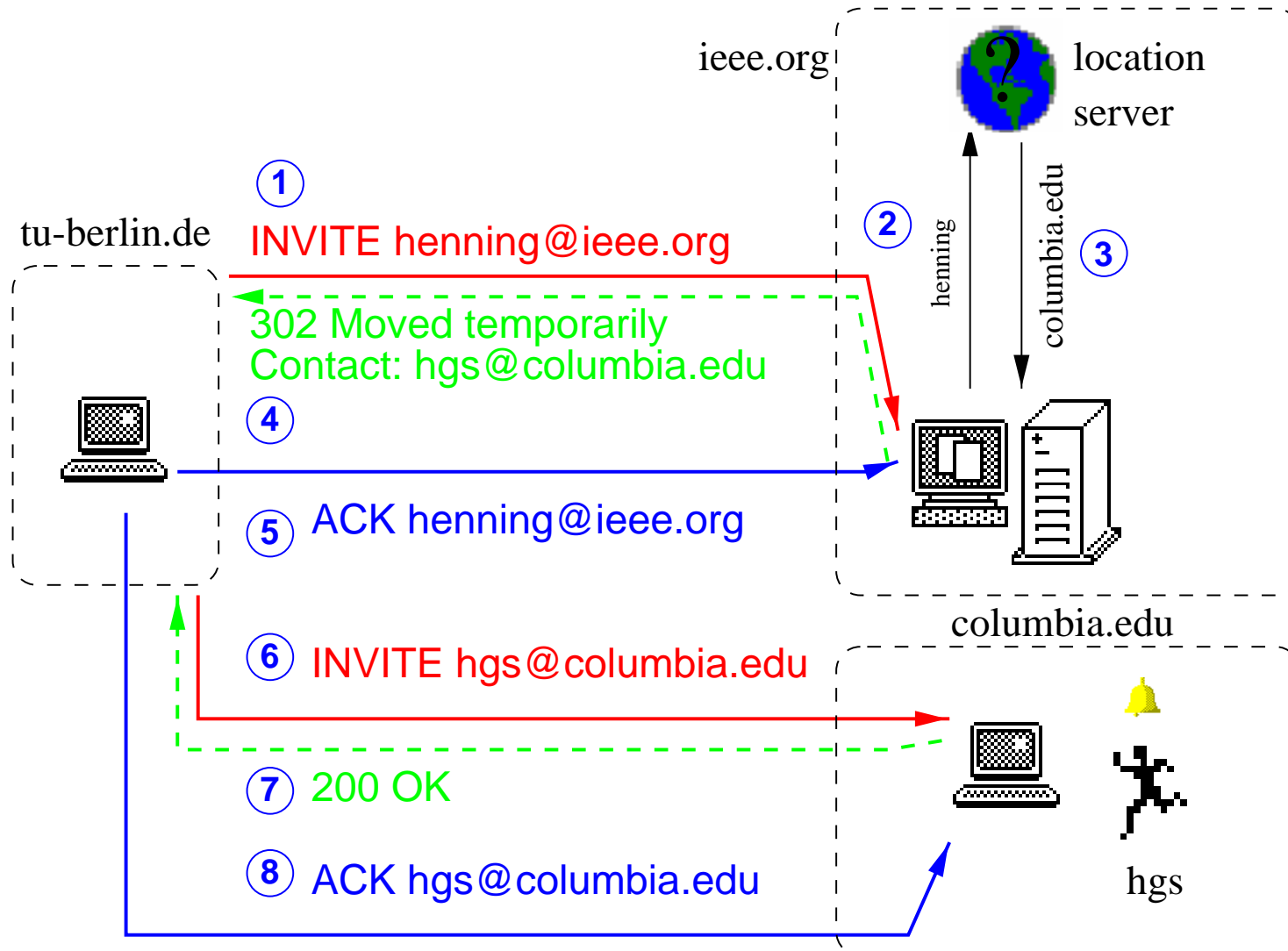
1. called server may map name to *user@host*
2. callee accepts, rejects, forward (→ new address)
3. if new address, go to step 2
4. if accept, caller confirms
5. ... conversation ...
6. caller or callee sends BYE

may “fork”

SIP Operation in Proxy Mode



SIP Operation in Redirect Mode



SIP Status

- IETF “Proposed Standard” (Feb. 1999), RFC 2543
- range of implementations: server, PC client, embedded systems (“Internet phones”)
- 2nd bake-off: about 15 implementations
- extensions planned for “buddy lists”

Aside: Where is Mobile IP Needed?

Not needed if short-lived, restartable client-server connections:

http	short, stateless
smtp	short, restartable
pop, imap	short, restartable
telnet	yes, but rarely used by mobiles (?)
ftp	restartable, rare
chat, irc	yes, but fixable (proxy, protocol)

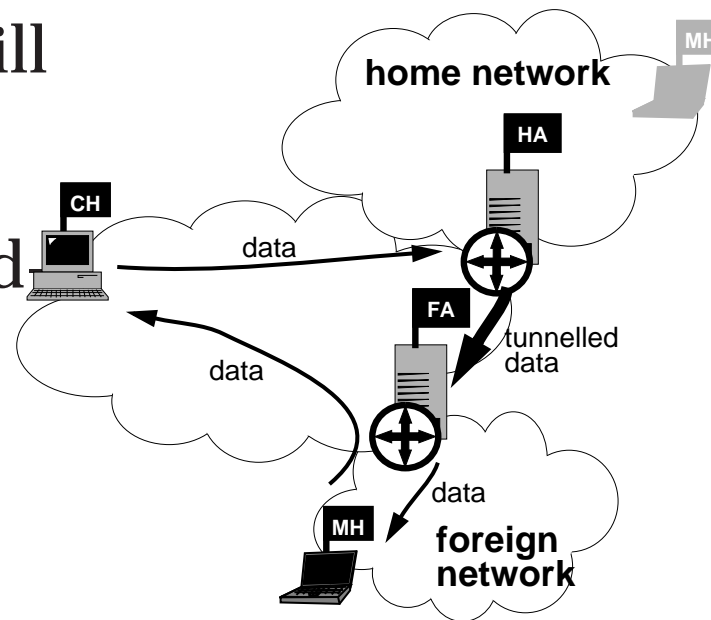
Requirements for VoIP Mobility

- fast hand-off, preferably without network support:
 - voice packet every 20–50 ms
 - FEC can recover 2–3 packets
- low packetization overhead:

headers	IP+UDP+RTP	40 bytes
G.729 payload	8 kb/s, 10 ms	$n \cdot 10$ bytes
- simple end systems

Mobile IP Issues

- encapsulation
- dog-legged routing
- binding updates still through HA
- may fail with IP address filters
- stack/infrastructure changes



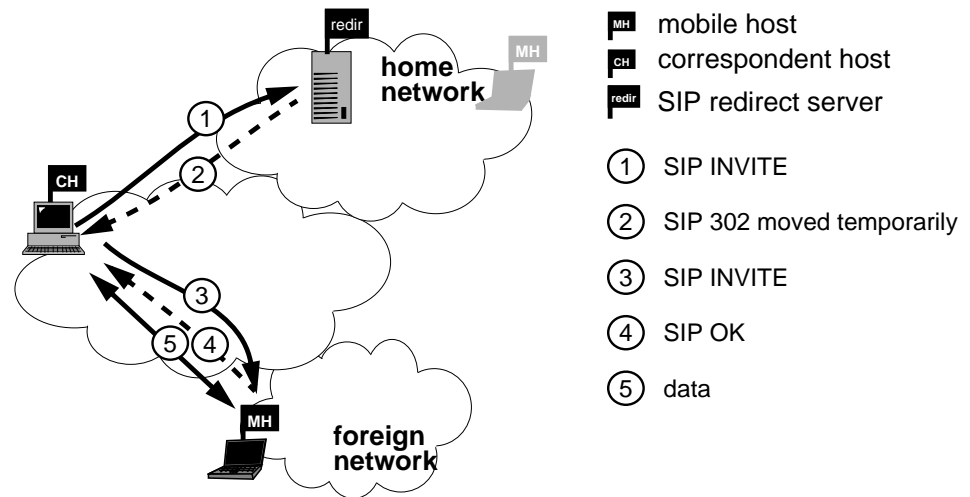
- MH** mobile host
- CH** correspondent host
- HA** router with home agent functionality
- FA** router with foreign agent functionality

SIP Mobility Overview

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

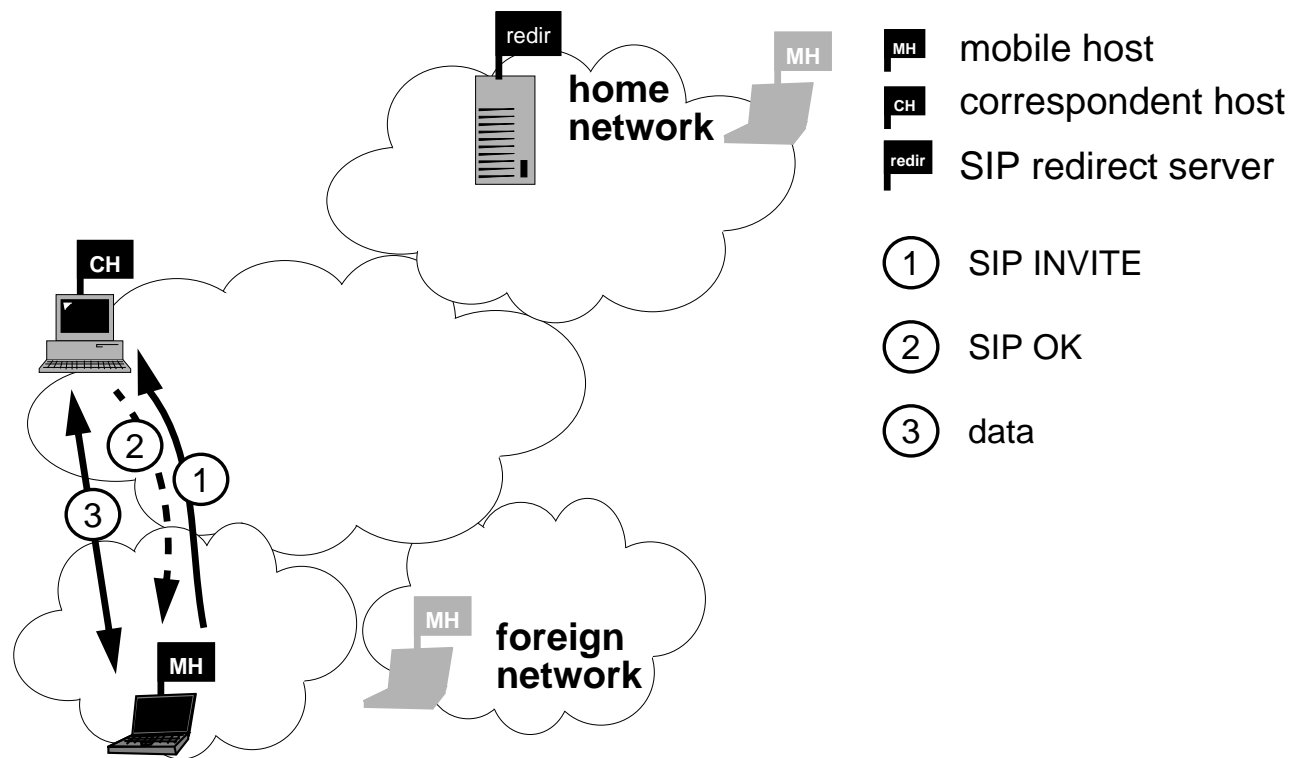
SIP mobility: pre-call

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



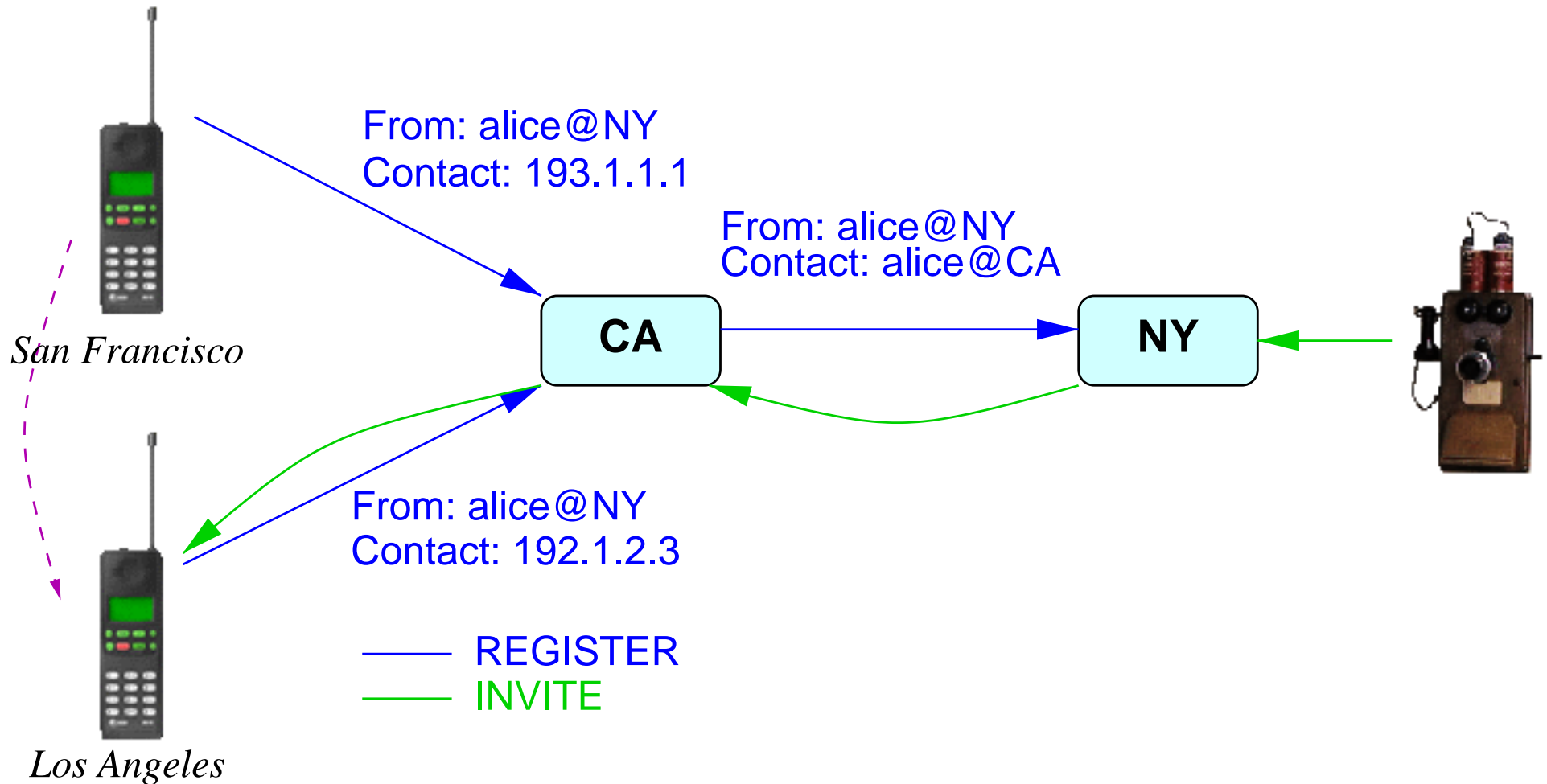
SIP Mobility: Mid-call

MH→CH: new INVITE, with Contact and updated SDP



SIP Mobility: Multi-stage Registration

Don't want to bother home registrar with each move



802.11 Movement Detection: Ad-Hoc Mode

no “access point” \Rightarrow regular station as BS

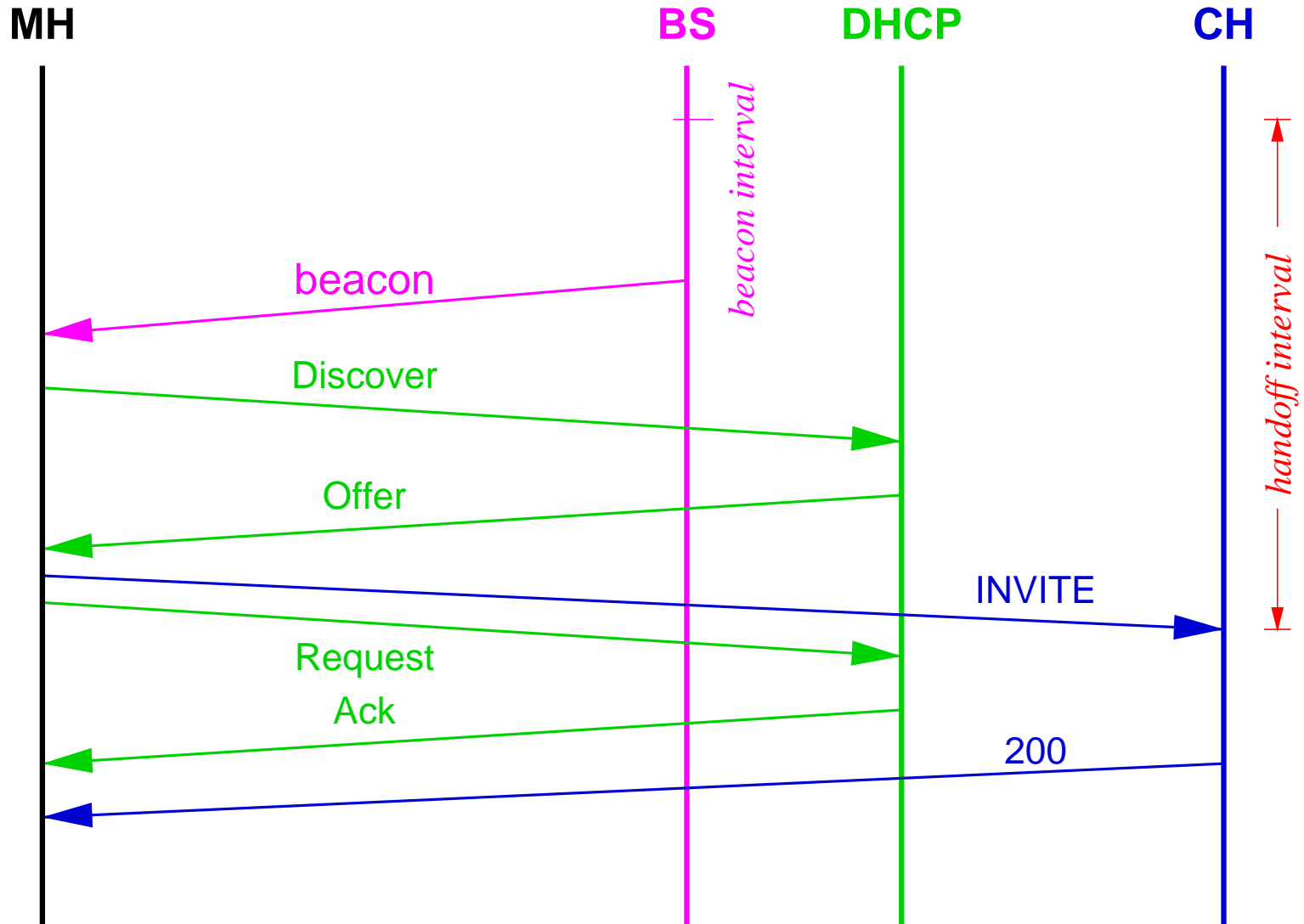
- BS serves as default router
- periodic multicast beacon
- pick best: driver provides SNR, strength
- could use regular multicast packets for quick BS discovery

802.11 Movement Detection: Infrastructure Mode

access point (AP) for BSS

- attachment handled by MAC layer, invisible to application
- BSSID is contained in 802.11 packet, but
 - BSSID not visible to application
 - driver doesn't get notified if MH attaches to new AP
- modified driver that polls hardware?

Handoff Performance



Open Issues

- handoff performance in a loaded network
- soft hand-off: IP-level vs. application proxies
- soft hand-off for 802.11 infrastructure mode possible?
- RTP issues: collision detection

Conclusion

- mobile telephony = most common mobile application
- all-IP network: can't punt hand-off
- terminal mobility as special case of personal mobility
- SIP-based mobility \Rightarrow immediate deployment

For more information...

Papers: <http://www.cs.columbia.edu/IRT>

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

SIP: <http://www.cs.columbia.edu/sip>