

SIP: Status and Directions

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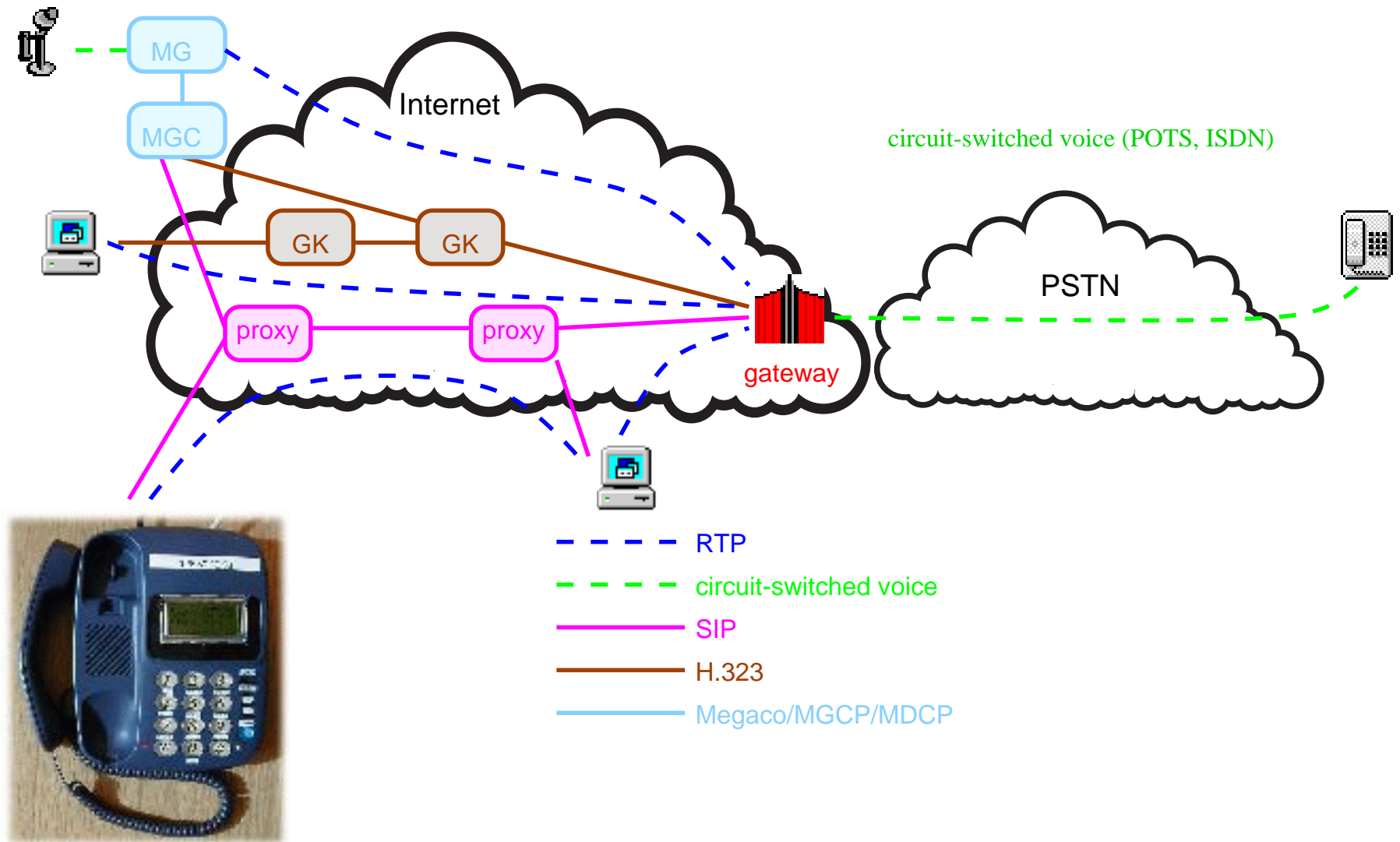
Nextone

December 14, 1999

Overview

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

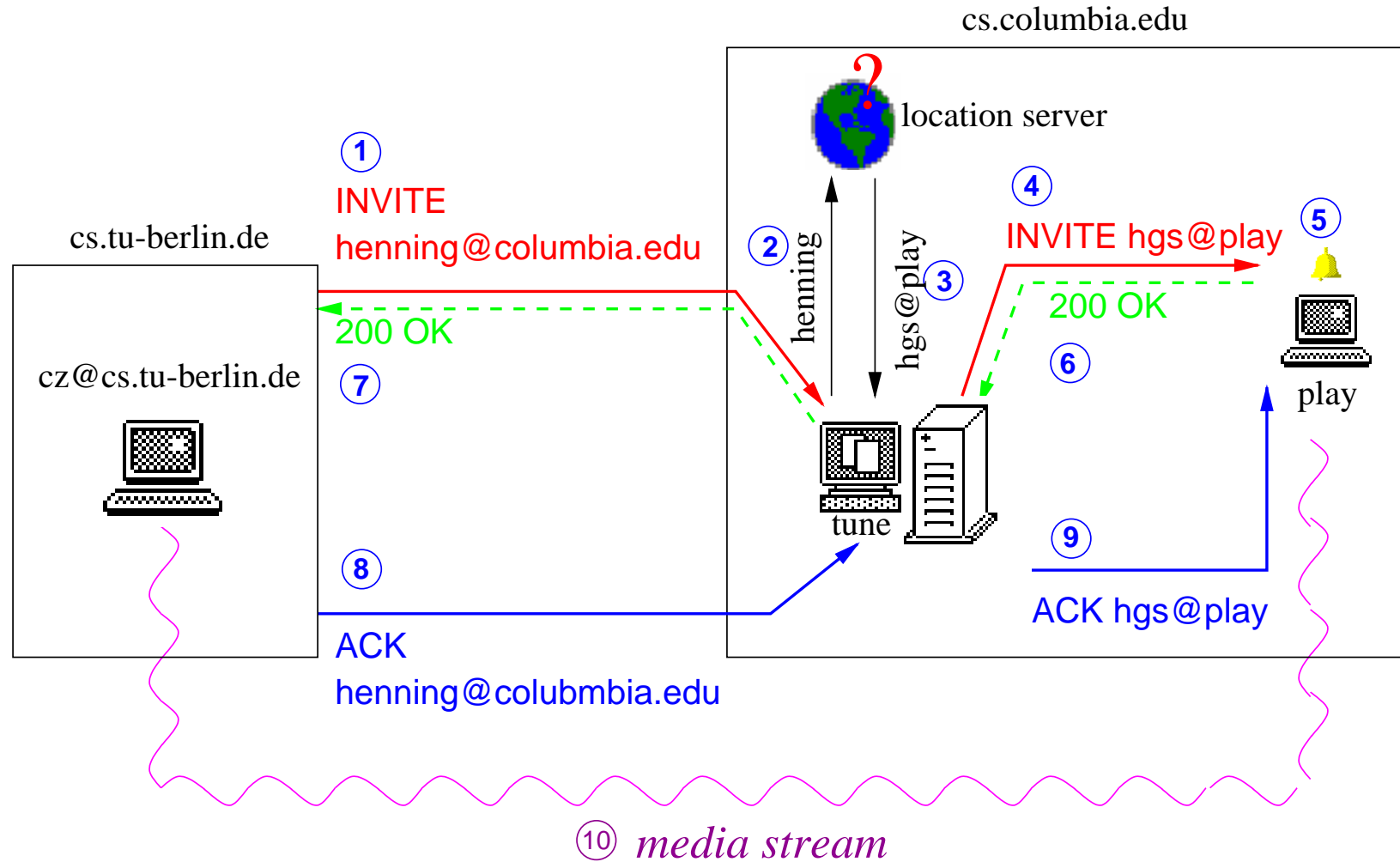
Architecture



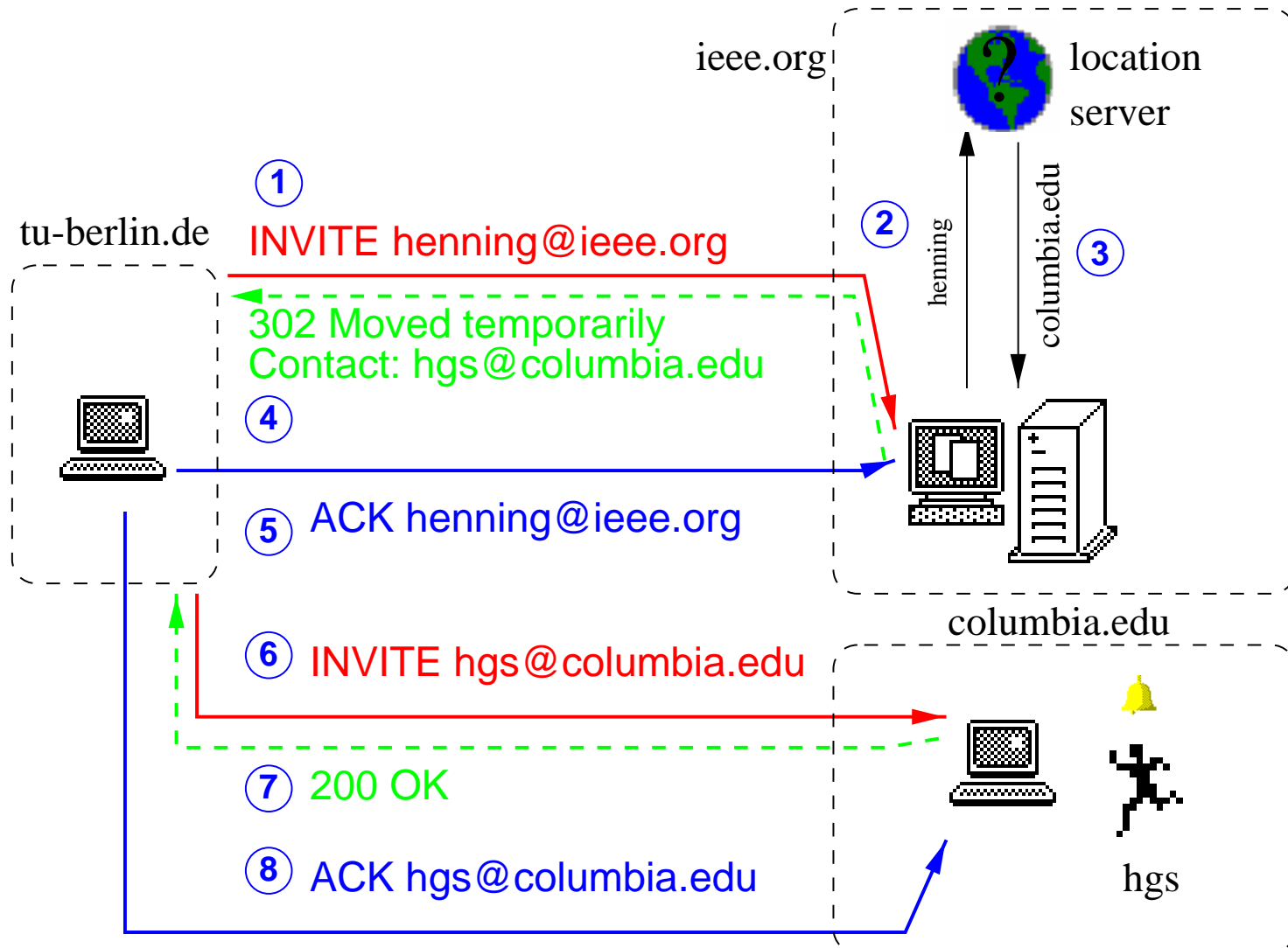
SIP 101

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 (→ 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 \Rightarrow 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 →
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	Telogy
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)

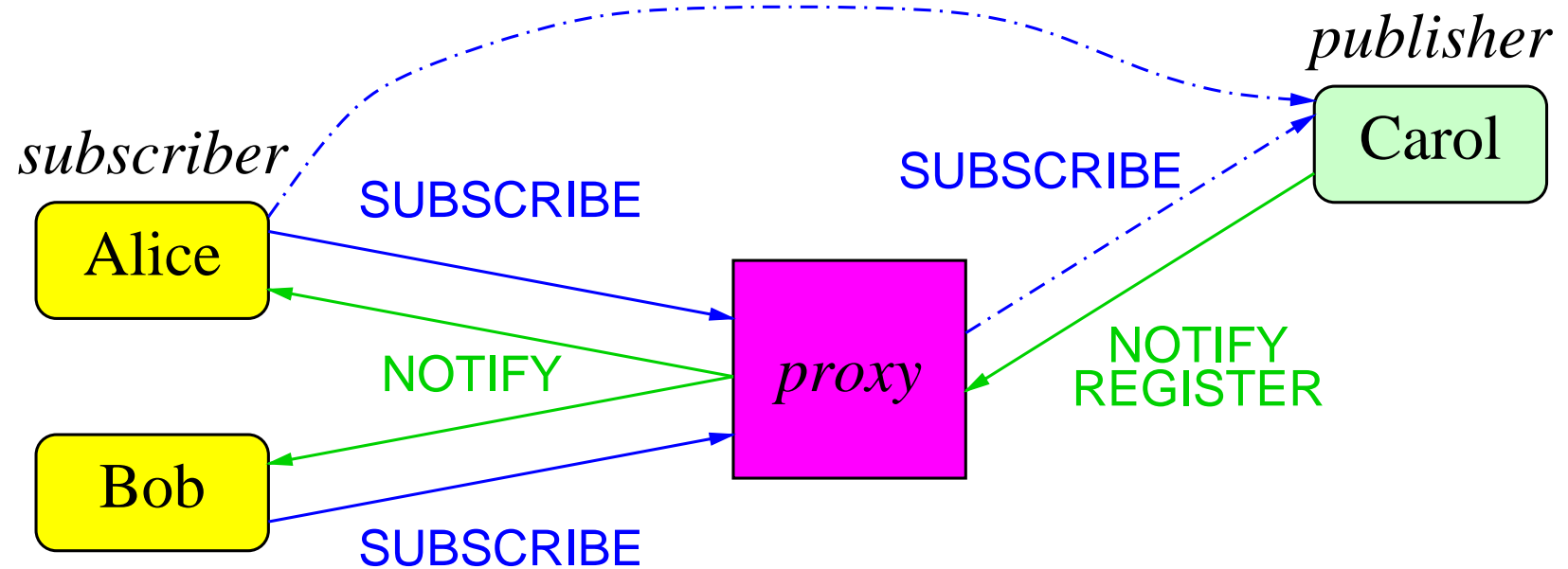
Integrating Signaling and Instant Messaging: Some Ideas

- “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

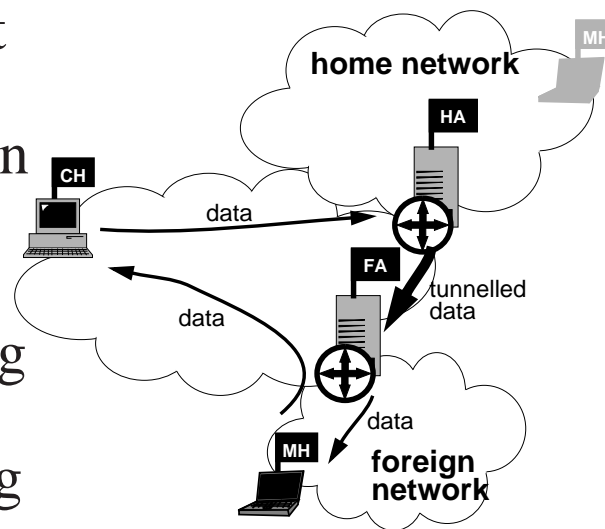
- add two methods: **SUBSCRIBE** and **NOTIFY**
- proxy server may intercept **SUBSCRIBE**
- use message body for event description
- default: presence, indicated by **REGISTER**
- one of *many* proposals for presence (IETF WG!)

SIP for Event Notification



Mobility

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



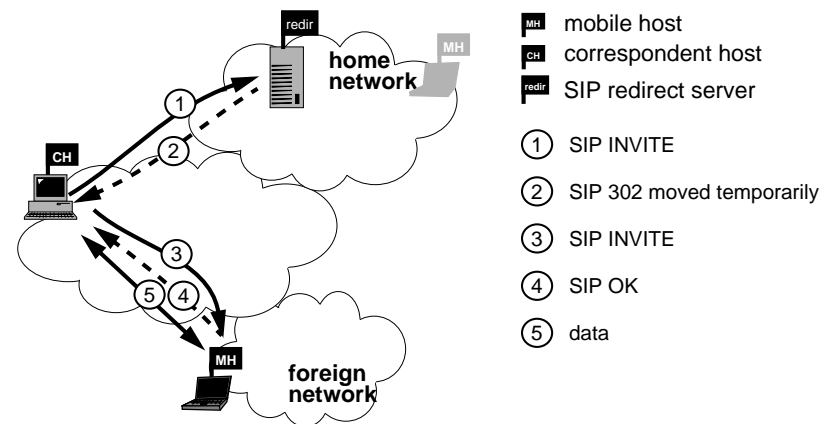
- MH** mobile host
- CH** correspondent host
- HA** router with home agent functionality
- FA** router with foreign agent functionality

SIP Mobility Overview

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

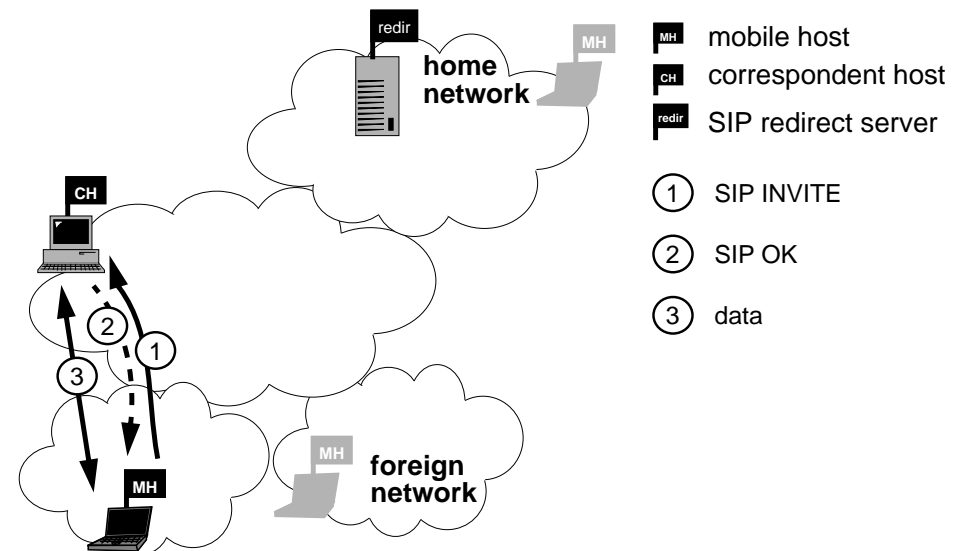
SIP Mobility: Pre-call

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



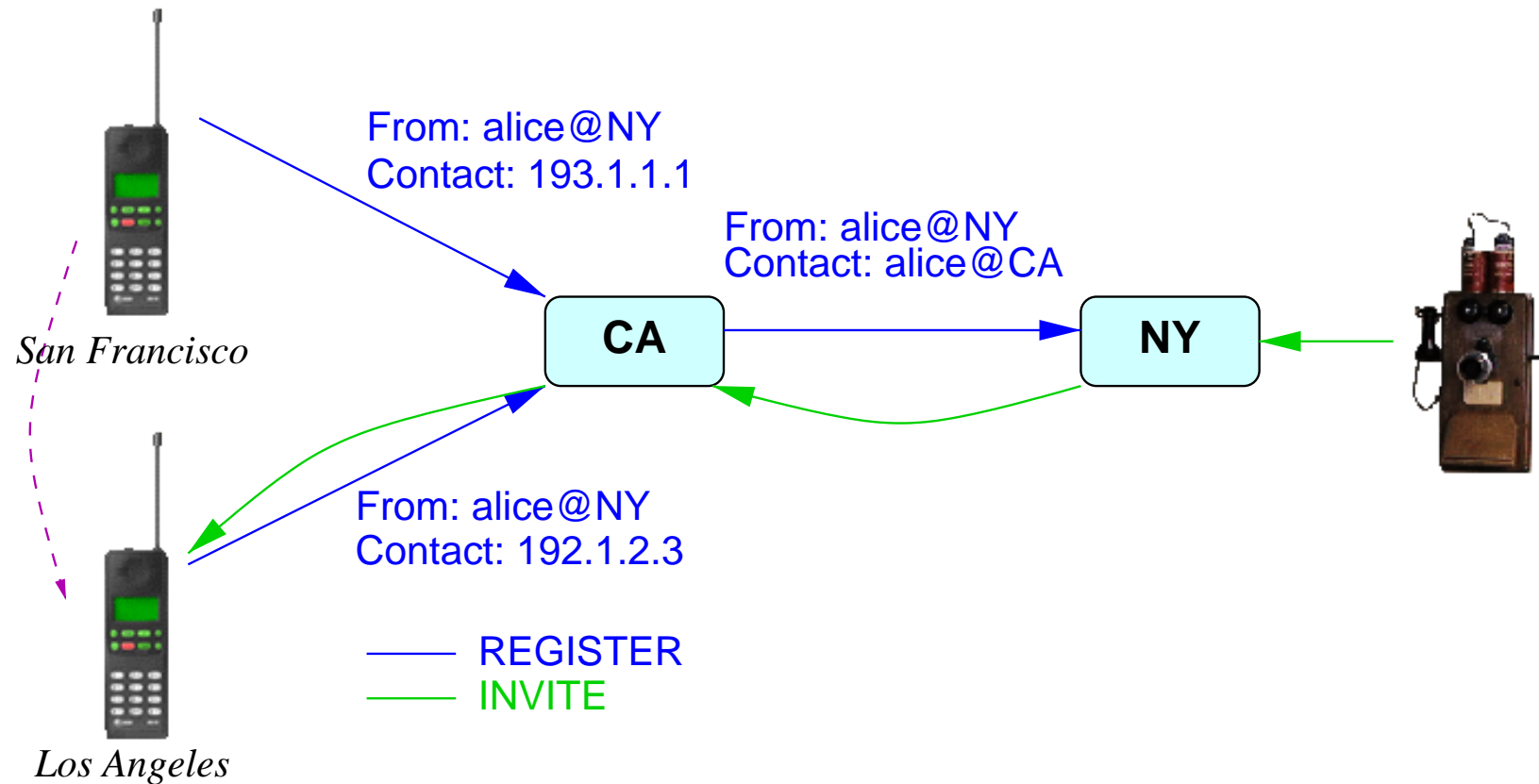
SIP mobility: mid-call

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