Application-Layer Mobility Using SIP

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Overview

- the Session Initiation Protocol (SIP)
- roaming users
- terminal mobility
- personal mobility
- service mobility
- session mobility

SIP Overview

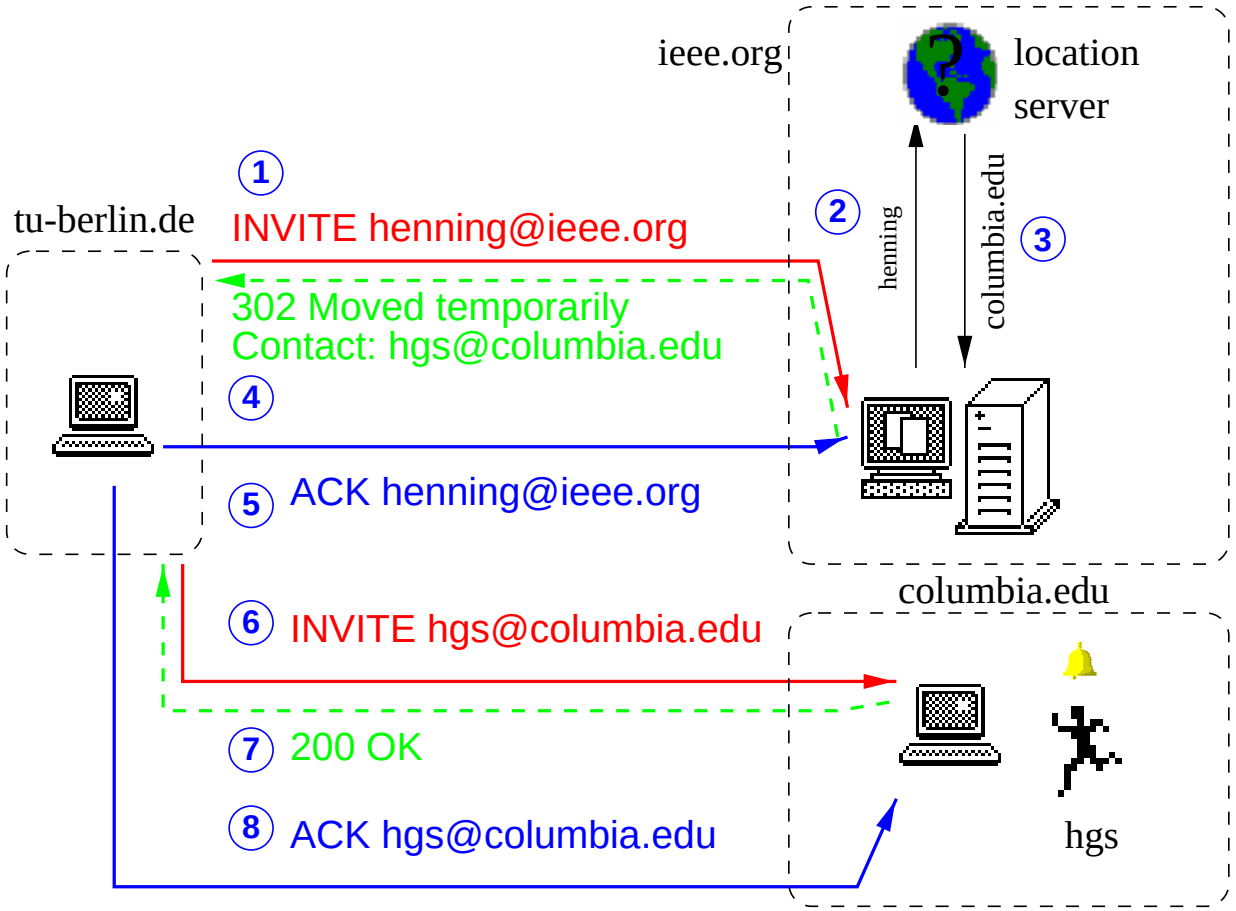
- protocol for establishing, modifying, tearing down (multimedia) *sessions*
- IETF Proposed Standard since March 1999
- multimedia = audio, video, shared applications, text, . . .
- also used for “click-to-dial” (PINT wg) and possibly Internet call waiting (SPIRITS wg)
- to be used for PacketCable Distributed Call Signaling and Third-Generation Wireless (3GPP, 3GPP2)
- also proposed for presence, instant messaging and event notification

SIP Components

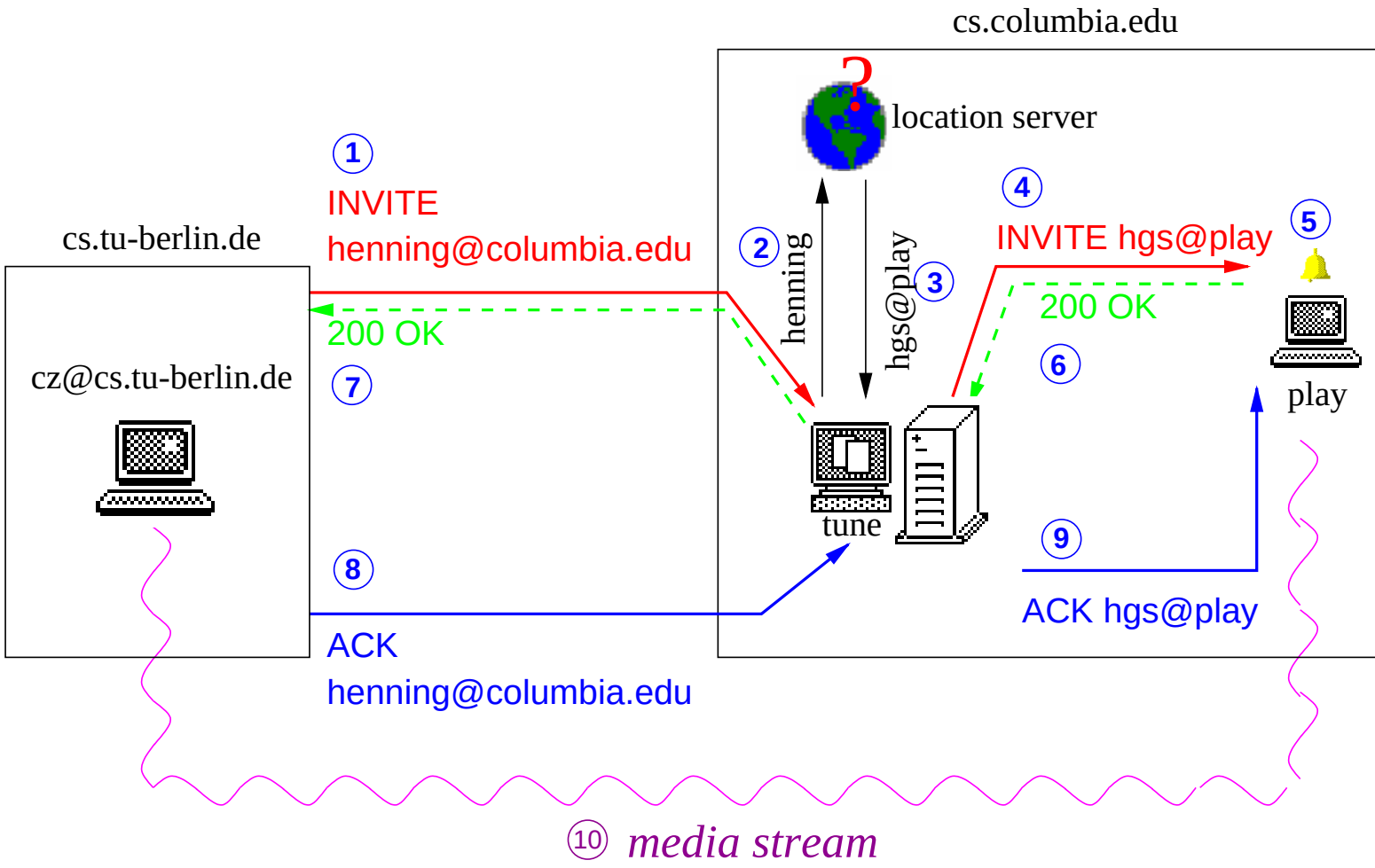
entity	does	examples
proxy server	forward calls	firewall controller, “call router”
redirect server		“application server”
user agent	end system	SIP phone, gateway, “softswitch”
registrar	location mgt.	mobility support

Roles are changeable, on a request-by-request basis

SIP Example: Redirection



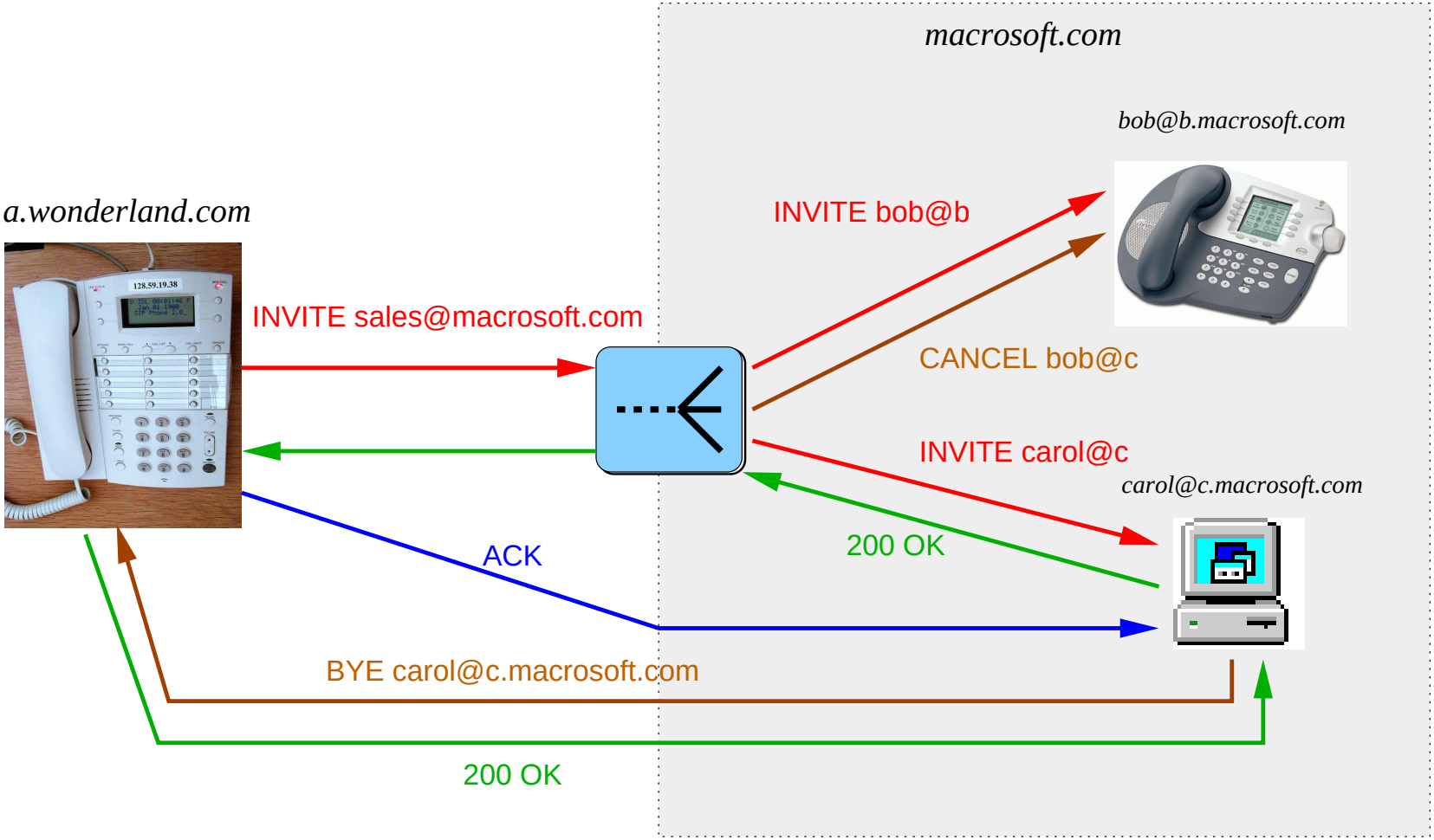
SIP Example: Proxying



SIP Advanced Features

- forking
- extensibility: new headers, methods, bodies
- security: web-like, PPP/CHAP or PGP
- multicast-capable
- support for personal, session, terminal, service mobility
- caller preferences: direct calls based on properties

SIP Forking Proxies



Mobility in an IP environment

Roaming users: logging in away from home network: hotel, home office

Terminal mobility: terminal moves between subnets

Personal mobility: different terminals, same address

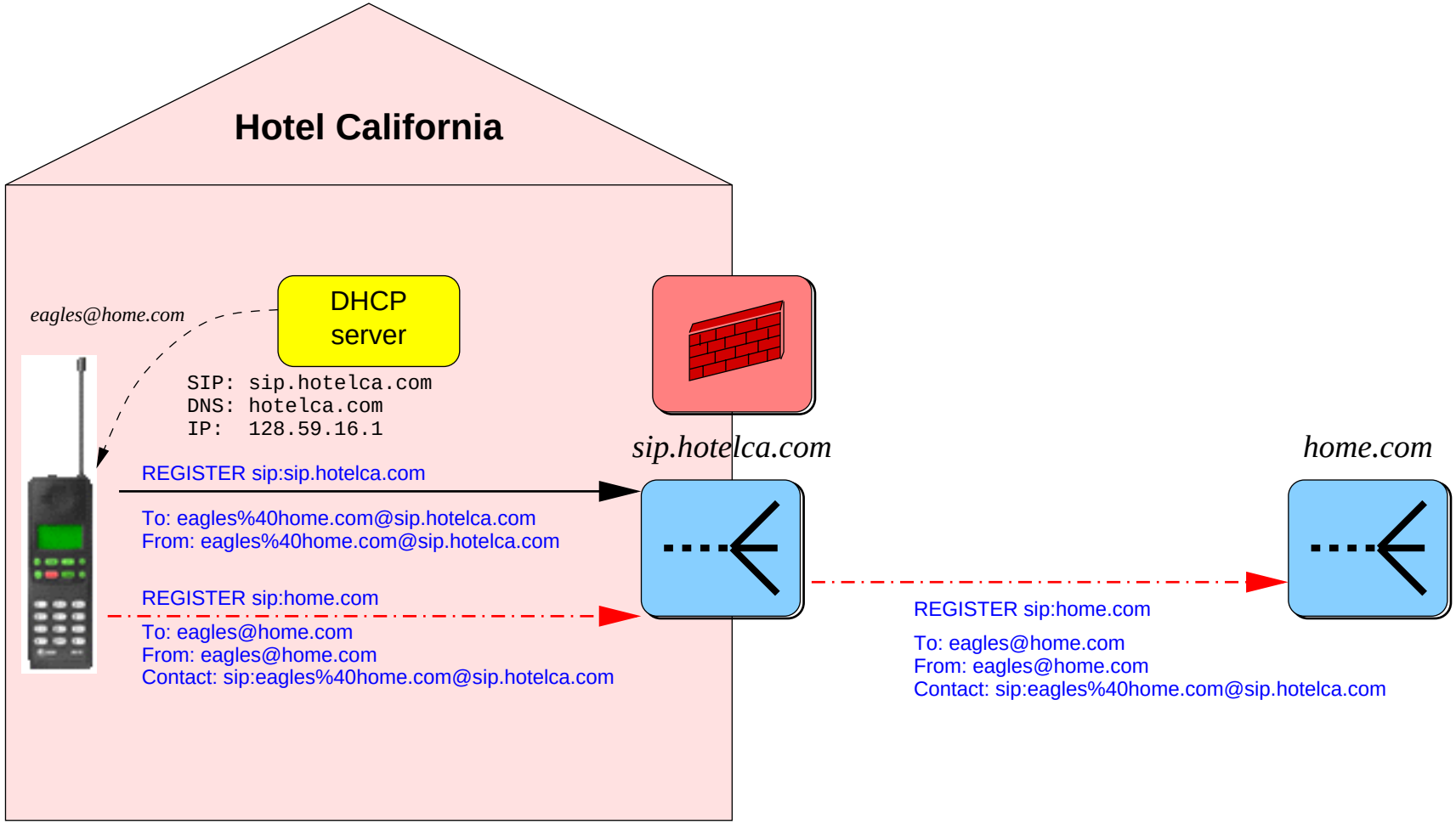
Service mobility: keep same services while mobile

Session mobility: move active session between terminals

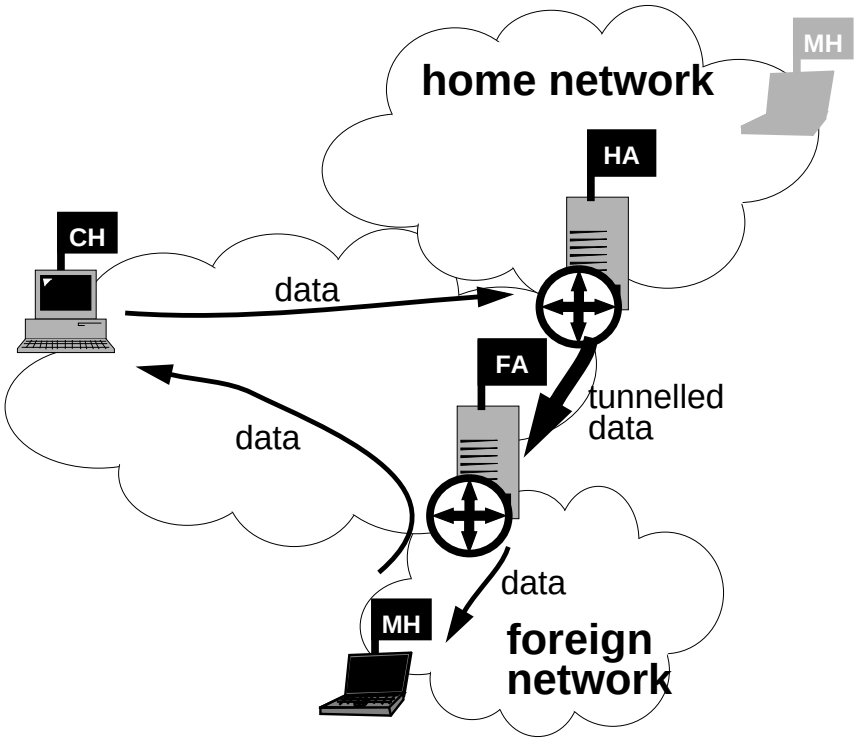
Simple mobility: roaming users

- users visit other networks: laptop, PDA, hotel phone, ...
- want to maintain external identity
- usually, just pass IP address to home registrar
- difficult if firewalls and NATs
  - requests need to use local proxy
  - thus, need to register locally

Roaming Users – Dual Registration



Terminal mobility – mobile IP



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

Terminal mobility – mobile IP difficulties

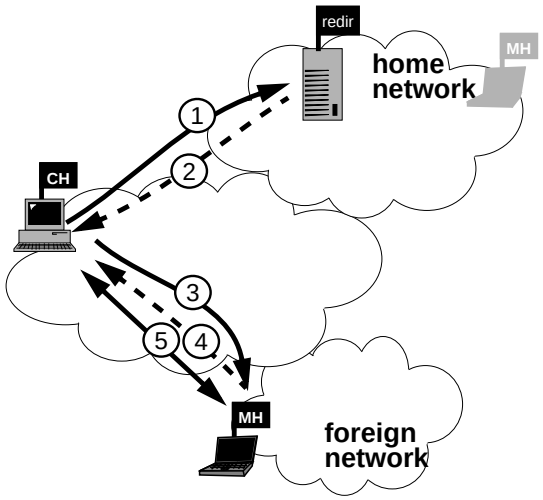
- domain of IEEE 802.11 (link layer), 3GPP (radio access network), mobile IP (network layer), . . .
- network-layer mobility has problems:
  - lack of deployment – home provider has no interest
  - need two addresses – home and visiting
  - dog-legged routing in IPv4
  - may not work with IP address filtering except through triangle routing
  - encapsulation overhead for voice: 8–20 bytes/packet for a 50-byte payload
  - authentication of redirection

SIP terminal mobility overview

- avoid audio packet encapsulation overhead
- one one-way delay handover, possibly with packet intercept
- pre-call mobility ➠ SIP proxy, redirect
- mid-call mobility ➠ SIP re-INVITE, RTP
- recovery from disconnection

SIP terminal mobility: pre-call

- MH acquires IP address via DHCP
- optional: MH finds local registrar via multicast REGISTER or uses designated SIP server (via DNS SRV)
- MH updates home SIP server – deregister old, register new
- optimization: hierarchical LR (later)

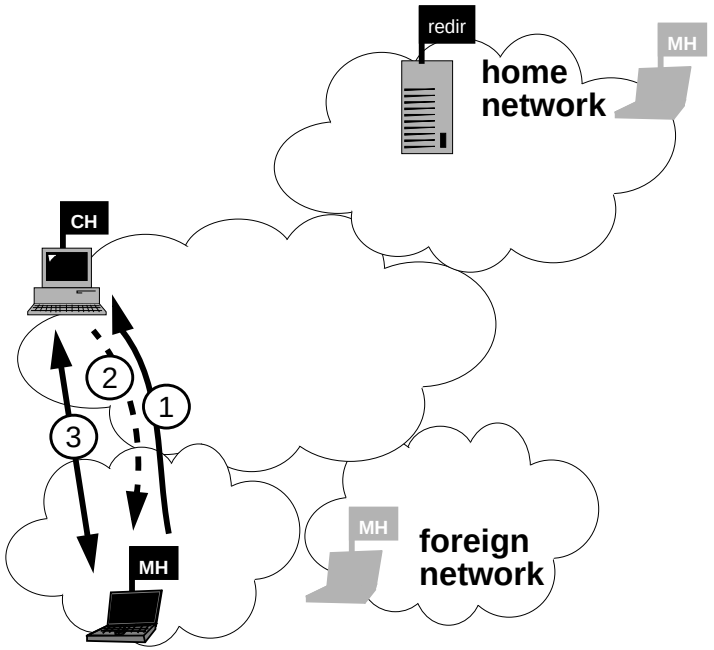


MH mobile host
CH correspondent host
redir SIP redirect server

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

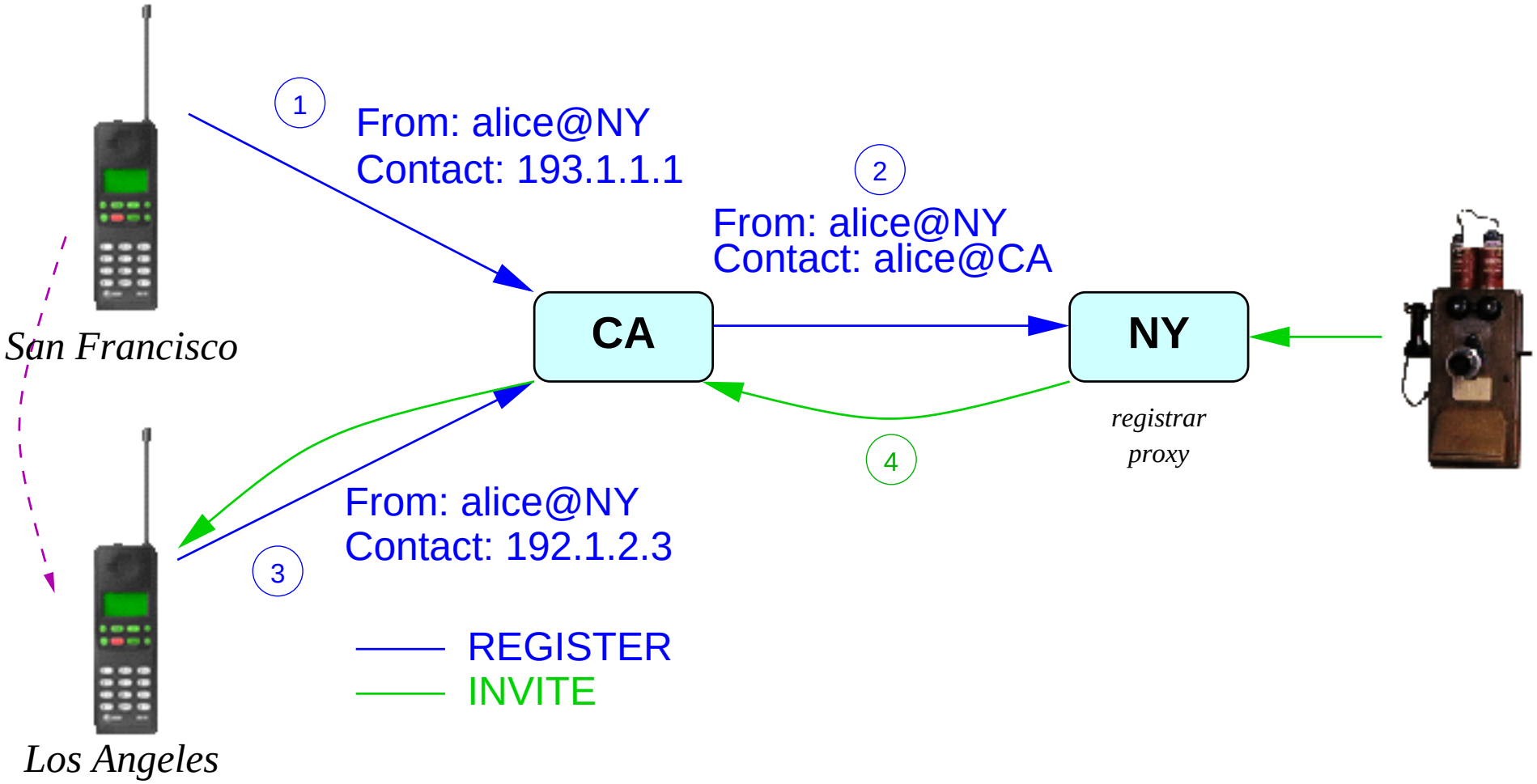
SIP terminal mobility: mid-call

- acquire new IP address
- MH→CH: new INVITE, with Contact header and updated SDP
- re-registers with home registrar
- requires one one-way delay

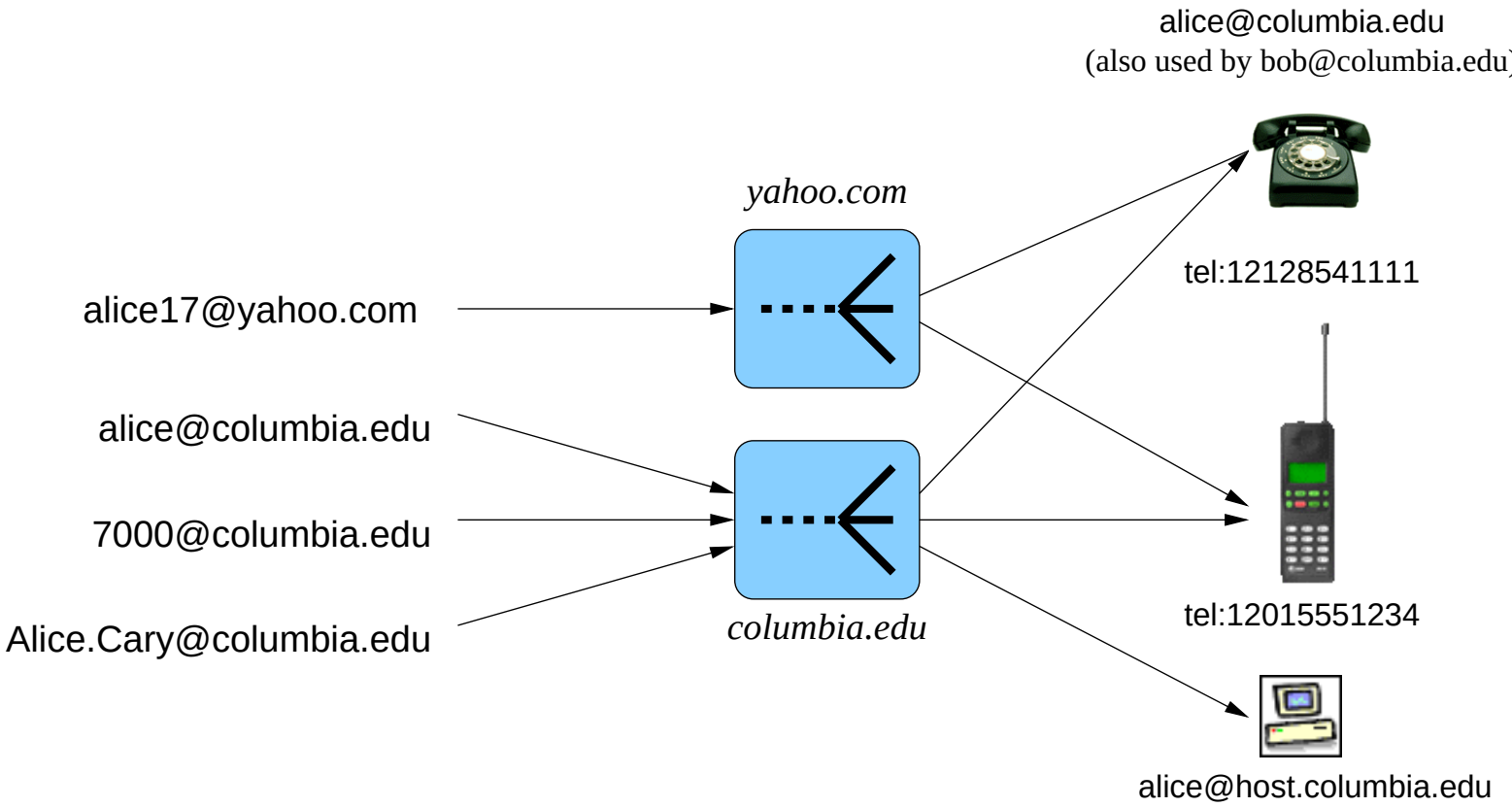


SIP terminal mobility: multi-stage registration

Don’t want to bother home registrar with each move



Personal mobility



Personal mobility

- switch between PDA, cell phone, PC, Ethernet phone, Internet appliance, . . .
- several “generic” addresses, one person/function, many terminals
- e.g., `tel:2129397042`, `hgs@cs.columbia.edu`, `schulzrinne@yahoo.com` or `support@acme.com`
- SIP is designed for that – proxying and redirection does translation
- but: need mapping mechanisms to recognize registrations as belonging to the same person
- some possible solutions:
  - dip into LDAP personnel database or `/etc/passwd` to match phone number and variations of name (*J.Doe*, *John.Doe*, *Doe*)
  - need dialing plan to recognize `7042@cs.columbia.edu` and `tel:2129397042` as same

Service mobility

Examples:

- speed dial & address book
- media preferences
- special feature buttons (voice mail, do-not-disturb)
- incoming call handling instructions
- buddy lists
- features in home provider server

$\longrightarrow$ independent of terminal (including pay phone!), across providers

Service mobility

- REGISTER can retrieve configuration information (e.g., speed dial settings, distinctive ringing or voice mail settings)
- but needs to be device-independent
- most such services (e.g., voicemail forwarding, call filtering) should remain on server(s)
- use SIP Route mechanism to direct path of outgoing calls via home server

```
Route: <sip:alice@home.net>, <sip:alice@services-r-us.com>
```

Service mobility – call handling

- need uniform basic service description model $\longrightarrow$ Call Processing Language (CPL)
- CPL = XML-based flow graph for inbound & outbound calls
- CPL for local call handling
- update CPL from terminal: add telemarketer to block list
- harder: synchronize CPL changes across multiple providers
- one possibility: REGISTER updates information, but device needs to know that it has multiple identities
- merging of call logs

SIP and mobility: issues

- doesn't work for TCP applications – solutions:
  - punt: "don't type and drive"
  - application-layer awareness: restart web, email, ftp transfer – need for deep fade anyway. . .
  - TCP redirect (Snoeren/Balakrishnan)
  - NAT-style boxes controlled by SIP (see Telcordia ITSUMO project)
- fast hand-off via SIP proxies with media translators
- but: works nicely for "vertical handoff" between different technologies - e.g., transfer call from mobile handset to office videophone when arriving at work

Conclusion

- network-layer mobility neither sufficient nor available
- many common services don’t need network-layer support
- application-layer mobility for sessions
- one SIP-based approach for multimedia sessions, presence & events