

# Application-layer mobility for Internet multimedia sessions

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## Overview

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- third-generation wireless networks
- using SIP for supporting facets of mobility
  - terminal mobility
  - personal mobility
  - session mobility
  - service mobility
- mobile code for Internet telephony

## Third-Generation Wireless Networks

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- 2G: GSM, CDMA narrowband
- 3G: different radio networks (wideband CDMA, EDGE, ...), but
  - up to 2 Mb/s bandwidth, but really only 144 kb/s
  - IPv6 to the end system
  - Session Initiation Protocol (SIP) for multimedia session setup
  - gateways to PSTN
  - being standardized in 3GPP (GSM), 3GPP2 (CDMA), MWIF (architecture) and IETF (IPv6, AAA, SIP)

## SIP (Session Initiation Protocol)

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- 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

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1. called server may map name to *user@host*
2. callee accepts, rejects, forward ( $\rightarrow$  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 Servers and Clients

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**UAC:** user-agent client (caller application)

**UAS:** user-agent server  $\Rightarrow$  accept, redirect, refuse call

**redirect server:** redirect requests

**proxy server:** server + client

**registrar:** track user locations

- user agent = UAC + UAS
- often combine registrar + (proxy or redirect server)

## Mobility in an IP environment

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**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

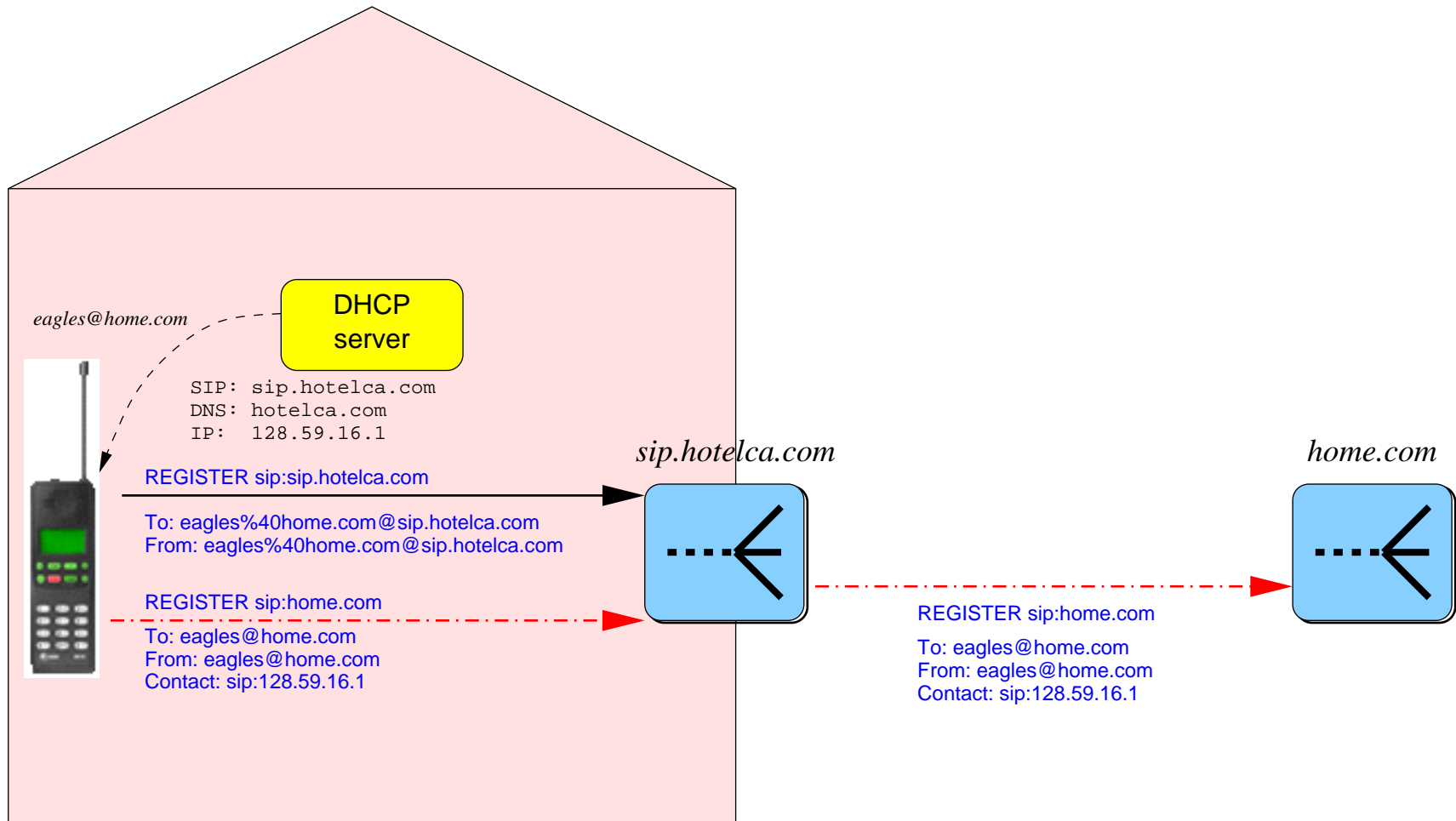
## Simple Mobility: Roaming Users

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- 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
- also may want to use home services while traveling



# Roaming Users



## Terminal mobility

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- domain of IEEE 802.11, mobile IP + hierarchical enhancements
- main problems:
  - handover performance
  - handover failure due to lack of resources in new network
  - authentication of redirection

## Aside: Where is Mobile IP Needed?

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**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)

Should make applications restartable to recover from deep fades

## Requirements for VoIP Mobility

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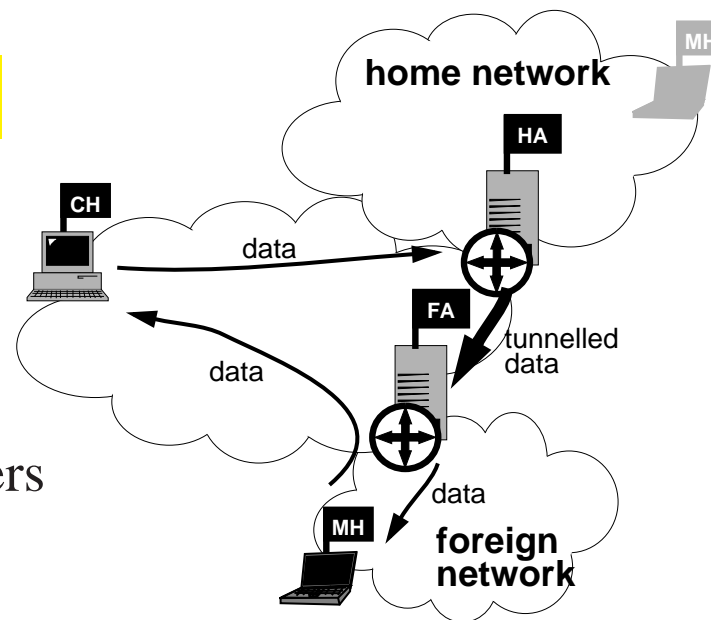
- fast hand-off, preferably without network support:
  - voice packet every 20–50 ms
  - FEC can recover 2–3 packets
- low packetization overhead:

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

## Mobile IP Issues

### Fixed in IPv6

- encapsulation
- need permanent IP address
- dog-legged routing
- binding upd. need 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

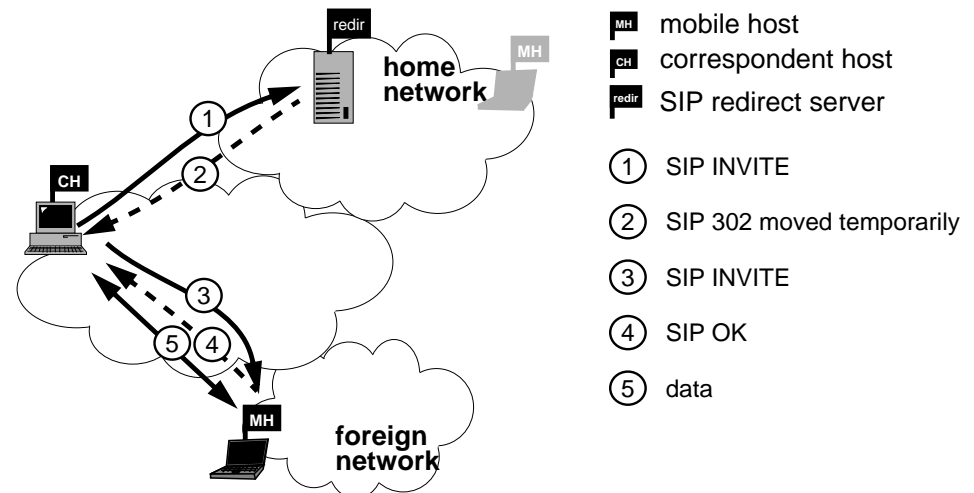
## SIP terminal mobility overview

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

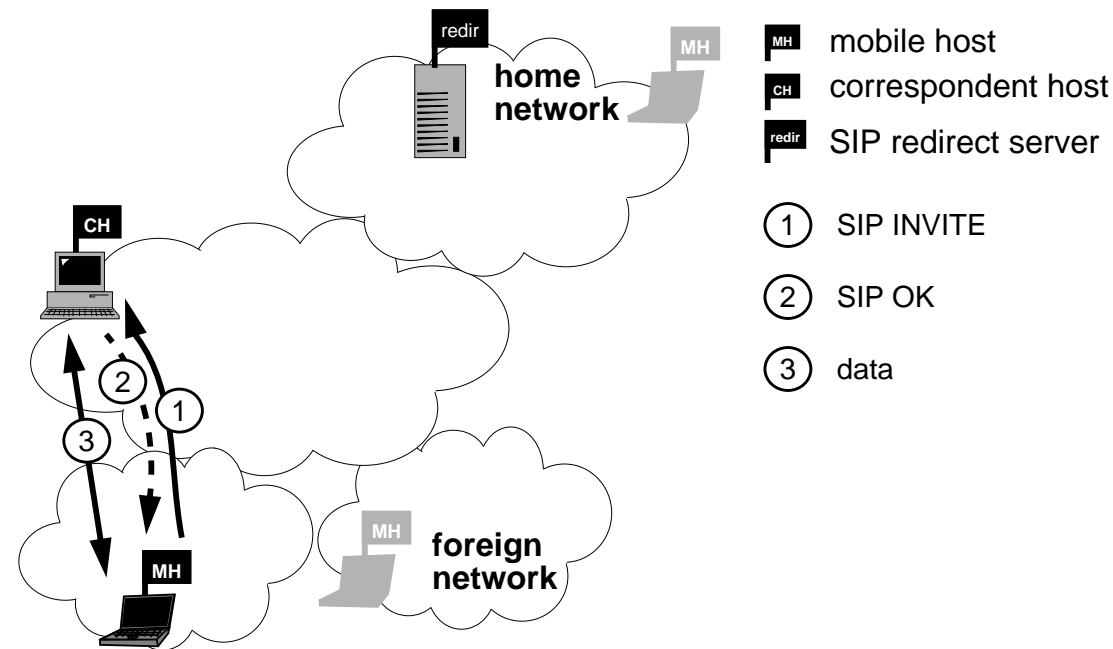
## SIP terminal mobility: pre-call

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



## SIP terminal mobility: mid-call

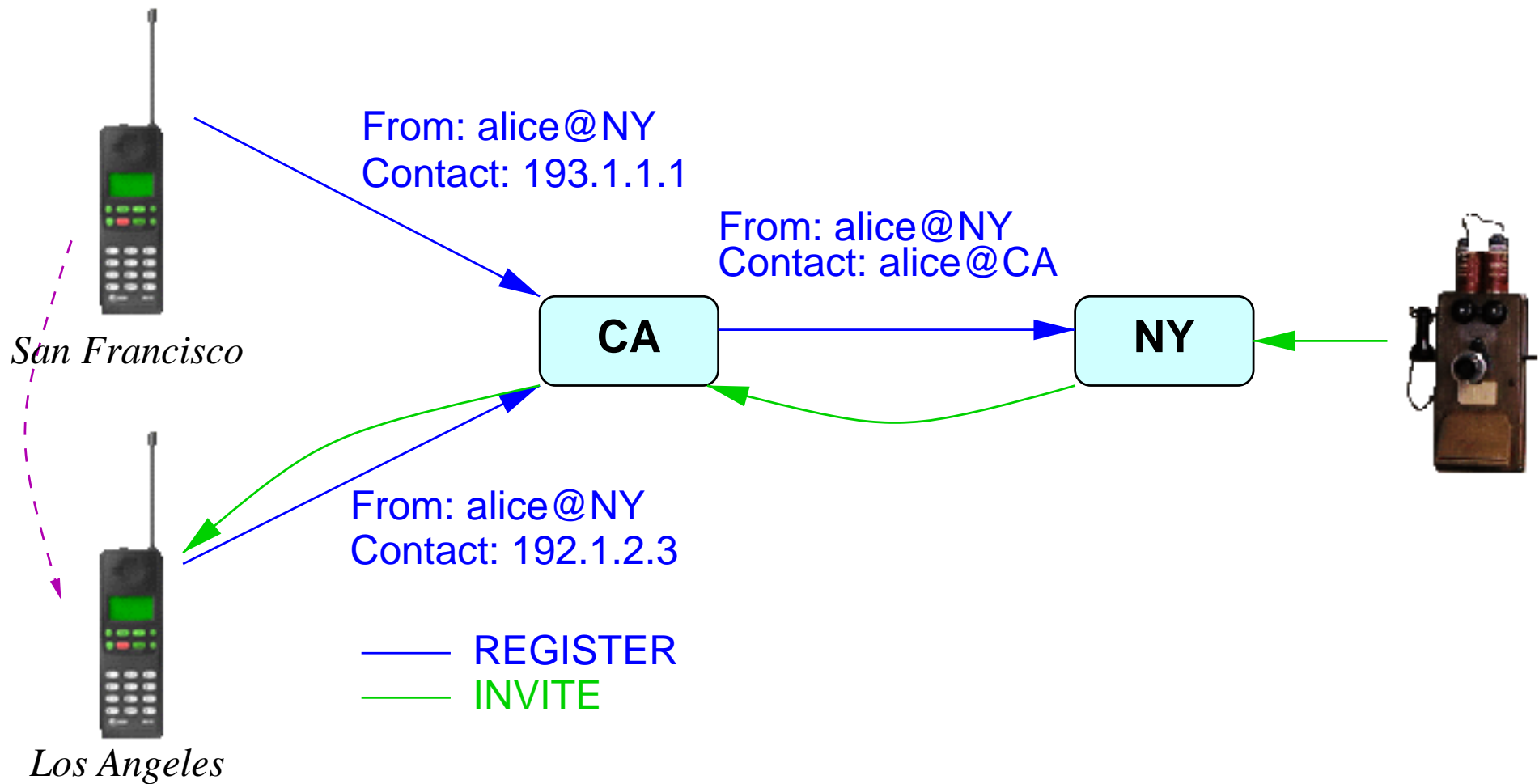
- MH→CH: new INVITE, with Contact and updated SDP
- re-registers with home registrar





## SIP terminal mobility: multi-stage registration

Don't want to bother home registrar with each move



## 802.11 Movement Detection: Ad-Hoc Mode

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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

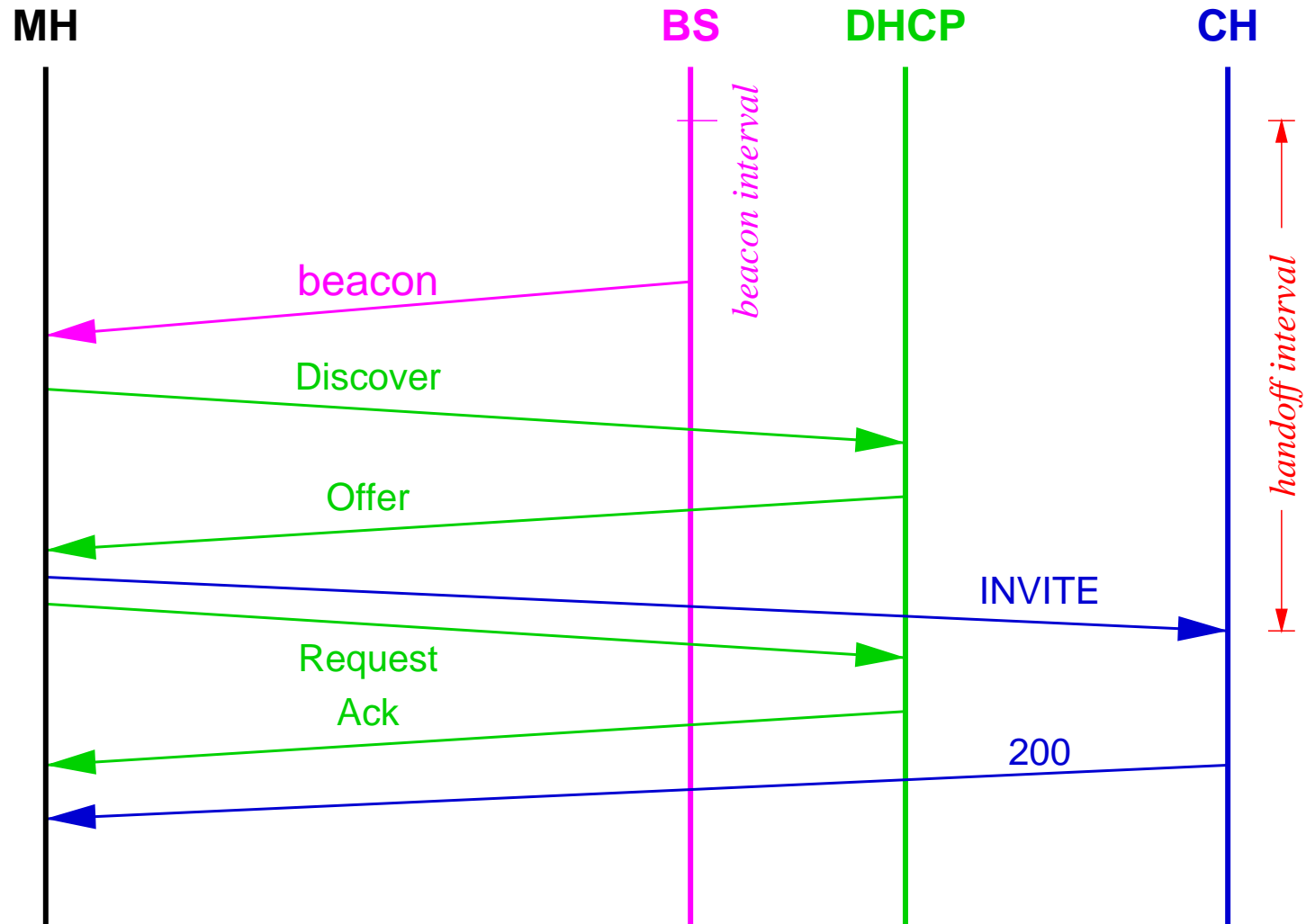
## 802.11 Movement Detection: Infrastructure Mode

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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



## SIP and mobility: issues

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- doesn't work for TCP applications – solutions:
  - punt: “don't walk while telnet'ing”
  - application-layer awareness: restart web, email, ftp transfer – need for deep fade anyway...
  - NAT-style boxes controlled by SIP (see Telcordia ITSUMO project)
- but: works nicely for “vertical handoff” between different technologies - e.g., transfer call from mobile handset to office videophone when arriving at work

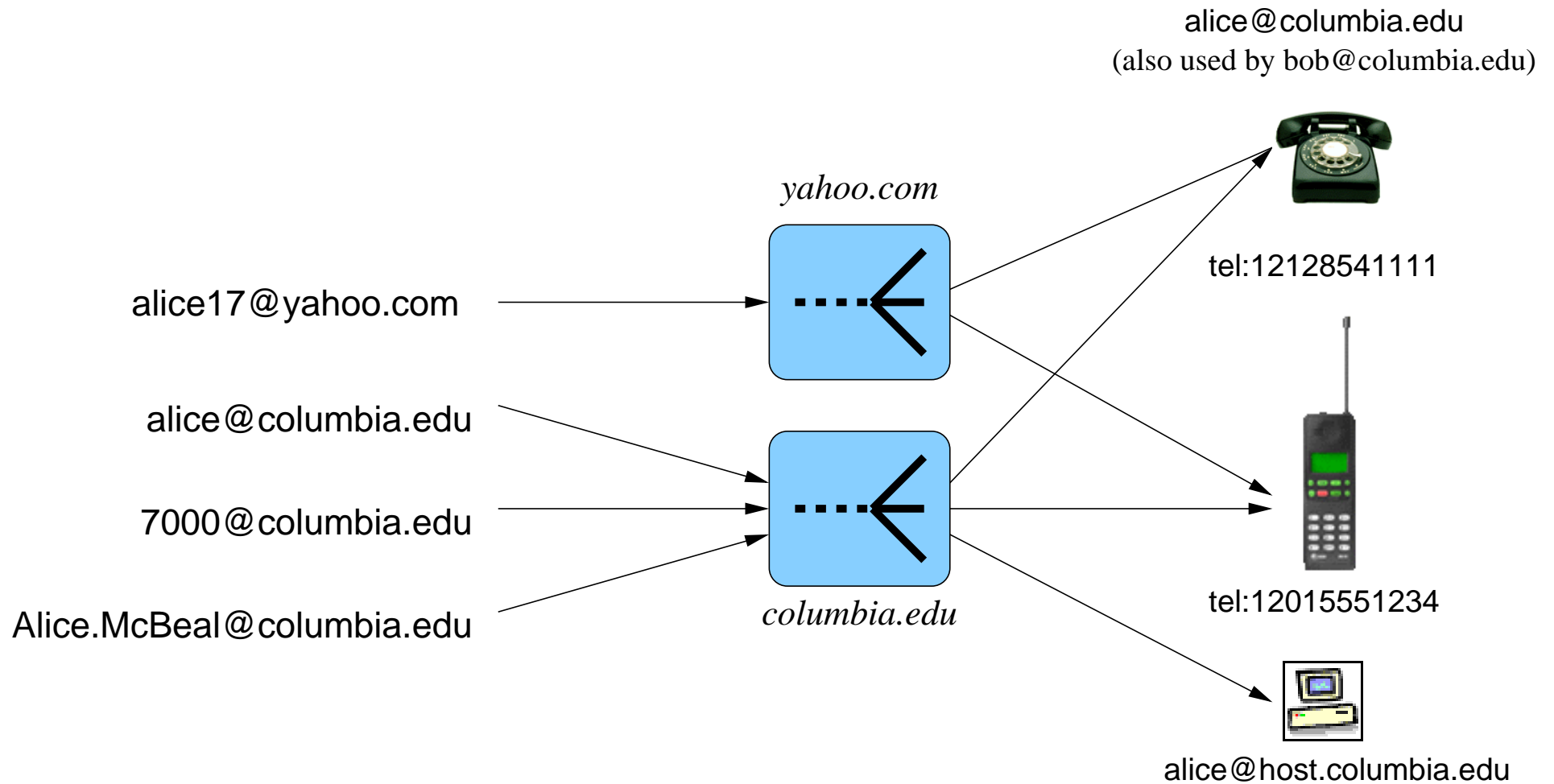
## When to use application-layer terminal mobility?

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- IPv4 network portions – latency, overhead
- mainly interactive multimedia applications
- address hiding for anonymity or media translation: SIP proxies with data intercept

# Personal mobility

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## Personal mobility

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- 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

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Examples:

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

→ independent of terminal (including pay phone!), across providers

## Service mobility

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- 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)

Separate issue: how does the payphone (or colleague's phone) recognize you?

- PDA (IR)
- i-button
- fingerprint
- speech recognition, ...

One device, but changing set of owners!

## Service mobility – call handling

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- need uniform basic service description model → Call Processing Language (CPL)
- want language similar to HTML, not PostScript → importable by user code generation tools
- safe, CPU-bounded, provable
- 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
- REGISTER updates information if device knows its multiple identities
- merging of call logs

## Textual representation

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```
<incoming>
  <address-switch field="origin" subfield="host">
    <address subdomain-of="example.com">
      <location url="sip:jones@example.com">
        <proxy>
          <busy> <sub ref="voicemail" /> </busy>
          <noanswer> <sub ref="voicemail" /> </noanswer>
          <failure> <sub ref="voicemail" /> </failure>
        </proxy>
      </location>
    </address>
    <otherwise>
      <sub ref="voicemail" />
    </otherwise>
  </address-switch>
</incoming>
</cpl>
```

## Mobility: Open Issues

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- hand-off performance
- simultaneous moves
- address hiding?
- co-existence with mobile IP
  - hand-off to non-MIP networks
  - avoiding IPv4 dog-legged routing for multimedia

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