Next-Generation Mobile Multimedia Systems

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Overview

- Internet telephony
- next-generation wireless systems
- the Session Initiation Protocol (SIP)
- mobility modes and application-layer mobility
- mobile code
- signaling, inter-domain events and messaging

What is Internet telephony

- = real-time communication using voice, video and other media across the Internet or private IP networks
- gateways to existing public phone network (PSTN)
- mostly PC to phone, but also between branch offices
- already 3.7 billion minutes for 2000 (out of about 200 billion)
- currently, about 3-10% of international traffic
- coming: PBX replacement

Why Internet telephony?

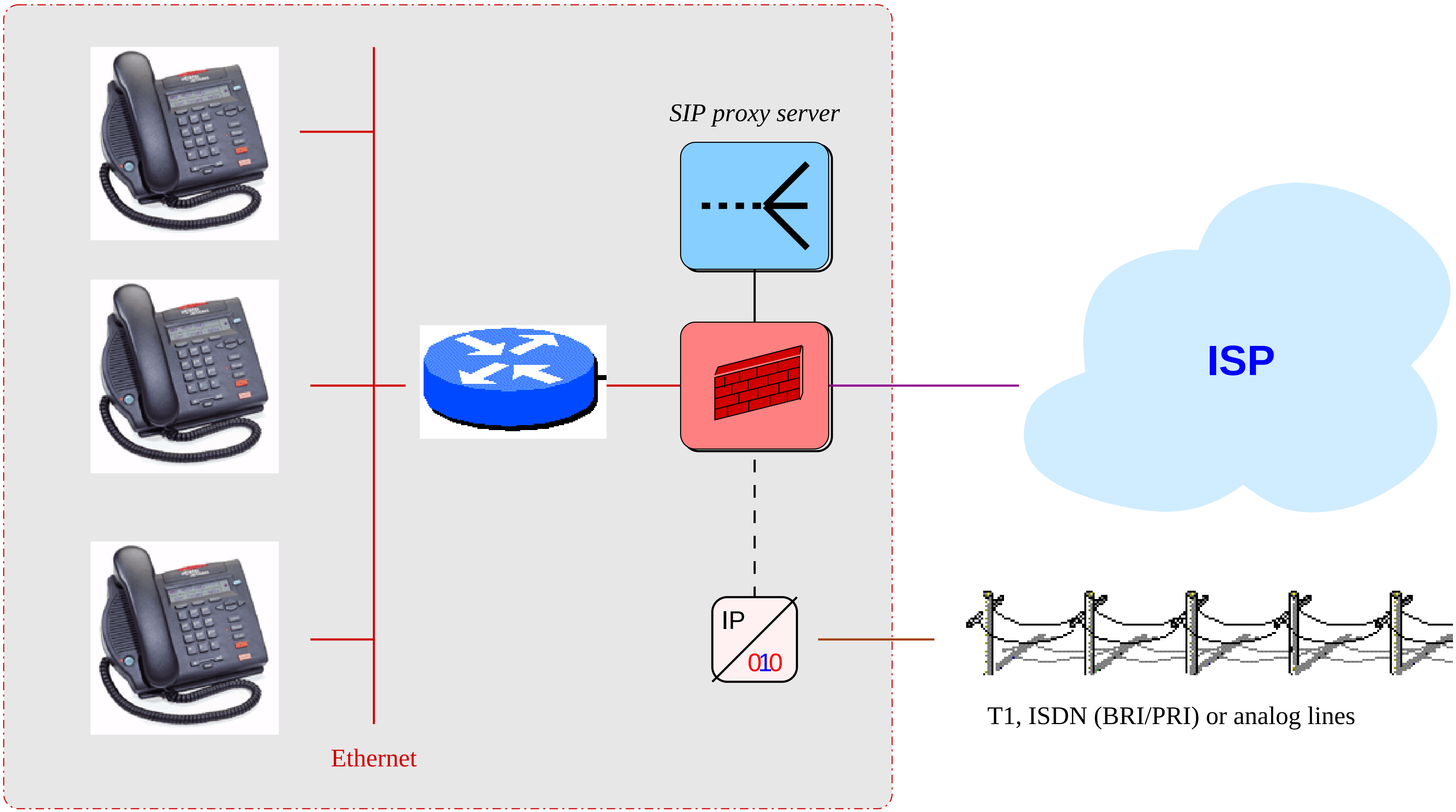
- new services: web-style programming, integration with web, email, presence
- transport efficiency due to compression: 64 kb/s $\searrow$ 8 kb/s
- avoid local access charges, but already reduced 7c $\searrow$ 2.4c
- lower switching costs: $175/circuit $\searrow$ $0.03
- cable plant integration (PBX, DSL, cable modem)
- improved network management
- bypass international telephony settlement

Internet Telephony Service Models

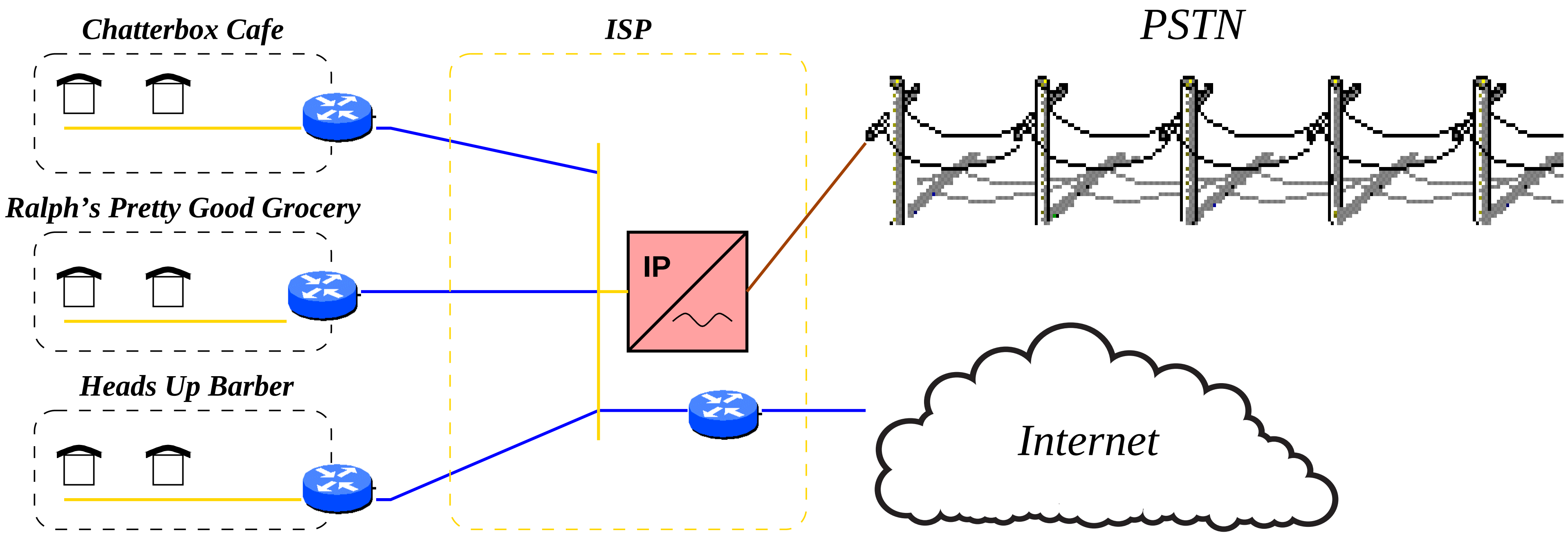
- Internet “PBX”
- Internet Centrex
- Internet Carrier

➠ same basic equipment, but size of gateway varies

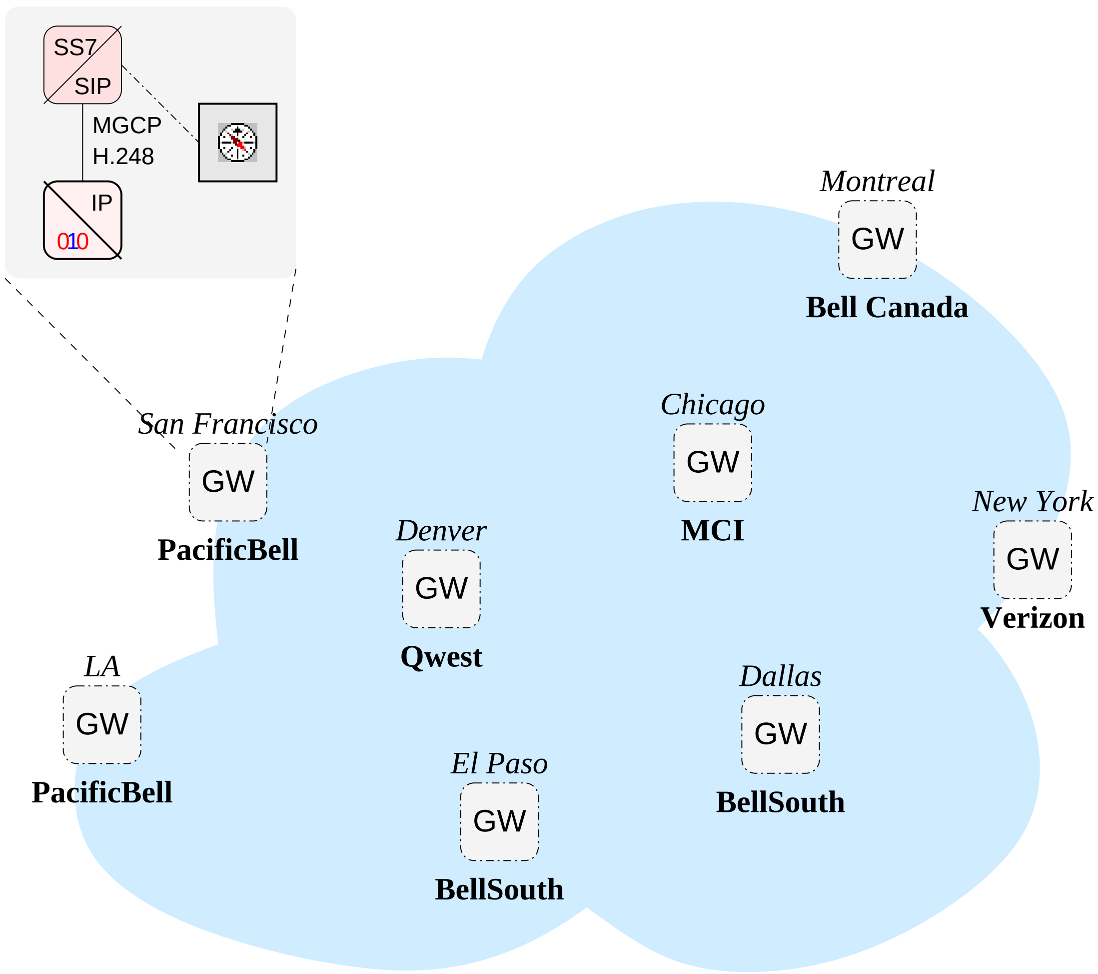
Internet PBX



IP centrex


Chatterbox Cafe
ISP
PSTN
Ralph's Pretty Good Grocery
IP
Heads Up Barber
Internet

IP carrier


SS7
SIP
MGCP
H.248
IP
010
Montreal
GW
Bell Canada
San Francisco
GW
PacificBell
Chicago
GW
MCI
New York
GW
Verizon
Denver
GW
Qwest
LA
GW
PacificBell
Dallas
GW
BellSouth
El Paso
GW
BellSouth

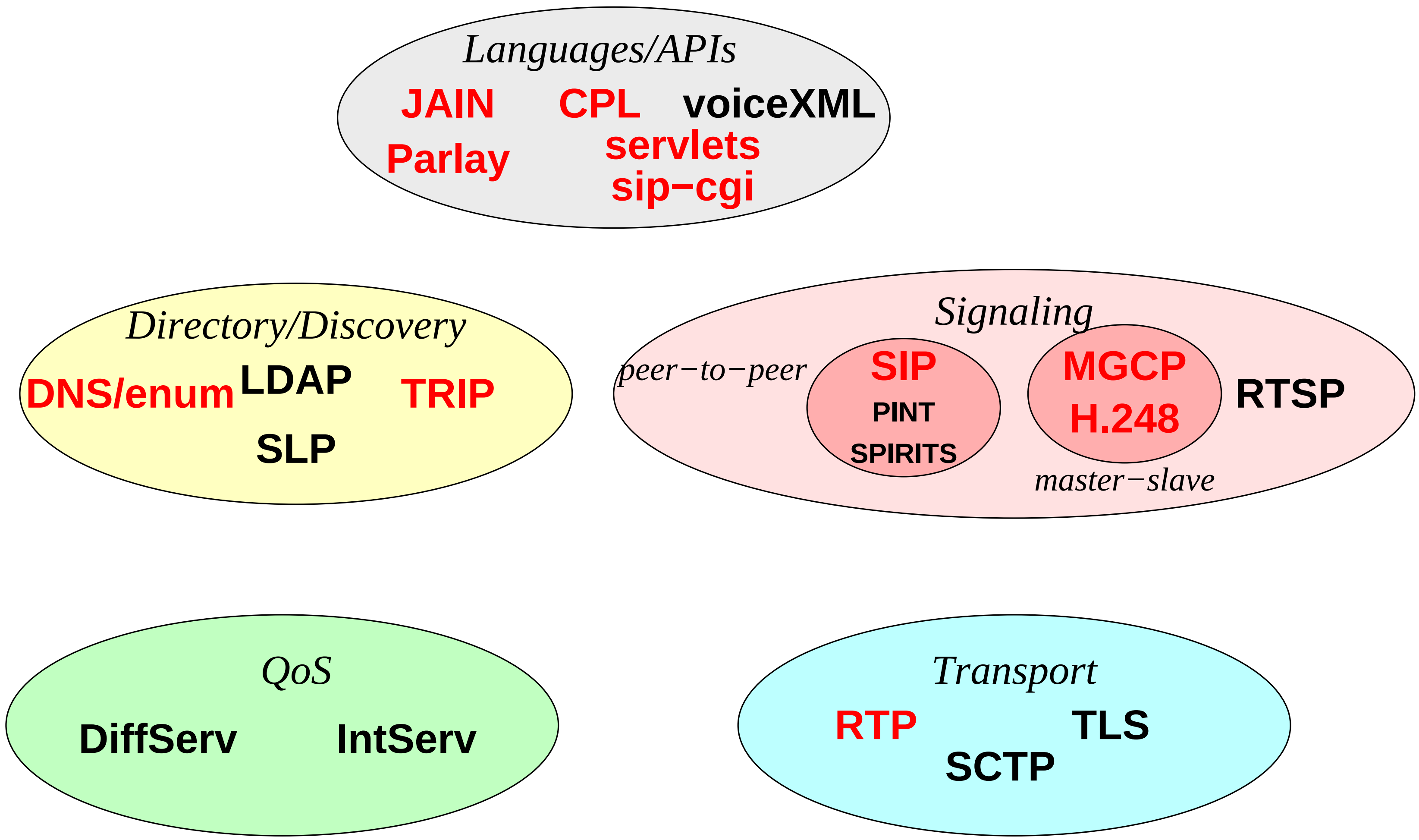
Technical problems

- quality of service (mostly admission control, not scheduling)
- reliability
- set-up and control
- creating new services, modeling old services
- mobility support

Third-Generation wireless

- goal: 144 kb/s moving, 384 kb/s stationary, 2 Mb/s indoors
- based on GSM or wideband CDMA
- implement IP(v6) in the hand set
- SIP as signaling system for voice calls in 3GPP
- in standardization now, deployment $\approx$ 2003

IETF VoIP protocol architecture



SIP overview

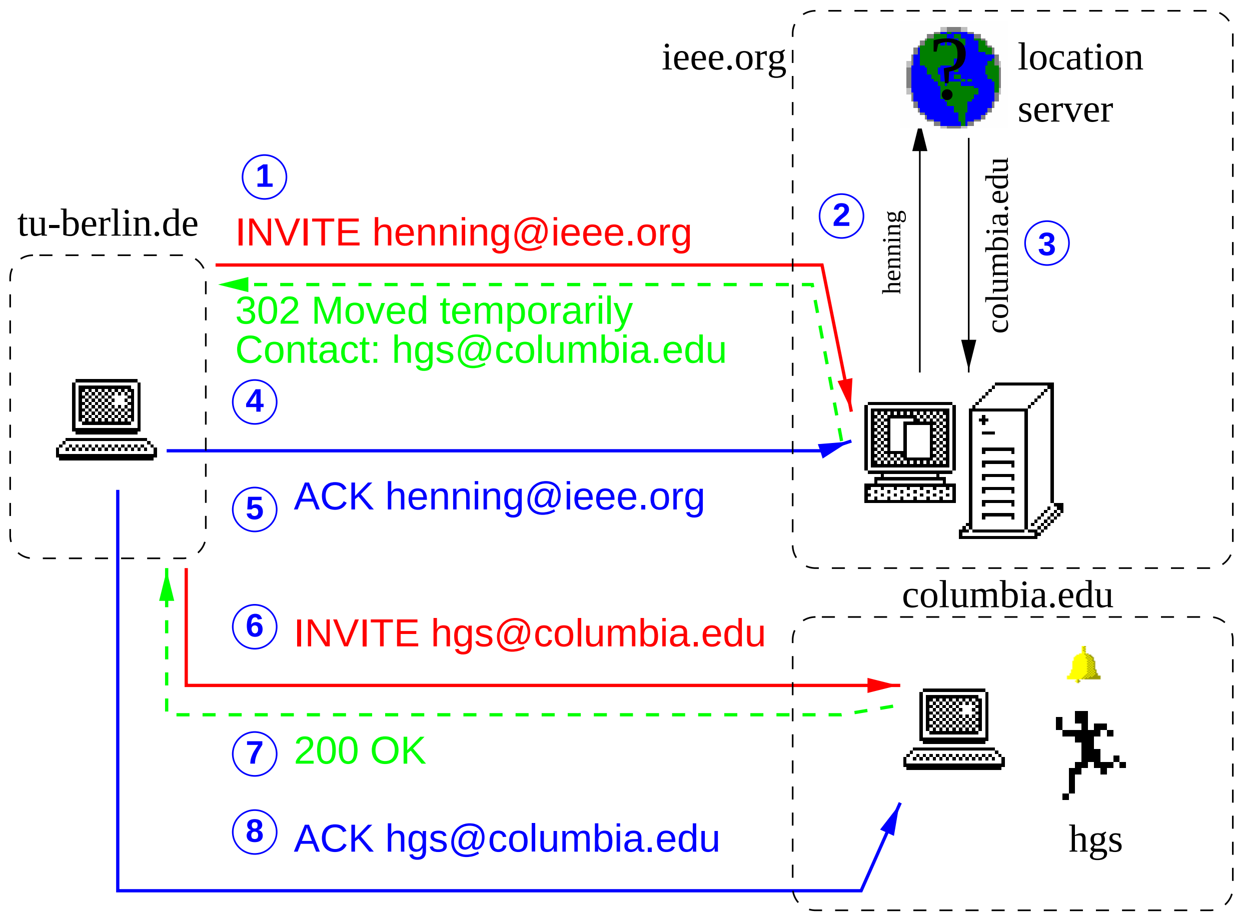
- protocol for establishing, modifying, tearing down (multimedia) *sessions*
- with registration to bind temporary locations to permanent names (e.g., 128.59.16.17 to `hgs@cs.columbia.edu`)
- IETF Proposed Standard since March 1999
- multimedia = audio, video, shared applications, text, ...
- also used 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)
- proposed for presence, instant messaging and event notification

SIP Components

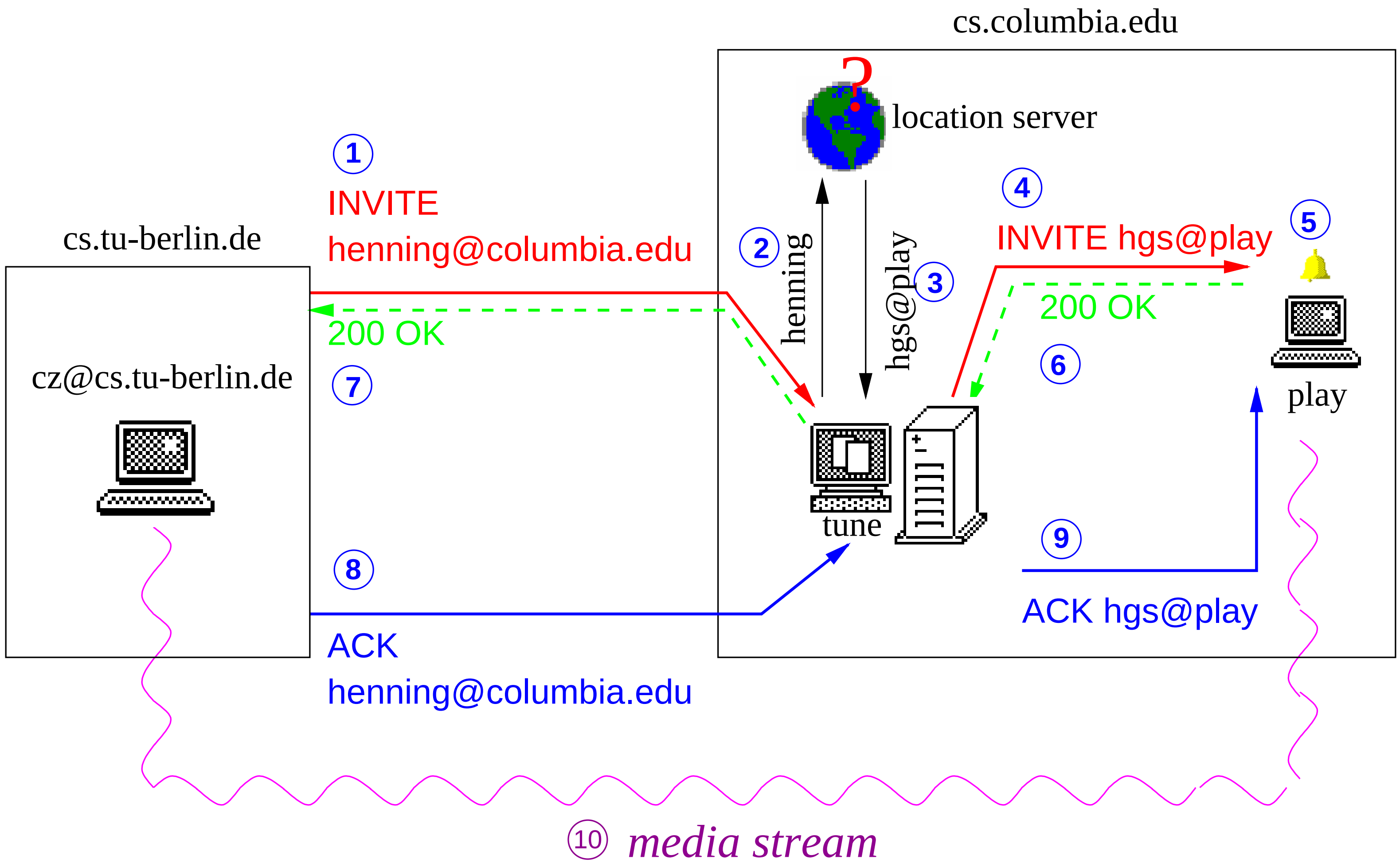
entity	does	examples
proxy server	forward calls	firewall controller, “call router”
redirect server		“application server”
user agent	end system	SIP phone, gateway, “softswitch”
registrar	location mgt.	mobility support

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

SIP example: redirection



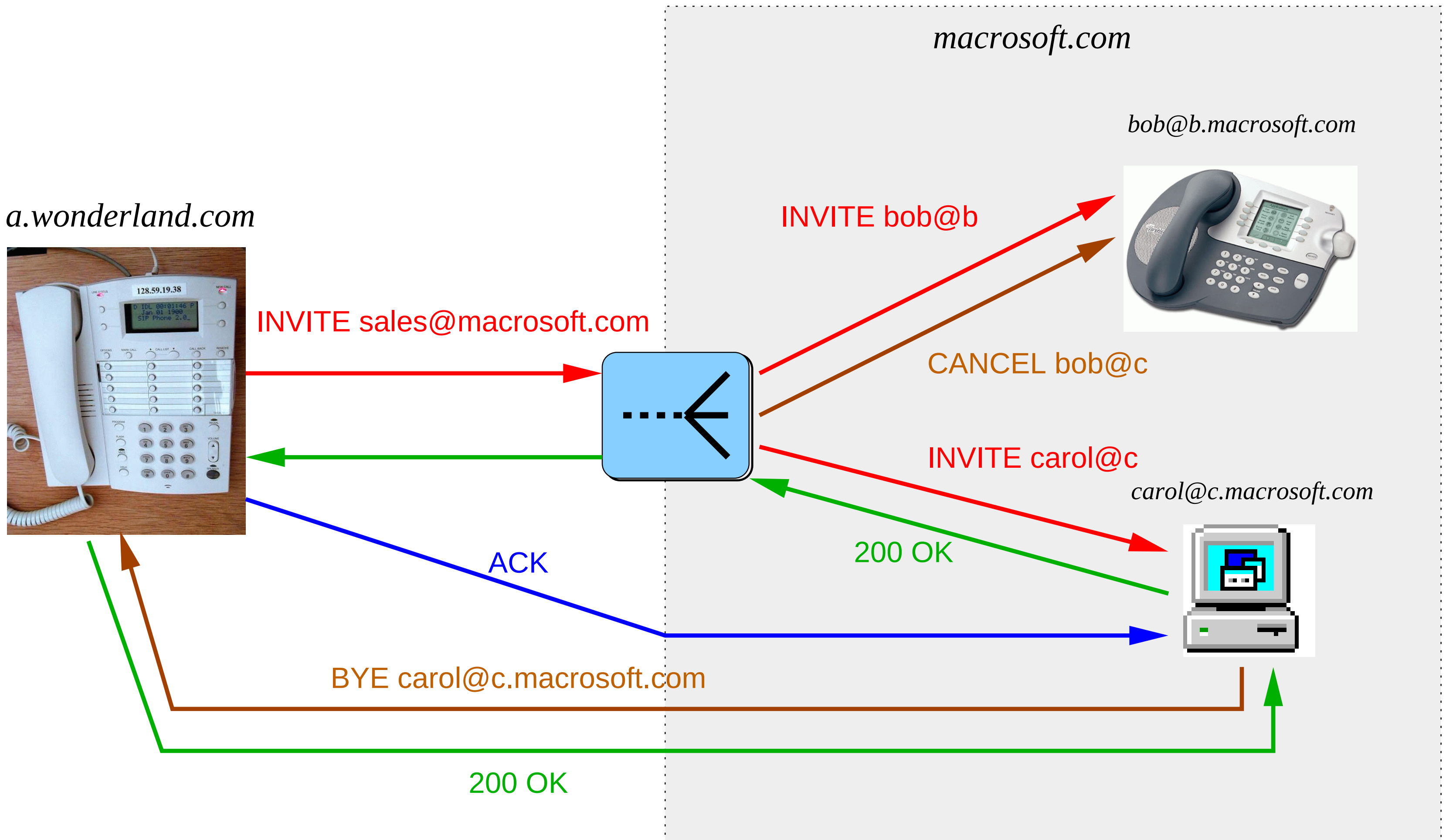
SIP example: proxying



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

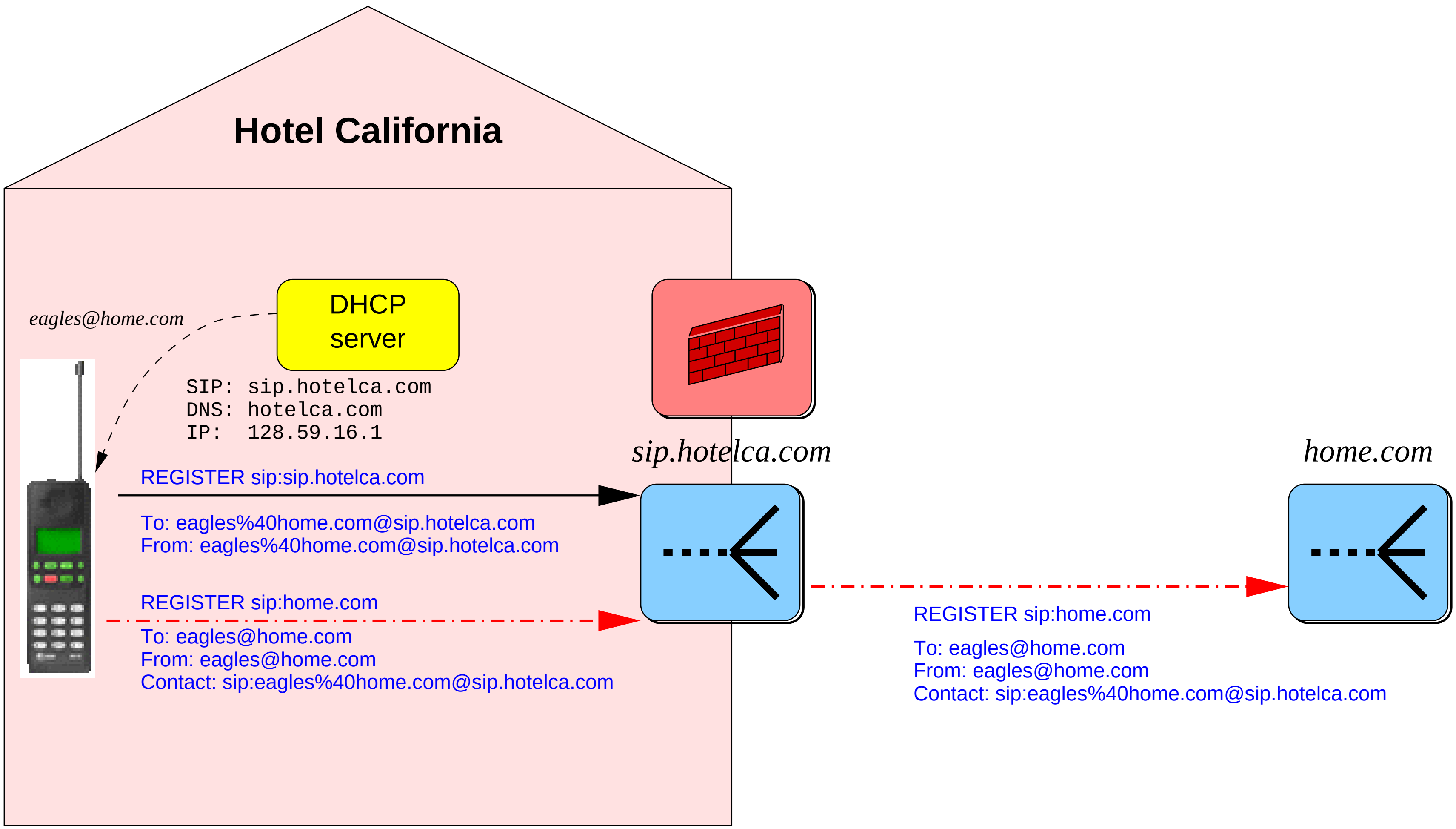
Service mobility: keep same services while mobile

Session mobility: move active session between terminals

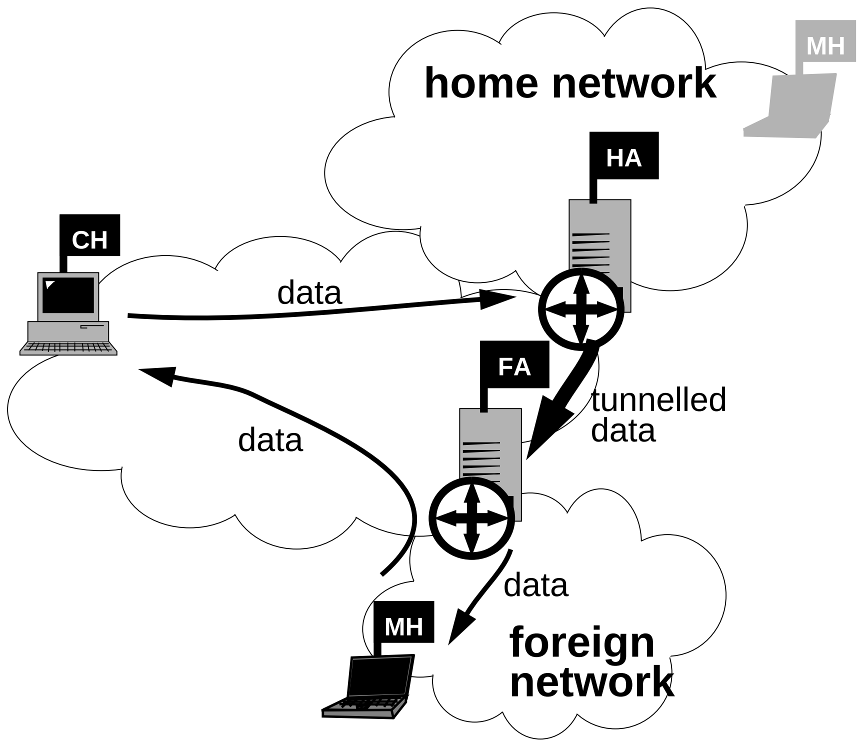
Simple mobility: roaming users

- 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

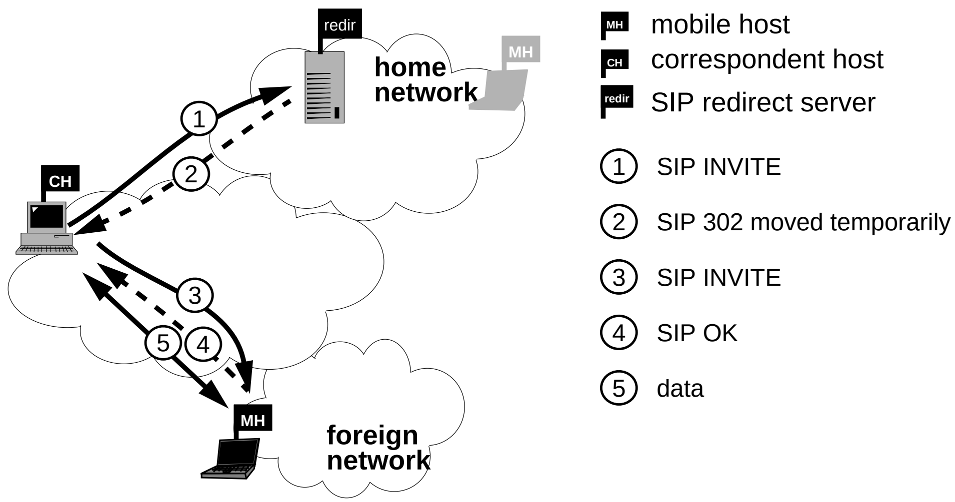
- domain of IEEE 802.11 (link layer), 3GPP (radio access network), mobile IP (network layer), . . .
- network-layer mobility has problems:
  - lack of deployment – home provider has no interest
  - need two addresses – home and visiting
  - dog-legged routing in IPv4
  - may not work with IP address 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

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

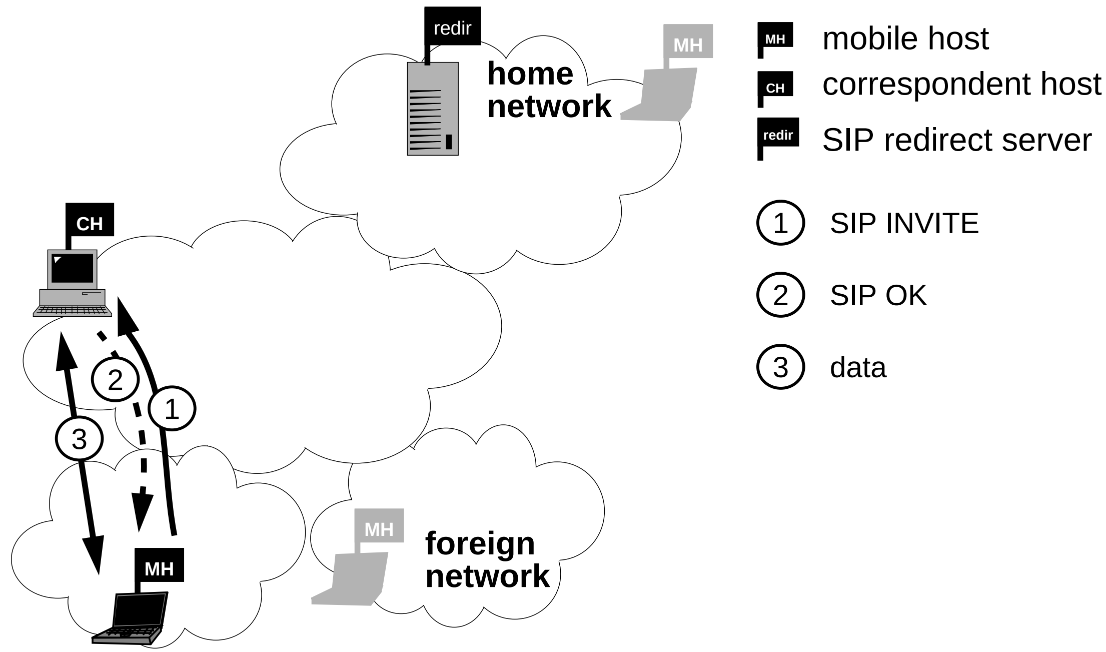
SIP terminal mobility: pre-call

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



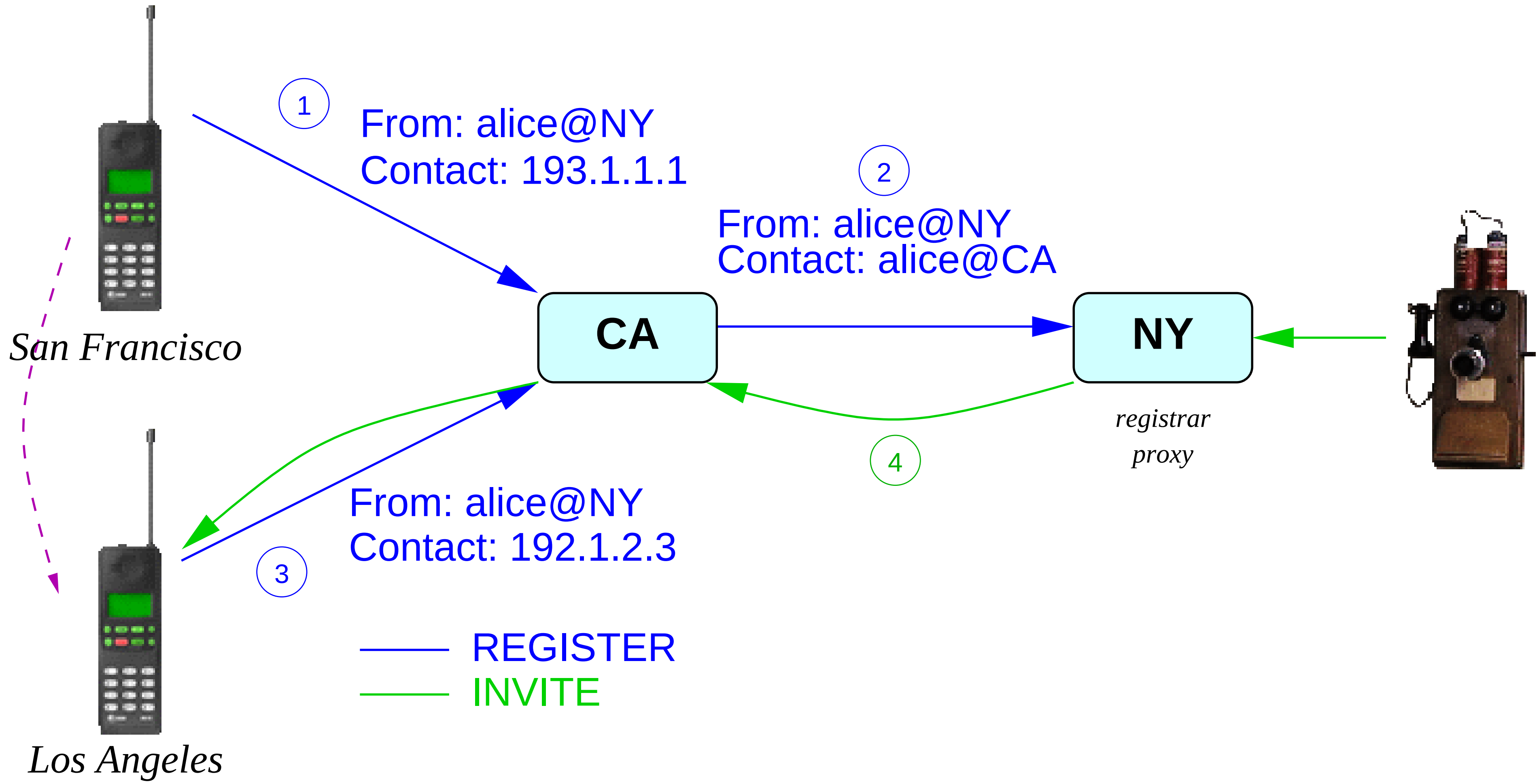
SIP terminal mobility: mid-call

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

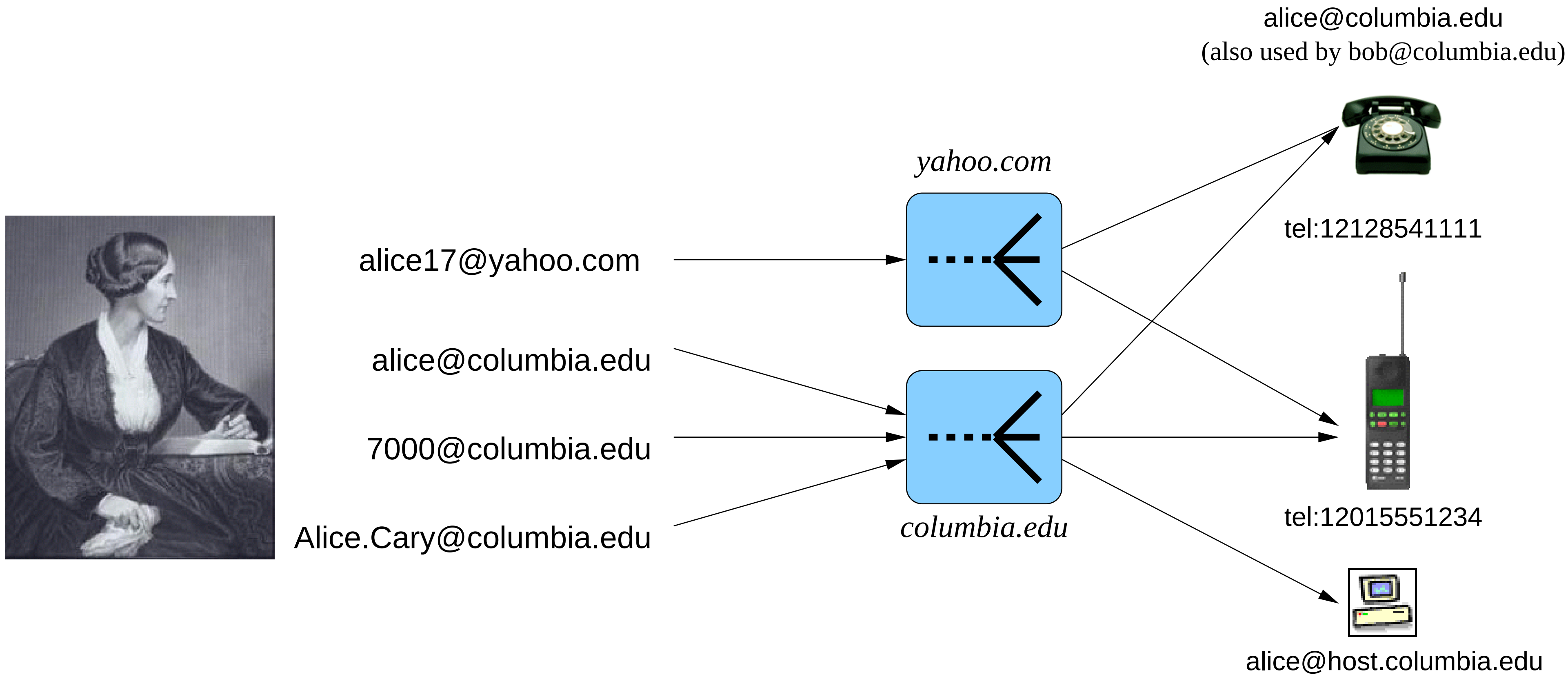


SIP terminal mobility: multi-stage registration

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>
```

Users creating services

- web: HTML $\longrightarrow$ ASP $\longrightarrow$ cgi, servlets
- Call Processing Language: similar role as HTML
  - XML DTD flow graph for inbound & outbound calls
  - generated by hand or graphical call flow editor
  - can be imported back into editor
  - could be generated by JavaScript, ASP
  - safe: execution model, bounded timing (no loops)
  - mobile code: uploaded via REGISTER from terminal to call-handling proxy
  - VoiceXML related, but for interactive voice response

Example Call Processing Language script

```
<?xml version="1.0" ?>
<!DOCTYPE cpl SYSTEM "cpl.dtd">

<cpl>
 <subaction id="voicemail">
  <location url=
    "sip:kns10@vm.cs.columbia.edu">
   <redirect />
  </location>
 </subaction>

 <incoming>
  <address-switch field="origin"
     subfield="host">
   <address
     subdomain-of="cs.columbia.edu">
    <location url=
      "sip:kns10@cbb.cs.columbia.edu">
     <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>
```

Programmable Internet Telephony

	APIs	servlets	sip-cgi	CPL
Language-independent	no	Java only	yes	own
Secure	no	mostly	no, but can be	yes
End user service creation	no	yes	power users	yes
GUI tools w/portability	no	no	no	yes
Call creation	yes	no	no	no
Multimedia	some	yes	yes	yes

Example: integration with iCal $\longrightarrow$ automatically export personal calendar to call handling

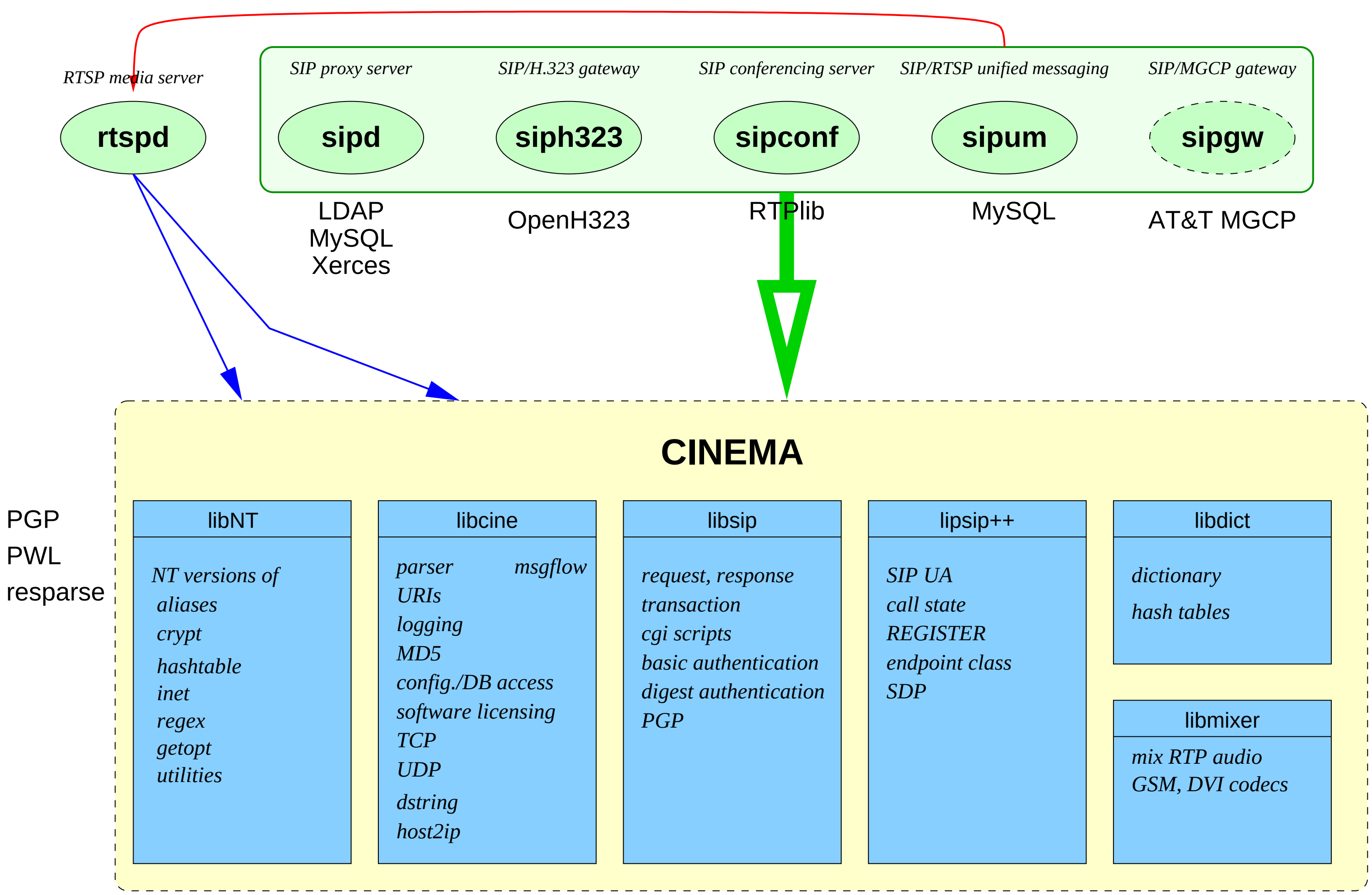
Service mobility – call handling

- need uniform basic service description model $\longrightarrow$ Call Processing Language (CPL)
- 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

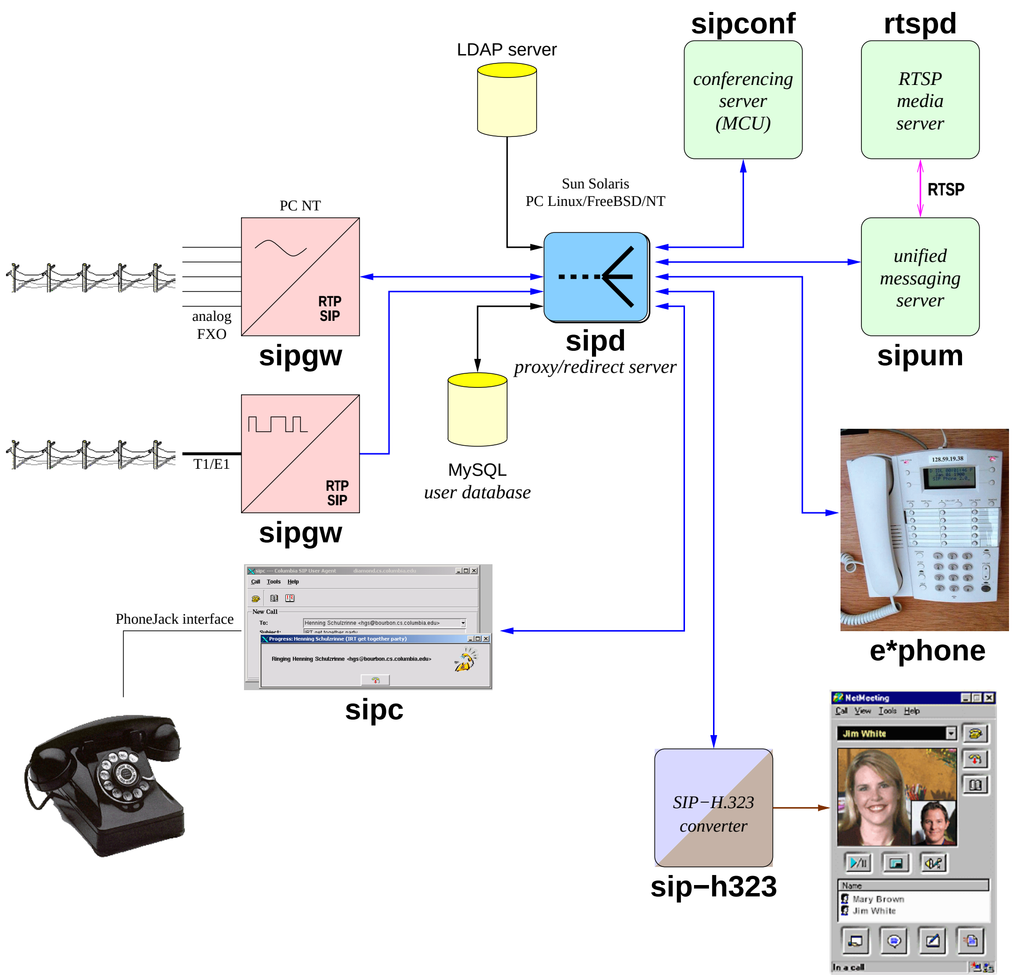
SIP and mobility: issues

- 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

CINEMA: Columbia Internet Extensible Multimedia Architecture



Columbia Internet telephony components



Scaling or how many calls can a SIP switch switch?

- 750,000 to 2.5 million busy hour call attempts for large class-5 switches = 3.6 ms/call
- AT&T: 280 million calls a day = 0.3 ms/call
- Yahoo: 780 million page views/day
- AOL: 110 million emails/day
- AOL: 500 million IM/day
- web server: about 1,500 to 3,000 static requests/second

Invisible Internet telephony

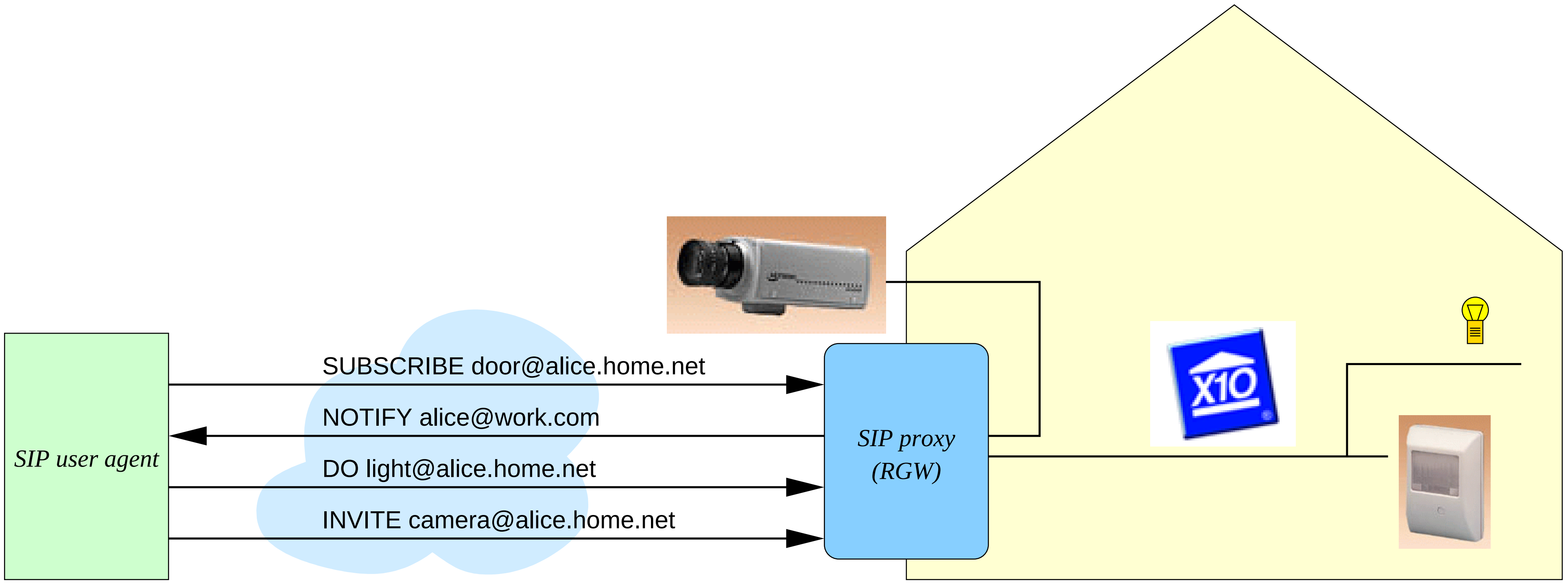
VoIP technology will appear in . . .

- Internet appliances
- home security cameras, web cams
- 3G mobile terminals
- fire alarms
- chat/IM tools
- interactive multiplayer games

Commonalities between signaling and events

- presence is just a special case of events: “Alice just logged in” $\approx$ “temperature in boiler exceeds 300° F”
- need to *locate* mobile end points
- may need to find several different destinations (“forking”)
- same addressing for users
- presence often precursor to calls
- likely to be found in same devices
- events already in VoIP: message alert, call events

Events: SIP for Appliances



Conclusion

- evolution from analog $\xrightarrow{1960s}$ circuit-switched digital $\xrightarrow{2000's}$ packet-based
- uniform solution for wired and wireless multimedia terminals
- 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