Internet telephony: What's Out There & How does it work?

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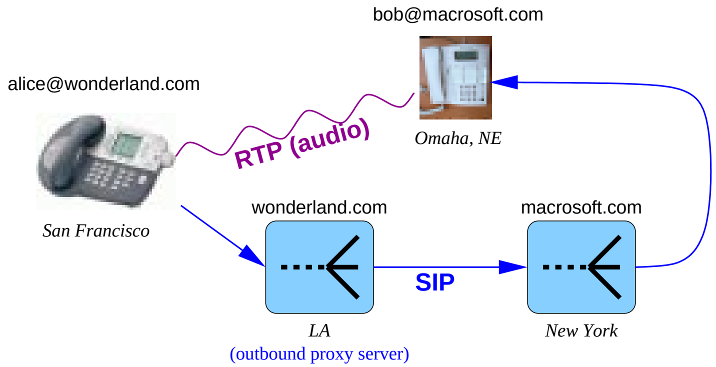
Overview

- VoIP is not just voice on packets
- where are we technically – protocol pieces
- emergency services
- mobility is more than just wireless terminals
- summary of open issues

VoIP is not just voice on packets

- strict separation of transport and services



- not just phone functionality with 12 buttons:
  - hybrids: PDA + phone, MP3 player + phone, . . .
  - richer interfaces: touch screens
- integrated with dominant Internet protocols: web, email, chat $\longrightarrow$ Session Initiation Protocol (SIP)
- treat PSTN as IP device with a very long microphone/speaker cable

Services

- old service model: enumerated services (“call forwarding”, “*69”, ...)
- new: customized call service logic
  - “only allow calls at dinner time from those who have called me before”
  - “if meeting scheduled in calendar, do not ring phone”
  - “try these three addresses in sequence”
- some services disappear: distinctive ringing, caller id, ...
- new services that integrate Internet services:
  - call voice mail $\longrightarrow$ get web page with messages
  - IVR $\longrightarrow$ web-based call routing
  - forward calls to email
  - click on link in email or web page to make call

Naming and routing in (SIP) networks

- long-term: email address = phone “number” (e.g., Telia, dynamicsoft)
- route is defined by DNS domains
- unlimited number of translations between names
- generalization of 8xx, 900 services and call forwarding

What services can carriers provide?

- accept *incoming* calls for existing numbers, terminate on IP
- IP centrex = SIP server outsourcing
- IP voice/video conference bridges
- anonymizer services
- identity certification services
- standards-based unified messaging
- emergency calling

Protocol pieces

- signaling
- device control
- number translation and gateway routing
- transport: RTP
- settlements: OSP
- related protocols: RTSP, VPIM for voice mail

Protocol pieces: signaling

Signaling + services: SIP or H.323

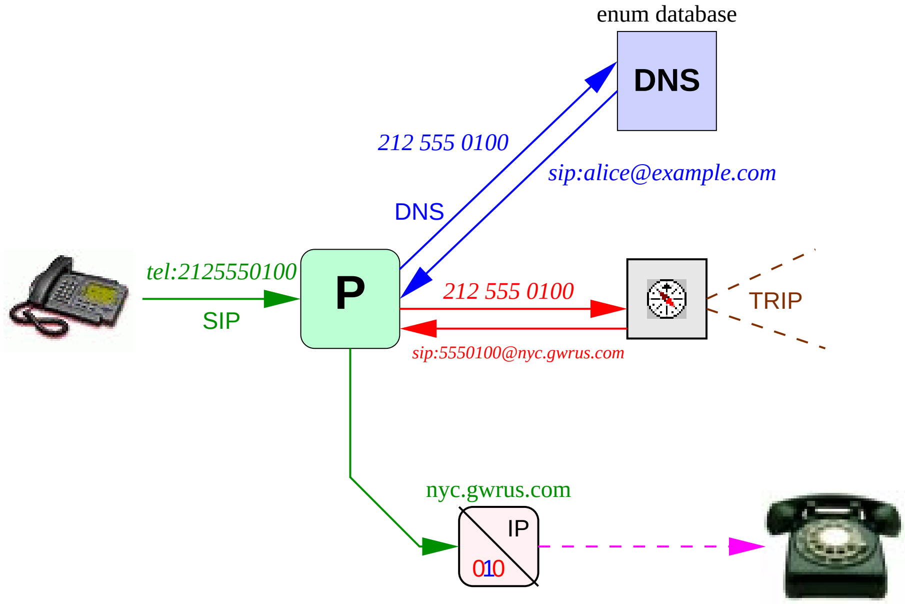
- many devices will support both
- translation is work in progress

Device control: local separation of DSP and signaling

- MGCP $\longrightarrow$ Megaco (= H.248)
- but: MGCP & Megaco are dead ends for all but black phones
- regulatory issues: forces single controller

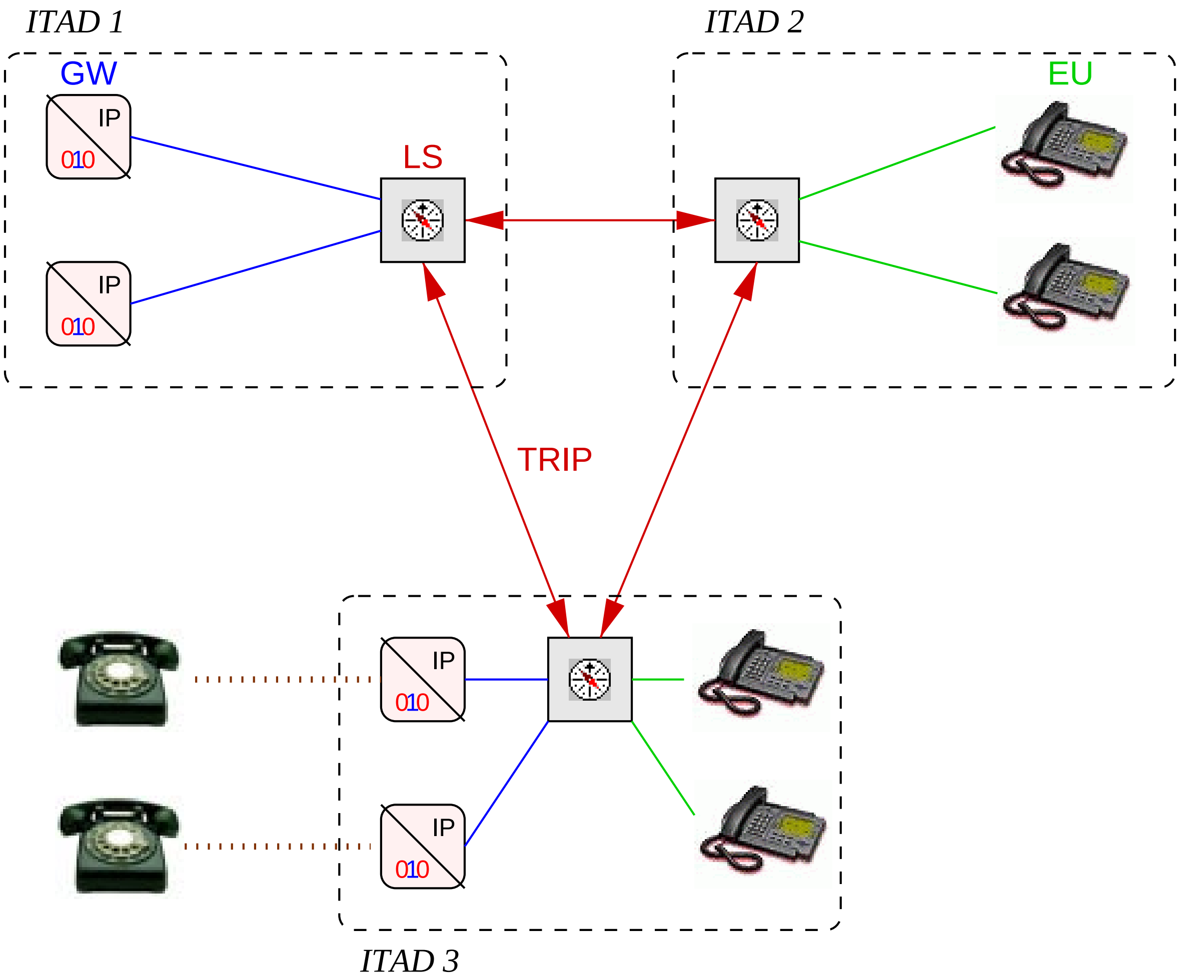
Protocol pieces: mapping numbers

enum = DNS entry for E.164 ⟶ SIP URL (e.g.)



- who owns and runs the `.e164.arpa` database?
- who gets to install entries in it?
- cheap mechanism for LNP…

Protocol pieces: finding gateways



Emergency services

- 911 can't just be a telephone service
- chat/IM, email, phone may blend into one application
- several problems:
  - needs to work internationally
  - intercept at outbound proxy?
  - designated address, e.g., "911@anydomain.com"?
  - authoritative identifying users
  - location of users

Mobility in an IP environment

Terminal mobility: terminal moves between subnets

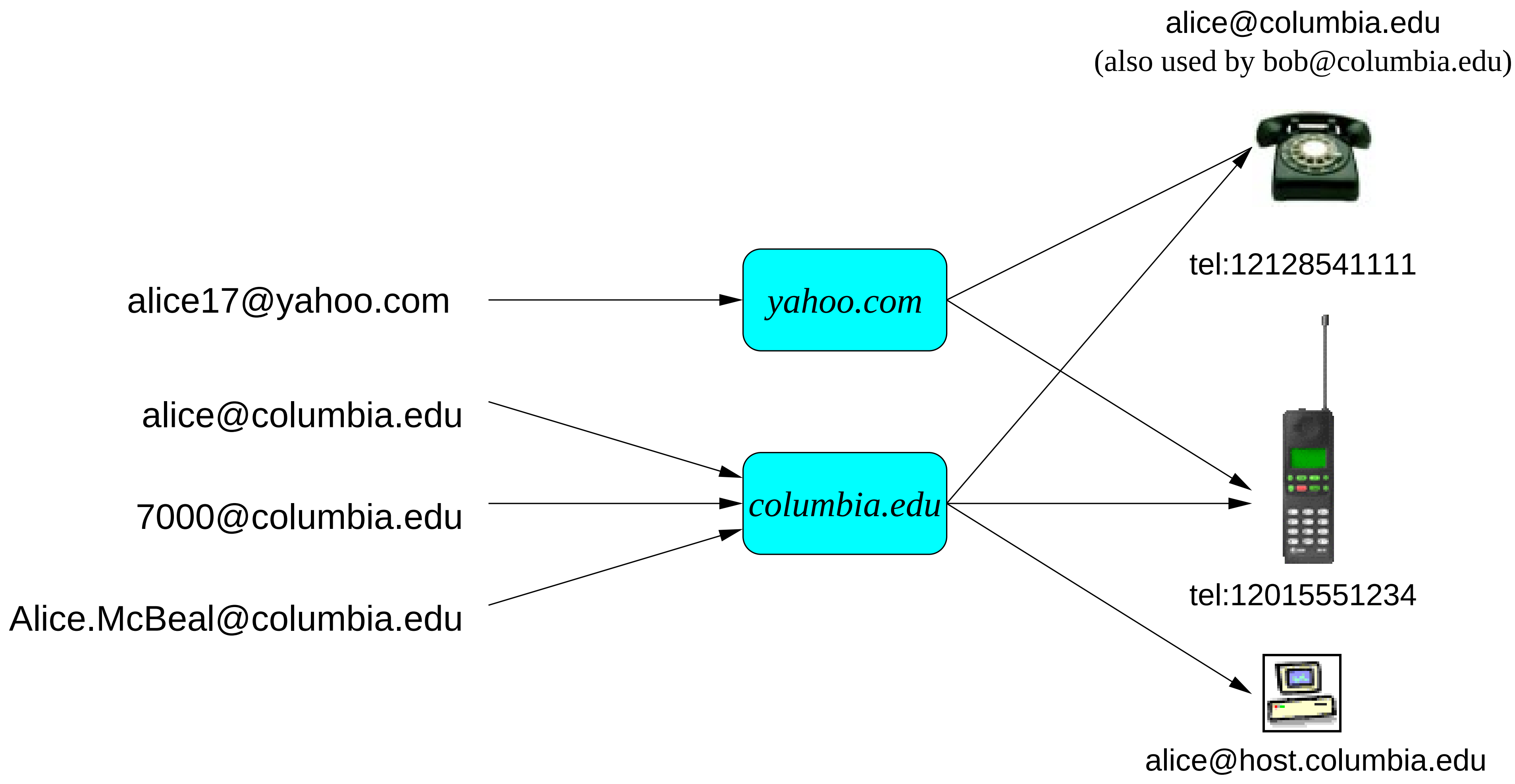
Personal mobility: different terminals, same address

Service mobility: keep same services while mobile

Terminal mobility

- domain of IEEE 802.11, 3GPP, mobile IP, ...
- main problems:
  - handover performance
  - handover failure due to lack of resources in new network
  - authentication of redirection

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

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

Service mobility

- during registration, 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, . . .

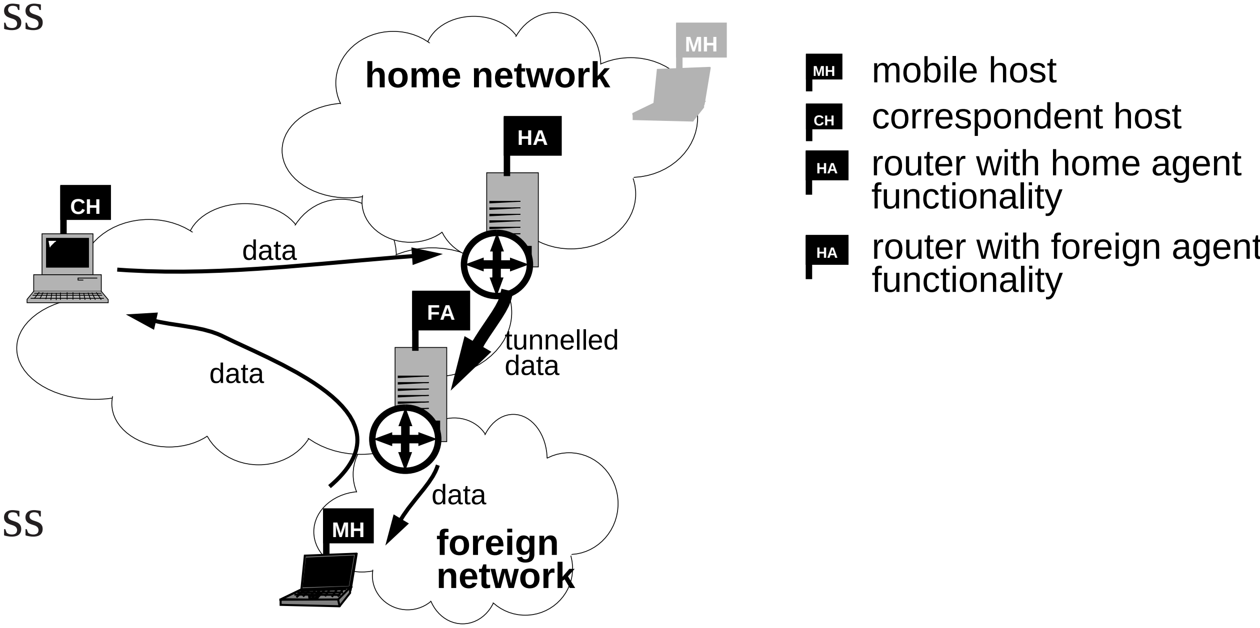
One device, but changing set of owners!

Service mobility – call handling

- 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: registration updates information, but device needs to know that it has multiple identities
- merging of call logs

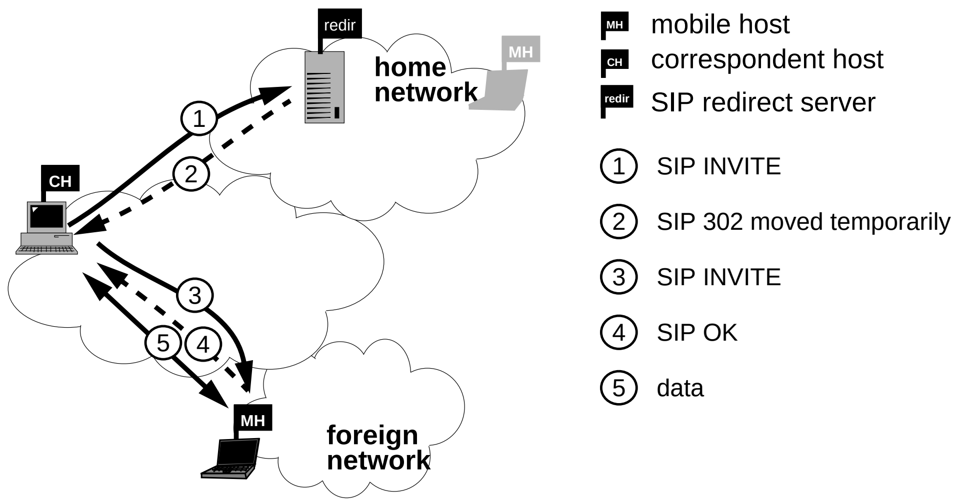
Terminal mobility – details

- move to new network ➠ IP address changes (DHCP)
- mobile IP hides address changes
- but: little deployment
- encapsulation overhead
- dog-legged routing
- may not work with IP address filtering

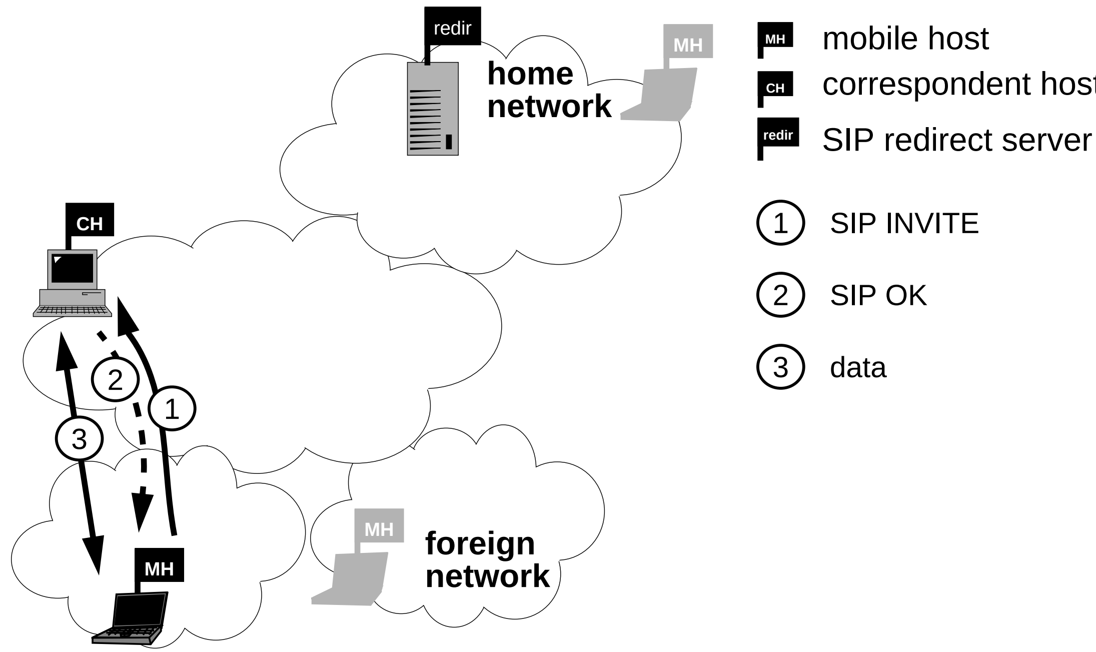


SIP terminal mobility: pre-call

- MH acquires IP address via DHCP
- optional: MH finds SIP server via multicast registration
- 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

Open Issues

- quality of service: from easy (voice diff-serv priority) to hard (inter-provider settlements for per-flow reservations)
- home networks: NATs, firewalls, asymmetric bandwidth, ...
- network reliability and protection
- emergency services
- service creation environments

Conclusion

- basic VoIP architecture pieces in place
- H.323 products (gateways, software, some phones) available
- SIP products emerging:
- operational and reliability issues
- mobility is more than just wireless handsets
- terminal, personal and service mobility
- SIP enables all three, but likely to be hybrid solutions

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

VoIP: http://www.cs.columbia.edu/˜hgs/internet