CSc 461/561
Multimedia Systems
SIP

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What we have so far …

- Network: IP and IP multicast
  - move packets from one host to other hosts
- Transport: RTP and RTCP
  - end2end media transport and control feedback
- Application: what we can do now
  - voice over IP (VoIP): yes!
  - IP telephony: not yet
    - what's missing?: *signaling*! (e.g., how to setup calls)

What we need: signaling support

- E.g., in telephone networks
  - in addition to speech path: voice
  - signaling path: out-of-band, packetized SS7
- E.g., in cellular systems (+mobility support)
  - handoff between base stations
  - roaming across service providers
- On the Internet
  - SIP: session initiation protocol

SIP: quick fact

- SIP is not limited to IP telephony
  - SIP is the Internet's signaling protocol
- SIP offers
  - setup calls (or sessions)
  - make changes to ongoing calls
  - terminate calls, and more (e.g., presence)
- SIP does not offer
  - media transport, QoS support, server control,etc

SIP design guidelines

- Client-server model
  - request-reply transaction
- HTTP+MIME-like format
- Common headers in plain text
  - request/response line (e.g., INVITE a@b.com SIP/2.0)
  - message headers (identification, routing, etc)
  - message body
    - e.g., session description (SDP)

SIP requests

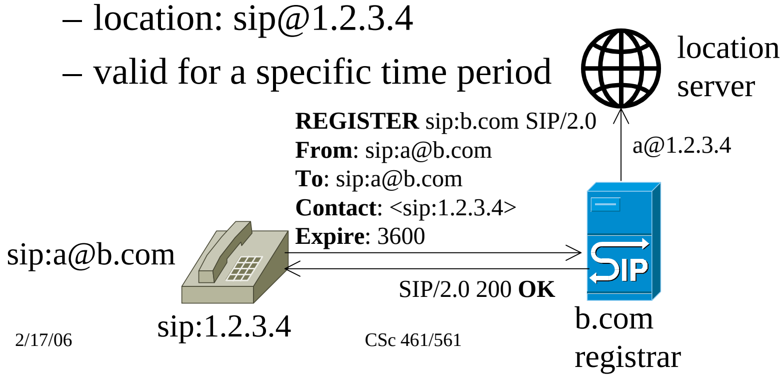
- REGISTER: register user agents
- INVITE: initiate calls
- ACK: confirm responses
- BYE: terminate or transfer calls
- Other methods:
  - CANCEL, OPTIONS, INFO, COMET, PRACK, SUBSCRIBE, NOTIFY, REFER
- SIP response: HTTP-like (e.g., SIP/2.0 200 OK)

SIP entities

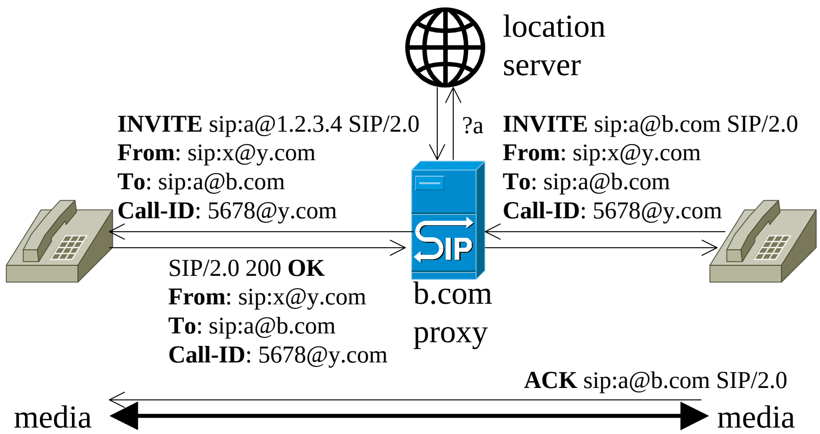
- User agent (UA)
  – UA client (UAC) and UA server (UAS)
- Proxy server
  – relay calls; chaining; forking
- Redirect server
  – redirect calls
- Registrar server
  – UA registration (UA whereabouts)

SIP: register

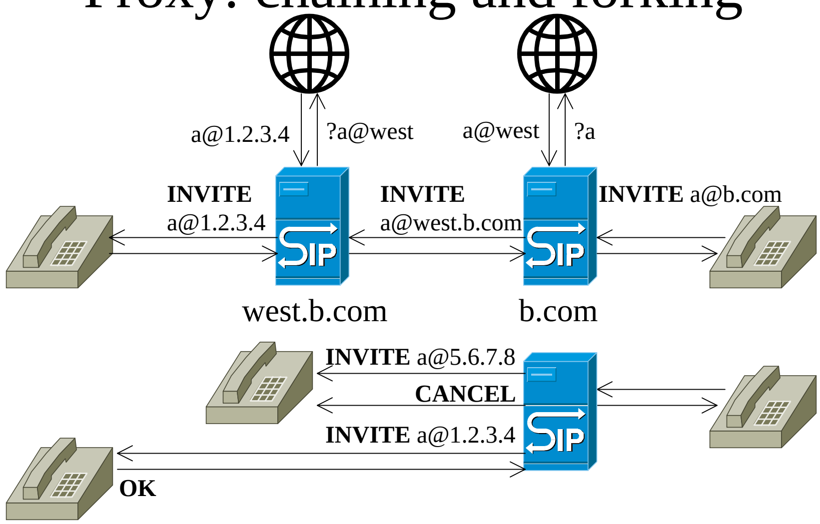
- Create UA name/location binding
  – name (URI): e.g., sip:a@b.com
  – location: sip@1.2.3.4
  – valid for a specific time period



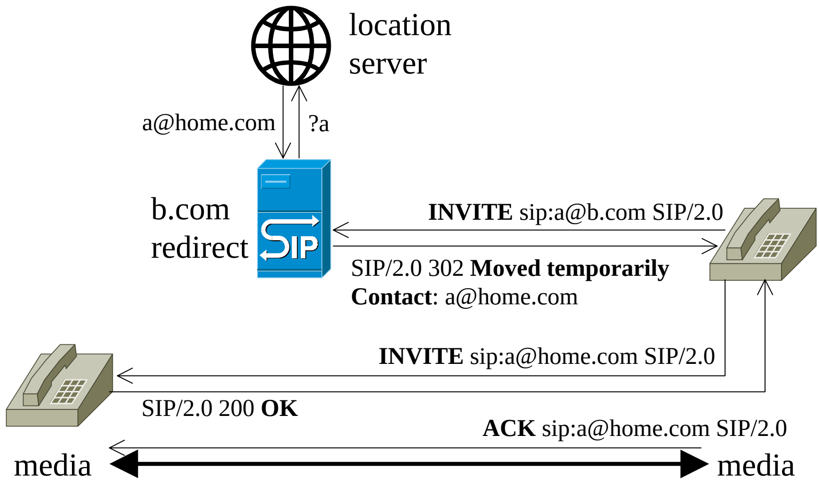
SIP: proxy mode



Proxy: chaining and forking



SIP: redirect mode



SIP headers

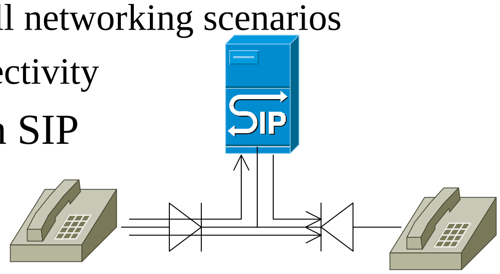
- From/To: caller/callee URI
- Via: proxy routing
  - request/reply: use the same route
  - may be different from the media route
- Call-ID: identification
  - unique at caller
- Content-Type/Length: payload info
  - e.g., Content-Type: application/sdp

SDP

- SDP: session description protocol
  - media type, network/transport parameters
  - e.g., media: media, port, protocol, format_list
    - m=audio 2000/2 RTP/AVP 0 98
    - a=rtpmap:0 PCMU/8000
  - connection: net_type, add_type, address
    - c=IN IP4 1.2.3.4/127/3
- Ref: http://www.ietf.org/rfc/rfc2327.txt

SIP and NAT

- NAT: network address translation
  - was to deal with IPv4 address shortage
  - now pervasive in all networking scenarios
  - “directional” connectivity
- Communications in SIP
  - UA<=>server
  - UA<=>UA



- Problem: when UAC and UAS behind NAT

IP telephony examples

- Vonage: proprietary VoIP infrastructure
  - good PSTN interworking
  - SIP compatible
    - phone adapter: SIP UA and more
- Skype: without specialized infrastructure
  - better NAT traversal capability
    - with the help of other users; voice encryption
  - proprietary protocols

This lecture

- SIP
  - SIP entities
    - UA, proxy/redirect/registrar server
  - SIP requests
    - register, invite
  - SDP
- Explore further
  - http://www.cs.columbia.edu/~hgs/sip/

Next lecture

- Multimedia QoS