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

<table>
<thead>
<tr>
<th>Entity</th>
<th>Does</th>
<th>Examples</th>
</tr>
</thead>
<tbody>
<tr>
<td>Proxy server</td>
<td>Forward calls</td>
<td>Firewall controller, “call router”</td>
</tr>
<tr>
<td>Redirect server</td>
<td>“Application server”</td>
<td></td>
</tr>
<tr>
<td>User agent</td>
<td>End system</td>
<td>SIP phone, gateway, “softswitch”</td>
</tr>
<tr>
<td>Registrar</td>
<td>Location mgt.</td>
<td>Mobility support</td>
</tr>
</tbody>
</table>

Roles are changeable, on a request-by-request basis
SIP Example: Redirection

1. INVITE henning@ieee.org
2. 302 Moved temporarily
   Contact: hgs@columbia.edu
3. ACK hgs@columbia.edu
4. INVITE hgs@columbia.edu
5. 200 OK
6. ACK hgs@columbia.edu
SIP Example: Proxying

1. INVITE henning@columbia.edu
2. henning
3. hgs@play
4. INVITE hgs@play
5. hgs@play
6. 200 OK
7. 200 OK
8. ACK henning@columbia.edu
9. ACK hgs@play
10. media stream
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

INVITE sales@macrosoft.com
carol@c.macrosoft.com

INVITE bob@b

CANCEL bob@c

INVITE carol@c
carol@c.macrosoft.com

ACK

BYE carol@c.macrosoft.com

200 OK

macrosoft.com

bob@b.macrosoft.com

a.wonderland.com
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

Hotel California

DHCP server

REGISTER sip:sip.hotelca.com
To: eagles%40home.com@sip.hotelca.com
From: eagles%40home.com@sip.hotelca.com

REGISTER sip:home.com
To: eagles%40home.com@sip.hotelca.com
From: eagles@home.com
Contact: sip:eagles%40home.com@sip.hotelca.com

IP: 128.59.16.1
SIP: sip.hotelca.com
DNS: hotelca.com

home.com

DHCP

Contact: sip:eagles%40home.com@sip.hotelca.com
REGISTER sip:home.com
To: eagles@home.com
From: eagles@home.com
Contact: sip:eagles%40home.com@sip.hotelca.com

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Terminal mobility – mobile IP

- mobile host
- correspondent host
- router with home agent functionality
- router with foreign agent functionality

- home network
- foreign network
- tunnelled data
- data
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 $\rightarrow$ 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 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)
### 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

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SIP terminal mobility: multi-stage registration

Don’t want to bother home registrar with each move

1. From: alice@NY  
   Contact: 193.1.1.1

2. From: alice@NY  
   Contact: alice@CA

3. From: alice@NY  
   Contact: 192.1.2.3

4. registrar proxy

Los Angeles  
San Francisco  
NY  
CA
Personal mobility

alice17@yahoo.com
alice@columbia.edu
7000@columbia.edu
Alice.Cary@columbia.edu

tel:12128541111
tel:12015551234

yahoo.com
columbia.edu

alice@host.columbia.edu
(also used by bob@columbia.edu)

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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

→ 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 → 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

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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