SIP for Mobility

Henning Schulzrinne
Dept. of Computer Science
Columbia University
New York, New York
(sip:)schulzrinne@cs.columbia.edu

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Overview

- next-generation wireless systems
- mobility modes and application-layer mobility
- mobile code
- signaling, inter-domain events and messaging
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 ≈ 2003
## 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></td>
<td>“application server”</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. columbia.edu
4. location server
5. ACK henning@ieee.org
6. INVITE hgs@columbia.edu
7. 200 OK
8. ACK hgs@columbia.edu

IEEE.org

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SIP example: proxying

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

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SIP forking proxies

INVITE sales@macrosoft.com

ACK

INVITE carol@c.macrosoft.com

CANCEL bob@c

200 OK

BYE carol@c.macrosoft.com

200 OK

macrosoft.com

bob@b.macrosoft.com

carol@c.macrosoft.com

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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

eagles@home.com

DHCP server

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

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

sip.hotelca.com

home.com
Terminal mobility – mobile IP

- **MH**: mobile host
- **CH**: correspondent host
- **HA**: router with home agent functionality
- **FA**: router with foreign agent functionality

The diagram illustrates the flow of data through the home and foreign networks. Mobile host (MH) communicates with correspondent host (CH) via routes involving home agent (HA) and foreign agent (FA). Data is tunnelled between these nodes to ensure seamless communication.
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

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

REGISTER
INVITE
Personal mobility

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

yahoo.com

columbia.edu

alice@host.columbia.edu
tel:12015551234
tel:12128541111

(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 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
Conclusion

- 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