SIP-based Mobility Management Scheme for Wireless Internet

Telcordia Technologies
Motivation

• Mobility and wireless are rapidly becoming the rule rather than exception.

• SIP is gaining acceptance as the signaling protocol for multimedia conferences and Internet telephony.

It is essential to support wireless mobile users in a SIP signaling and control environment.

• Current Wireless Standard efforts using SIP
  – IETF
  – 3GPP (Third Generation Partnership Project)
  – MWIF (Mobile Wireless Internet Forum)
  – 3GPP2
Outline

- Objective
- Mobility Management Requirement
- Existing mobility solutions
- SIP based mobility
- Performance
- Wireless Internet Testbed Implementation
- Issues and Summary
Why is Mobility Management Difficult?

- Goals of mobility support in Internet:
  - allow a mobile device to move between different subnets and domains
  - preserve an ongoing session between the mobile device and its counterpart alive while moving
  - Ability to provide same service irrespective of network attachment

- Several protocols and mechanisms have been developed, broadly divided into:
  - Network Layer Mobility
    - MIP, CIP, HAWAII, TeleMIP, MIP-LR, MIPv6
  - Application Layer Mobility
    - SIP based Mobility Management Scheme
Multimedia Protocol Stack

Signaling
- H.323
- SIP
- RTSP
- MGCP
- LDAP

Quality of Service
- RTP
- RTCP
- RSVP
- DNS
- IGMP
- ICMP

Media Transport
- media encaps (H.26x, MPEGx)

Application Daemon

Network
- TCP
- UDP
- IPv4, IPv6, IP Multicast
- ICMP
- IGMP
- MIP
- MIP variant
- MIP
- CIP
- PPP
- AAL3/4
- AAL5
- CDMA
- Ethernet
- V.34

Physical
- SONET
- ATM
- 802.11
- ATM
- Ethernet
- V.34

Kernel
Service Profile for all IP wireless network user

<table>
<thead>
<tr>
<th>Services Requirements</th>
<th>Multimedia</th>
<th>Voice</th>
<th>Data</th>
</tr>
</thead>
<tbody>
<tr>
<td>Delay</td>
<td>Stringent</td>
<td>Stringent</td>
<td>Tolerant</td>
</tr>
<tr>
<td>Loss/error</td>
<td>Tolerant - Stringent</td>
<td>Tolerant</td>
<td>Stringent</td>
</tr>
<tr>
<td>Bit rate (outdoor)</td>
<td>Pedestrian ≤ 384 kb/s, Vehicular ≤ 144 kb/s</td>
<td>≤ 64 kb/s</td>
<td>Pedestrian ≤ 384 kb/s, Vehicular ≤ 144 kb/s</td>
</tr>
<tr>
<td>Bit rate (indoor)</td>
<td>≤ 2 Mb/s</td>
<td>≤ 64 kb/s</td>
<td>≤ 2 Mb/s</td>
</tr>
<tr>
<td>Example applications</td>
<td>Video streaming, video conferencing</td>
<td>Mobile telephony</td>
<td>File Transfer (e.g., ftp) to mobile</td>
</tr>
</tbody>
</table>
## Qualitative comparison with Mobility approaches

### QUALITATIVE COMPARISON OF DIFFERENT APPROACHES

<table>
<thead>
<tr>
<th></th>
<th>Intra-domain encapsulation</th>
<th>Inter-domain encapsulation</th>
<th>Changes to end-systems</th>
<th>Triangle routing</th>
<th>Infrastructure change</th>
<th>Fast handoff</th>
</tr>
</thead>
<tbody>
<tr>
<td>MIP</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>MIP-RO</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
</tr>
<tr>
<td>MIP-RR</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>MIP-FF</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>CIP</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>HAWAII</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>MIP-LR</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>TeleMIP</td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>SIP</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
</tr>
</tbody>
</table>
Objective

• To develop a mobility management scheme for wireless IP networks based on SIP signaling scheme
  – Support for all types of mobility
  – Support global roaming
  – Independent of underlying wireless technology
  – Support for real-time and non-real-time multimedia applications (both TCP and UDP/RTP based applications)
  – Inter-work with today’s 1G/2G telephony smoothly
## Technical Issues for SIP Mobility

<table>
<thead>
<tr>
<th>Functions</th>
<th>Requirements. Should:</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hand-off</td>
<td>• Support cell, subnet (intra-domain) and domain hand-off.</td>
</tr>
<tr>
<td></td>
<td>• Utilize the soft hand-off feature of CDMA technology, or virtual hand-off</td>
</tr>
<tr>
<td></td>
<td>• Be wireless “technology independent”.</td>
</tr>
<tr>
<td>Registration</td>
<td>• Be completed in less than a few seconds.</td>
</tr>
<tr>
<td></td>
<td>• Support Hierarchical Registration</td>
</tr>
<tr>
<td>Configuration</td>
<td>• Be done in fractions of a second for roaming users (e.g., IP address, DNS server.)</td>
</tr>
<tr>
<td>Address Binding</td>
<td>• Allow a user to maintain a universal identifier regardless of its point of attachment to the network.</td>
</tr>
<tr>
<td>Location Management</td>
<td>• Be up to date, accurate, and confidential.</td>
</tr>
</tbody>
</table>
**SIP Mobility Advantages**

- Easier Interaction with associated standard IETF protocols
  - DNS, HTTP, LDAP for location management
  - SLP for service discovery
  - AAA protocol (e.g.; Diameter) for inter-domain mobility
  - DHCP/DRCP for IP address configuration
  - tftp for firmware upload
  - SDP for providing session parameters (e.g., change mid-call parameters)
  - RTP/UDP for transport, RTSP for stream control application (e.g., IP Telephony, voice mail, streaming)

- Elimination of triangular routing and IP-IP encapsulation associated with other mobility approaches such as MIP
  - Reduces delay
  - Saves network overhead

- End hosts should be equipped with SIP-UA
- Suitable for real-time multimedia traffic such as voice over IP and/or video streaming
  - Can be used for RTP/UDP based application as is
  - SIP extensions for Non-real-time application

- Complements IPV6 mobility
- Can co-exist with MIP, Cellular IP and other Micro-mobility approaches
SIP Mobility Basics

• Supports end-to-end mobility by means of application layer signaling meant for multi-media/multi-party sessions
• SIP based mobility can also be termed as Application Layer mobility
• More than just hand-off
  – supports various types of mobility
  – provides flexible services
• Compensate for lack of Mobile IP deployment
• Less reliance on underlying transport network of the ISPs
• Supports application-layer equivalent of Mobile IP registration
• Fast-handoff
• Paging
Types of Mobility supported by SIP

• Terminal Mobility
  – Pre-session mobility (Micro/Macro/Domain)
    • pre-session mobility by means of unique URI (ability for a user to be reached under the same identifier, using different terminals)
    • use of SIP proxy, redirect, registrar
    • Hierarchical registration for faster registration update

• Mid-session mobility
  • Move between cells, subnets, domains, supports handoffs
  • Real-Time (RTP/UDP)
    – SIP Re-invite, RTP SSRC/IP address
    – Hierarchical proxy and RTP translator for fast hand-off within a domain
    – Duration limited multicast between subnet handoff
    – use of RTSP to control multi-media stream server
Mid-session mobility for TCP based application

- **Mobility problem with TCP applications**
  - TCP socket - bound to source and destination address
  - One of these addresses change => connection breaks
- TCP applications: ftp, telnet, web
- **Application Layer restart and recovery capabilities**
  - connection: close header into HTTP request
  - FTP variants (e.g., bullet-proof ftp)
- Multi-homing feature of SCTP (IETF)
- TCP-Migrate Option
- SIP-eye enabled in the end-hosts - keeps track of the TCP end-points of SIP
- SIP Mobility Proxy (Columbia U.)
  - an interceptor to forward data
Basics of SIP Mobility

• Personal Mobility
  – Use of one logical address to address a single user located at different terminals
    • One address to many potential terminals
  – Many addresses reaching one terminal
  – Use of forking Proxy, a user can be reached at any of the devices

• Service Mobility
  – Allows users to maintain access to their services while moving or changing devices and network service providers
  – Maintain speed dial list, address books, buddy lists, incoming call handling
  – As part of registration message (on a routine basis or upon network change) it conveys
    • current network address
    • Properties of the device (media supported, call priority etc.)
    • Other configuration elements

• Session Mobility
  – Allow a user to maintain an ongoing media session even while changing terminals
  – Use of MGCP/Megaco
  – Third-party Call control
  – Refer Mechanism
**SIP Mobility - Mobile IP**

**Plain Mobile IP**

1. SIP INVITE
2. 302 client moved
3. SIP INVITE
4. SIP OK
5. Data

**SIP Pre-session Mobility**
SIP Mid-session mobility

1. MN moves (gets new IP)
2. MN re-invites (send new IP to CN)
3. SIP OK
4. Data
SIP Mobility

When both terminals move

[Diagram showing the movement of CNs and MNs between Home and Foreign Networks, with SIP Servers involved in communication.]
Evaluation Model for SIP and Mobile IP

Caller’s Network

Callee’s Home Network

Callee’s Foreign Network

High-speed link

Low-speed link

N hops

M hops

P hops

MIP

SIP

CH

FA

MH

MH
SIP-Mobile IP Transport Delay vs. Packet size

SIP/MIP Latency vs. Packet size

Packet Size in bytes

Latency in msec

SIP-SD
MIP-SD
MIP-D
SIP-D

D - only data
SD – Signalling + Data
Bandwidth Efficiency Gain

SIP/MIP bandwidth gain

Packet size in bytes

Bandwidth Efficiency Gain

SIP b/w efficiency gain
SIP Mobility - Handoff

By sending SIP re-INVITE message from new location, CN starts sending its voice packet to the new location and Communication continues seamlessly.
SIP Mobility - Handoff

Corresponding Node at IP0

Mobile Node at IP1

SIP signaling

RTP

Invite user@domain
Contact user@IP2

SIP signaling

RTP

SIP UA

IP2

RAT

Mobile Node -> IP2
SIP Personal Mobility

Bob@hotmail.com

Bob@columbia.edu

7010@columbia.edu

Bob.Cattani@columbia.edu

Hotmail.com Server

Columbia.edu

Fixed Phone

Mobile Phone

Host
Session Mobility using Call transfer

1. REFER Bob@fixed
   Referred-By: B1

2. INVITE Bob@fixed
   Referred-By: B1

3. BYE Alice@Wonderland

Bob@mobile

Session Transferred

Bob@fixed

Bob@mobile

Alice@wonderland

Third party control

INVITE from 2

ACK

SDP(4)

INVITE with no SDP

200 OK

RTP

ACK

SIP-CONF-PULVER-2001
Registration with local SIP Registrar

- Visited Network Registrar needs to map Alice’s URI to a canonical name
- Only first registration in CA needs to go all the way to NY
- All SIP messages to Alice need to go through SIP server/registrar for address translation.
References

• www.research.telcordia.com/sip-mobile
• Application Layer Mobility Using SIP, MC2R
  
  *Henning Schulzrinne, Elin Wedlund*

• Application Layer Mobility Management Scheme for Wireless Internet, 3G Wireless Conference
  
  *Dutta, Vakil, Baba, Chen, Tauil, Schulzrinne*