SIP-based Mobility Architecture for Next Generation Wireless Networks

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Outline

- Introduction
- Overview of SIP
- Problem Description
- Proposed Architecture
- Performance Measurements
- Conclusion
- References

Introduction

- The handoff procedure with SIP suffers from delay and packet loss which is detrimental for real-time services with stringent quality of service (QoS) requirements
- We propose a SIP based architecture that supports soft handoff for IP centric wireless networks alleviating the problem of packet loss
- In our architecture, the soft handoff is achieved at the IP layer with the help of SIP signaling, so that it is independent of the underlying radio access technology
- The proposed architecture ensures that there is no packet loss and the end-to-end delay jitter is kept under control

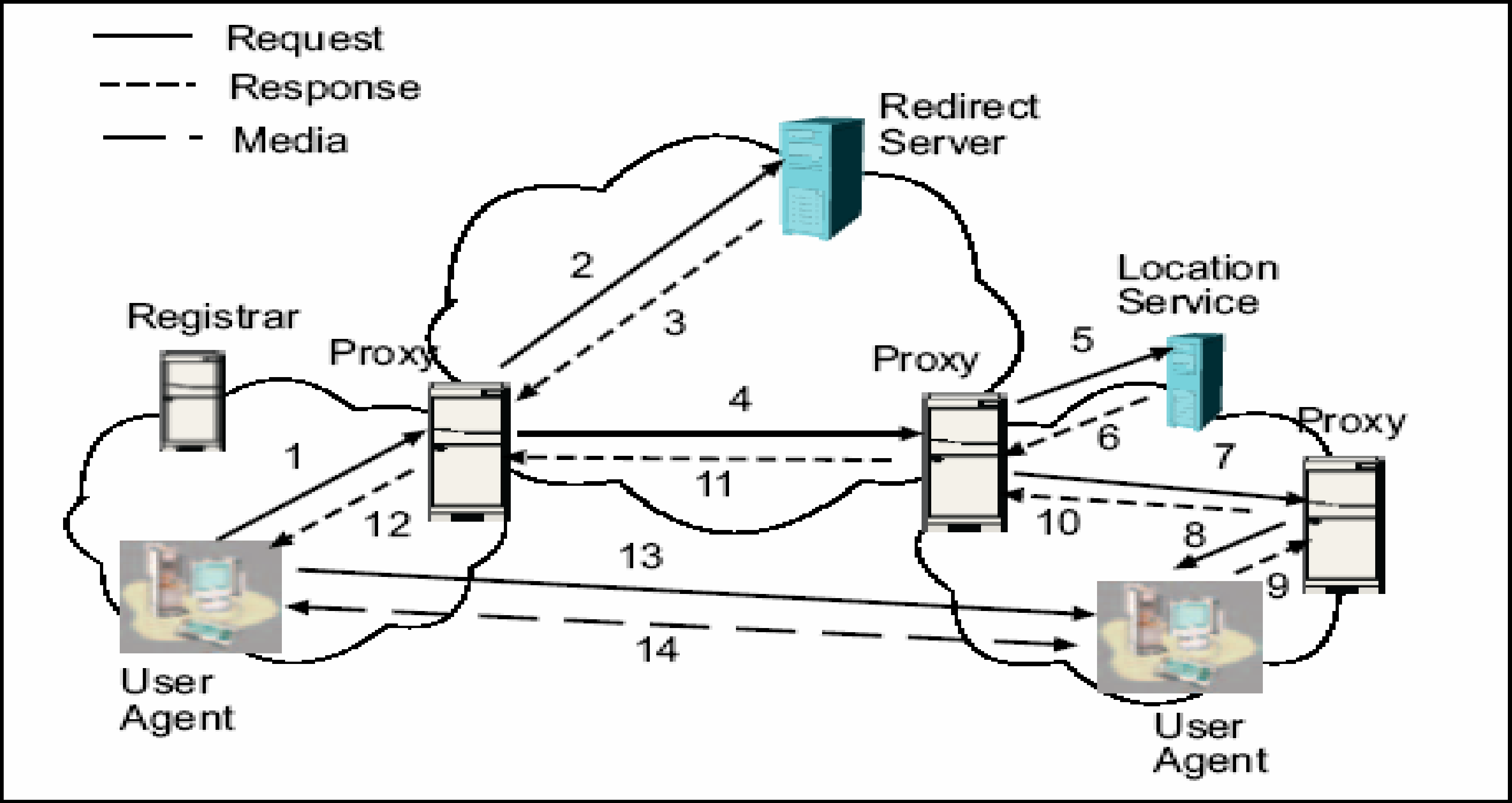
Why we need SIP

- Mobile IP suffers from the triangular routing problem
- Mobile IP route optimization has been proposed to solve triangular routing problem, but again suffers from the following drawbacks such as encapsulation overhead and IP stack change
- In several mobility protocols proposed as a remedy to these problems, SIP has the least dependency on the access networks [1]

Session Initiation Protocol (SIP)

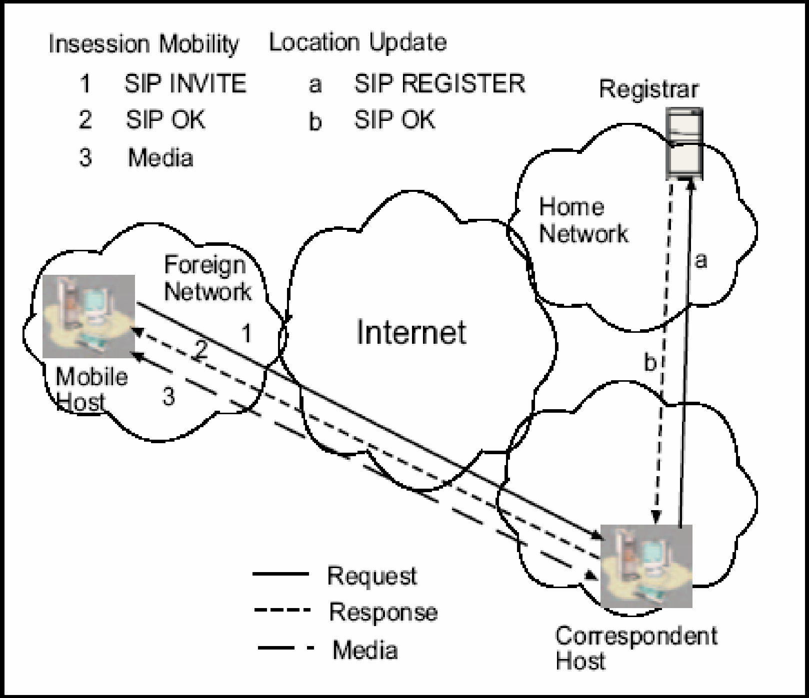
- SIP is accepted by 3GPP as an application layer signaling protocol for setting up real-time multimedia sessions
- Although SIP based mobility management solves the problem posed by Mobile IP route optimization, for some cases it introduces unacceptable handoff delays for multimedia applications [2]

SIP Architecture



SIP Mobility Management

- *Pre-call mobility*
  - each time the MH changes location it registers with the home network's registrar service
- *Mid-call mobility*
  - for ongoing sessions, the MH sends a re-INVITE message to the CH using the same call identifier
  - the CH starts sending data to the new location as soon as it gets the re-INVITE message
  - the handoff delay is essentially the one-way delay for sending an INVITE message from the MH to the CH



mid-call mobility

Problem Description

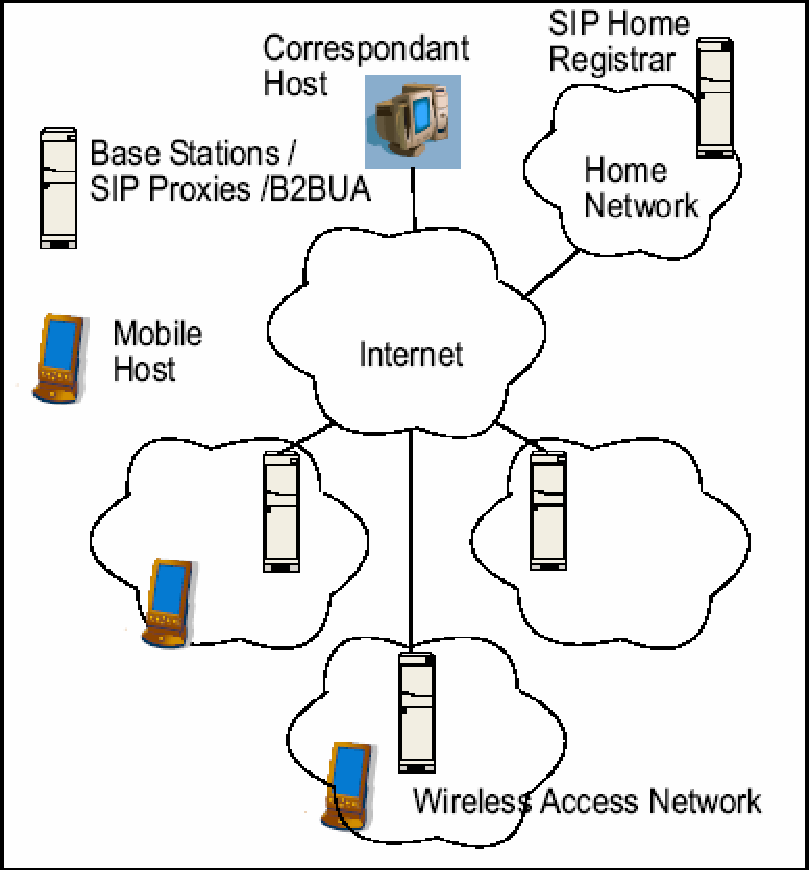
- Large handoff delay cause considerable packet loss which seriously affects the quality of the multimedia streaming applications
- Example:
  - approximately 4-5 voice packets are dropped with a handoff delay of 1 sec for a 16 Kbps stream with 64 bytes voice packets
  - 200000 packets are lost for a 1.5 Mbps MPEG-4 stream with 1050 bytes of packet size
- Such packet dropping has serious consequence on the video quality because of the propagation of error in MPEG-4, particularly to the dependent frames or the I-frames [3]

Problem Description

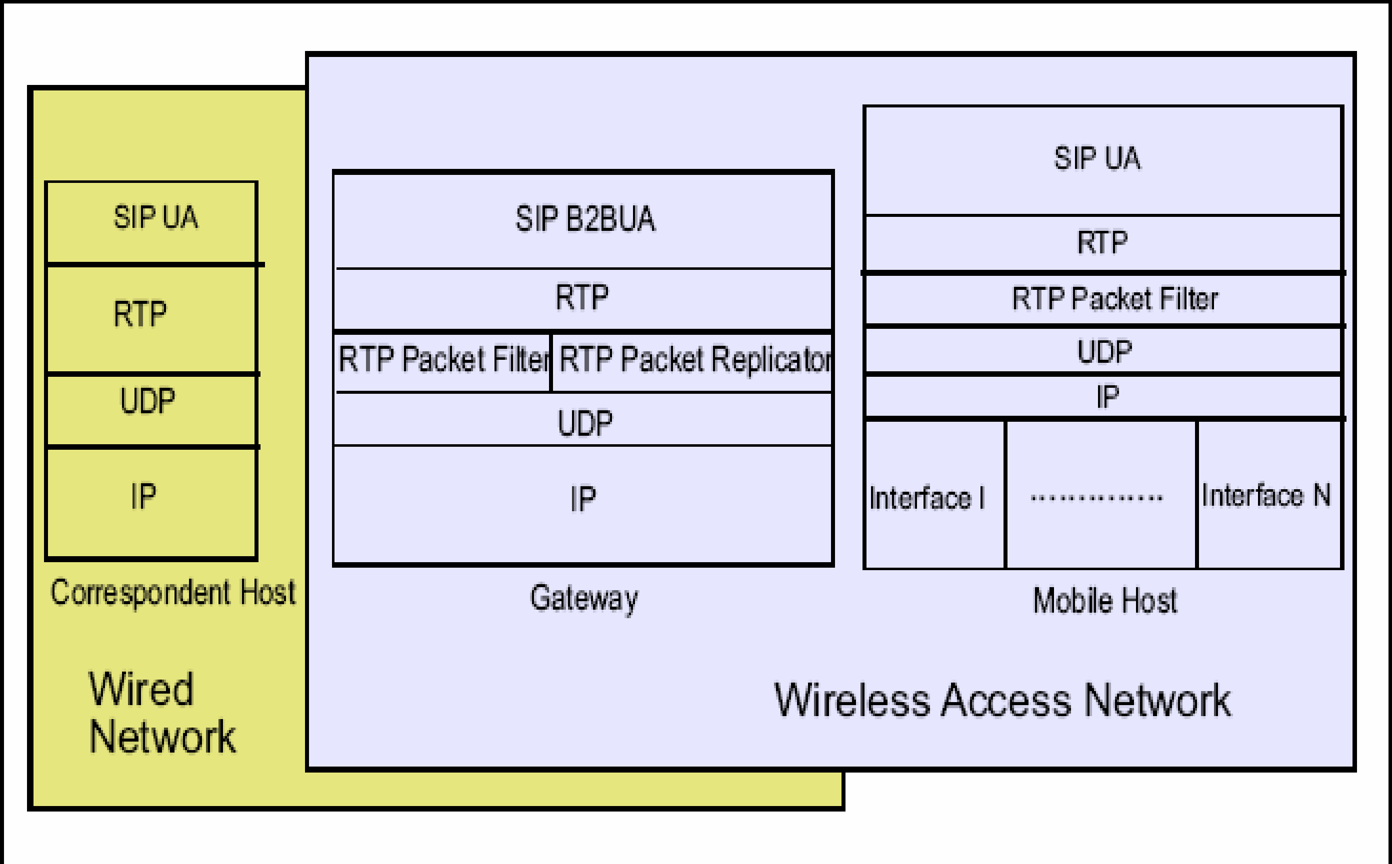
- The handoff delay in SIP based mobility is essentially one-way delay from the MH to the CH, but several different operations need to be completed before the INVITE message could be transported. These are:
  - Detection of the new network by the MH
  - The MH needs to acquire an IP address by a procedure specific to the access networks (ex:DHCP)
- Analytical study reveals that the handoff delay can be more than 1 sec for low bandwidth access network, for which hard handoff, has considerable effect on the application quality

Proposed Architecture

- The MH is equipped to interface through different types of access technologies and can receivce/transmit packets through more than one of these interfaces simultaneously
- The soft handoff procedure is initiated by the MH but is executed at the base stations
- Each base station is equipped with a SIP B2BUA and a SIP proxy server



Proposed protocol architecture

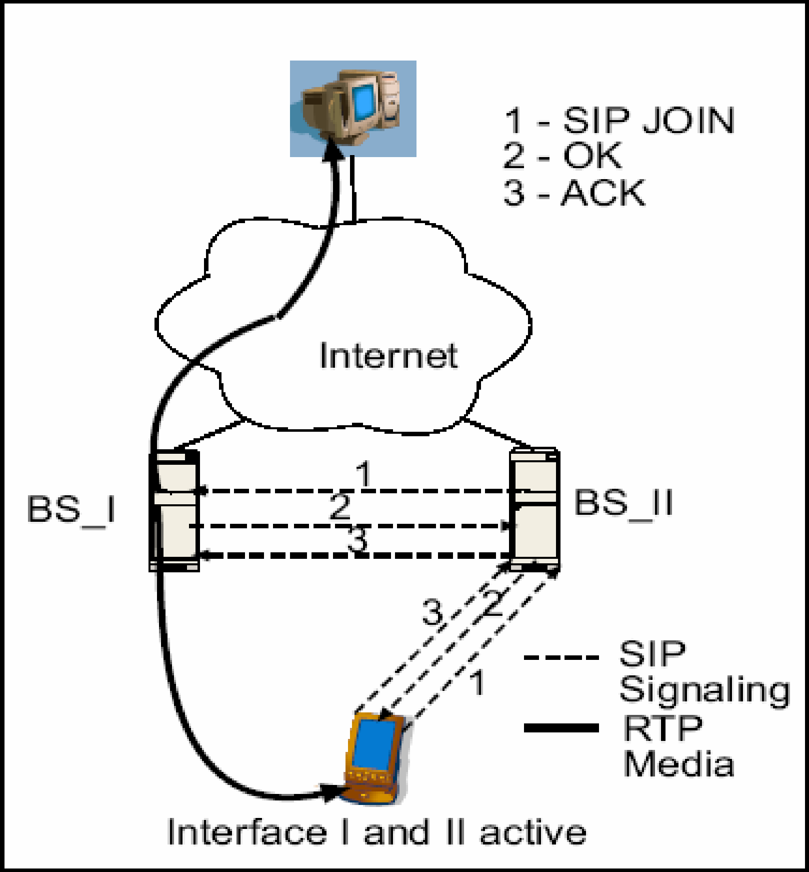


SIP-based Soft Handoff

- A B2BUA is a logical entity that receives a request and processes it
- The B2BUA is coupled with a media gateway that acts as a proxy, forwarding the RTP packets
- The media gateway has the dual functionality of a RTP *packet replicator* and a RTP *packet filter*
- The MH has a packet filter only
- The packet replicator duplicates an RTP packet and sends it to a different IP address
- The packet filter filters RTP packets received at the media gateway and sends a single copy of the RTP packet to the destination

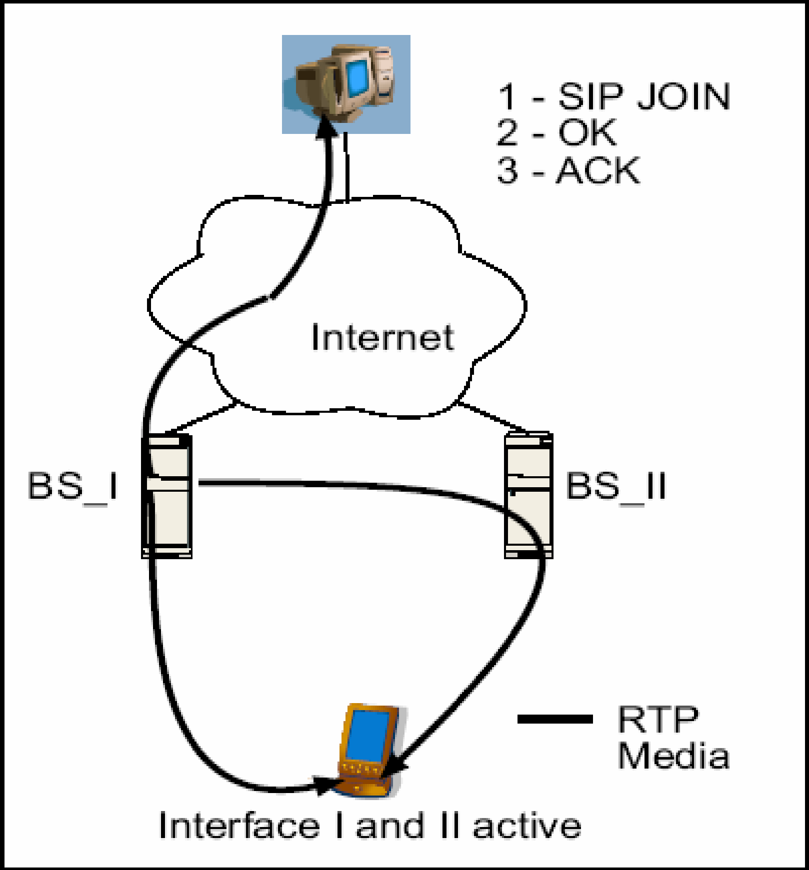
Sending of JOIN message to initiate the soft handoff

- During the handoff, a new network interface gets activated, the MH sends an INVITE message with the JOIN header to the SIP B2BUA proxy servers [4]
- For this operation, the MH only requires to know about the available network interfaces



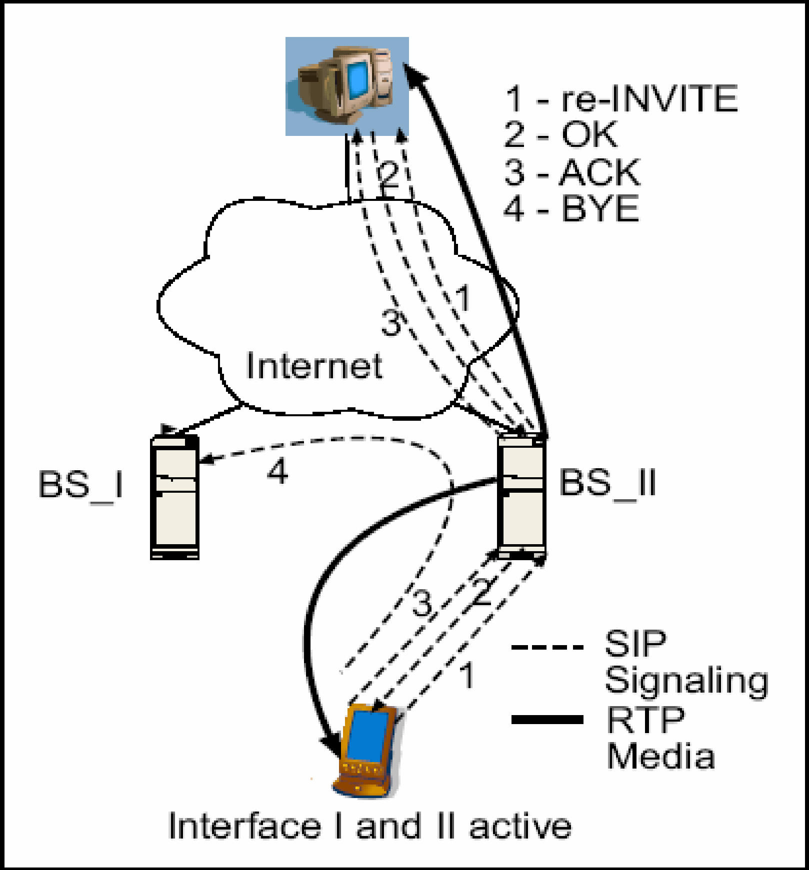
Splitting of RTP stream - soft handoff procedure

- The B2BUA send a copy of all packets directed towards the old interface of the MH to the newly activated interface
- Once the packets reaches the MH through the newly activated interface, a re-INVITE message is sent to the CH with the new IP address and the corresponding contact information
- And then MH discards the duplicate RTP packets

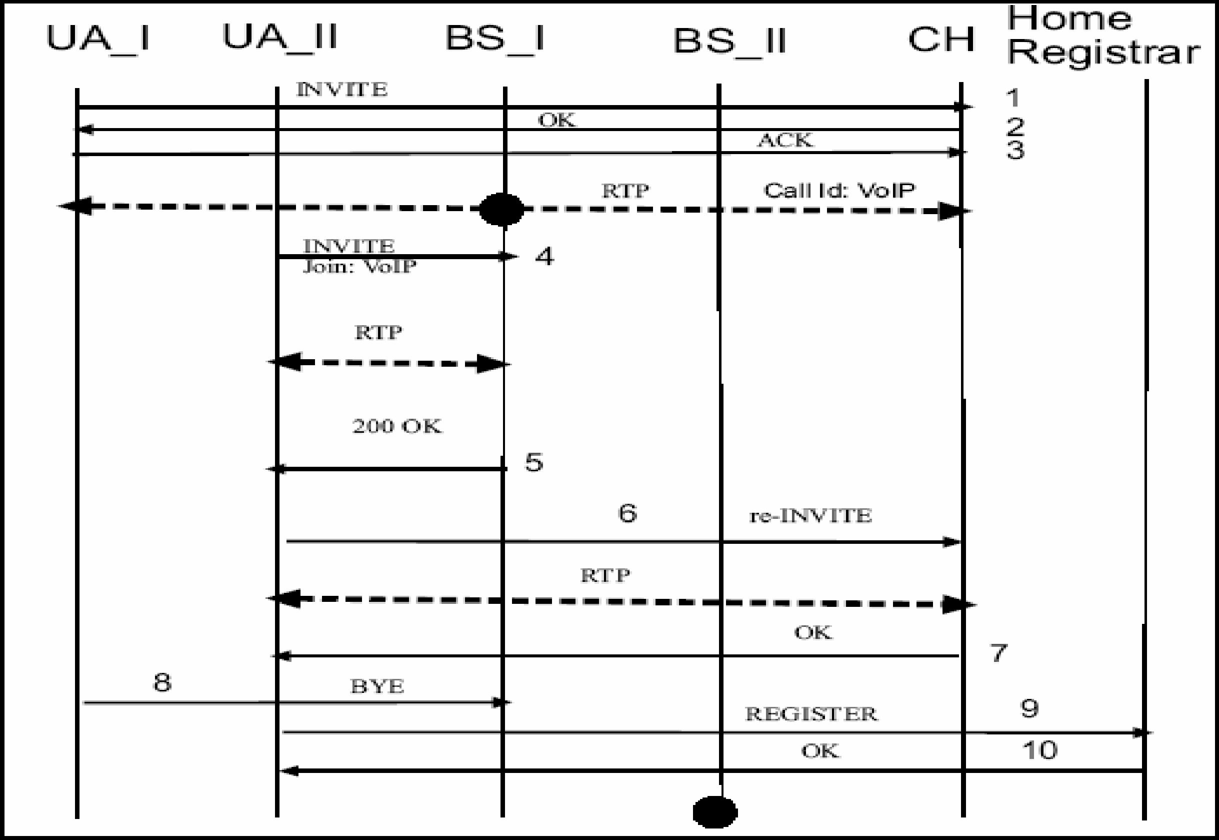


Signaling to update the ongoing session parameters on account of the change in MH's IP address

- The call parameters are re-negotiated, with the selection of a new intermediate SIP proxy server and B2BUA orresponding to the newly activated interface
- Once the call re-negotiation is complete, MH terminates the call-leg through the initial interface,
- Finally, the MH registers its new location information with the home network's registrar server



Message Diagram

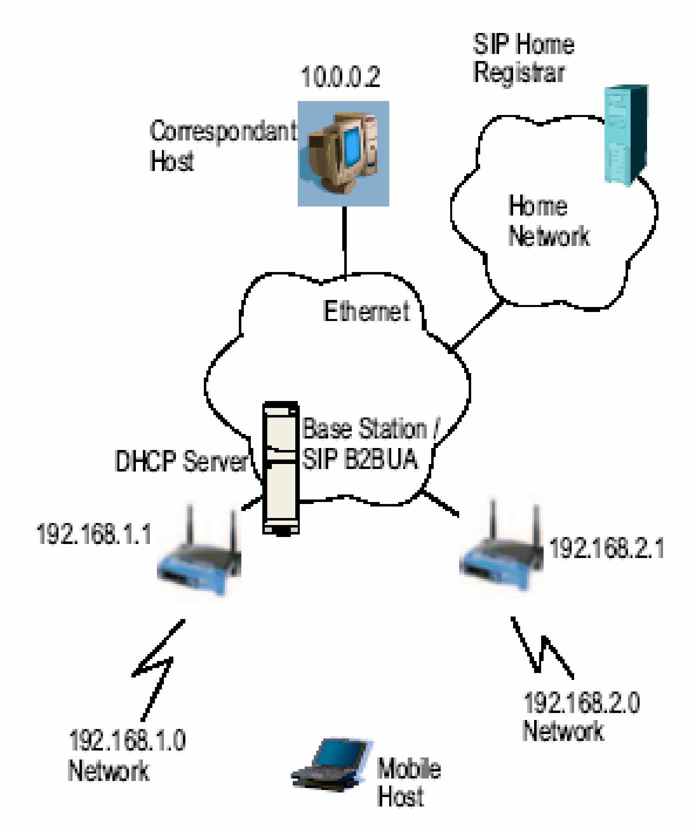


Handoff delay

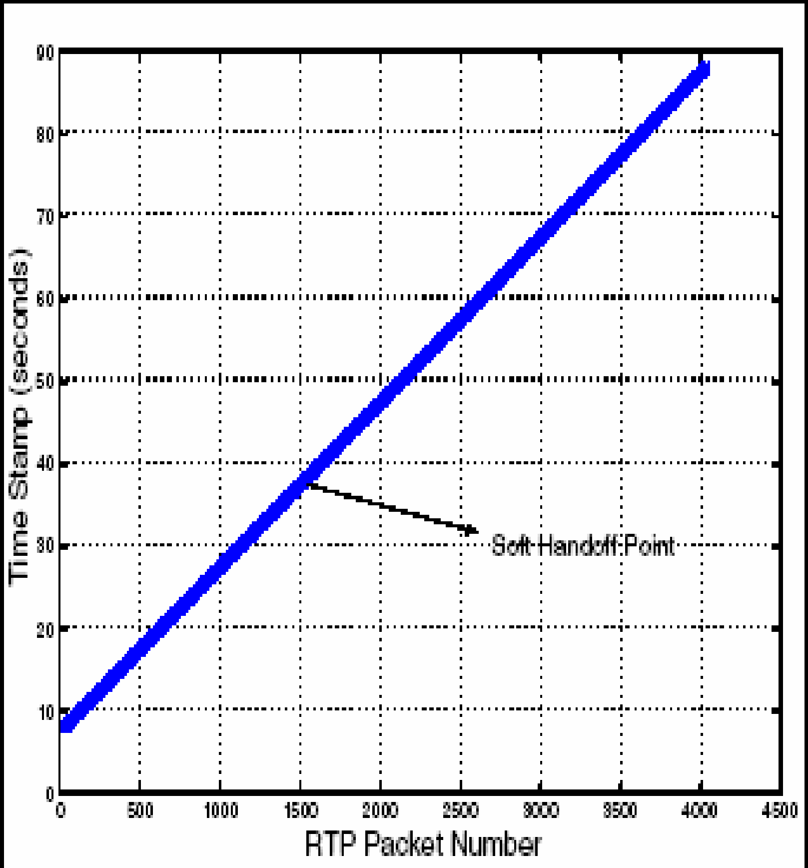
- The handoff delay is composed of the following major operations:
  - Network detection and address configuration operation performed by the MH (t_{attach})
  - Sending the INVITE message with the JOIN header to BS I (t_{join})
  - Sending the re-INVITE message to update the session with the new location parameters ($t_{re-invite}$)

Experimental testbed setup

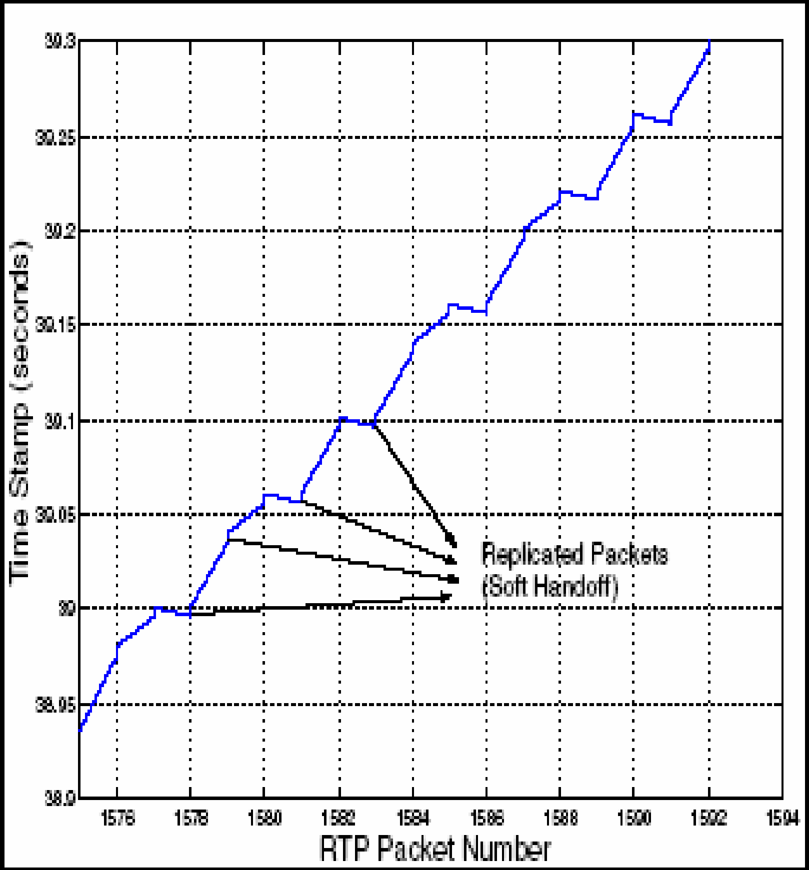
- Two different subnets with wireless access are created for testing the performance of the SIP based terminal mobility with soft handoff
- They are obtained as an average over 10 different handoff events:
  - *tattach* = 23.95369231 secs
  - *tjoin* = 3.618 msecs
  - *tre−invite* = 359.84 msecs



Performance Measurements

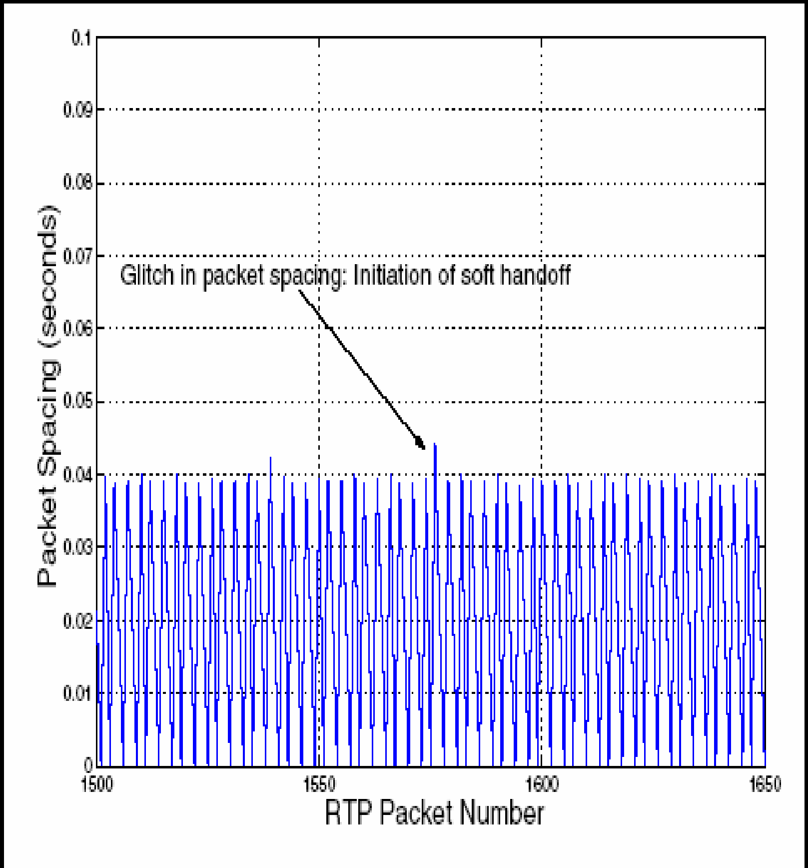


RTP stream at MH

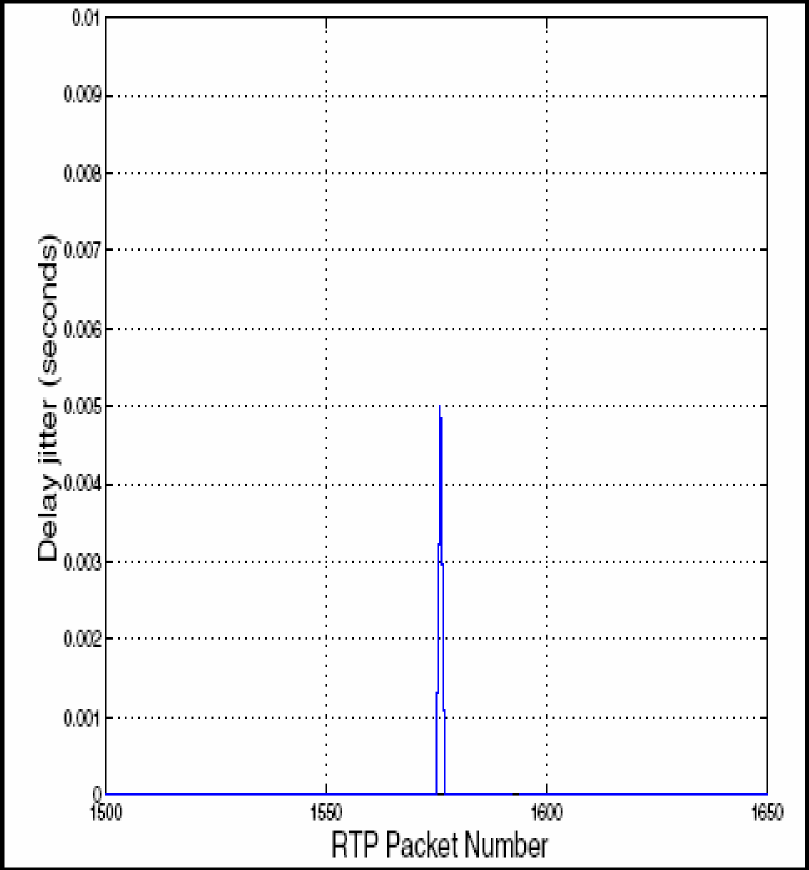


Soft handoff with replicated RTP packets

Performance Measurements



Spacing between the RTP packets



Delay jitter for RTP packets

Conclusion

- SIP provides an elegant application layer mobility support that solves the problems associated with lower layer mobility protocols in next generation heterogeneous wireless access networks
- However, the handoff delay in SIP may be substantial causing considerable packet loss, which affects the quality of voice or video streams seriously
- We have proposed a SIP based soft handoff mobility architecture for next generation wireless networks ensuring zero packet loss and controlled delay jitter

References

- [1]N. Banerjee,W.Wu, S. K. Das, and S. Dawkins, and J. Pathak, “Mobility Support inWireless Internet”, IEEE Wireless Communications Magazine, Vol. 10, No. 5, Page(s): 54-61, 2003
- [2] N. Banerjee, W. Wu, K. Basu, and S. K. Das, “SIPBased Mobility Management in 4G Wireless Networks”, *Journal of Computer Communications, special issue on Research Directions in 4th Generation Wireless Networks*, Vol. 27/8, Page(s): 697-707, 2003
- [3] N. Feamster and H. Balakrishnan, ”Packet Loss Recovery for Streaming Video”, *Packet Video Workshop*, April 2002
- [4] R. Mahy and D. Petrie, “The Session Inititation Protocol (SIP) ”Join” Header”, draft-ietf-sip-join-03.txt, Feb 2004, Work in progress.