SIP-based Mobility Architecture for Next Generation Wireless Networks

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Abstract

Application-level protocol abstraction is required to support seamless mobility in next generation heterogeneous wireless networks. Session Initiation Protocol (SIP) provides such an abstraction in providing mobility support in such networks. However, the handoff procedure with SIP suffers from undesirable delay and hence packet loss for some cases, which is detrimental to applications such as Voice over IP (VoIP) or streaming video with stringent quality of service (QoS) requirements. In this paper, we propose a SIP based architecture that supports soft handoff for IP centric wireless networks alleviating the problem of packet loss. The proposed architecture ensures that there is no packet loss and the end-to-end delay jitter is kept under control, thus maintaining two important parameters dictating the QoS for streaming multimedia applications.

1 Introduction

Seamless mobility in converged IP centric networks provides an important paradigm for uninterrupted services in pervasive/ubiquitous computing environments. Seamless services require network and device independence that allow the users to move across different access networks and change computing devices. IP convergence has led to the co-existence of several IP based wireless access technologies (e.g., GPRS, CDMA 2000 Wireless LAN [23]) and the emergence of other next generation technologies (e.g., UMTS) along with the diverse range of mobile devices make the seamless service provisioning an extremely non-trivial problem. The problem is further complicated for multimedia streaming applications (e.g. Voice over IP or VoIP, video streaming, etc.) having stringent QoS requirements such as minimum bandwidth, delay, jitter, and loss rate.

Mobility management protocols are in general responsible for supporting services seamlessly across heterogeneous access networks that require connection migration from one network to another. This is known as the *vertical handoff*. Thus, in addition to providing location transparency, the mobility management protocols in this case also need to provide network transparency. A number of research work has been directed towards solving the vertical handoff problem for IP based networks [28, 4, 33]. Most of these works are based on Mobile IP [18]. However, Mobile IP suffers from the problem of triangular routing which is detrimental to real-time traffic like streaming multimedia, where the important issues are fast handoff, low latency and minimal packet loss. Mobile IP route optimization [19] has been proposed to solve the problem particularly, but route optimization again suffers from the following drawbacks: the IP stack needs change to implement route optimization and the home agent can only initiate the route optimization which introduces additional delay. Besides, mobile IP encapsulation adds 8–20 bytes of overhead for each data packet. Several mobility protocols have been proposed as a remedy to these problems [18, 20, 30, 8, 6, 27, 21]. Based on the layer of their operation, these protocols can be broadly classified as those operating in the network layer [18], transport layer [27] and application layer [21]. The dependency of these mobility protocols on the access networks reduces progressively as we move up on the protocol stack [3]. Among them, Session Initiation Protocol (SIP) [21] has been standardized by the Internet Engineering Task Force (IETF) [29] as an signaling protocol for multimedia session setup in IP based networks. In addition, SIP is capable of supporting not only personal mobility but also terminal, session and service mobility at the application layer. Application layer protocols, however, are transparent to the network (or lower layer) characteristics. For example, an

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application layer protocol sending UDP packets does not need to know how an underlying GPRS or a CDMA 2000 network transport the packet. They maintain the true end-to-end semantics of a connection and are expected to be the right candidate for handling mobility in a heterogeneous network environment. Indeed, SIP has been accepted by the third Generation Partnership Project (3GPP) as an application layer signaling protocol for setting up real-time multimedia sessions. Moreover, SIP has gained widespread acceptance from several commercial vendors like as Sprint PCS, Verizon to provide several important services such as Instant Messaging, push-to-talk, etc. Thus, SIP based mobility management could potentially use a readily available operational infrastructure, which would facilitate its fast deployment. Therefore, SIP seems to be an attractive candidate as an application layer mobility management protocol for vertical handoff [3]. Although SIP based mobility management solves the problem posed by Mobile IP route optimization, for some cases it introduces unacceptable handoff delays [2] for multimedia applications with stringent QoS requirements. Moreover, SIP entails application layer processing of the messages which may introduce additional delay.

In this paper, we propose an architecture for IP-layer based soft handoff scheme with SIP for next generation wireless infrastructure networks. Soft handoff ensures minimal packet loss and handoff delay variation, which are critical requirements for providing QoS to multimedia applications. The proposed architecture is inspired by works which introduce some level of data redundancy to solve the problem of packet loss. A multicast based architectures for host mobility [16] was proposed to reduce the handoff delay and minimize packet loss. This approach needs the deployment of IP multicast infrastructure. However, IP multicast being not that successful, leaves this approach subject to doubts regarding the performance efficiency and deployment feasibility. Transport and network level bandwidth aggregation [5, 10, 15], where multiple interfaces are used during handoff, were proposed to attain the same goal. Soft handoff at the IP level for SIP based mobility management was first hinted in [24]. A similar approach, based on CDMA's soft handoff mechanism, has been proposed in [31], for optimized fast handoff schemes with SIP in CDMA networks. However, this study utilizes the multiple concurrently received signals in CDMA networks to achieve the soft handoff. In contrast, 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 has been implemented in a testbed environment as a proof of concept and has been evaluated to measure the performance efficiency. It has been shown with the help of experimental results that the architecture performs efficiently in terms of packet loss and delay jitter.

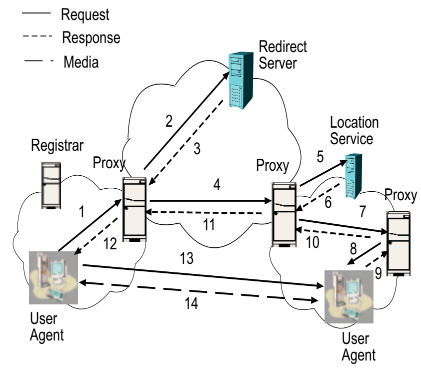

Figure 1. SIP architecture

The rest of the paper is organized as follows. An overview of SIP and its mobility support is presented in Section 2. The problem under consideration is described in Section 3. The proposed architecture is presented in Section 4. The details of a testbed setup as a proof of concept and for performance evaluation are described in Section 5. Section 6 concludes the paper.

2 Overview of SIP and mobility support

SIP is a control protocol that allows creation, modification and termination of sessions with one or more participants. It is used for both voice and video calls either for point-to-point or multiparty sessions. It is independent of the media transport which for example, typically uses Real-time Transport Protocol (RTP) over UDP [25]. It allows multiple end-points to establish media sessions with each other. This includes terminating the session, locating the end-points, establishing the session and modifying the media session after the session establishment has been completed. Recently, SIP has gained widespread acceptance and deployment among wireline service providers for introducing new services such as VoIP; within the enterprises for Instant Messaging and collaboration; and amongst mobile carriers for push-to-talk services. Industry acceptance of SIP as the protocol of choice for converged communications over IP networks is thus highly likely. As shown in Figure 1, a SIP infrastructure consists of user agents, registration servers, location servers and SIP proxies deployed across a network. A user agent is a SIP endpoint that identifies services such as controlling session setup and media transfer. User agents are identified by SIP URIs (Uniform Resource Identifiers), which are unique HTTP-like URIs of the form `sip:user@domain`. All user agents register their IP addresses with a SIP registrar server (which can

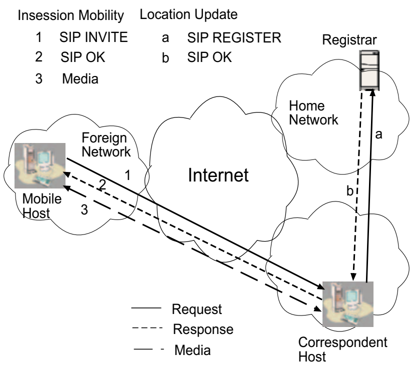

Figure 2. SIP based mid-call terminal mobility management

be co-located with a SIP proxy). Details of the SIP protocol can be found in [21]. SIP defines a set of messages, such as INVITE, REGISTER, REFER etc., to setup sessions between the user agents. These messages are routed through SIP proxies that are deployed in the network. The DNS Service records help in finding SIP proxies responsible for the destination domain.

A session or dialog is setup between two user agents following a client-server interaction model, where the requesting user agent client (UAC) interacts with the target user agent server (UAS). A logical entity formed by concatenating a UAC and a UAS that keeps all the dialog information and intercepts all the messages within a dialog to participate in the same, is known as a back-to-back user agent (B2BUA). All requests from an originating UAC, such as an INVITE are routed by the proxy to an appropriate target UAS, based on the target SIP URI included in the `Request-URI` field of the INVITE message. Proxies may query location and redirect servers for SIP service discovery or to determine the current bindings of the SIP URI. Signaling messages are exchanged between user agents, proxies and redirect/location servers to locate the appropriate services or endpoints for media exchange. For reasons of scalability, multiple proxies are used to distribute the signaling load [13]. A session is setup between two user agents through SIP signaling messages comprising of an INVITE (messages 1,2,4,7, and 8 in Figure 1), an OK response (messages 9-12 in Figure 1) and an ACK (message 13 in Figure 1) to the response [21]. The call setup is followed by media exchange using RTP. The session is torn down through an exchange of BYE and OK messages.

Apart from the session setup function SIP inherently supports personal mobility and can be extended to support service and terminal mobility [24, 32]. Personal mobility enables a user to be found independent of the location and network device. Terminal mobility, on the other hand, enables a user to change location or IP address during the traffic flow of an ongoing session. It can be explained with an example of an ongoing session between a mobile host (MH) and a correspondent host (CH) as follows. Each MH belongs to a home network with a SIP server providing a registrar service. Each time the MH changes location it registers with the home network's registrar service. This is in principle similar to Mobile IP home registration. For ongoing sessions, the MH sends a re-INVITE message to the corresponding CH using the same call identifier as in the original setup. The former procedure takes care of *pre-call mobility*, while the latter enables *mid-call mobility*. High level messaging of SIP based mid-call mobility management is depicted in Figure 2. The new contact information (e.g., URI for future contact) is put in the `Contact` field of the SIP message to redirect the subsequent SIP messages to the current location. The data traffic flow is redirected by updating the transport address field in the Session Description Protocol (SDP) [9] part of the re-INVITE message. For mid-call mobility, the CH starts sending data to the new location as soon as it gets the re-INVITE message. Hence, the handoff delay is essentially the one-way delay for sending an INVITE message from the MH to the CH.

3 Problem Description

Mobility is the most important feature of wireless networks that makes continuous service possible in pervasive/ubiquitous environments. Seamless service is usually achieved by supporting handoff. Handoff is the process of changing parameters (e.g., endpoint address, channel etc.) associated with the current connection. For UDP based connections the major parameters are the source and destination IP addresses, which can be changed by the movement of an MH, either within a network or across different networks. The former scenario initiates horizontal handoff, whereas the latter initiates vertical handoff. Handoff is again divided into two broad categories: *hard* and *soft* handoffs. They are also characterized by "break before make" and "make before break." In hard handoffs, current resources are released before new resources are used and in soft handoffs, both existing and new resources are used during the handoff process. For soft handoff the MH should be capable of communicating through multiple network interfaces.

Usually, a mobility management protocol, operating at the control plane independent of the data plane supports handoff. As described earlier, SIP provides vertical handoff support in IP centric networks for multimedia applications. Although the data plane protocols provide QoS to the applications, it is the responsibility of the mobility management protocol to maintain the QoS during the handoff period. For multimedia streaming applications, the most important

QoS parameters are (i) end-to-end delay, (ii) delay jitter or variation of end-to-end delay between the packets, and (iii) packet loss. Of these three parameters, the first two depend on the network condition in the path of the data traffic. Generally, the issues related to these parameters can be resolved by providing a playout and jitter buffer. The handoff delay causes only a glitch as far as these two parameters are concerned and has no long-term effect. However, large handoff delay cause considerable packet loss which seriously affects the quality of the multimedia streaming applications. For 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 and 2×10^5 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 [7]. For voice streams, the packet loss usually results in annoying popping and clicking sounds.

Despite the advantages of SIP in providing mobility support in IP based heterogeneous networks, there are some issues that need to be resolved for proper QoS provisioning to multimedia applications. The handoff delay in SIP based mobility is essentially the time required by the re-INVITE message to reach the CH from the MH, but several different operations need to be completed before the INVITE message could be transported. These are: (i) Detection of the new network by the MH. This depends on the networking technology (e.g., periodic beacons from the access points are used in WLANs to intimate a mobile device about the presence of the network) as well as on the operating system in the MH. (ii) The MH needs to acquire an IP address by a procedure specific to the access network. This may be DHCP address configuration for WLAN or Attach and PDP Context Activation for GPRS networks. Analytical study [2] reveals that the handoff delay can be more than 1 sec for low bandwidth access network, for which hard handoff, according to the previous discussion, has considerable effect on the application quality. So, the mobility management protocol needs to employ some mechanism to counter the harmful effect of the handoff delay. Soft handoff technique provides such as a mechanism to deal with the large handoff delays and consequent packet drop.

4 Proposed Architecture

In this section, we have proposed an architecture for SIP based mobility management supporting soft handoff at the IP layer in next generation heterogeneous wireless networks. As illustrated in Figure 3, an MH can move between various wireless networks with different access technologies such as GPRS, CDMA, WLAN, etc. 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. In each wireless access network the MH communicates through the base stations, acting as the gateways to the Internet. Each MH is SIP-enabled and SIP takes the responsibility of session setup and the provisioning of seamless mobility. According to the SIP architecture, each MH has a home network with a registrar service containing the latest location information of the MH. Typically, the CH that wants to setup a session with the MH contacts the registrar service at the MH's home network and gets the latest contact information for the MH. As described earlier, when an MH moves to a different network, acquiring a new IP address, its SIP client initiates a handoff procedure by sending a re-INVITE message with updated SDP parameters to the CH as well as to the home network's registrar service. Handoff can also be base station assisted, but we have adopted an MH-initiated handoff as it has the best knowledge on the currently active network interfaces and hence is the best candidate to initiate the handoff. As discussed in Section 3, the handoff procedure introduces a connection disruption for a considerable duration resulting in packet loss. The IP layer soft handoff has been proposed based on SIP signaling to avoid this packet loss.

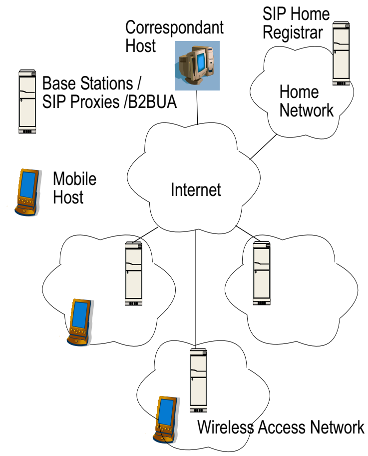

Figure 3. Next generation wireless network architecture

4.1 SIP-based Soft Handoff

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. A B2BUA is a logical entity that receives a request and processes it as a user agent server (UAS). It maintains dialog state and par-

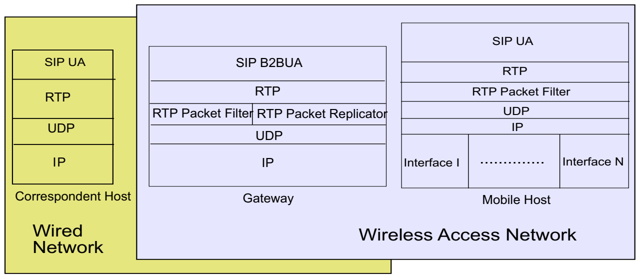

Figure 4. Proposed protocol architecture

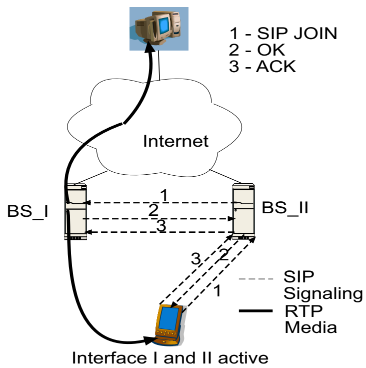

Figure 5. Sending of JOIN message to initiate the soft handoff

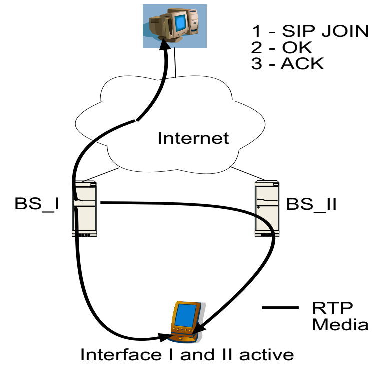

Figure 6. Splitting of RTP stream - soft handoff procedure

ticipates in all requests sent on the dialogs it has established. All the SIP messages are directed through the outbound proxy at the base station using the `Record-Route` field of the message header, so that the B2BUA is able to capture the ongoing dialog information. 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, on the other hand, has a packet filter only [1]. The packet replicator duplicates an RTP packet and sends it to a different IP address while the packet filter filters RTP packets received at the media gateway and sends a single copy of the RTP packet to the destination. In principle, the B2BUA agent and the media gateway can be physically decoupled from each other.

When an MH is in transition from one network to another (i.e., during the handoff period), more than one network interfaces become active and the MH is capable of communicating through them. Now, SIP does not enforce any restriction to the use of the network interface while sending the SIP messages. In fact, any of the available network interfaces can be used by a SIP user agent to send the messages and this facility is available in almost all of the SIP client implementations. During the transition period when a new network interface gets activated, the SIP UAC at the MH sends an INVITE message with the JOIN header [12] to the SIP B2BUA proxy server. Note that, for this operation the SIP client only requires to know about the available network interfaces during the handoff period and requires no other support from the network layer. Thus although the soft handoff takes place at the IP layer, it is entirely controlled at the application layer. The JOIN header contains all the relevant information about the ongoing call. The B2BUA being a statefull entity, is able to identify the call and accordingly configures the packet replicator and the packet filter. The B2BUA essentially, configures the packet replicator at the media gateway to send a copy of all packet directed towards the old interface of the MH to the newly activated interface. During the transient handoff period the MH sends and receives the packets through both the interfaces. The packet filters at the media gateway and the MH discards the duplicate RTP packets. As soon as the packets reaches the MH through the newly activated interface, a re-INVITE message is sent to the CH with the IP address for the newly active interface and the corresponding contact information. As a result, the call parameters are re-negotiated on a an end-to-end basis, with the selection of a new intermediate SIP proxy server and B2BUA belonging to the base station corresponding to the newly activated interface. Once the call re-negotiation is complete, a BYE message is sent to terminate the call-leg through the initial interface, as soon as a duplicate packet reaches the newly activated interface. Finally, the MH registers its new location information with the home network's registrar service by using REGISTER

[1]The proposed architecture has been designed primarily for downstream traffic from the CH. The MH would typically need an RTP replicator for implementing soft handoff for upstream traffic as well.

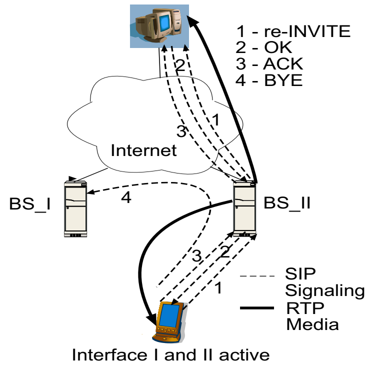

Figure 7. Signaling to update the ongoing session parameters on account of the change in MH's IP address

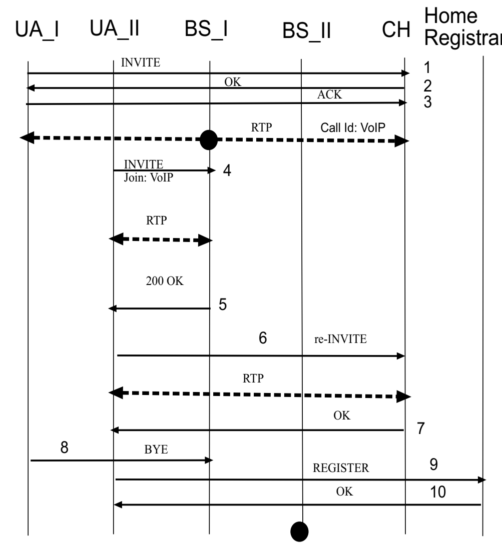

Figure 8. Message Diagram

message. The soft handoff procedure is further illustrated with the following example.

4.2 An Example

The SIP-based soft handoff is illustrated with an example in Figures 5-7. Figure 5 shows that a session has already been in progress between the correspondent host (CH) and the mobile host (MH). The CH and MH belong to different subnet domains with SIP URIs, `CH@correspondent.com` and `MH@home.com`, respectively. The MH moves between two domains, *viz.* `visited_I.com` and `visited_II.com`. The corresponding base stations for the two domains are denoted as BS_I and BS_II with URIs as `BS_I@visited_I.com` and `BS_II@visited_II.com`, respectively. The MH has two interfaces, *viz.* UA_I and UA_II, through which it acquires IP address pertaining to the two domains.

When the MH moves from domain `visited_I.com` to `visited_II.com`, UA_II gets activated and it acquires an IP address by a mechanism specific to that particular network. The MH SIP UA, on detecting the newly activated UA_II interface, then sends an INVITE message, with a JOIN header option, to BS_I through interface UA_II. The INVITE message has the new contact address `MH@visited_II.com` for the MH in the `Contact` field. The SDP parameters are also updated with the newly acquired IP address. The JOIN header contains information (call-id, to-tag, and from-tag), which is used by B2BUA at BS_I to match the existing SIP dialog. BS_I then configures the RTP packet replicator and the filter for the particular ongoing dialog to send a copy of packets directed toward UA_I to UA_II and filter duplicate packets coming from the MH via the two interfaces. At the same time a SIP OK message is sent to UA_II. Therefore, for a transient period, the RTP packets reach both interfaces of the MH. The duplicate RTP packets at the MH are filtered by the packet filter and delivered to the upper layers, while those at the media gateway are filtered and sent to the CH. This is shown in Figure 6.

As soon as the MH starts receiving the packets through UA_II, it sends a re-INVITE message to the CH to renegotiate the session parameters on an end-to-end basis, with changed end points. As a result of session renegotiation, the path of the media packets gets straightened out and the CH communicates with the MH through BS_II. This is illustrated in Figure 7. As soon as a duplicate packet reaches the interface UA_II, the connection from UA_I is released by sending a BYE message to BS_I, so that it can delete the dialog information pertaining to the SIP dialog going through BS_I. The timing diagram for soft handoff is depicted in Figure 8 and the detailed description of each of the messages is given in Figure 9.

The handoff procedure is composed of the following major operations, each of which contributes to the handoff delay: (i) Network detection and address configuration operation performed by the MH. It depends on the networking technology and the MH's operating system. (ii) Sending the INVITE message with the JOIN header to BS_I. (iii) Sending the re-INVITE message to update the session with the new location parameters. The corresponding delays are denoted by t_{attach}, t_{join}, and $t_{re-invite}$, respectively. As mentioned before, these delays cause considerable packet loss, which aversely affects the QoS of multimedia streaming applications. The objective of this work is to nullify the effect of these delay components with soft handoff.

```
Message 1:  CH -> UA_I

INVITE sip:MH@home.com SIP/2.0
Via:  SIP/2.0/UDP <IP Address of CH>:5060
To:  <MH@home.com>
From:  <CH@correspondent.com>;tag=001
Call-Id:  VoIP
CSeq 1 INVITE
Contact:  <sip:CH@correspondent.com>
Record-Route:  <sip:BS_I@visited_I.com;lr>


Message 2:  UA_I -> CH

SIP/2.0 200 OK
To:  <MH@home.com>;tag=002
From:  <CH@correspondent.com>;tag=001
Call-Id:  VoIP
CSeq 1 INVITE
Record-Route:  <sip:BS_I@visited_I.com;lr>


Message 3:  CH -> UA_I

ACK sip:MH@visited_II.com SIP/2.0
To:  <MH@home.com>;tag=002
From:  <CH@correspondent.com>;tag=001
Call-Id:  VoIP
CSeq 1 INVITE

Message 4:  UA_II -> BS_I

INVITE sip:BS_I@visited_I.com SIP/2.0
Via:  SIP/2.0/UDP <IP Address of UA_II>:5060
To:  <sip:BS_I@visited_I.com>
From:  <sip:MH@home.com>;tag=003
Call-Id:  VoIP
CSeq:  2 INVITE
Contact:  <MH@visited_II.com>
Join:  VoIP;to-tag=001;from-tag=002

Message 5:  BS_I -> UA_II

SIP/2.0 200 OK
To:  <sip:BS_I@visited_I.com>;tag=004
From:  <sip:MH@home.com>;tag=003
Call-Id:  VoIP
CSeq 2 INVITE

Message 6 UA_II -> CH

INVITE sip:CH@correspondent.com SIP/2.0
Via:  SIP/2.0/UDP <IP Address of UA_II>:5060
To:  <sip:MH@home.com>;tag=002
From:  <sip:CH@correspondent.com>;tag=001
Call-ID: VoIP
CSeq:  3 INVITE
Contact:  <sip:MH@visited_II.com>

Message 7:  CH -> UA_II

SIP/2.0 200 OK
To:  <sip:MH@home.com>;tag=002
From:  <sip:CH@correspondent.com>;tag=001
Call-Id:  VoIP
CSeq 3 INVITE

Message 8 UA_II -> BS_I

BYE sip:BS_I@visited_I.com SIP/2.0
Via:  SIP/2.0/UDP <IP Address of UA_II>:5060
To:  <sip:MH@home.com>;tag=002
From:  <sip:CH@correspondent.com>;tag=001
Call-ID: VoIP
CSeq:  1 BYE

Message 9 UA_II -> Home Registrar

REGISTER sip:registrar.home.com SIP/2.0
Via:  SIP/2.0/UDP <IP Address of UA_II>:5060
To:  <sip:registrar@home.com>
From:  <sip:MH@home.com>;tag=005
Call-ID: abcd
CSeq:  1 REGISTER
Contact:  <sip:MH@visited_II.com>

Message 10 Home Registrar -> UA_II

SIP/2.0 200 OK
To:  <sip:registrar@home.com>;tag=006
From:  <sip:MH@home.com>;tag=005
Call-ID: abcd
CSeq:  1 REGISTER
Contact:  <sip:MH@visited_II.com>
```

Figure 9. Detailed Message Description

5 Proof of Concept

5.1 Testbed Setup

The proposed architecture has been implemented on Linux kernel 2.4.20 and user environment Redhat 9.0. A schematic diagram of the testbed is shown in Figure 10. Two different subnets with wireless access are created for testing the performance of the SIP based terminal mobility with soft handoff.

For demonstration purpose, both the wireless access networks are implemented with IEEE 802.11 based wireless LANs. Two Linksys 2.4 GHz access points are used with wired interface configured at 100 Mbps and the wireless interface at 54 Mbps. A 1 GHz, Pentium 4 desktop is configured as the base station and a 1 GHz AMD Athlon laptop is used as the MH with two PCMCIA Orinoco gold wireless cards. The SIP stack used to implement the SIP-based mobility and soft handoff is GNU `oSIP2 2.0.6` [17]. Note that `oSIP2` provides an API for the SIP message parser, SDP message parser, and library to handle "SIP transactions" as defined by the SIP standard. `Linphone 0.12.2` [14], a Gnome based SIP soft phone built on `oSIP2` stack is used as the SIP UAC and UAS for session management. `Linphone` allows the selection of network interface for SIP message communication, a feature which is required to initiate the soft handoff in our proposed architecture. The SIP B2BUA at the base station is implemented by modifying `siproxd 0.5.4` [26], a stateful SIP proxy also built on `oSIP2` stack. The proxy has been modified to

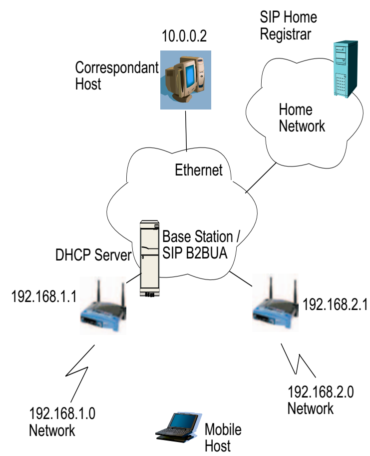

Figure 10. Experimental testbed setup

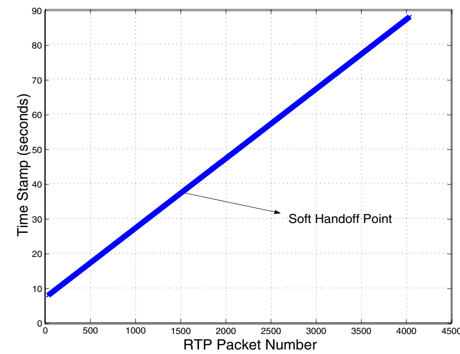

Figure 11. RTP stream at MH

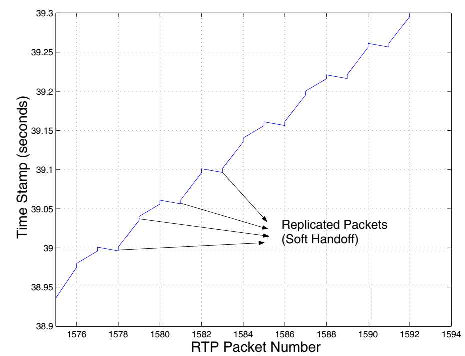

Figure 12. Soft handoff with replicated RTP packets

understand the semantics of the JOIN message in the context of soft handoff and to activate the packet replication rules. Although the latter is the job of the media gateway, in our testbed we have used a co-located SIP B2BUA and media gateway. Thus, both the functionalities are implemented by modifying `siproxd`. The packet replication function has been implemented using the `iptables` framework of linux 2.4 kernel. `iptables` is a generic table structure for the definition of rulesets which dictate, among other features, packet replication. The filtering of duplicate packets are done by RTP translators [25] based on the SSRC identifier values in the RTP packets, i.e., packets with identical SSRC identifier values are discarded.

5.2 Performance Measurements

The performance of the proposed soft handoff architecture has been measured in the above testbed describe above using a captured voice stream coded with Speex 8000 codec. Typical observed values of the parameters are as follows: $t_{attach} = 23.95369231$ secs, $t_{join} = 3.618$ msecs, and $t_{re-invite} = 359.84$ msecs, respectively. They are obtained as an average over 10 different handoff events. Figure 11 illustrates the RTP packet stream as recorded at the MH. To demonstrate the effect of soft handoff, the MH was moved into a new subnet after 15 secs. Because of the handoff delay components, the soft handoff procedure could not be initiated before 38.9572 secs. The soft handoff initiation point, indicated in Figure 11 is shown in further details in Figure 12. The vertical notches in the plot imply duplicated RTP packets received at MH, which are subsequently filtered out by the packet filter. The packet replication continued till the re-INVITE message updates the session parameter which enable the CH to redirect the packets directly to the MH at its new IP address. As expected, no packet loss was observed in the RTP stream.

Figure 13 shows the spacing in seconds between the consecutive RTP packets. For the purpose of clarity, a portion of the stream (from packet 1500 to 1650 only) is shown with a glitch in the inter-packet spacing, which indicates the point at which the MH stops accepting packets through the old interface and starts accepting them through the newly activated interface. The glitch results from the different routes taken by the packets directed toward the old and the new interfaces. The delay jitter is typically a measure of the difference in the end-to-end delay along the two different routes corresponding to the two network interfaces and is shown as the single spike in Figure 14. However, other than these glitches, the jitter remains under control all the time and has no long term effect on the streaming RTP traffic. The standard deviation of the observed packet spacings is 17.125 msecs and that for the delay jitter is only 0.112 msec. As mentioned before, such spikes in delay jitter can

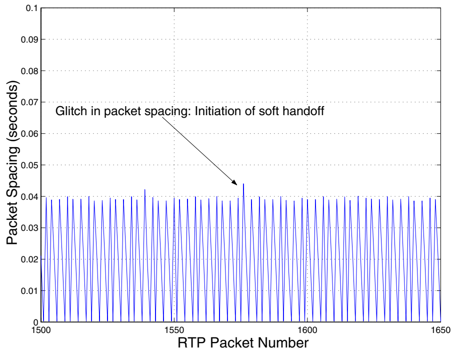

Figure 13. Spacing between the RTP packets

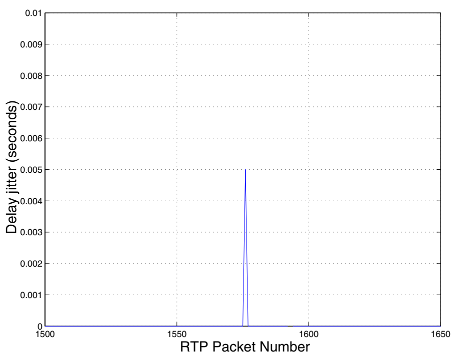

Figure 14. Delay jitter for RTP packets

be nullified by using a playout and jitter buffer at the terminal device, without any support, like soft handoff, from the network infrastructure.

6 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. To alleviate this problem, we have proposed a SIP based soft handoff mobility architecture for next generation wireless networks. A testbed has been setup to measure the efficiency of the proposed architecture. Experimental results show that the architecture is capable of ensuring zero packet loss and controlled delay jitter. Although in this paper we have considered vertical handoff only, SIP based mobility can also be potentially used to support horizontal handoff. However, the comparative performance against link layer handoff protocols is subject to further study. In the proposed architecture, we have assumed the presence of SIP proxy and the B2BUA in the base stations. For certain networks, this may not be possible and hence the SIP proxy and the media gateway locations need to be selected after careful studies. We would like to investigate into this issue further in the future by taking up case by case access network scenarios. Also, for the sake of developing a proof of concept we have used IEEE 802.11 based networks only, whereas the architecture has been proposed primarily for heterogeneous networks. We plan to test the architecture in a truly heterogeneous environment with different types wireless access technologies.

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