

Experiments and Analysis of Voice over Mobile IP

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ABSTRACT

Supporting mobility in real-time applications such as *Internet telephony* is in demand as mobile Internet becomes prevalent. A simple solution is to use *Mobile IP* in order to provide transparent handoffs to upper layer applications. However, due in part to the Mobile IP's limitations such as triangular routing delay and encapsulation overhead, application-layer solutions for mobile Internet telephony has been developed by extending *Session Initiation Protocol (SIP)*. This work presents experimental results on Internet telephony based on Mobile IP and examines its feasibility by analyzing delay factors and increased network loads. We found that packet size in terms of number of frames per packet is important in optimizing the packet transmission delay and network load without hindering the interactivity in voice conversations. Our major contribution is to give a real experience on voice over Mobile IP and to raise a need of designing delay-aware and/or load-aware scheme for Internet telephony.

Keywords

Voice over IP, IP Telephony, Session Initiation Protocol, Mobile IP, delay-aware, load-aware.

1. INTRODUCTION

Internet telephony is one of feasible Internet-based multimedia services with current Internet technologies, which can provide value-added features to existing Web-based services. *H.323* [1] and the *Session Initiation Protocol (SIP)* [2] have been proposed for implementing the Internet telephony. However, with the increasing popularity of PDA, laptop computers, and wireless LANs, Internet telephony users want to remain online even though they are on the move.

A simple solution to support mobility is to use *Mobile IP* [3]. However, the major weakness of Mobile IP is the *triangular routing*; i.e., packets have to pass through a *home agent* and then a *foreign agent* before they reach a mobile host. In addition to the latency problem, delay due to *handoffs* and *encapsulation overhead* in Mobile IP should be minimized to support real-time traffic.

Approaches based on SIP have recently gained widespread acceptance. *Moh et al.* [4] addressed some implementation problems for including mobility support to SIP. *Wedlund and Schulzrinne* [5] extended SIP to propose an application level approach for real-time mobile communication. If a mobile

terminal roams into a foreign network, the correspondent terminal can get the address of the mobile terminal by requesting to the *SIP redirect server* using *DHCP*. By moving the mobility handling to the application layer, they eliminated the need for *tunneling* required in Mobile IP.

However, their mobility support is limited to SIP-aware applications and SIP-aware correspondent hosts. Another requirement for using SIP-based mobility support is that networks should support DHCP because the SIP client needs a DHCP client to obtain an IP address. Also, when Mobile IP is used for other applications than SIP-aware ones, a certain coordinator is required. *Wedlund and Schulzrinne* considered the *Mobile Policy Table* [6] that maintains whether Mobile IP or SIP is used for each application. Since each host runs both Mobile IP-based and SIP-based applications, resources are wasted to keep duplicated information both in the SIP server and in the home agent. Moreover, researchers [7, 8, 9] believe that all mobility-related functionality should be handled at the IP (network) layer to enable the deployment of a homogeneous, IP-based wireless access network that is independent of the different wireless interfaces and applications.

Our objective in this work is to present an experience in Internet telephony over Mobile IP. *Real-time transport protocol (RTP)* is used for real-time multimedia data traffic. Various factors that affect delay, packet loss, and network load are investigated to maximize the bandwidth utilization and, at the same time, to keep the perceived quality at the receiver in terms of packet transmission delay of voice data. We found that packet size plays an important role in this regard, and optimal number of frames per packet has been explored via experiment. To the authors' best knowledge, there has been little work reported on Mobile IP experiments with real time applications.

The rest of this paper is organized as follows. In Section 2, some related backgrounds on Mobile IP and SIP are provided. Section 3 and Section 4 present the testbed and experimental results of Internet telephony over Mobile IP, respectively. Finally, conclusions are given in Section 5.

2. BACKGROUNDS

In this section, we briefly describe the basic mechanism of Mobile IP and explain essential parts of SIP.

2.1 Mobile IP

Mobile IP [3] allows a mobile host (MH) to transparently communicate with other nodes in spite of changes in network attachment. A MH recognizes its new attachment when it receives an *agent advertisement message* from a *foreign agent (FA)* not from its *home agent (HA)*. The MH, then, contacts the FA to obtain a *care-of address* and registers its HA with the information on the new physical location. When a *correspondent host (CH)* sends packets destined to the MH, the HA intercepts the packets and redirects them to the MH's care-of address by tunneling as shown in Figure 1(1) and (2). When the MH sends packets to CH, however, they are sent directly to CH as shown in Figure 1(3). This packet routing mechanism is called *triangular routing*.

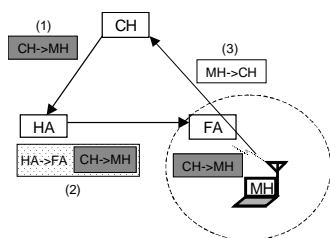


Figure 1. Triangular routing.

2.2 Session Initiation Protocol

The *Session Initiation Protocol (SIP)* [2] allows two or more participants to establish a session consisting of multiple media streams such as audio and video. The SIP also has been used in practical applications such as distributed games, shared applications, shared text editors and white boards.

Like *H.323*, SIP can be used for Internet telephony. While *H.323* is devoted to embrace more traditional circuit-switched approach to signaling protocol based on the *ISDN Q.931* and earlier *H-series*, SIP rather favors more lightweight Internet approach based on *HTTP* [10]. In SIP, callers and callees are identified by SIP addresses, email-like addresses. When making a SIP call, a caller first locates an appropriate server and then sends a SIP request, called *INVITATION*. Instead of directly reaching an intended callee, a SIP request may trigger a chain of new SIP requests by proxies. Users can register their location(s) with SIP servers. In this paper, we consider only peer-to-peer communications without any SIP server.

3. VOICE OVER MOBILE IP

In this section, we introduce our experimental environment for Internet telephony over Mobile IP. Then, we explain the packet format for voice traffic and the *codec*.

3.1 Testbed

Figure 2 depicts our experimental environment. It consists of a HA, two FAs, a router, a number of CHs, and a MH. *Dynamics* [12], a Mobile IPv4 implementation at *Helsinki University of Technology (HUT)* has been used in our testbed with *Linux OS*. *ORiNOCO's* 11 Mbps wireless *PCMCIA* cards are used for the wireless network interface cards. For voice communication, *Linphone* SIP application [13] is installed at two end-hosts, CH and MH.

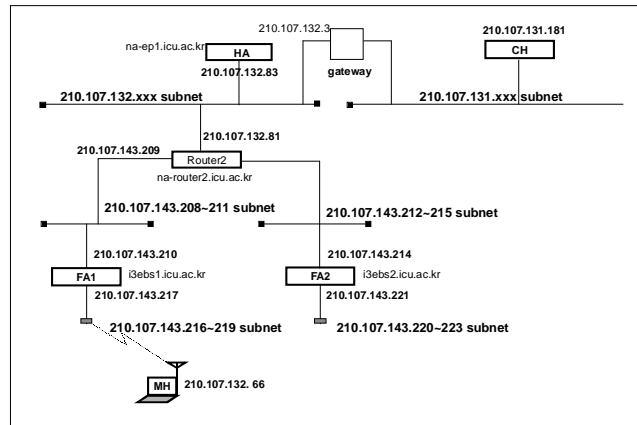


Figure 2. Testbed configuration.

3.2 Packet Format for Voice Traffic

Real-time transport protocol (RTP) is used for voice traffic of Internet telephony. Since RTP has a very compact header format, it adds little overhead to the application's data. This is an important feature in real-time applications because they tend to need a lot of bandwidth, and rely on small message sizes to reduce delay.

In our experiments, the length of an IP packet is 87 bytes including 20 bytes of IP header, 14 bytes of IP options, 8 bytes of UDP header, and 45 bytes of RTP message. 45 bytes of a RTP message consist of 33 bytes of voice data following the 12 bytes of RTP header. This is because *GSM codec*, which we have chosen as used in *Linphone* SIP application [13], produces 33 bytes of compressed frame at a time. This codec sends one frame per packet.

4. Experiment Results and Discussions

In this section, we show our experimental results of Internet telephony over Mobile IP by measuring interarrival time of voice traffic and loads on the network. Network load and end-to-end delay are measured with different packet sizes in order to find the appropriate number of frames per packet.

4.1 Interarrival Time of Voice Traffic

Interarrival time of voice traffic has been measured for both normal IP and Mobile IP in three different ways of calling; one-way, two-way, and two-way with background traffic.

In the first experiment (one-way), an Internet telephony client makes a call to another client, which answers automatically like an answering machine. The answering message is played for 14 seconds, and it corresponds to 700 packets with GSM codec that sends a packet every 20 msec. Figure 3 shows packet interarrival times with normal IP (Figure 3(a)) and Mobile IP (Figure 3(b)). Mobile IP exhibits wider range but 99% of interarrival times are quite similar: 19.95~20.05 msec and 19.91~20.09 msec with normal IP and Mobile IP, respectively.

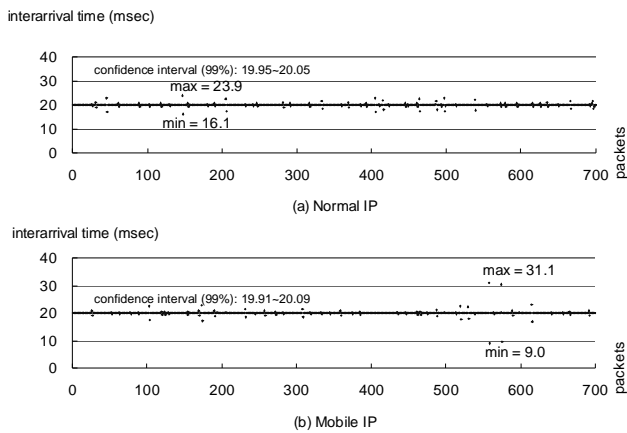


Figure 3. Interarrival times of one-way calling.

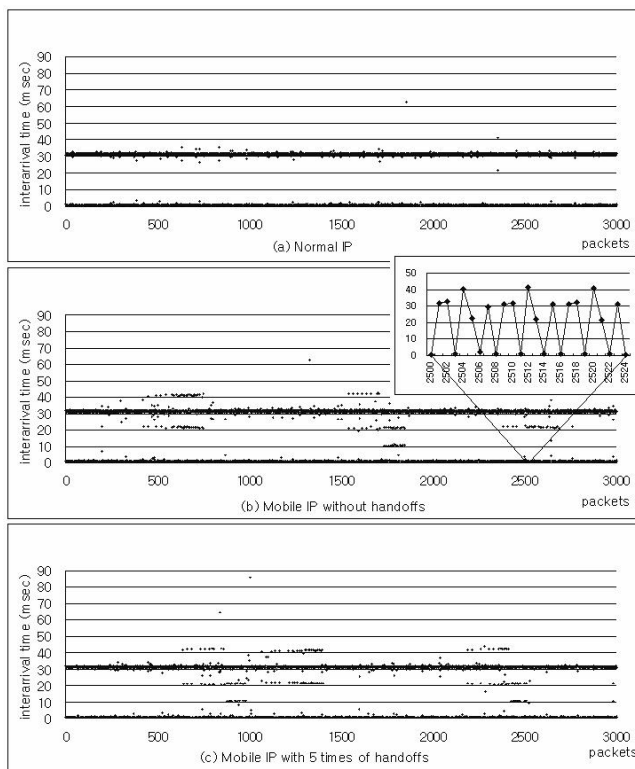


Figure 4. Interarrival times of two-way calling.

In the second experiment (two-way), interarrival times for bi-directional voice conversation have been measured for 60 seconds. Figure 4 shows the results for three cases; (a) normal IP, (b) Mobile IP without handoffs and (c) Mobile IP with five handoffs. As shown in the figure, two interarrival times, 2 msec and 32 msec, are dominant in those three cases. In case of Mobile IP, most packets arrive within 42 msec. In our experiment, very few packet losses are observed with Mobile IP because the MH on the move can receive packets from old FA as well as new FA. Note that it is possible because of large overlap area between two adjacent cells.

In the third experiment (two-way with background traffic), background traffic has been added to the network in order to see the feasibility of Internet telephony over Mobile IP in more

realistic environment. In this configuration, five more CHs generate traffic in the network, communicating with the MH as if there are five voice sessions. Each host sends the same number of packets at the same rate. Figure 5 shows the frequency of each packet's interarrival times for both normal IP and Mobile IP during three calls to answering machine that has a fourteen-second answering message (700 packets) for each call. It is observed that, between the results with normal IP and those with Mobile IP, there is no significant difference in terms of quality degradation due to packet loss and dispersiveness of packets' interarrival times. In both cases, there are a few packet losses (five with normal IP and six with Mobile IP), which make the interarrival times of the subsequent packets double as shown at the vicinity of 40 msec of interarrival times in Figure 5. Except those few cases, the longest interarrival time is 26 msec and 32 msec with normal IP and Mobile IP, respectively. Although the difference seems not negligible, over 98 percent of interarrival times are between 18 and 22 msec in both cases. Consequently, Internet telephony with Mobile IP is shown to be feasible in the context of our testbed configuration.

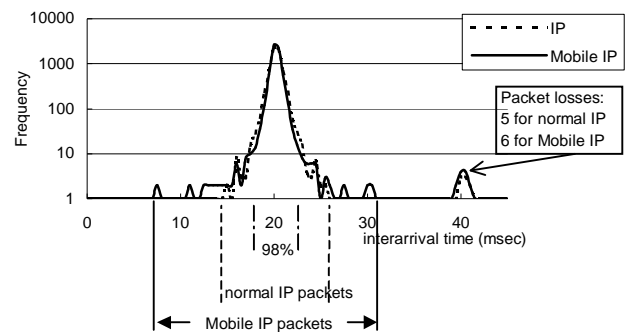


Figure 5. Interarrival times of two-way calling with background traffic (frequency).

4.2 Optimal Number of Frames in a Packet

In the last section, packet interarrival times have been analyzed but another important performance metric is network load. With the increasing number of MHs, Mobile IP network will suffer from scalability problem and thus, it is highly demanded to reduce the network load and to save the bandwidth of Mobile IP network especially for wireless links. In this section, network load and end-to-end delay during phone calls are observed with variable packet sizes other than of one frame (33 bytes).

Aggregation of more than one *frame per packet (f/p)* can save the network bandwidth by reducing the packet header overhead. In addition to the basic headers, in Mobile IP, every packet is encapsulated with an additional header of 20 bytes for tunneling from HA to FA. However, with the large number of frames per packet, users will suffer from delays.

Figure 6 shows the total data size and the total number of packets in the network during a three-minute voice call with different *f/p*'s. The total data size of one-way, raw voice data is 297 Kbytes (9216 frames), which doesn't vary with *f/p*. Consider the raw voice data of three frames (99 bytes) with *f/p*=1 and *f/p*=3. With *f/p*=1, the CH must send three 87-byte packets (54 bytes for

RTP, UDP, and IP headers, and 33 for a frame) to HA, where each packet becomes 107 bytes with the encapsulation header totaling 321 bytes. With $f/p = 3$, the CH sends one 153-byte packet (54 bytes for RTP, UDP, and IP headers and 99 bytes for three frames) to HA, where it becomes 173 bytes between HA and FA. With $f/p=3$, the total data size is reduced (58% between CH and HA, and 53% between HA and FA) but the latency becomes large. Therefore, the optimal number of frames in a packet must be carefully determined especially with Mobile IP.

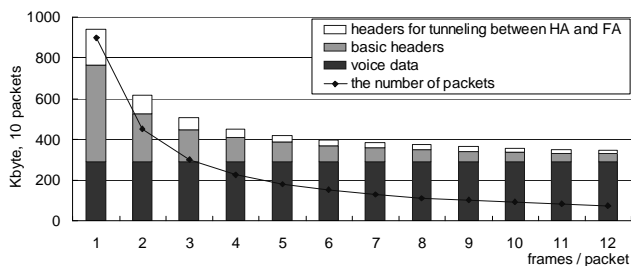


Figure 6. Total data size and number of packets with different frames per packet.

Now, the desirable number of f/p in Mobile IP is discussed with the consideration both of network load and latency. End-to-end delay is the accumulation of transmission processing and queuing delays in routers, propagation delays, and end-system processing delays along a path from source to destination. A study [11] shows that for highly interactive audio applications, such as Internet phone, end-to-end delays less than 150 msec are not perceived by a human listener; delays between 150 and 400 msec can be acceptable but not ideal; and delays exceeding 400 msec can seriously hinder the interactivity in voice conversations.

End-to-end delay is largely divided into encoding latency and packet transmission delay, where the former increases linearly (by 20 msec in GSM codec) as the number of frames per packet increases. If the maximum allowable end-to-end delay is 150 msec, the maximum packet transmission delays permitted can be easily deduced as plotted in Figure 7. The other line represents the network load in terms of bandwidth save with variable f/p 's based on the experiment of one-minute conversation with background traffic.

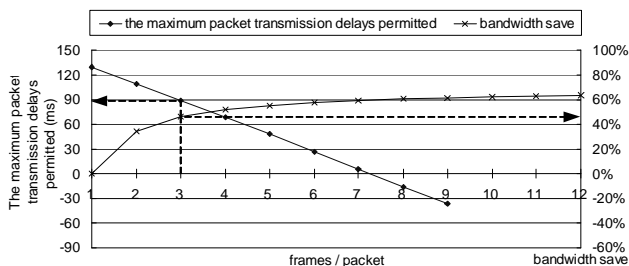


Figure 7. Desirable number of frames per packet.

Taking all information discussed above, if an Internet telephony application set the number of frames per packet to three (latency becomes 60 msec), the packet transmission delay up to 90 msec does not affect human perception. Regarding loads in the network, this can reduce 46% of the loads and makes it possible

to conversation with only one-third of the packets compared with the case of sending one frame per packet.

5. CONCLUSION AND FUTURE WORK

This paper investigated the feasibility of Internet telephony over Mobile IP. It is shown that Mobile IP's interarrival times do not vary much. Thus, it can be concluded that the Internet telephony over Mobile IP is feasible and it is desirable to send three frames per packet to reduce loads on the campus-sized network.

As a future work, we plan to study various kinds of codecs in the same context and to design a delay-aware and/or load-aware scheme for Internet telephony.

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