# Performance Evaluation of Two Layered Mobility Management using Mobile IP and Session Initiation Protocol

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***Abstract* – In this paper we present results, which have obtained by extensive simulations for Mobile IP and Session Initiation Protocol from the perspective of VoIP service in wireless Internet access. After illustrating the problem in these two protocols for diverse cases of mobility management, we propose an integrated model, to reduce the handover latency and packet loss during handover. This combination of network and application layer mobility management model reduces the global signaling load and provides fast handoff for ongoing conversations. The proposed approach needs no modification to the existing SIP message set and Mobile IP.**

**Simulation results presented in this paper are based on the NS2 Mobility Software[12]. However, since the current version of NS2 does not include SIP model for VoIP service, we add a suite of new features and procedures that are specific to this paper. The simulations results show that our proposed mechanisms achieve better performance than other protocols.**

## I. INTRODUCTION

In recent, we have seen a rapid growth in cellular mobile telecommunications and Internet penetration. Another important trend over the past few years is the emergence of the Voice over IP(VoIP) services and its rapid growth. The natural evolution of these technologies is towards a wireless Internet, which will provide access not only to real-time, but also to non-real time services from anywhere at anytime.

For the endpoint to take full advantage of mobility afforded by the wireless network, the host should be able to physically roam to any point on the wireless network while still maintaining any ongoing calls. In addition, a mechanism must exist for future incoming calls to reach the mobile node at its new address.

Several protocols and mechanisms have been developed to support inter-domain mobility and intra-domain mobility for multimedia services in the Internet. Most of the existing micro-mobility protocol proposals(such as Cellular IP, Hawaii or Mobile IP Regional Registration adopt a hierarchical mobility management architecture) assume Mobile IP[1] as global mobility protocol. In this paper we limit our attention to macro mobility.

Currently, there exist two basic approaches to support macro mobility in the VoIP services. The first one seeks to solve the mobility in the network layer by using Mobile IP and related proposals. Although Mobile IP is not directly related to VoIP applications, mobility support for VoIP service can be realized via Mobile IP. The other approach is to solve the mobility problem in application layer by augmenting the existing VoIP protocols such as H.323 and SIP.

A well-known problem with mobile IP is the triangular routing which adds delay in the network in classical Mobile IP case. These also require tunneling which adds the overhead for bandwidth constrained wireless link. While the SIP-based approach offers several advantages over a corresponding MIP-based solution, it continues to suffer from certain drawbacks. Main consideration in this paper is that handoff mechanism in SIP can cause call disruption if the new SIP session is not created completely while the mobile terminal is in the overlapped area. Due primarily to the amount of time it takes to perform the DHCP IP address renewal, there is a perceivable period of silence during call handoff.

Our main theme here is to compare the network layer solution(i.e. Mobile IP) with the application layer solution(i.e. SIP) to support terminal mobility in VoIP services and then propose a integrated model, to reduce the inter-domain handoff(also known as macro-mobility) delay in VoIP services in a mobile environments. This is a continuation of our on-going work in wireless/mobile networks.

Unlike Mobile IP, however, the proposed approach limits tunneling to TCP connections that are active during a movement. Also, for SIP traffic, it limits tunneling to traffic that are sent to mobile node until the re-INVITE is completed

In the rest of this paper, we provide some related background knowledge on mobility in VoIP service in Section 2. Section 3 is devoted to our solutions to mobility support in SIP. Section 4 shows simulation results run on ns2 to present that proposed approach can efficiently support real-time communications. Finally we give some conclusion remarks in Section 5.

## II. IP MOBILITY SUPPORTING FOR VOIP TRAFFIC

In this section, we describe the previous work related to mobility supporting for VoIP service.

### A. Mobile IP

Mobile IP is the oldest and probably the most widely known mobility management proposal[2]. Mobile IP consists of three major operations: Agent discovery, Registration, and Tunneling. Agent discovery is used to advertise the ability of mobility agents for services on each link. Registration is for mobile node to register with its home agent, and for mobility agents to provide registration service. When a mobile node is away from home, a care-of address is temporarily assigned to the visiting mobile node, either by the foreign agent, or by other means such as DHCP. Tunneling is suggested for mobility agents to forward packets destined to the mobile node to be routed correctly.

A well-known problem with Mobile IP is triangular routing. Route optimization solves this by sending binding updates to inform the sending host about the actual location of the mobile node. but, because of the requirements that are put on the correspondent hosts, it cannot be expected that route optimization will be widely employed in a near future. Several other drawbacks of route optimization are suggested in [3].

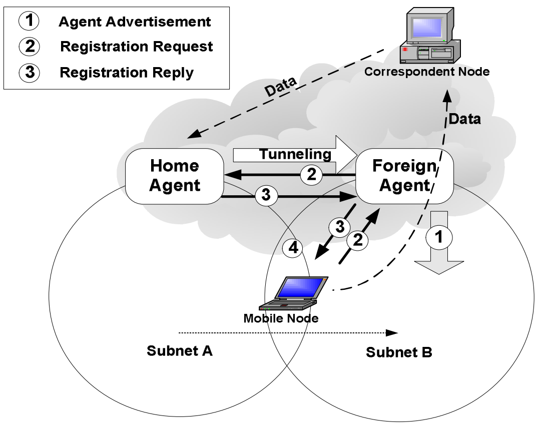


Fig .1. Mobile IP

**B. SIP Mobility**

Even though the original SIP protocol did not consider the mobility of the end nodes, there have been ongoing research efforts to support mobility in the current SIP protocol.

Wedlund and Schulzrinne proposed mobility support in the application layer protocol SIP where applicable, in order to support real-time communication in a more efficient way[3].

If the mobile node moves during an active session, first it obtains a new address from a DHCP server(or a variant of it), and then sends a new session invitation to the correspondent host(Fig. 2). With this new invitation, it tells its new IP address so as to forward packets properly. Actually, this invitation is nothing more than updating the current ongoing session description. While the SIP-based approach offers several advantages over a corresponding MIP-based solution, it continues to suffer from several drawbacks.

The most significant drawback is the absence of a mobility management for long-term TCP connection TCP.

Second, it can cause call disruption if the new SIP session is not created completely while the mobile terminal is in the overlapped area. As opposed to a mobile node using Mobile IP(when mobile node detect movements, it can obtain CoA from a foreign agent), a mobile node using SIP-mobility always needs to acquire an IP address via DHCP, which depending on implementation, can be a major part of the overall handoff delay. In [8][9], a empirical results show that some common DHCP implementations resulted IP address renewal time of more than 2 seconds, but if DAD(Duplicate

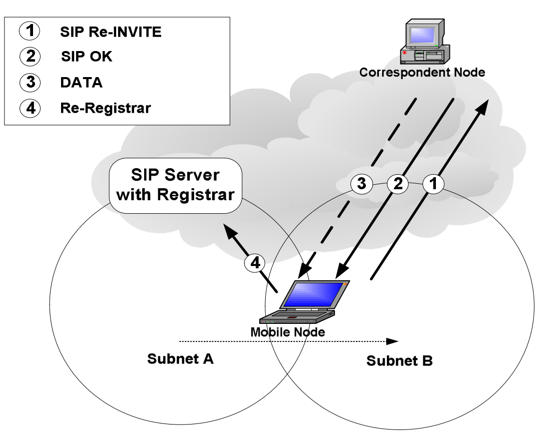


Fig .2. SIP Mobility

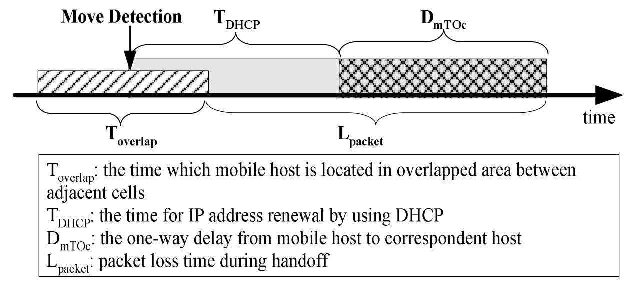


Fig. 3. The relation between packet loss and handoff delay

Address Detection) were removed DHCP delay was decreased to about 0.1s. For our approach, this disruption time(including DHCP delay) of SIP-mobility during a call re-establishment of SIP- mobility is complemented by using mobile IP. Fig3 shows the affect of the dwell time in overlapped area and $D_{mTOc}$ for packet loss during handoff in SIP-mobility.

## III. Proposed Architecture

The basic concept of the proposed architecture is that the SIP mobility support approach does not necessarily exclude the Mobile IP approach, rather it may work to complement based on the kind of application [3]. The network architecture of the proposed location management scheme is shown in Fig. 4. It uses a SIP network server, and a Mobility Agent with SIP Registrar to facilitate location management. While the SIP network server handles call/session delivery, the Mobility Agent with SIP Registrar is used for handling location registration, location updates, and location queries.

**A. Mobility Agent with Registrar**

The purpose of the registration process in mobile IP is to inform mobility agent of a mobile node's new IP address and update the binding information between home address of mobile node and the care-of address. This allows TCP or non-SIP connections to be maintained without a disruption. On the other hand, the aim of SIP session re-establishment is to inform location of a mobile node's new IP address to correspondent host(i.e., SIP UA). This allows correspondent hosts to communicate with mobile nodes directly. We propose the extension in home agent as the way for utilizing above two advantages.

For the purposes of our paper, it is useful to consider two types of binding. "user address binding", analogous to binding in an SIP Registrars, is the mapping between user-level identifier and a temporary IP address of host name, and "IP address binding", roughly the binding between a permanent IP address identifying a host to a temporary care-of address.

Fig. 5 shows a mobility binding table in home agent. The purpose of this table is to map a mobile node's *home address* with its *Current Location*(or *collocated care-of address*) and forward packets accordingly. Also, it is used to support the binding between a *mobile user's permanent identifier* and *his/her computer's actual IP address*. The former is designed for IP address binding, and the latter is defined for user address binding.

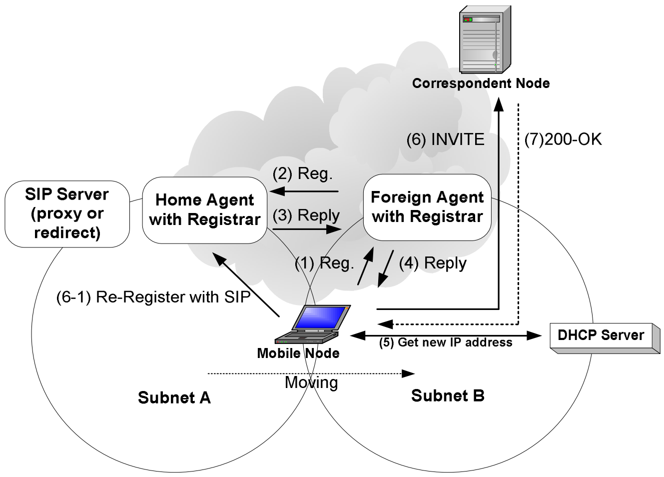


Fig. 4. The architecture of the proposed approach

In addition it may store other user profiles such as QoS requirements. This information is stored on a home agent using SIP REGISTER message when a host is first registered on the network. Also, if a user changes the device being used, it updates binding information, such as *UserURI*, *ContractURI*, *Home Address*, and *Current Location* by using SIP Re-REGISTER message.

However, if a mobile node being used by a SIP User Agent changes its location, it first registers its location(*CoA*) with home agent using a registration message in mobile IP. The home agent then updates *CoA* or *Current Location*(in case with Route Optimization). Then, after a mobile node acquires new IP address, a mobile node updates binding information between *UserURI* and *Current Location*. If home agent receives a registration message which is set 'D' bit, it updates *Current Location* instead of *CoA*.

By using this approach, the home agents do not tunnel data packets on behalf of mobile nodes after completing handoff for ongoing VoIP call, which represents a major difference compared to Mobile IP.

### B. Handoff

For a successful handoff, a mobile node registers with the affiliated home agent to indicate its location, followed by a dynamic join and departure of the multi-party conference for handoff. A location update may be performed upon the mobile crossing a service area boundary as described in the previous subsection.

Figure 6 shows the exchange of handoff-related message when a mobile node moves into a new domain while call in progress. The process of using proposed approach to handoff is outlined below. The mobile node not only obtains an initial care-of address, but also acquires globally unique Care-of Address(i.e. co-located care of address). When a mobile node is away from home, and it's connected to a foreign network, it detects is on a different network and sends a registration request through the foreign agent to the home agent requesting mobility binding update for a period of time. The home agent updates its mobility binding table and sends a

| *UserURI* | *ContactURI* | *Home Address* | *Current Location* | *CoA* | *Lifetime* |
|---|---|---|---|---|---|

Fig. 5. Mobility Binding Table in Home Agent

registration reply back to the mobile node through the foreign agent allowing or denying that registration. At the same time, mobile node must contact the DHCP server to obtain a new IP address.

Finally, to register its new IP address, a mobile node sends a SIP re-REGISTER message to the home agent, although this does not factor into the handoff delay. The Request-URI field of the REGISTER message contains the home domain of the mobile node, the To and From headers are both filled with the SIP address(the NAI) of the mobile user, while the Contact header is used to store the new IP address of the mobile node. In Fig. 6, it is note that, unlike classical Mobile IP, a mobile node does not send any more registration update message in Mobile IP to home agent after acquiring the new IP address. Instead, the location information for the mobile node is updated by SIP's re-REGISTER(6-1 procedure in Fig 6 update "user address binding" and CCoA of the mobile node after IP renewal). Also, the tunneled route in mobile IP is only used for those sessions that were initiated before mobile host moved to the new network. That is, until a new SIP call is completed, all packets for old call are tunneled via home agent. Sessions initiated after the moving use mobile host's new IP address and take the direct path to/form mobile host. The Fig 6 ignores any AAA(Authorization, Authentication, Accounting) delays that would be incurred on inter-domain hand-offs as these will strongly depend on the architecture chosen.

## IV. Performance Evaluation

In this section we present our simulation results. Simulations are conducted to investigate two critical performance issues, i.e. packet loss during handoffs and handoff latency. The simulations are run using NS2(v2.1b9a) from Lawrence Berkeley National Laboratory(LBNL), which is widely used in the networking community to study IP networks. Because the current version of NS2 does not support SIP for VoIP service, we add a suite of new features(user agent, redirect server, proxy server, registrars) and procedures that are specific to this paper.

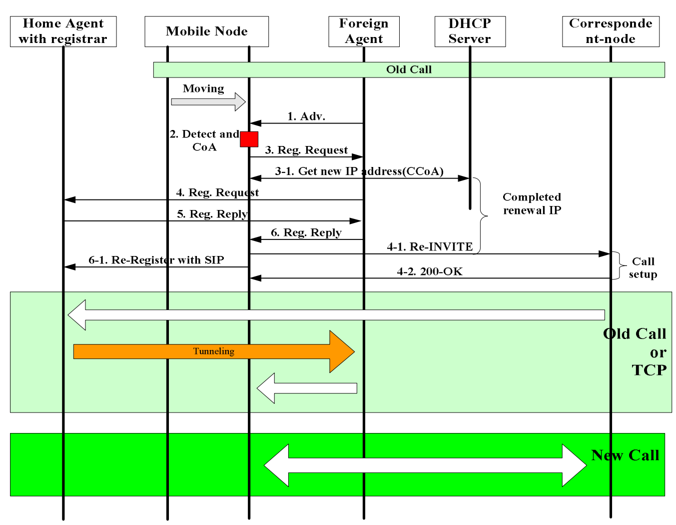


Fig. 6. The timeline for handoff procedures

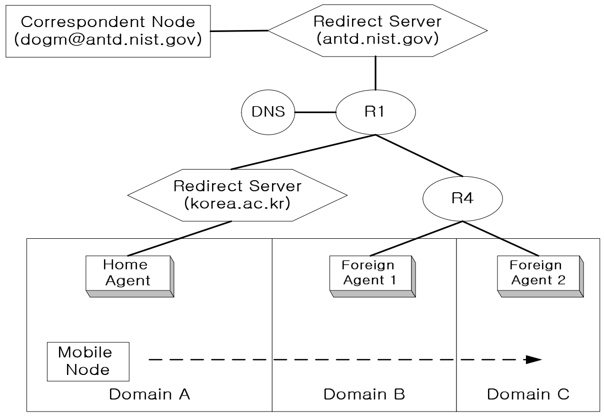


Fig. 7. The simulated network topology

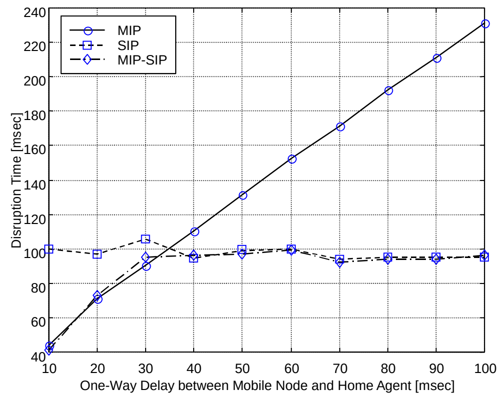


Fig. 8. Disruption Time vs. Delay between MN and CN

### 4.1 Implemented Simulation Model

The SIP simulation model is based on the latest description of the SIP protocol[1]. We implemented both Redirect Server mode and Proxy Server mode. That is, implemented User Agent, Redirect Server, Proxy Server, Registrar, DNS for SIP). And, two kind of VoIP traffic generator over SIP simulation model are implemented(one is CBR(Constant Bit Rate), second one is Exponential distributed traffic with mean values for the talk/silent times).

The SIP-mobility simulation model is based on the description provided in [3][4]. We used the re-INVITE message in SIP[1] for supporting direct communication. While SIP during handover need to move detection, related papers have mentioned that it is depend on low layer functionality such as network layer or link layer. In our simulation model, move detection function in SIP is implemented by network layer(i.e. mobile IP) functionality for fairness of movement detection.

For our simulation, we modified the existing mobile IP model in NS2 to support co-located care of address and extended mobile routing model to delivery packet between access point and mobile node without mobile IP. Also, we implemented two kind of binding mechanisms(see section 3.1) in Home Agent. Nodes are modeled without constraints on switching capacity or message processing speed.

All simulations are performed using the network topology shown in Fig. 7. The simulation environment consists of a correspondent node(CN) streaming audio(or VoIP) data over UDP to a MN, Home Agent with Registrar and Redirect Server. In this paper because we are interested in packet loss during call in progress and handoff latency, the CN acts as a CBR source, producing fixed length packets(200 bytes: payload of 160 bytes and a header(RTP+UDP+IP) of 40 bytes– typical of PCM coding schemes like G..711) at a rate of 64Kbps(corresponding to 20ms frames as, eg, in IS-95/cdma2000). The MN acts as a sink, receiving packets that arrive at a constant packet inter-arrival rate.

All routers in the simulated topology utilize a drop-tail queuing strategy. Also, A Mobile node connects to access points(APs) using the ns-2 carrier sense multiple access with collision avoidance(CSMA/CA) wireless link model where each AP operates on a different frequency band. And, the bandwidth and the link delay between base station and mobile node are set to 11Mbps and 2ms, respectively.

### 4.2 Simulation Results

In this section we will analyze the VoIP service performance decay involved by Mobile IP, SIP-mobility, and MIP-SIP integrated model in wireless access networks. We present simulation results for the hard handoff performance of each mobility protocol. An example of such a technology is the IEEE 802.11 standard for Wireless Area Networks[8].

In this paper we focus our attention on reducing the handover completion delay. Therefore, in our simulation we assume that the movement detection mechanisms of mobile IP uses ECS(Eager Cell Switching, to register with new foreign agents as soon as they are discovered) and SIP-mobility depends on the movement detection in network layer for the fairness of simulation.

In this paper, we consider three factors that affect the handoff delay and packet loss over wireless networks:(1) $T_{overlap}$: the dwell time which mobile node is located in overlapped area between old cell and new cell after detecting a movement. (2) $D_{oTOn}$: the delay to send a message between the New Foreign Agent and Old Foreign Agent. (3) $D_{mTOc}$: the one-way delay from mobile node to correspondent node. In this paper, we ignores the renewal delay of IP address(that is, CoA delay of mobile IP and DHCP delay of SIP-mobility) since we interested in the disruption time and packet loss caused from location update delay.

For VoIP traffic the hard handoff interval simply means a service disruption equivalent in length to the handoff interval.

Fig 8 illustrates the handoff disruption time as the delay $D_{oTOn}$ increases. Here the $D_{mTOc}$ are set to be 30ms. We plot the disruption time of macro-handoff in Mobile IP, SIP

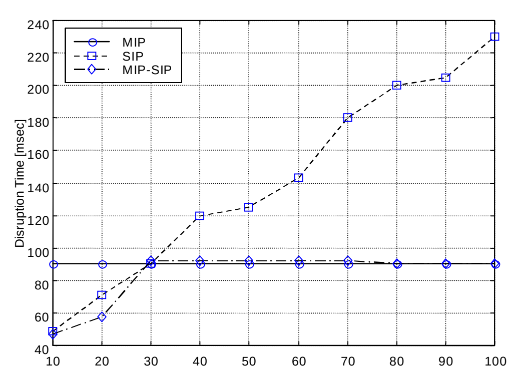


Fig. 9. Disruption Time vs. Delay Between MN and HA

approach, and the integrated mobility management proposed in the paper respectively. Each data point corresponds to the average of more than 100 independent handoff events. Obviously, the disruption during macro-handoff of SIP-mobility becomes shorter than that of Mobile IP approach as the distance between the MN and its home network increases since the SIP-mobility handoff mainly depends on $D_{mTOc}$. Thus, its disruption time is fixed about 95ms(about 1.8 packet loss, packet loss is roughly about 1.8 during 100ms in our simulation). However, the disruption time in mobile IP increases directly proportional to $D_{oTOn}$ since mobile node must register its IP address with home agent whenever it moves a new subnet. Finally, the integrated mobility management proposed in this paper shows more effective results since it depends on the more fast method between mobile IP and SIP-mobility.

Fig 9 shows the disruption time as the delay $D_{mTOc}$ increases. Here the $D_{oTOn}$ are assumed to be 30ms. As SIP mobility in only depends on the distance between the MN and CN, the disruption time of SIP-mobility increases according to the delay $D_{mTOc}$. Likewise, in Figure 8, we can see that our proposed approach reduce the average of disruption time. That is, our proposed model can complement the packet loss time of SIP-mobility during a call re-establishment of SIP-mobility by using mobile IP.

The Fig. 10 illustrates the Round Trip Time between MN and CN. The objective of this simulation is to find out whether our scheme can optimize the route between the MN and CN while the mobile node is in the midst of a VoIP session during handoff for three mobility protocol. From simulation results, it is clear that the proposed model is able to achieve route optimization by direct communication between MN and CN, and support more short disruption time than SIP-mobility.

## V. Conclusion

In this paper, we proposed an efficient approach to deal with handoff during ongoing call over VoIP service. Unlike the previous work, the proposed approach uses integrated procedures of mobile IP and SIP mobility for real-time communication over UDP.

Compared with other proposals, our proposal has three primary advantages. First, the proposed mechanism can reduce packet loss, and handoff latency by compensating mobile IP and SIP-mobility shortcomings mutually. Second, the only modification to the existing infrastructure is the extension in the home agent and the addition of a database to find registrars. Third, the complexity of the proposal occurs only in registration, call setup shares the single-lookup efficiency of SIP and is therefore relatively fast. Fourth, because it is integration of the two registration databases, there is no possibility for data to become inconsistent. Finally, this mechanism can reduce signaling overhead(the dual registration procedure imposes significantly more signaling overhead than Mobile IP registration alone, since SIP registrations must be refreshed frequently). That is, unlike classical Mobile IP, a mobile node does not send any more registration update message in Mobile IP to home agent after obtaining the new IP address. Instead, the location information for the mobile node is only updated by SIP's re-REGISTER.

The simulation results shows that the proposed approach outperforms the existing approach in most cases. We believe that the work presented here is an important step towards supporting VoIP service over wireless Internet.

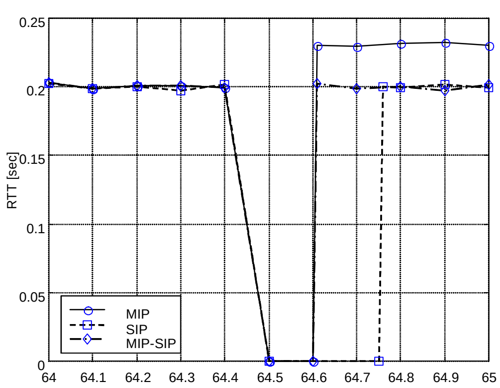

Fig. 10. Round Trip Time between MN and CN

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