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Efficient Architecture and Handoff Strategy used for VoIP Sessions in SIP Based Wireless Networks

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Abstract. In the near future, the Internet is likely to become an All-IP network that provides various multimedia services over wireless networks. Although the earliest VoIP applications did not consider the end-node mobility, researchers have attempted to support mobility in current VoIP protocols, such as Session Initial Protocol (SIP)-based mobility. The SIP-based mobility is considered because it can readily support mobility. However, calling disruptions may occur in traditional SIP mid-call terminal mobility because handoff procedure may be required, depending on the implementation and the real network deployment considerations. In any case, issues in the combined SIP/RSVP for guaranteeing QoS of VoIP service under mobile environment are also considered to be crucial. Therefore, this study describes the solutions by devising novel hierarchy network architecture. Also, the mechanisms including help with neighboring users in adjacent cells and the third party call control to overcome those issues are included. The simulation results indicate that the proposed technique is practical and better executive than conventional schemes.

Keywords: VoIP, IP mobility, SIP, RSVP, QoS

1. Introduction

Recently, the growth of Internet has led to an All-IP network to provide various multimedia services over wireless nodes. Researchers have attempted to support mobility in current VoIP protocols. Currently, two basic approaches are known for supporting mobility in VoIP services. The first approach attempts to support mobility in the network layer by using Mobile IP and its related proposals, while the other framework relies on the application layer by augmenting existing protocols, such as Session Initial Protocol (SIP). Although mobility for VoIP services can be supported via Mobile IP, there are some potential shortcomings, such as routing inefficiency, overhead problems, handoff latency, and stability problems as pointed out in [1–3]. Additionally, other studies [4, 5] have had demonstrated differences in performances between SIP-based mobility and Mobile IP. Most of them conclude that the Mobile IP approach can support non-real-time sessions while SIP-based mobility better supports real-time sessions, such as VoIP. Therefore, SIP based mobility seems to be the most suitable method to handle VoIP sessions for mobility support.

This study presents solutions to lower handoff delay time; thereby, reducing resulted packet loss, RSVP path re-establishment time and resources usage in SIP-based mid-call terminal mobility. Additionally, an RSVP incorporated environment is proposed by devising hierarchy network architecture and methods including helping neighboring users and the third party call control. The aim of the proposed scheme solicits some users in neighboring cells for assisting the mobile node (MN). The proposed scheme also aims to handle and achieve SIP mid-call terminal mobility procedures involving location update, and rapid RSVP path re-establishment by using third party call control mechanism during handoff.

Section 2 introduces related works. Section 3 describes the proposed approach and the simulation model. Section 4 presents the simulation results. The conclusion and the future work are given in Section 5.

2. Related Works

Herein, pertinent literature is surveyed. In [6], describes the structure of the SIP multicast mobility that is independent of the IP layer and extends SIP functions to support SIP multicast mobility. The proposed method avoids problems exiting in the IP multicast mobility by shifting the concept of multicasting and mobility from IP layer to session and application layers. In [3], two composite mobility management architectures based on SIP and Mobile IP are proposed to support multimedia services in a seamless fashion. The first approach uses SIP with IP encapsulation measures on correspondent node (CN) to support mobility for all traffic from or toward the MN. However, the second approach segregates traffic and adopts SIP with the NAT mechanism to support mobility for real-time traffic over UDP with Mobile IP supporting mobility for non-real-time traffic mainly TCP-based applications [3]. Also compares mobility schemes with a table, which shows the advantages of both the proposed methods over Mobile IP including the support of personal and session mobility, the lack of a single point of failure, and mobility awareness above the IP layer. Therefore, SIP-based mobility appears to be practical and possesses many useful features of supporting multimedia services in all-IP networks. Yet, whether to use SIP supporting mobility for all services, or to use SIP and Mobile IP for supporting real-time and non-real-time applications, respectively, is a question that should be further considered. Fortunately, [4, 5] perform simulations or laboratory experiments to solve this question. It shows the simulation and experimental results that SIP provides better throughput and performance than mobile IP owing to latency improvement and increased packet size utilization in real-time (RTP/UDP) traffic.

The study in [7] applies SIP-based mobility mainly to dominate mobility management for VoIP sessions as Mobile IP has some flaws in real-time application. However, calling disruptions may still occur in the SIP mid-call mobility. Thus, the proposed mechanism uses Mobile IP only to tunnel data packets initiated before the mobile host moves to the new network, while the sessions initiated after moving with the mobile host's new IP address and taking the direct path to/from the MN. However, Mobile IP must be integrated with SIP; therefore, the home agent (HA) or SIP proxy should be configured to store both "user address binding" (UserURI, and ContactURI) and "IP address binding" (Home Address, and care-of address information). This configuration may complicate the implementation and design the home agent or SIP proxy. Moreover, end-users also need to be aware of (what) and support both SIP and Mobile IP operations. However, our proposed method uses only a SIP related mechanism called third party call control. With the help from neighboring users, this method can handle the problems mentioned in previous section, but there is no non-SIP-based mobility approach that can be added to the system.

In short, previous works clearly show that SIP-based mobility is an appropriate mobility approach for supporting real-time multimedia services such as the VoIP service. However, calling disruptions may occur during standard SIP mid-call terminal mobility procedures. Also,

Table 1. Comparison of mobility management schemes

	Mobility approach					
Ability/feature	MIP	MIP with route optimization	SIP-based	Hybrid SIP/MIP	Multicast mobility in SIP layer	Third party call control with neighboring users help (proposed scheme)
Optimized routing	No	Yes	Yes	Yes	Yes	Yes
Personal and session mobility	No	No	Yes	Yes	Yes	Yes
No IP stack modifications	Yes	No	Yes	No	Yes	Yes
Seamless handoff support	Possibly	Possibly	Possibly	Possibly	Possibly	Yes
Suitable for which type of service	Non-real-time App.(TCP-based)	Non-real-time App. (TCP-based)	Real-time App. (UDP-based)	Real-time App. (UDP-based)	Real-time App. (UDP-based)	Real-time App. (UDP-based)
Need additional mobility approach other than itself based added to it	No	No	No	Yes	Yes	No
Concerning QoS guarantee also	Possibly (not specified specifically)	Possibly (not specified specifically)	Possibly (not specified specifically)	Possibly (not specified specifically)	Possibly (not specified specifically)	Yes

deploying the combined SIP and RSVP architecture can ensure QoS for VoIP applications, but when employed in wireless/mobile networks, it may also incur issues, such as RSVP path re-establishment. The next subsection discusses both these issues. Table 1 compares the mobility management methods discussed in this section.

2.1. Problems in SIP Mid-call Terminal Mobility

In conventional SIP mid-call terminal mobility [1, 7], as seen in Figure 1, the MN first detects its movement to the new cell, possibly a new wireless IP subnet, resulting in L3 handoff after entering the overlapped area. Then, the MN requests a new contact IP address from the DHCP server, depending on implementation and real network deployment considerations. However, this procedure can contribute significantly to overall handoff delay. Then, the MN can re-invite the CN (Macro-mobility) or re-register to the outbound proxy server (Micro-mobility) [1, 8, 9] to update its location. The new location thus becomes reachable by other hosts after handing off to the new cell. Nevertheless, if the new SIP session is not created completely while the MN is in the overlapped area [7], calling disruptions may occur. Figure 2 shows this phenomenon.