A Mobility Management Scheme using SCTP-SIP for Real-time Services across Heterogeneous Networks

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ABSTRACT

The Session Initiation Protocol (SIP) has been enhanced to support seamless mobility in heterogeneous networks. However, it still suffers from significant handoff delays and packet losses unacceptable for real-time multimedia services. In this paper, we propose a mobility management scheme for real-time services across heterogeneous networks that combines SIP and Stream Control Transmission Protocol (SCTP). Our scheme ensures transport-layer continuance through the multi-homing functionality of SCTP while maintaining location and network transparency through SIP without the use of re-invite message signaling. An evaluation of our approach demonstrates that it provides seamless mobility while satisfying the Quality of Service (QoS) requirements for real-time multimedia services without any changes in the underlying network infrastructure.

Categories and Subject Descriptors

C.2.1 [**Computer Communication Networks**]: Network Architecture and Design – *Wireless communication*

General Terms

Management, Design, Experimentation

Keywords

Vertical Handoff, SCTP, SIP, SIP over SCTP, RTP over SCTP

1. INTRODUCTION

In recent years, with the emergence of the 4th generation networks, various next-generation wireless access technologies have been introduced. Some of these include WiMax (IEEE 802.16) and the Universal Mobile Telecommunications System (UMTS). Together with the currently available wireless access technologies, provisioning of seamless mobility among these proliferated wireless networks has become an important issue in providing for real-time multimedia services (e.g., VoIP and MoIP).

In regard to mobility support, there are a number of protocols [1,2] available. Mobile IP [1] supports mobility in network layer through triangular routing. The Session Initiation Protocol (SIP) [2] provides application-layer mobility support through the use of signaling to update and maintain an application-level session. However, both protocols suffer from significant handoff delays and packet losses, failing to meet the Quality of Service (QoS) requirements for real-time services, due to triangular routing in Mobile IP and session re-establishment in SIP [3].

In addition to the network-layer and application-layer approaches that use mobility management protocols, other approaches [4, 5] have been proposed that use a transport layer protocol called the Stream Control Transmission Protocol (SCTP) [10]. In these approaches, the multi-homing feature of SCTP has been suggested for handling handoffs across heterogeneous networks. However, since SCTP was defined as a transport-layer protocol, it requires users to be aware of change of location and network type.

In this paper, upon taking these issues into account, we propose a vertical handoff scheme that combines SIP and SCTP in providing location and network transparency while ensuring the continuance of media transport. This paper is organized as follows: Section 2 introduces the background and related works. Section 3 presents our proposed scheme. Experimental results based on prototyping are depicted in Section 4. Lastly, in Section 5, we conclude this paper with suggestions for future work.

2. RELATED WORK

In order to provide seamless mobility across heterogeneous networks, several approaches have been proposed. In [11], the authors have evaluated the use of Mobile IP, including a mechanism that initiates a handoff between two different network interfaces. In their approach, Mobile IP initiates a handoff at the network-level, resulting in a noticeable handoff delay due to triangular routing.

In [3, 7], the authors have evaluated the use of SIP for a vertical handoff using bi-casting. By locating a base station between end hosts in order to send a duplicated media transmission to both interfaces during a handoff, the authors have proven that seamless mobility with zero packet loss can be achieved. However, their

SAC'09, March 8-12, 2009, Honolulu, Hawaii, U.S.A.

approach has the following disadvantages. First, a sudden spike in end-to-end jitter occurs due to the duplicate transmission of media traffic. Second, additional components within the communication of end hosts are required due to bi-casting, leading to infrastructural changes.

In addition to the network-layer and application-layer approaches above, the authors in [4, 5] proposed the use of SCTP to provide seamless mobility through the utilization of SCTP's multi-homing functionality and dynamic address reconfiguration (DAR) extension [6] proposed in mobile SCTP (mSCTP). Through the dynamic binding of a newly acquired IP address to a current session, the authors propose ways for how a handoff between two interfaces can be achieved. However, since SCTP is a transport protocol, it cannot achieve nor provide network and location transparency, requiring the user to be aware of a change of location and network type. Furthermore, the necessary measures regarding its use as a transport, especially for delay-sensitive real-time services have not been analyzed.

In summary, none of the approaches presented above are actually an applicable mobility scheme for real-time multimedia services across heterogeneous networks.

3. MOBILITY MANAGEMENT SCHEME USING SCTP-SIP

3.1 Overview

The goal of the mobility management scheme using SCTP-SIP is to ensure application-level session continuance without using the re-invite message signaling of SIP for a session re-establishment between end hosts. This is achieved through the multi-homing functionality of SCTP, where the IP address of a newly connected interface is dynamically associated to the current media transmission, thus eliminating the need for an application-level session re-establishment.

In our scheme, prior to a vertical handoff, the handoff decision of whether to switch among different interfaces currently attached to each network is analyzed through the use of preference based on the network characteristic and Signal-to-Noise Ratio (SNR) obtained for each interface. This, unlike a horizontal handoff, is because a Received Signal Strength Indication (RSSI), such as an SNR, does not necessarily relate to the quality of each network due to the diverse characteristics of heterogeneous networks.

As a handoff procedure is initiated, an MH uses a newly acquired IP address from the new interface. This is achieved through the multi-homing functionality of SCTP, where the Correspondent Host (CH) recognizes that a new IP address has been associated with the MH. After association, the MH assigns the CH to switch its destination address for sending a media transmission to the newly acquired IP address of the MH. As the incoming media transmission of an ongoing session is detected on the new interface of the MH, the MH assumes that a proper handoff for the CH has occurred. Then, the MH also starts to send its media transmission to the new interface, completing the vertical handoff without a transport-layer and application-layer session re-establishment. This transport-layer vertical handoff replaces the handoff of the conventional SIP. Thus, the application-layer session re-establishment is unnecessary. However, SIP still takes an important role in mobility management as it provides location and network transparency to an application by updating the SIP location and registrar server with the newly associated IP address. As the Uniform Resource Identifier (URI) of each client represents its identity in SIP, an update of the IP address matching with the MH's URI enables other clients to locate the MH even after a handoff.

Our scheme, unlike a handoff using the bi-casting approach for SIP proposed in [3, 7], does not require the involvement of additional components during handoff nor does it require the duplication of media transmission, which causes a sudden spike in end-to-end jitter. Additionally, our scheme ensures zero packet loss due to the handoff procedures wherein the CH and MH initiate the handoff of media transmission upon the certainty that the new interface is associated with the current transmission.

3.2 Handoff Decision

Our scheme uses a policy-driven handoff decision where a handoff is initiated based on user preference on the priority amongst network cost, speed, or battery usage together with RSSI. Our mechanism, which is similar to the approach proposed in [8, 9, 12], uses the maximum bandwidth of a particular interface and SNR as a factor for deciding on when to initiate a handoff.

In our scheme, as an MH detects an overlapped region of the multiple networks, the MH measures the SNR of each network's Access Point. If a newly attached network has lower priority, a handoff is not considered until the SNR of the current interface is decreased to a cell search threshold (CST). However, if the newly attached network has higher priority with an SNR higher than the fixed threshold, the MH initiates a vertical handoff immediately.

As an example, the following scenario can be considered. An MH currently attached to a WLAN (Interface 1) with a maximum bandwidth capable of 54Mbps moves into a WiMax (Interface 2) where the maximum capable bandwidth is 18Mbps for a downlink and 6Mbps for an uplink. Since the network performance for a WLAN is much greater than that of a WiMax in most conditions, as described in Figure 1, the MH maintains a connection with the WLAN until the CST. And as the SNR of the WLAN weakens below its CST, the MH assumes a handoff occurring toward the WiMax network and prepares for a handoff. Finally, when the SNR of the WiMax increases to its CST, the handoff is initiated.

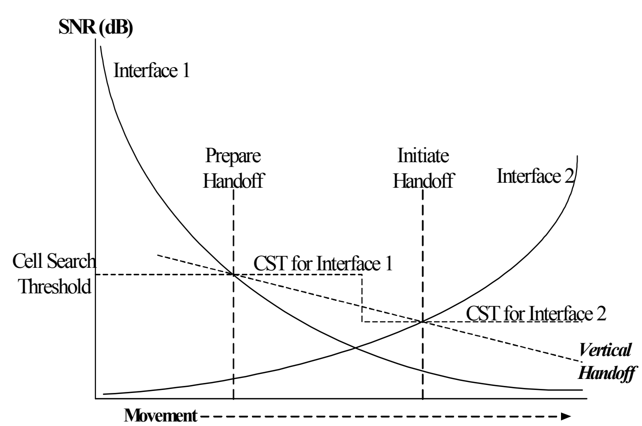


Figure 1. SNR relation regarding a vertical handoff

3.3 Handoff Procedure

This section provides the detailed handoff procedures of our scheme. Unlike the conventional SIP handoff procedure, in our scheme, a handoff is processed while the media transmission is ongoing using SCTP for transport. In our procedure, a handoff is carried out in three stages. In Stage 1 (Handoff Preparation), the MH and CH are notified with a new association. In Stage 2 (Peer Handoff), the MH assigns the CH to initiate the handoff where the destination address for the MH is changed to a new interface. In Stage 3 (Local Handoff), the MH changes its source interface to a new one, completing the handoff procedure.

In comparison with the handoff procedure described in [4, 5], our procedure differs from their single homing configuration where Stage 1 and Stage 2 of our scheme is viewed as a single handoff stage. In our scheme, we consider Stage 1 and Stage 2 as a separate handoff procedures since data exchange between two stages can still occur as long as MH maintains its handoff state to Stage 1. Though the overall delay caused from SCTP's handoff may be the same, our scheme decentralizes the handoff delay into two separate stages thus providing delay more viable to real-time services under conditions wherein combined delay in Stage 1 and Stage 2 causes packet drop. In addition, since the gap between two stages depends on MH's handoff decision, the impact of handoff delay can be reduced by offering real-time services enough time to accommodate the delay caused in each stage.

3.3.1 Stage 1 – Handoff Preparation

As an MH decides to initiate a vertical handoff, it prepares the handoff as shown in Figure 2. First, the MH associates the new IP address of the new interface attached to the current media transmission by (1). This is done through the use of a dynamic address reconfiguration extension for SCTP [6]. As a new IP address is bounden to the current media transmission using SCTP_BINDX_ADD_ADDR (new IP Address: xxx.xxx.xxx .xxx), in (2), an ASCONF message is sent to the CH notifying its media transmission that the new IP address on the MH has been associated. Then, the CH includes the new IP address to the current association and confirms through notification by sending back an ASCONF_ACK message by (3). Now, the CH is associated with both IP addresses of the MH for a single transport session. This means that when the CH receives media data from the new IP address, it will assume that the data belongs to the media session associated with the prior IP address of the MH and pass it to the application layer. This process eliminates the need for a new session establishment similar to that of TCP where a new IP address of an MH means a new transport session.

The preparation stage also provides additional benefit. After the preparation stage, if a sudden handoff occurs, SCTP within the CH will recognize its handoff through the missed arrival of the SACK message from the MH. Then, the CH will try to send media to the new IP address of the MH associated during the preparation stage. However, the preparation stage will not be informed to SIP since a change in an actual media transmission has not occurred.

3.3.2 Stage 2 – Peer Handoff

Since the association of the new IP address from a new interface has been established during the preparation stage, a handoff on the CH is initiated as illustrated in Figure 3. (1) The MH instructs the CH to change its destination IP address to its new IP address. This is done through a SET_PRIMARY_ADDR (new IP Address: xxx.xxx.xxx.xxx) instruction of SCTP wherein an ASCONF message is sent instructing the CH to change its primary destination address to the new IP address by (2). The CH confirms this instruction by sending back an ASCONF_ACK message by (3). However, even after a peer handoff stage, the old IP address of the MH is still associated since media from the MH to CH is still sent using the old IP address.

During the peer handoff stage, upon transition of a media transmission at the CH, SIP on the CH is informed that a change in association of media transmission has occurred by (4). SIP on the CH, then replaces the old IP address for the MH's URI with the new IP address informed based on SCTP by (5). However, this information is only updated on the CH and not on the SIP Registrar since the handoff procedure is still incomplete.

3.3.3 Stage 3 – Local Handoff

Illustrated in Figure 4, the MH initiates its local handoff using an SCTP_BINDX_REM_ADDR (old IP Address: xxx.xxx.xxx.xxx) instruction, wherein new media is sent from its new IP address of the new interface through the removal of the old IP address from its association by (1). Then, the MH notifies the CH with an ASCONF message instructing the CH to remove the old IP address of the MH from its association by (2). The CH, after removing the old IP address of the MH, sends a confirmation notice with an ASCONF_ACK message by (3).

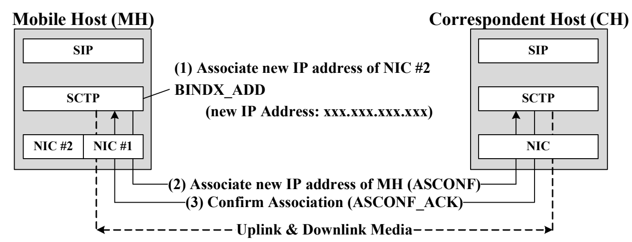


Figure 2. Stage 1 – Preparation

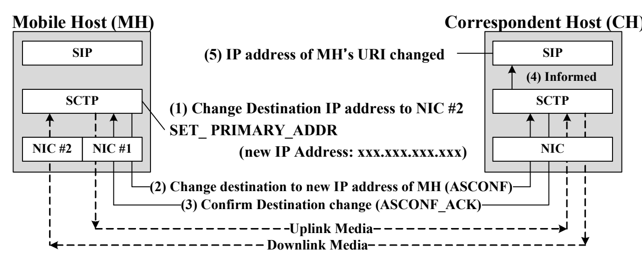


Figure 3. Stage 2 – Peer handoff

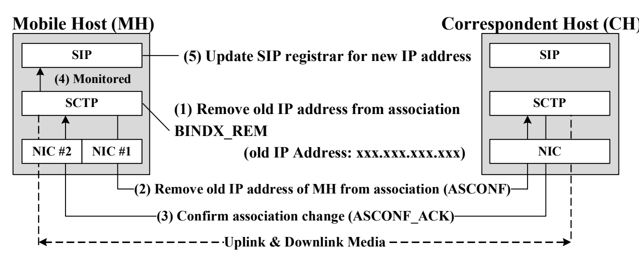


Figure 4. Stage 3 – Local handoff

As confirmation-acknowledgement ASCONF_ACK arrives from the CH, this information is informed to SIP by (4). Then, SIP updates its IP address to the SIP registrar through re-registration process where the SIP registrar updates the URI of the MH with a new IP address for the entire SIP components by (5). This process completes the vertical handoff procedure initiated by the MH. Upon completion, a media transmission is only handled by the new interface, and SIP components including SIP Location Server will recognize the new IP address as the destination for the MH.

4. PROTOTYPING AND EXPERIMENTAL RESULTS

4.1 Prototyping

As heterogeneous wireless networks for a vertical handoff, an MH is equipped with two wireless network interface cards that follow IEEE 802.11b and 802.11g standards. The first interface, namely NIC #1, connects to AP_1, while NIC #2 connects to AP_2. Also, we have set a higher priority for NIC #2 which has a maximum bandwidth of 54Mbps, while NIC #1 has 11Mbps. As illustrated in Figure 5, when the MH moves along to an overlapped region, a handoff is initiated. The maximum bandwidth for both interfaces was set to 2Mbps in order to evaluate the performance of our scheme for real-time services under variable conditions.

Our proposed scheme is based upon the use of SCTP and SIP. For SCTP, there are two prototypes available: SCTPLIB [17] and LKSCTP [15]. SCTPLIB is a user-level implementation of SCTP, while LKSCTP is a kernel level implementation of SCTP. In our experimentation, we use LKSCTP since, in [5], the authors have concluded that LKSCTP showed better performance.

For SIP, we used an application called Kphone [16] for a user agent to initiate a single VoIP session between the MH and CH. Kphone is a Multimedia over IP (MoIP) application that implements SIP for location transparency. However, since Kphone alone does not support SCTP for a media transport, we have modified the Kphone to both send a media transmission through SCTP and monitor the SCTP process to interact with SIP. Figure 6 illustrates the architecture of our prototype. In summary, SIP signaling messages are delivered over UDP, while a media transmission is delivered over SCTP.

4.2 Experimental Results

We have used the conventional SIP handoff approach used in a horizontal handoff as a comparison to our scheme for a vertical handoff. For evaluating the performance of our scheme, we measure the delay until a new media transmission arrives to the MH after a vertical handoff which, for our scheme would match to Stage 2. In our experiments, the media transmission is interrupted from the time the MH sends a re-INVITE message to the CH and the CH responds with a 200 OK. As illustrated in Figure 7(a), the results show that the handoff delay is about 100ms on an environment where the average RTT is approximately 15ms. A number of packets were dropped due to a timeout within Kphone.

As Figure 7(b) illustrates, a clear spike in packet spacing is observed due to a handoff. When a vertical handoff occurs, packet spacing between the last packet received from the CH before handoff and the first packet received from the CH after handoff increases marginally due to session re-establishment procedure on CH wherein CH queues the ongoing media session as it awaits acknowledgement for the 200 OK response message with session description protocol (SDP) message sent in accordance to re-Invite message arrived from MH.

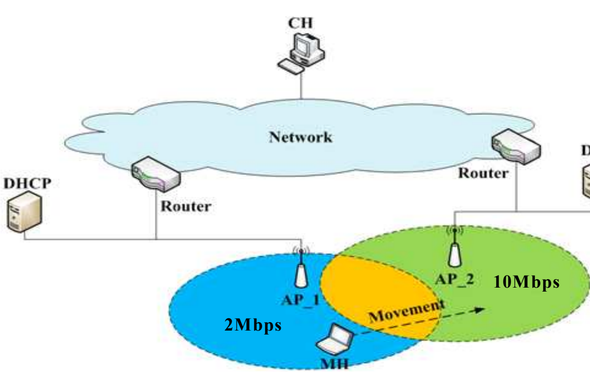


Figure 5. Vertical handoff scenario

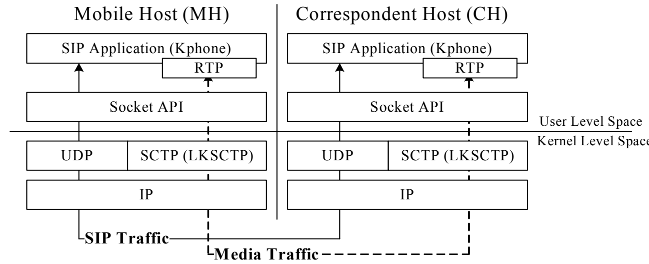


Figure 6. Software configuration

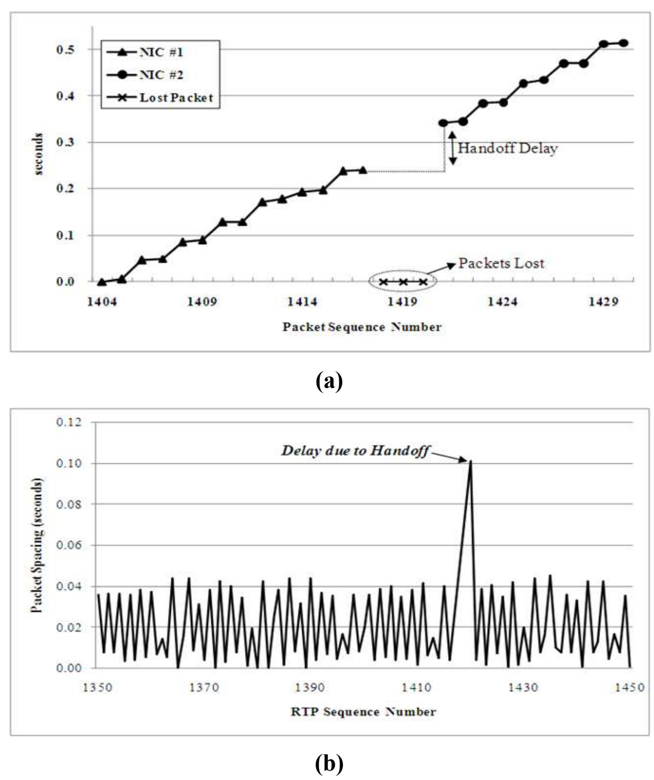
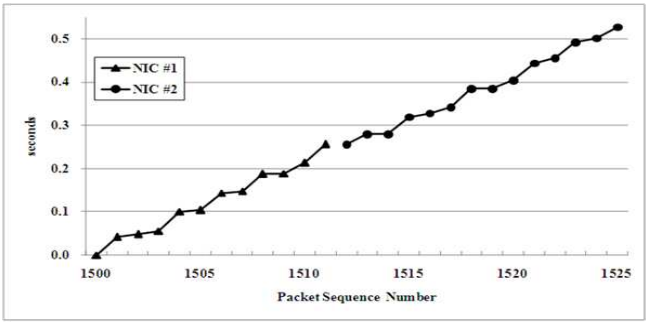
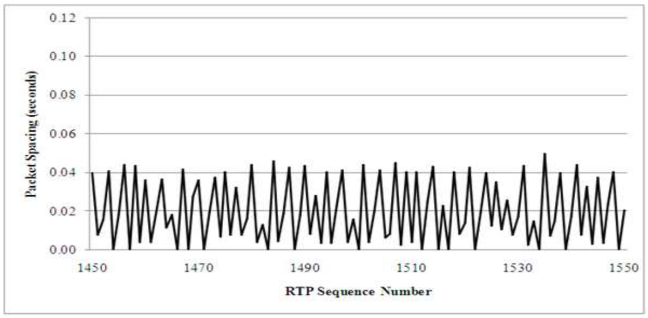


Figure 7. Handoff delay (a) and packet spacing (b) between RTP packets in conventional SIP at MH



(a)



(b)

Figure 8. Handoff delay (a) and packet spacing (b) between RTP packets using SCTP-SIP scheme at MH

In our proposed scheme, however, as Figure 8(a) illustrates, there was no queuing delay in the media transmission since the handoff procedure is carried out within media traffic. Thus, the handoff delay observed was equal to approximately that of a single RTT for Stage 2 of our scheme. In addition, as Figure 8(b) illustrates, packet spacing during the handoff showed no marginal spike during media transmission.

5. CONCLUSION AND FUTURE WORK

In this paper, we have proposed a mobility management scheme using SCTP-SIP for real-time services across heterogeneous networks. Our scheme employs the multi-homing feature of SCTP in stage by stage case bind together with the location and network transparency provided by SIP without the re-invite message signaling in order to provide efficient mobility management. In addition, our scheme does not require any additional components in the underlying network infrastructure.

Our evaluation has verified that the use of SCTP can achieve a performance nearly equal to that of UDP in terms of the QoS requirements of real-time multimedia services. Though the concept of the utilization of the multi-homing functionality of SCTP for a handoff may not be new, we would like to emphasize that SCTP alone cannot be viable for mobility management since it cannot guarantee the location transparency required for mobility management amongst peers. Hence, our combined use of SIP and SCTP provides seamless mobility for real-time services across heterogeneous networks where SCTP provides seamless handoff amongst media session while SIP provides appropriate update of peer's location to either server or the opponent peer. This, in comparison with the conventional SIP, gives advantage of avoiding the need for media session re-establishment on CH which causes certain delay as observed in our experiments. We plan to do further research for providing seamless mobility across heterogeneous networks utilizing different aspects of mobility management protocols.

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