Handoff Performance of Mobile Host and Mobile Router Employing HMIP Extension

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Abstract* – Mobility support for mobile networks will be more important as the mobile Internet becomes increasingly popular. To support mobile networks, the concept of prefix scope binding (PSB) is being discussed in IETF; however, by only applying this concept to Mobile IPv6 (MIP), the problem of packet loss still remains. In this paper, we apply the PSB concept to our proposed protocol, the Hierarchical Mobile IPv6 extension with buffering function (HMIP-B), in which mobility anchor points buffer packets destined to the mobile hosts during handoff. We compare MIP and HMIP-B based on the handoff performance related to mobile networks by computer simulation. The simulation results indicate that our proposal improves the handoff performance in both TCP and UDP communications.

I. Introduction

The vast address space of IPv6 will enable mobile objects such as cars, buses, trains, airplanes, or ships to carry an IPv6 network, in which many kinds of information devices can act as an IPv6 host having the IPv6 address. If at least one of the routers in the IPv6 network connects to a router on the Internet, any host in the IPv6 network can communicate with any host on the Internet. In this paper, we refer to such an IPv6 network and the router connecting to a router on the Internet as a mobile network and a mobile router (MR) respectively. The mobile network moves along with the mobile object while the MR is changing its point of attachment to the Internet. Such points are hereinafter referred to as access routers (ARs).

To support the mobility of mobile hosts (MHs), Mobile IPv6 (MIP) [1] is proposed in IETF, in which a MH must inform its home agent (HA) of the binding of its home address (HoA) and the current care of address (CoA). The HA forwards packets destined to the MH using an IP tunneling scheme. In order to support packet routing among mobile networks and the Internet with MIP, the prefix scope binding (PSB) concept is proposed in [2]. In that operation, the MR informs the HA of the binding of the prefixes used in the mobile network and the current CoA of the MR. The HA then forwards the packets that are destined to a node in the mobile network to the current CoA of the MR. However, MIP with PSB is supposed to incur severe packets loss when the MR performs a handoff similarly to MIP, which is supposed to do so when the MH performs a handoff.

In this paper, we introduce the PSB concept to our proposed protocol, in which a mobility anchor point (MAP) [3] buffers packets for the MH during its handoff [4], and evaluate the impact of the proposal on the handoff performance.

In Section II, we describe the handoff latency in conventional MIP. In Section III, we review our proposal and introduce the PSB concept to it. In Section IV, we review related work. In Section V, we present the details of the simulation models. In Section VI, we evaluate by computer simulation the impact of the proposal on the communications quality of a MH connected to a mobile network. Section VII concludes the paper.

II. Handoff Latency

MIP is originally designed to have no assumptions concerning underlying Layer 2 (L2), so that it has the widest possible applicability for L2 access technologies. However, such a clean separation of Layer 3 (L3) and L2 results in a time period referred to as the handoff latency [5], during which the MH is unable to send or receive any packets. Therefore, many packets are lost when the MH performs handoffs.

MIP has two main factors that cause the handoff latency. One is the disconnection period during L2 handoff. Generally, it takes a short time for the physical interface of the MH to change its single connection from the old AR to the new AR. Needless to say, during this period, the MH cannot send or receive any packets. The other is the signaling latency for a binding update packet (BU) from the MH to the HA. The HA continues to route the packets destined for the MH to the old CoA until receiving a BU informing the HA of the new CoA.

When we use MIP with PSB for mobile network mobility, these factors also bring about severe packet loss during the MR handoffs.

III. Proposal

A. Mobility management for MH

In order to reduce the handoff latency of a MH, Hierarchical Mobile IPv6 (HMIP) [3] has been proposed in IETF. The HMIP concept is introduced mainly in order to minimize BU signaling latency. In HMIP, packets destined to the MH are routed to the MH via the HA and a MAP. The MH has only to inform the MAP of the new CoA after handoffs within a MAP domain. This architecture can reduce BU signaling latency since it will take less time to update the MAP than it does the distant HA. However, this is not a perfect solution for packet loss. No matter how short the interval between the MH and the MAP, the L2 disconnection period may bring about non-zero packet loss.

To address this, in [4], we proposed a HMIP extension (HMIP-B) that employs a buffering function at the MAP, which is intended to prevent packet loss completely. In HMIP-B, 1-bit field (buffering flag) is added to the BU for a buffering request. The MAP buffers packets destined to the MH while it performs a handoff. Figures 1a and 1b show the operation of HMIP-B. The sequence of the handoff from AR_2 to AR_3 is described below.

1. Immediately before an L2 handoff, the MH sends a BU to MAP_1 with a buffering flag set. MAP_1 returns a binding acknowledgement (BA) to the MH.
2. On receiving the BU with the buffering flag set, MAP_1 starts buffering packets for the MH.
3. On receiving the BA, the MH performs an L2 handoff from AR_2 to AR_3.
4. On completion of the L2 handoff, the MH sends a router solicitation (RS) and receives a router advertisement (RA) from AR_3 and configures the new CoA.
5. The MH sends a BU containing the new CoA to MAP_1 in which the buffering flag is not set. MAP_1 returns a BA.
6. On receiving the BU in which the buffering flag is not set, MAP_1 quits buffering and sends to the MH all the buffered packets for the MH with the new CoA.

B. *Mobility management for mobile networks*

We can apply the same sequence to MR mobility by introducing the PSB concept to HMIP-B. All BUs should include prefixes used in the mobile network. Thus, the HA tunnels packets whose destination is within the mobile network to the address of the MAP and the MAP tunnels them to the CoA of the MR again or buffers them during the MR handoffs.

IV. Related Work

Before describing the simulation results, we present a review of related work. Due to the limited space here, we can provide only a brief overview.

The concept in itself of packet buffering for a loss-less handoff was proposed earlier. In [6], a buffering function is equipped in a foreign agent (FA) of Mobile IPv4 (MIPv4) [7]. According to [6], the buffering mechanism reduces UDP packet loss and improves TCP throughput compared to the original MIPv4. The same scheme for MIPv6 is also proposed in [8].

In particular, packet buffering at an FA is often studied to improve TCP throughput. Both [9] and [10] suggest the need for packet buffering in TCP communications. The delay caused by buffering, however, tends to degrade real-time communications. To address such a delay, in [4], we proposed the HMIP-B concept in which buffering is performed by the MAP because a packet buffered by an aggregation router (i.e. MAP) has a shorter delay than does that buffered by an edge router (i.e. AR) due to the elimination of a redundant route. We have also evaluated the handoff performance of HMIP-B in [11] and [12].

However, the effectiveness of packet buffering in a handoff related to a mobile network has not yet been evaluated sufficiently.

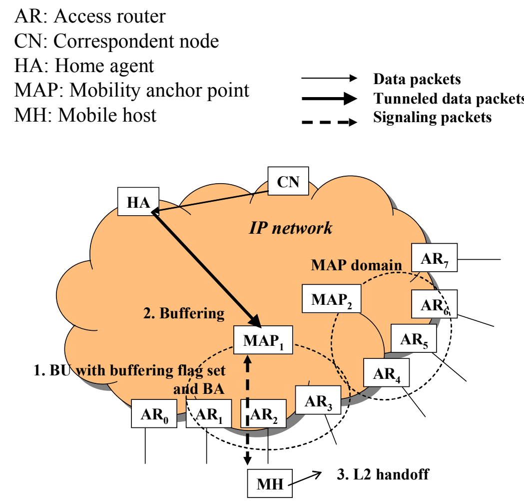

Figure 1a. HMIP-B Operation (a).

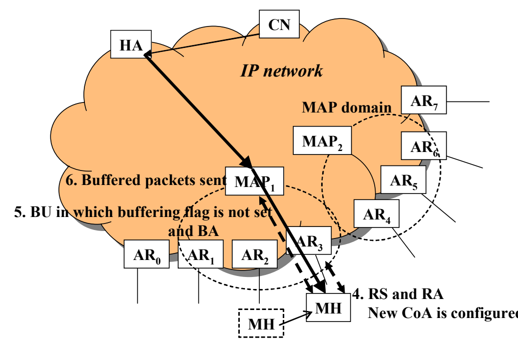

Figure 1b. HMIP-B Operation (b).

V. Simulation Models

A. *Simulation outline*

To evaluate the improvement by the proposed method, we compare MIP and HMIP-B by analyzing the performance in some types of handoffs related to a mobile network along the following scenario. In this scenario, a MH user travels on a bus that has a mobile network, the bus travels to the next stop, and the user debuses there.

In this scenario, 3 types of handoffs occur, which are given in Table I. A type A handoff takes place when the user boards the bus. The MH performs the handoff from the AR in a fixed

TABLE I
Handoff Types

Handoff type	Specification
Type A	Handoff performed by MH from an AR in a fixed network to AR in a mobile network.
Type B	Handoff performed by MR between ARs in a fixed network while the MH exists in the MR mobile network.
Type C	Handoff performed by MH from an AR in a mobile network to AR in a fixed network.

network to the AR in the mobile network. While the bus is moving, the MR performs a type B handoff between ARs located along the road. This handoff affects the communications quality of the MH in the mobile network. A type C handoff occurs when the user debuses, which means a handoff occurs from the AR in the mobile network to the AR in a fixed network. We analyze the communications quality of the MH for each of the 3 types of handoff.

B. Evaluation objectives

The communications quality in the handoff can be evaluated using the following objectives.

1. UDP communications

- Packet loss

In common real-time applications, packet loss concealment is adopted. When a packet is lost on the network, this technique interpolates the missing gap based on history data samples. However, due to handoffs, burst packet losses often take place. It is almost impossible even for packets loss concealment to hide such packet loss from the user because packets are lost in groups.

- Packet delay

Packet delay is also critical for real-time communications using UDP. For instance, ITU-T Recommendation G.114 [13] specifies 150 msec as the one-way delay to achieve high-quality voice and 400 msec as the acceptable upper bound. In the simulations, we monitor the delay of each packet on the network (from the physical interface of the sender to that of the receiver), which is not allowed to surpass 400 msec.

2. TCP communications

We monitor TCP bulk data flow in the handoffs. Some popular versions of TCP, e.g., Tahoe and Reno, are not originally designed to perform fast recovery from multiple packets losses within one congestion window [14]. Consequently, in the case where subsequent packets are lost in groups due to a handoff, a slow start algorithm can be observed.

C. Scenario components

We have eight scenarios for analyzing any handoff case. Scenario specifications are given in Table II. All the scenario components are described.

1. Network model

All scenarios use the common network model shown in Fig. 2. In this network, IP networks A, B, C, and D are connected to each other through the Internet using backbone links (shown as thick lines). HA_1 in IP network A is for the MH and the HA_2 in IP network B is for the MR. The correspondent node (CN) is located in IP network C. IP network D has 4 ARs and MAP_1. The mobile network comprises the MR, MAP_2, and an AR. The delays through the Internet, IP networks A-D, and the mobile network are 50 [15], 10, and 1 msec, respectively. The one-way delay of each backbone link is 1 msec. The bit rate of wireless links between the ARs and the MH is 20 Mbps [16] and that of the wired links is so high (100 Mbps) that it cannot be considered a bottleneck for any communications. AR_1 through AR_4 send RAs on their wireless link indicating the IP address of the MAP_1, and AR_5 similarly sends RAs on its wireless link regarding the MAP_2.

In all scenarios, the MH performs a type A handoff from AR_1 to AR_5 around 46 sec into the simulation. The mobile network starts to move with the MH and the MR performs a type B handoff from AR_2 to AR_3 around 122 sec. After that, the MH performs type C handoff from the AR_5 to AR_4 around 188 sec.

TABLE II
SCENARIO SPECIFICATION

Scenario	Traffic model	Mobility management protocol	L2 disconnection period (msec)
1	UDP	MIP	0
2		(with PSB)	50
3		HMIP-B	0
4		(with PSB)	50
5	TCP	MIP	0
6		(with PSB)	50
7		HMIP-B	0
8		(with PSB)	50

AR: Access router
CN: Correspondent node
HA: Home agent
MAP: Mobility anchor point
MH: Mobile host
MR: Mobile router
RT: Router

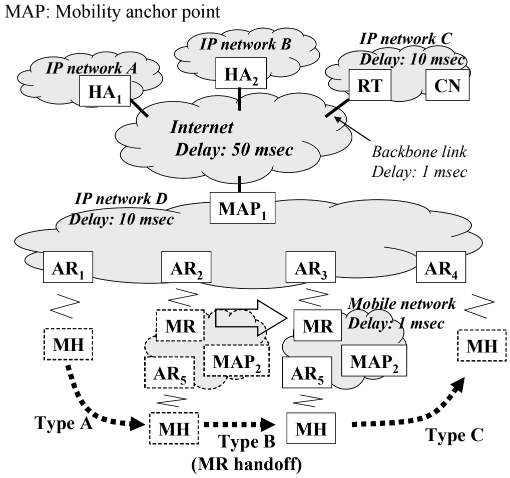

Figure 2. Network model.

TABLE III
TRAFFIC MODELS

Traffic model	Specification	Objectives
UDP communication	Application: VoIP (G.729) [17] Direction: CN => MH Packet rate: 50 pps Packet size: 80 bytes (IP header: 40 bytes, UDP header: 8 bytes, RTP header: 12 bytes, Data: 20 bytes)	Packet loss Packet delay
TCP communication	Application: FTP Direction: CN => MH TCP: Reno (on both ends) MH TCP receive buffer size: 8760 bytes	Data flow

2. Traffic model

Out of the eight scenarios, the first four scenarios are for analysis of UDP communications and the others are for TCP communications. The specifications of these traffic models are given in Table III.

3. Mobility management protocol

For each traffic model, we test MIP and HMIP-B. In Scenario 1, 2, 5, and 6, the MH uses MIP and the MR uses MIP with PSB. In the other scenarios, the MH uses HMIP-B and the MR uses HMIP-B with PSB.

4. L2 disconnection period

We set the L2 disconnection periods to 0 msec or 50 msec. For example, in Scenario 1, the MH and the MR require 0 msec to change the L2 connections from an old AR to a new AR.

VI. Simulation Results

A. UDP communications

Figure 3 shows the end-to-end delays of all packets in Scenarios 1-4. The horizontal axis represents the simulation time when each packet is sent out by the CN. The vertical axis represents the end-to-end packet transmission delay that each packet gains. As stated previously for these 4 scenarios, before the type A handoff and after the type C handoff, the MH is connected to an AR in a fixed network. During these periods,

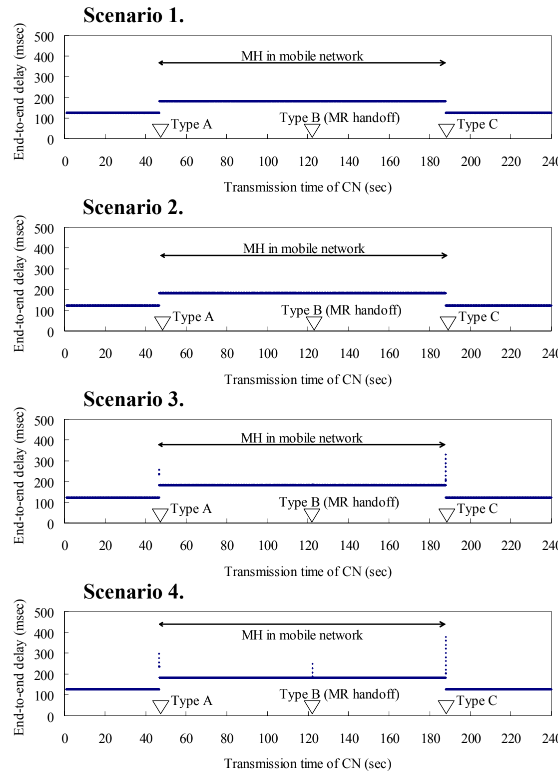

Figure 3. End-to-end transmission delay.

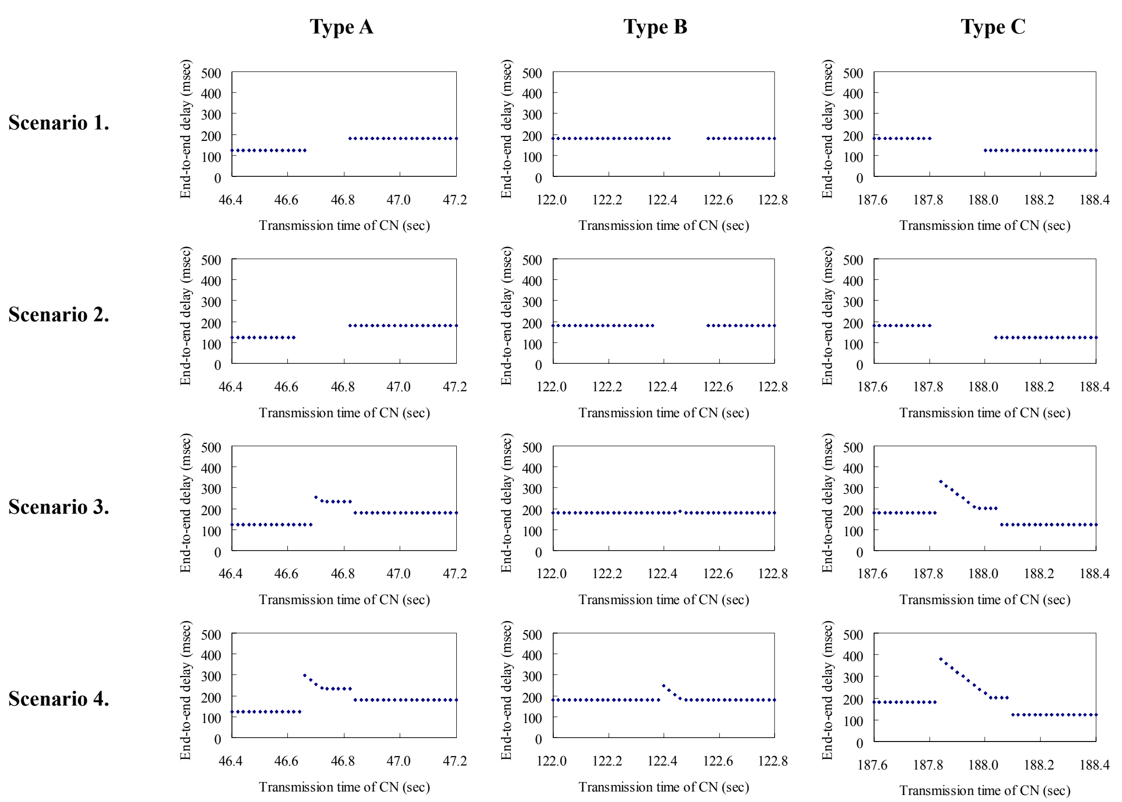

Figure 4. End-to-end transmission delay during each handoff.

packets destined for the MH are transmitted via HA_1 and the delay is approximately 120 msec. On the other hand, between the type A handoff and the type C handoff, the MH is in the mobile network. During this period, packets are transmitted via HA_1, HA_2, and MR; thus, the delay is approximately 180 msec. Note that, in Scenarios 3 and 4, some packets during handoffs suffer a longer delay than others.

Figure 4 is a close up of the time of each handoff in Fig. 3. In these figures, a gap represents the packet loss. The number of packets lost and the longest delay pertaining to each handoff are shown in Table IV.

In Scenario 1, due to type A, type B, and type C handoffs, 7, 6, and 9 packets are lost, respectively. In Scenario 2, the longer L2 disconnection periods generate more packet loss than in Scenario 1. These are undesirable results because more than 4 sequential packets lost causes a disruption in the streaming voice of longer than 80 msec, which corresponds to one phoneme in human speech [18].

On the other hand, Scenarios 3 and 4 realize lossless handoffs. We must note here that packets buffered in MAPs incur a longer delay than other packets; moreover, in Scenario 4, the longest delay reaches nearly 400 msec due to the L2 disconnection period. As described previously, this is close to the acceptable upper bound of the end-to-end delay of VoIP. However, the network model represents a very severe condition, in which it is difficult to make the end-to-end transmission delay sufficiently short for voice communications because both HAs are very far from IP network D. In actual situations, the distances between MH-HA1 and MR-HA2 will be usually shorter than those of this network model. The end-to-end delay then would be shorter than those in Scenarios 3 and 4.

B. TCP communications

Figure 5 shows the segment sequence number sent by the CN and Fig. 6 shows a close up of each handoff in Fig. 5. As shown in these figures, in Scenarios 5 and 6, the TCP data flow is disturbed by each handoff. These represent instances in which the handoff causes retransmission of TCP data segments with a slow start algorithm, not with fast recovery. This is because, as shown in the analysis of UDP communications, MIP incurs a service disruption period in which packets destined for the MH would be lost. This service disruption period often causes sequential TCP data segment loss; consequently, TCP Reno is urged to perform congestion avoidance with the slow start algorithm. In Scenarios 7 and 8, none of the handoffs disturb the TCP data flow since there is no packet loss in any handoff due to the buffering function at the MAPs. Even a 50-msec L2 disconnection period in Scenario 8 is sufficiently short that it does not cause a retransmission time out.

VII. Conclusion

In this paper, we confirmed the importance of mobility support for mobile networks. We reviewed the problems of Mobile IPv6 (MIP) and its extension with the prefix scope binding (PSB). Furthermore, we review our proposed protocol (HMIP-B) with PSB and evaluated the handoff performance of MIP and HMIP-B by computer simulation. The simulation results indicate that the proposed method improves the communications quality of both UDP and TCP over handoffs related to mobile networks.

TABLE IV
Packet Loss and Maximum Delay

Packet loss	Type A	Type B	Type C
Scenario 1	7	6	9
Scenario 2	9	9	11
Scenario 3	0	0	0
Scenario 4	0	0	0

Max. delay (msec)	Type A	Type B	Type C
Scenario 1	(182)	(182)	(182)
Scenario 2	(182)	(182)	(182)
Scenario 3	256	187	329
Scenario 4	296	246	379

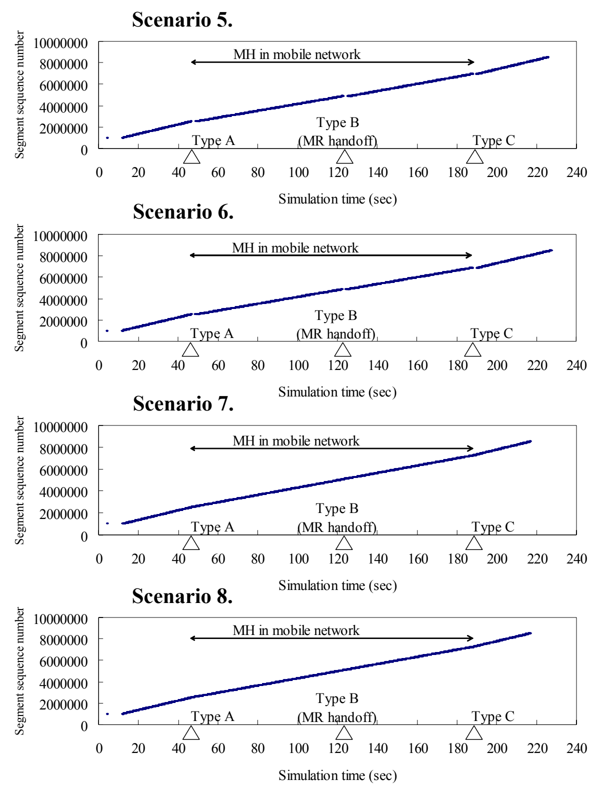

Figure 5. Sent segment sequence number.

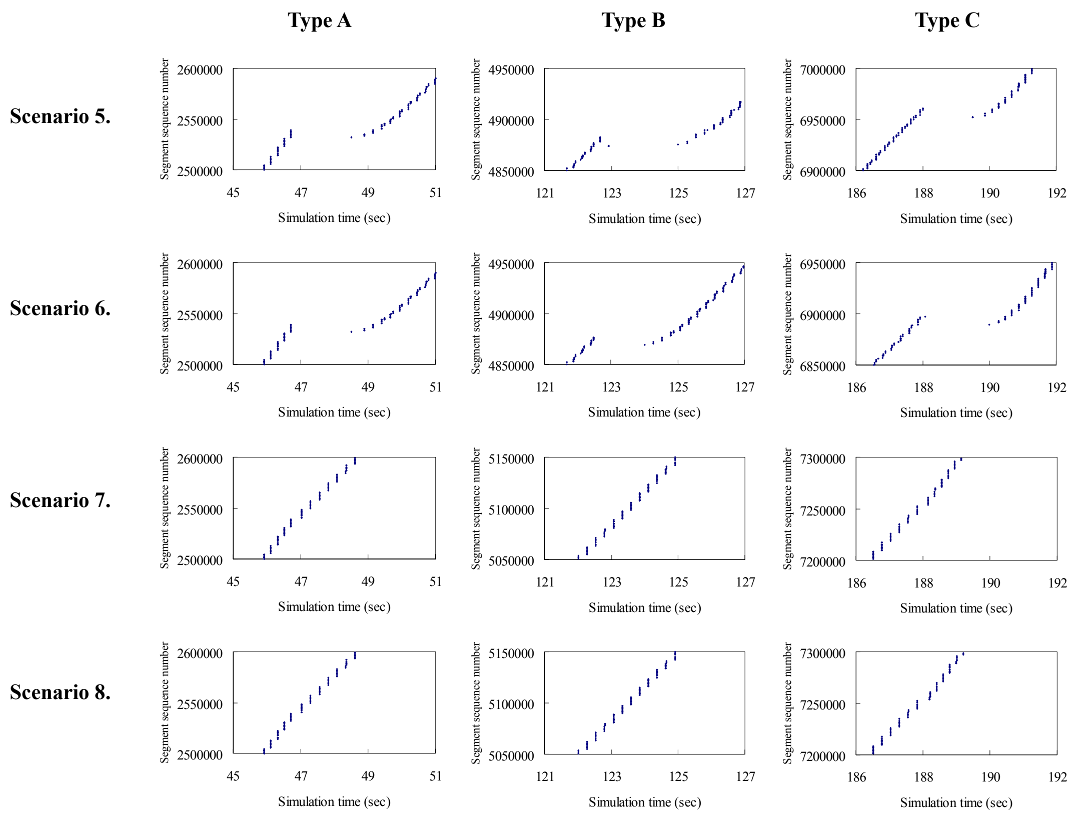

Figure 6. Sent segment sequence number during each handoff.

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