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Network-based Fast Handover for IMS Applications and Services†

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Abstract — In this paper we address the handover scheme based on the Session Initiation Protocol (SIP) for IMS applications and services. The existing SIP handover tends to suffer from the poor handover performance due to larger handover latency. To solve this problem, we propose a fast handover scheme based on network agent. In the proposed scheme, the SIP handover latency could be more reduced by using a new PRE-INVITE method with the help of the network agents. The proposed scheme is analyzed and compared with the existing scheme in the performance perspective.

Keywords — IMS, SIP, Handover, Mobility, Network-based

1. Introduction

Session Initiation Protocol (SIP) [1] is an application layer protocol used for establishing and tearing down the multimedia sessions of the IP Multimedia Subsystem (IMS) applications such as Voice over IP or multimedia conference [2]. It was originally defined by IETF and then adopted for mobile networks by 3GPP and 3GPP2. In the viewpoint of mobility management, SIP does not provide the complete functionalities. It supports the location management only, but it does not support seamless handover. Handover is the key feature to the mobility support. In order to provide full mobility service to the customers, we need an extension of SIP for seamless handover. Another limitation of SIP for mobility control comes from the application layer protocol feature. Compared to IP or transport layer protocols, the application layer protocols for mobility control suffer from long handover latency in nature [3]. There have been many researches that have tried to extend SIP to a full mobility protocol. Through these researches, SIP can support session mobility and slow soft handover service [1, 2, 5]. But that is not sufficient. Existing solutions use to adopt Re-INVITE concept to support session mobility but they still cannot support fast handover because the new SIP session is established after mobile node have associated to the new network. Re-INVITE eventually cause long handover latency. In this paper we propose a novel way to reduce long latency of SIP handover. We propose a network-based fast handover for IMS applications and services. Especially, we extend the SIP for seamless handover in the following principles:

1. Fast handover,
2. Cross-layer interaction,

In order to support fast handover service we use Pre-INVITE instead of Re-INVITE. Pre-INVITE is a simple concept that is to make a new session before the underlying network is changed. We also need to interact with the lower layer event for fast handover. It is noted that the IEEE 802.21 Media Independent Handover Services (MIH) [4] provides link layer intelligence and other related network information to upper layers to optimize handovers between heterogeneous media. So, we use event, information and command services of the MIH protocol. Most of handover operations are based on network entities. Mobile terminal just start up and complete the handover operations according to corresponding MIH events.

The rest of this paper is organized as follows. Section 2 describes the problems on existing mobility solution of SIP in the viewpoint of handover latency. Section 3 gives a detailed procedure of proposed scheme, network-based fast handover for IMS services and application. In Section 4, we analyze the handover performance of proposed scheme compared with the existing scheme. Finally, Section 5 concludes this paper.

2. Problems on Existing SIP Handover

Most of current SIP handover solutions can be summarized as Re-INVITE concept and its extension. To describe the problems on Re-INVITE scheme, we consider the IMS-based network framework as shown in Figure 1. In the figure, we simplify the network entities. Some of entities are removed because they do not take part in handover operation directly, i.e., application server (AS), home subscriber server (HSS), etc. Note that there is some overlapping area between two networks and the correspondent node CN has only one network interface card (single-homed).

Here is a typical SIP handover scenario: The user equipment (UE) which is connected to old network moves to the new network while it communicates with CN. After the network is changed, UE sends another INVITE message (Re-INVITE) to
CN. If the session is updated successfully, two hosts communicate with each other through the new network. However, they have to suffer from long handover latency. Sometimes this latency may bring significant result to the two hosts. This scenario is described well in the figure 2.

In this paper, we use handover latency as a difference between the time when the last application message is sent (or received) through the old network and the time when the first application message is sent (or received) through the new network.

In the figure 2, we can easily see why traditional SIP handover suffers from long latency. SIP handover latency is mainly caused by following three parts: (1) network link-switching time, (2) IP address configuring time and (3) session re-initiation time.

During link-switching time UE suffers from both handover delay and loss. Although we assume that there is some overlapping area between two networks, the new link can be up only after the old link goes down (Remind CN is single-homed).

New IP address configuring time is another reason for long handover latency. Current IP address can be configured by either state-full or state-less fashion. However, both ways are not suitable for fast handover because their timeout policy and duplicate address detection mechanism can cause long handover latency.

Moreover, typical SIP handover scheme have additional weakness, session re-initiation time. It takes at least 2RTT for UE to receive application data from CN through the new network.

For this reason, we focused on above three parts and tried to reduce the handover latency for each part basis. Proposed scheme is described in next section.

3. Proposed Scheme

In this section, we show the key ideas to reduce SIP handover latency and then propose the integrated solution for IMS-based applications and services.

3.1 Key Ideas to Reduce SIP Handover Delay

First of all, we have to reduce the link-switching delay. In the viewpoint of an application layer protocol, a mobile host that is about to change its network link to another one often experiences packet loss and service delay because it realize the truth that its link has been down only after it found any application data lost. Actually UE doesn’t have to wait the timeout event of the application data to know the network to be down. By using MIH functionalities, link switching delay can be reduced. MIH provides event services such as “Link Detected,” “Link Up,” “Link Going Down,” “Link Down,” and so on. If UE detects Link Going Down event, it has only to start its handover procedure.

We can also reduce session re-initiation delay by making new session before the old network goes down. We propose new Pre-INVITE scheme. By using Pre-INVITE we can make a new session before the old network is disconnected. However, we must have an IP address for the new network so that we can use our Pre-INVITE scheme. Unfortunately, we can’t wait for configuring the IP address for new network because it can cause quite long delay on IMS-based services, i.e., VoIP. This is the limitation of the SIP, the application layer protocol. Because the SIP operates at the application layer, its operations are done by end-to-end basis. So, there is nothing to do but all of the handover steps run sequentially. If we use SIP as a trigging protocol for handover and the network entities take part in the handover procedure actively, overall handover latency can be reduced greatly. By using network-based approach it is possible that handover functions are carried out simultaneously. For this way UE just try to switch the underlying networks while the network entities make a session for the new network.

3.2 Network-based SIP Fast Handover Scheme

We propose the integrated scheme for fast SIP handover. Our proposal includes the solutions described in previous section. The network topology and environment are exactly same to the things that are shown in figure 1.
We extend the SIP to the fast and seamless handover protocol. To support full mobility service, we defined several new protocol messages shown in following table:

<table>
<thead>
<tr>
<th>Message</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>HO.Req (SIP)</td>
<td>Request message that triggers the handover procedure. This message can include context information of UE, data buffering option, and so on.</td>
</tr>
<tr>
<td>HO.Ack (SIP)</td>
<td>Acknowledge for HO.Req. UE can know its new address, the address of next router and proxy server after receiving this message.</td>
</tr>
<tr>
<td>HO.Complete (SIP)</td>
<td>Complete message for overall handover procedure.</td>
</tr>
<tr>
<td>HI</td>
<td>Initiation message for network-based handover functionalities. HI contains all the information from UE. After receiving HI, the router acquires the candidate IP address of UE.</td>
</tr>
<tr>
<td>HAck</td>
<td>Acknowledge for HI. The proxy obtains UE’s new IP address from HI message and then makes a new session by sending Pre-INVITE to CN.</td>
</tr>
<tr>
<td>HC</td>
<td>Handover complete message used to transfer buffered data to UE.</td>
</tr>
<tr>
<td>Pre-INVITE (SIP)</td>
<td>Session initiation message. Almost same to INVITE message.</td>
</tr>
</tbody>
</table>

Overall handover steps can be classified into four steps: (1) detecting new network and preparing required information using MIH, (2) triggering handover procedure and acquiring new IP address, (3) making a new session and buffering application data and (4) completing the procedure and receiving the buffered data.

Figure 3 shows the detailed handover procedure.

In the figure, UE has connected to the old network and communicates with CN. When it receives Link Detected event from the MIH layer, it can retrieves information about available neighbor networks using the MIH Information service. Although the MIH specification defines many elements in this neighbor information [4], we use the minimum functionalities like an IP address of the candidate next router (GGSN_{NEW}). The Information Server and the detailed procedures for information query are not presented in the figure.

When the signal strength of the old link gets weaker than the predefined threshold, UE detects Link Going Down event from the MIH layer and triggers the handover procedure by sending HO.Req message to the old SIP proxy server (P-CSCF_{OLD}). The context information of UE must be included in HO.Req such as the identifier, MAC address, IP address of GGSN_{NEW}, security information like a session key, etc.

After received HO.Req, P-CSCF_{OLD} sends HI message to the GGSN_{NEW} with the received context information. And then GGSN_{NEW}, instead of UE, configures the new IP address for UE by DHCP or Auto-configuration mechanism. We assume that P-CSCF_{OLD} can configure UE’s address. We also assume that GGSN_{NEW} knows the IP address of the next SIP proxy server (P-CSCF_{NEW}). Configured IP address and IP address of P-CSCF_{NEW} are delivered to the UE through HAck and HO.Ack message.

When UE gets its new address and P-CSCF_{NEW} address, it attaches the new address into its network interface. It is noted that most operating systems allow attaching multiple IP addresses to one network interface card. After successful address attachment, UE can switch underlying network. At the same time, GGSN_{NEW} makes a new session for UE using Pre-INVITE message. If UE requested handover with data buffering option, GGSN_{NEW} should keep application data from CN until receiving HC message from P-CSCF_{NEW}.

Now, UE can associate to the new network link by using MIH LinkConnect command when it detects Link Down event. As soon as possible UE detects Link Up event, it have to send HO.Complete message to P-CSCF_{NEW}. Finally, it can receive all the buffered data and new application data from CN through the new network.

4. Performance Analysis

To show the feasibility, we have evaluated the proposed handover scheme by rough numerical analysis. Especially, we have compared the handover latency of two schemes.

In the traditional Re-INVITE approach, the handover latency, $T_{SIP\_HO}$, can be calculated as:

$$T_{SIP\_HO} = t_{LS} + t_{AC} + t_{SIP}. \quad (1)$$

In the equation, $t_{LS}$ represents the link switching delay taken for UE to perform the L2 handover, $t_{AC}$ is the address configuration delay required for configuration of a new IP
address via DHCP or Auto-configuration mechanism, and $t_{SGP}$ is the delay for exchanging the SIP Re-INVITE and OK messages [6].

In the case of the proposed Pre-INVITE approach, the handover latency, $T_{SIP\_HO\_NEW}$, can be calculated as:

$$T_{SIP\_HO\_NEW} = t_{LS} + t_{UP} + t_{GP}. \quad (2)$$

In the equation 2, $t_{UP}$ represents the link delay between UE and P-CSCF and $t_{GP}$ is the delay between GGSN and P-CSCF. We assume the two networks have similar characteristics for simplicity. Because establishing a new SIP session and switching underline network link of UE are performed almost simultaneously, $t_{SGP}$ is not even presented in the equation 2. Moreover, $t_{AC}$ doesn’t affect overall handover by help of MIH. For brief comparisons, we assume that all network links have the same round trip time (RTT). Then we can summarize

$$T_{SIP\_HO} \approx 3 \times RTT + t_{LS}, \quad (3)$$
$$T_{SIP\_HO\_NEW} \approx 1 \times RTT + t_{LS}. \quad (4)$$

It is noted that $t_{AC}$ implies two round-trip delays for DHCP message interactions [2] and $t_{SGP}$ implies one round-trip delays for exchanging the Re-INVITE and OK messages [6]. We ignore the effect of the duplicate address detection during the DHCP operation. Following figure shows the handover delay of two schemes according to equation 3 and 4.

![Figure 4. Handover latency of normal SIP and proposed Scheme](image)

In the figure, we can easily see that proposed scheme can reduce the handover delay effectively although we have assumed several conditions. As the average network round-trip delay increases, handover latency of the proposed SIP increases three times slowly than the traditional SIP handover scheme.

5. Conclusions and Future Work

In this paper, we have suggested a network-based fast handover based on Session Initiation Protocol for IMS applications and services such as Voice over IP or multimedia conference. The network-based mobility enables IP mobility basically without any mobile node’s participation like Proxy Mobile IPv6. In our scheme, the old SIP proxy server has the responsibility of handover procedure. When the mobile host receives the link layer event from MIH layer, it just triggers handover procedure using newly defined SIP messages. And the proxy server performs new IP configuration and new session establishment instead mobile host. Simultaneous operation of processing handover steps and switching the network link is the key point to fast handover. The result of analytic comparison shows that the proposed handover scheme can reduce handover delay effectively, compared to traditional SIP. We don’t have to hold on the quite old Re-INVITE scheme any more.

However, proposed scheme have several limitations to solve. First, the old proxy server has to configure IP addresses of the mobile host. Current implementations of DHCP and IP Auto configuration cannot support that. Second, both mobile terminals and network entities have to support MIH functionalities. We never win the fast handover until we get cross-layer interactions. MIH is useful tool for interacting with each layer. Finally, we have assumed there is some overlapping area, but we haven’t suggested the time duration for the mobile host to be laid on this area. In fact, the movement patterns and velocity of the mobile host affects the proposed handover scheme. If the mobile host moves so fast, the link down event occurs before it receives the handover acknowledge (HO.Ack) from the old proxy server. Then it detects the network has changed and tries to obtain new IP address. To make SIP as a full and fast mobility protocol, listed problems must be solved. We remain listed issues as our future works.

REFERENCES


