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Performance Analysis of Handover Schemes in Heterogeneous Networks*

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Internet protocol (IP)-based mobile systems are ushering new and faster technologies in wireless mobile networking. Therefore, the expectations from these network services with respect to data transfer rate and quality of service (QoS) are high. As such, meeting these requirements is the recent trend in modern wireless technologies. An important aspect regarding such improvements is the modification of handoff schemes between different networks. In this paper, we are focusing on the recent trends based on seamless handoff scheme in heterogeneous networks such as worldwide interoperability for microwave access (WiMAX) and long term evolution (LTE). The development of the session initiation protocol (SIP) Prior-INVITE scheme is an improvement on the earlier used SIP Re-INVITE method, which comes a long way in decreasing the average handoff delay. The performance analysis using software simulation on account of various parameters, such as handoff delay, cost of signaling and packet loss rate are accomplished in this work. The performance analysis demonstrates that the proposed scheme outperforms the ordinary cross-layer scheme and noncross-layer scheme in a vertical handover scenario.

Keywords: Delay; handover; IMS; LTE; MIP; SIP; WiMAX.

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1. Introduction

Even with all the development of voice call in telecommunication, the demands in this industry increased with the introduction of the internet and the services offered by it. The challenge, now faced by the telecom majors is providing a seamless service during vertical handovers. For the past several years, Voice over internet protocol (VoIP) is seen as a viable technology to meet the increasing demands in this sector, overcoming the constraints of a circuit-switched network. This broadband network system is actually a group of systems that use packet switching, i.e., Internet Protocol (IP) for transmitting information. Technically speaking, digital signals or information are formed into packets and then transmitted over a channel. Since mobile data is also converted into packets and transmitted digitally, VoIP brings uniformity in the whole network, thus enabling the seamless delivering of data and calls between different access technologies. Many broadband technologies use VoIP in order to provide multimedia services, executed using IP multimedia systems (IMSs). The new generation wireless networks are striving to achieve seamless integration of existing heterogeneous networks without making too many modifications in schemes with respect to protocol and signaling, rather than going for the development of completely new networks. This is achievable by integrating various wireless network systems. Two prime examples of such networks are: worldwide interoperability for microwave access (WiMAX) and long term evolution (LTE).

Even with supreme channel capacities in these networks, users experience delay and traffic congestion while a session is transferred from one network to another, which hinders seamless transmission. Hence, a seamless handover mechanism is necessary for good quality of service (QoS) in this regard. Wireless networks support two types of handovers: horizontal and vertical. When switching is between the access points or nodes in a single wireless access network, then it is called as horizontal handover. When it is between different access networks, then it is called as vertical handover; for example, the handover within LTE is horizontal while the handover from LTE to WiMAX is vertical. In heterogeneous networks, handover management protocols help in seamless handover.

In this paper, we have implemented a Prior-INVITE scheme which has evolved packet core (EPC) as the backbone architecture. The core network (CN) of the WiMAX and LTE access technologies are interconnected through the IMS architecture. The performance of the Prior-INVITE scheme is compared with respect to the ordinary cross-layer scheme and the noncross-layer scheme in a vertical handover scenario.[1] The vertical handover scenario is considered in this work where the mobile node (MN) is moving from a WiMAX to a LTE network. The performance of the proposed handover scheme is analyzed for parameters such as handoff delay, signaling cost and packet loss using Matlab simulation. The obtained results are compared against the existing schemes and it is inferred that the proposed scheme produced a marked improvement over the existing counterparts.

The rest of the paper is organized as follows. Section 2 reviews on the related works corresponding to handover techniques. Section 3 provides an overview of the network architecture. Section 4 describes the proposed handover method. Section 5 details about the mathematical modeling of the proposed system. Results and discussions are presented in Sec. 6, Section 7 provides concluding remarks.

2. Related Work

The advent of new access technologies in the communication domain offers a considerable increase in the QoS and other performance parameters. The percentage of customer share has been decreasing per access technology with the increase in the number of technologies. Thus, there is a need for a solution to deal with the interoperability between large numbers of access technologies. A detailed comparison of the session initiation protocol (SIP) handover delay between WLAN and UMTS has been done.[2] In Refs. 3, 4 and 5 the author has dealt with the problem of interoperability between WiMAX and 3G using SIP-based IMS signaling. LTE and WiMAX are currently the technologies offering the highest data rates and market growth. Thus, there is a need for the seamless integration of the two architecture. In Ref. 1, a new cross-layer approach with the integration architecture of LTE and WiMAX has been proposed. The performance evaluation was done in terms of handover latency and signaling cost. The packet loss rate is an important QoS criterion and was not considered in the cross-layer approach. In Ref. 4, an integrated architecture for LTE and WiMAX was proposed, including the packet loss rate analysis using the cross-layer SIP handover. The handover latency from WiMAX to LTE for five hops was found to be 150 ms, whereas the recommended value for latency is between 50 ms and 200 ms. Thus, there is a need for further improvement in the handover management protocol even after the integration of the cross-layered approach. In Refs. 6 and 7, the authors have proposed a completely new make before break approach in vertical handover signaling. Media Independent Handoff (MIH) portion of the signaling is completed even before the link goes down, and the number of messages after the link goes down is reduced considerably. We have applied this new make before break technique using the cross-layer approach for vertical handover signaling in the WiMAX–LTE environment in our proposed SIP Prior-INVITE scheme.

3. Network Architecture Overview

This section gives an overview of the handover scheme on network architecture, which is an integration of the EPC and IMS.

3.1. *EPC*

The core EPC network controls the UE and establishes the bearing network and servers. The important nodes of the EPC architecture are PDN gateway (P-GW),

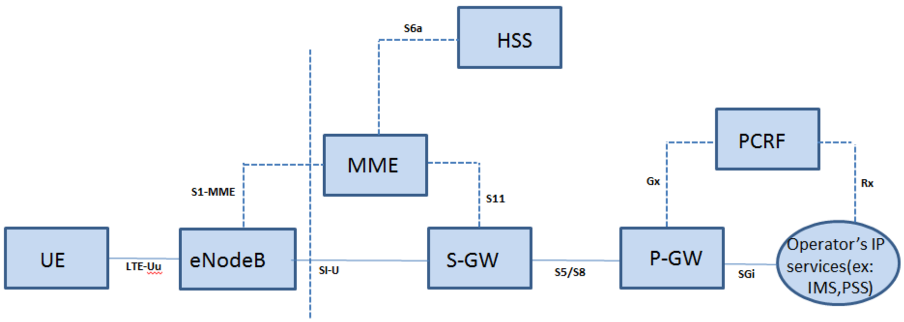

Fig. 1. EPC network elements.

serving gateway (S-GW) and mobility management entity (MME). Along with these elements, EPC has other important elements like home subscriber server (HSS) and the policy control and charging rules function (PCRF). The multimedia applications (VoIP) are controlled by the IP Multimedia Subsystem (IMS) in confluence with EPC.[8,9] Figure 1 shows the CN elements of EPC.

3.2. *IP multimedia subsystems*

IP multimedia subsystem combines and synchronizes the packet and circuit switching technologies. It is an application layer network. It was developed by 3rd generation partnership project (3GPP). Based on the architecture components, the IMS network includes IMS CN and CN elements.[10] These components combine to provide multimedia services, session setup, control of session, etc. The mobile user can connect to the IMS core using any access network, provided it has IP-based connectivity i.e., the IMS is independent of the wireless technology used (UMTS, PSTN, WLAN, broadband, etc.).[11–14] As shown in Fig. 2, IMS is integrated with EPC to get a fully functioning IP-based network. Figure 3 shows the CN elements of the IMS network.

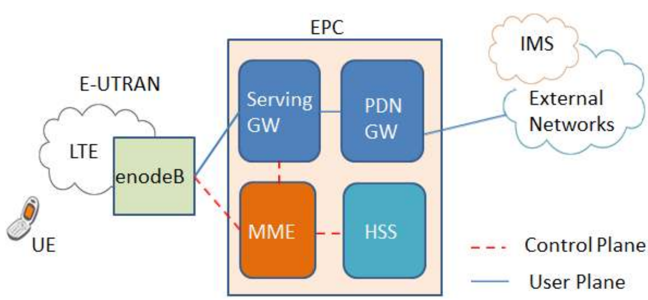

Fig. 2. EPC architecture in confluence with IMS network.

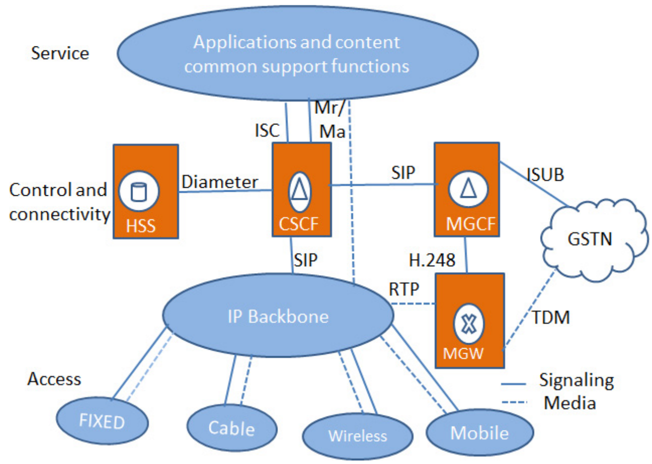

Fig. 3. IMS CN.

3.3. *Handover schemes and procedures*

The basic procedure and the importance of the various handover schemes for different networks are explained in detail. In this paper, a comparison is made between the proposed Prior-INVITE scheme with Re-INVITE cross-layer and Re-INVITE noncross-layer schemes.[4] The Re-INVITE schemes provide seamless handover and support the QoS by integrating the MIP and SIP protocols. For the Re-INVITE noncross-layer scheme, MS performs MIP registration and IMS registration independently during handover.[1] Here, the Re-INVITE noncross-layer and the Re-INVITE cross-layer schemes are compared, where the latter gives low signaling cost, delay and packet loss. In this proposed Prior-INVITE scheme, the signaling cost, delay and the packet loss can be further reduced as the number of messages exchange during handoff is minimized. The various INVITE schemes are discussed in the following sections.

3.3.1. *Cross layer design*

The communication networks in this design are realized using a layered protocol stack. In the earlier noncross-layer design, adjacent layers can communicate with each other but not with the noncontiguous layers.[5,15,16] This problem has been resolved using a cross-layer architecture, which uses media independent handover function (MIHF), which facilitates communication between noncontiguous layers independently shown in Fig. 4.[6,17,18] Thus, messages need not pass through all the intermediate layers. This helps in reducing the signaling overhead. In our case, all the SIP messages need not originate at the MN, reducing the number of hops a message needs to cross. The cross-layer design with the SIP Re-INVITE messaging sequence is given in Fig. 5.[4,19,20]

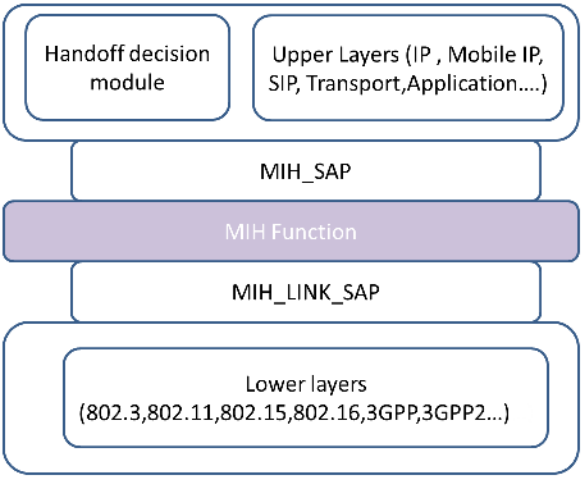

Fig. 4. MIHF layered architecture.

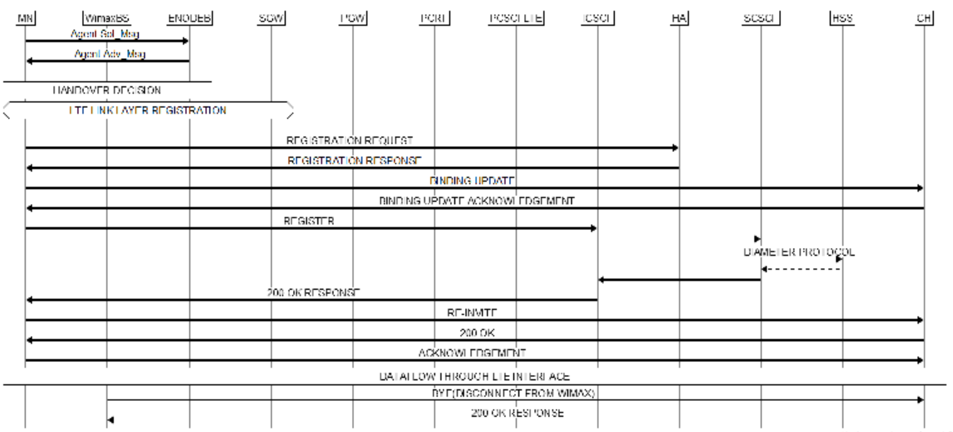

Fig. 5. Cross-layer with SIP Re-INVITE messaging sequence.

4. Proposed SIP-Prior-Handover

The proposed scheme uses the SIP Prior-INVITE in place of the SIP Re-INVITE. Here, a Make before break approach is used. The message sequence of the SIP Re-INVITE and SIP Prior-INVITE are shown in the Figs. 5 and 6, respectively. It can be seen from the figure that the communication of the MN with the new network begins even before the link with the serving network goes down. In fact, the MIP message sequence begins as soon as the Link-detected event happens and finishes before the Link going down event is registered. This process is completed with the

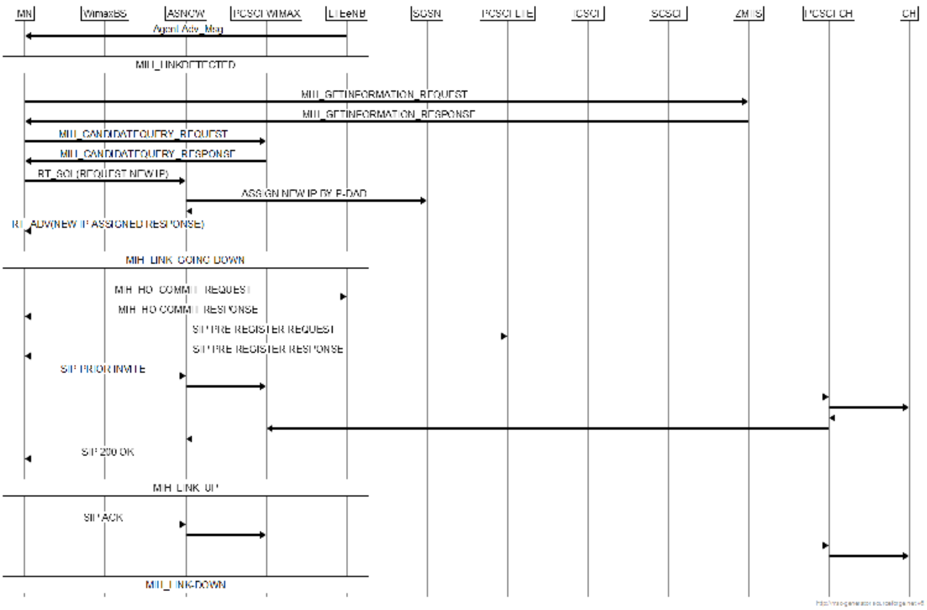

Fig. 6. SIP Prior-INVITE messaging sequence.

help of a Prior-duplicate address detection (P-DAD) procedure, bringing down the signaling cost and delay. After this, the SIP sequence is initiated. The SIP messaging sequence for this scheme is given in Fig. 6. The complete P-DAD procedure is described below.

The P-DAD procedure is given as,

- This procedure is used for obtaining the new IP-address for the target network before the network going down event occurs.
- As soon as the event MIH_link_detected is received by the MN, it queries the Z-MIIS in order to obtain the information about possible point of attachments (PoA).
- Then, the selected PoA's (LTE) address is received from the Z-MIIS and is added to the new care of address request (NCoA). A request message is forwarded to the S-GW by the MN.
- The S-GW sends the NCoA.Req message to the target gateway.
- After receiving the NCoA request message, the target gateway replies with a NCoA response message with a new configured IPv6 address to the MN with the network prefix of the target PoA.
- After getting the new IP address, it is configured to the LTE interface by the MN and re-registered with the IMS CN only after the Link going down event is detected.

5. Mathematical Modeling

In this section, we present the mathematical model to evaluate the performance of the proposed SIP Prior-INVITE scheme for the LTE–WiMAX network. This mathematical model is capable of providing performance analysis of delay, signaling cost and packet loss rate.

5.1. *Delay analysis*

Delay or the vertical handover latency is the time taken by the network to exchange mobile IP (MIP) and SIP messages with the corresponding host (CH) after finding the target network. The delay is analyzed in the networks against the number of hops that occur in the network. The delay is also calculated against the bandwidth of the network. The total delay in the proposed scheme mainly comes from the transmission delay, processing delay and queuing delay. The total delay is estimated as

$$D_{\text{setup}} = D_{(T-\text{setup})} + D_{(P-\text{setup})} + D_{(Q-\text{setup})}\,, \tag{1}$$

where D_{setup} is the total IMS handover delay, $D_{(T-\text{setup})}$ is the transmission delay, $D_{(P-\text{setup})}$ is the processing delay and $D_{(Q-\text{setup})}$ is the queuing delay. All the delays are calculated for the duration of the session setup.

5.1.1. *Transmission delay*

Transmission delay is the propagation delay during the signaling message propagation. In a packet switched network, transmission delay is the time required to put all the packet's frames onto the transmission line. The following is the formula we have used for calculating transmission delay[4,21]:

$$D_{(T-\text{setup})} = \frac{S}{B_{\text{wl}}} + K(\text{wl}) + h_{(x-y)} \times \left(\frac{S}{B_w} + K(w)\right), \tag{2}$$

where S denotes size of packet in bytes, B_{wl} and B_w are the bandwidths of wireless and wired links considered to be 70 Mbps and 100 Mbps, respectively, $K(\text{wl})$ and $K(w)$ are the latencies of the wireless and wired links, taken as 0.002 s and 0.0005 s, respectively, $h_{(x-y)}$ is the number of hops between x and y in the wired link. The MIP and SIP signaling messages are involved during the session setup for the noncross-layer and cross-layer schemes.[4] The equation for signaling messages of the proposed Prior-INVITE scheme is (see Tables 1 and 4)

$$\begin{aligned} D_{(T-\text{setup})} = {} & \text{dmih\_ho\_comm.req}(\text{MN} - \text{LTE}.enB) \\ & + \text{dmih\_ho\_comm.res}(\text{LTE}.enB - \text{MS}) \\ & + \text{dprerg}(\text{MS} - \text{PCSCFLTE}) \\ & + \text{200ok}(\text{PCSCFLTE} - \text{MS}) + \text{dpreinv}(\text{MS} - \text{CH}) \\ & + \text{200ok}(\text{CH} - \text{MS}) + \text{dsip.ack}(\text{MS} - \text{CH})\,. \end{aligned} \tag{3}$$

Table 1. Delay components involved during SIP and MIP registration.

Delay components	Messages
dprerg	Pre-register request
200 OK	Ok response
dpreinv	Prior-INVITE
dmih_ho_comm.req	Handoff commit request
dmih_ho_comm.res	Handoff commit response
dsip.ack	Sip-acknowledgement

5.1.2. *Processing delay*

Processing delay is mainly the delay incurred in a server due to the encapsulation and de-capsulation of data. It is assumed to be equal for all the servers/gateways in the architecture. In order to find the relation between the number of hops and total processing delay, the processing delay for each message was calculated and then multiplied with the number of hops. Since the number of messages received at the MN and number of messages received at the corresponding host (CH) are fixed, we need not multiply the delay in these messages with the number of hops. The rest of the messages will undergo multiple hops in a network; therefore, their delays are multiplied by the number of hops. Now, assuming the same processing rate for all the servers, we multiply the processing rate by each messages size and then by the number of hops.

The number of hops for some messages like agent solicit, agent advertisement, register, OK response, commit request and response messages are considered to be constant and equal to 2. This is so because these messages do not communicate with the other nodes that contribute to the increase in hops. The rest of the messages have variable hops. First, the processing delay for noncross-layer scheme is described below.

For a noncross-layer scheme:

$$D = \sum (Sz(i) \times D_{(\mathrm{proc})} \times 2)\,, \tag{4}$$

$$D' = \sum (Sz(i) \times D_{(\mathrm{proc})} \times h_{(x-y)})\,, \tag{5}$$

where D is the processing delay for agent solicit and agent adv. messages, D' is the processing delay for the rest of the messages.

$$D_{(P-\mathrm{setup})} = D + D'\,. \tag{6}$$

The calculated processing delay for the noncross-layer scheme is found to be high. Moreover, the objective is to reduce the overall handover delay, let us now describe the necessary equations producing the delay in the cross-layer scheme.

For a cross-layer scheme:
For the cross-layer scheme, we can find $D_{(P-\text{setup})}$ using Eq. (6), where D is the processing delay for agent solicit, agent adv., register and 200 OK response messages, D' is the processing delay for the rest of messages.

For a Prior-INVITE scheme:
For the Prior-INVITE scheme, we can find $D_{(P-\text{setup})}$ using Eq. (6), where D is the processing delay for commit req and response Messages, D' is the processing delay for the rest of messages. D_{proc} is taken as 1 Mbps for all delays.[22]

5.1.3. *Queuing delay*

Queuing delay can be considered to be the delay because of the queuing of the packets at network servers. The queuing delay for the messages at all servers is found to be equal except for the messages at the CH server. In order to find the relation between the total queuing delay and the number of hops for the messages, we use this relation. As in the calculation of the processing delay, we observe that the number of messages received at the MN server and CH server (meaning their queuing delays too), is fixed. The queuing delay in the rest of the messages is multiplied by the number of hops.

The mathematical modeling of queuing delay is listed in Tables 2 and 3.[2,3,14] The delay of the CH D_{ch} is given by the formula $D_{\text{ch}} = (((1 - \varrho_s - \varrho_n)/\mu_{\text{CH}}) + R)/((1 - \varrho_n) + (1 - \varrho_s - \varrho_n))$. In this D_{ch} equation, the parameter R is equal to $\lambda_n E[X_1^2] + \lambda_{\text{CH}} E[X_2^2]/2$. Moreover, $E[X_1^2]$ and $E[X_2^2]$ are the second moments of μ_n and μ_{CH}, respectively. In Table 3, the rest of the formulae to find the queuing delay are also given.

For the noncross-layer scheme:

$$D = D_Q \times 2 \times 2\,, \tag{7}$$

$$D' = D_Q \times 9 \times h_{(x-y)}\,, \tag{8}$$

$$D_{(Q-\text{Setup})} = 3 \times D_{\text{MN}} + 3 \times D_{\text{CH}} + D + D'\,, \tag{9}$$

Table 2. Parameter definitions.

Parameter notation	Definition
μ_{MN}	Rate for SIP messages processing at MN
μ_{CH}	Rate for SIP messages processing at CH
μ	Rate for nonSIP messages processing at CH
λ_{MN}	Rate of arrival for SIP messages at MN server
λ_{CH}	Rate of arrival for SIP messages at CH server
λ_n	Rate of arrival for nonSIP messages to CH
ϱ_s	CH and CSCF servers load
ϱ_n	CH and CSCF servers load for nonSIP messages

Table 3. Formula for queuing delay.

Parameter	Formulae	Delay component
ϱ	λ/μ	Network delay
D_{MN}	$\dfrac{1}{\mu_{\mathrm{MN}} - \lambda_{\mathrm{MN}}}$	Queuing delay for messages at MN
$D_{P-\mathrm{CSCF}} = D_{S-\mathrm{CSCF}} = D_{I-\mathrm{CSCF}}$	$\dfrac{\varrho_s}{\lambda \times (1-\varrho_s)}$	Queuing delay for messages at P-CSCF, S-CSCF, I-CSCF
D_{CH}	$\dfrac{\dfrac{1-\varrho_s-\varrho_n}{\mu_{\mathrm{CH}}} + R}{1-\varrho_n+1-\varrho_s-\varrho_n}$	Queuing delay for messages at CH
$D_{\mathrm{ASN-GW}}$	$\varrho_s/(\lambda \times (1-\varrho_s))$	Queuing delay for messages at ASN-GW
D_{SGSN}	$\varrho_s/(\lambda \times (1-\varrho_s))$	Queuing delay for messages at SGSN

where D_Q is queuing delay for messages at each server except MN and CH, D is the queuing delay for agent solicit and agent adv. messages, D' is the queuing delay for the rest of the messages.

For the Cross layer scheme:

$$D = D_Q \times 4 \times 2\,, \tag{10}$$

$$D' = D_Q \times 7 \times h_{(x-y)}\,, \tag{11}$$

$$D_{(Q-\mathrm{Setup})} = 3 \times D_{\mathrm{MN}} + 3 \times D_{\mathrm{CH}} + D + D'\,, \tag{12}$$

where D is the queuing delay for agent solicit, agent adv., register and 200 OK response messages, D' is the queuing Delay for the rest of the messages

For the Prior-INVITE scheme:

$$D = D_Q \times 4 \times 2\,, \tag{13}$$

$$D' = D_Q \times 3 \times h_{(x-y)}\,, \tag{14}$$

$$D_{(Q-\mathrm{Setup})} = 3 \times D_{\mathrm{MN}} + 2 \times D_{\mathrm{CH}} + D + D'\,, \tag{15}$$

where D is the queuing delay for the commit request and response messages, D' is the queuing delay for the rest of the messages

For the noncross-layer and Cross-layer delays:

$\mu_{\mathrm{MN}} = 2895.5$ packets/s, $\mu_{\mathrm{CH}} = \mu_{\mathrm{MN}}$.
$\lambda_{\mathrm{MN}} = 50$ packets/s, $\lambda = 500$ packets/s.
$\varrho_s = 0.173$, $\varrho_n = 0.7$.
From formulae given earlier, we calculate
$D_{\mathrm{MN}} = 0.35$ ms.
$D_{\mathrm{CH}} = 0.102$ ms.
$D_Q = 0.42$ ms.

For the Prior-INVITE scheme:
$\mu_{\mathrm{MN}} = 4008\,\text{packets/s}$, $\mu_{\mathrm{CH}} = \mu_{\mathrm{MN}}$.
$\lambda_{\mathrm{MN}} = 50\,\text{packets/s}$, $\lambda = 500\,\text{packets/s}$.
$\varrho_s = 0.125$, $\varrho_n = 0.7$.
From formulae given earlier, we calculate
$D_{MN} = 0.25\,\text{ms}$.
$D_{\mathrm{CH}} = 0.092\,\text{ms}$.
$D_Q = 0.285\,\text{ms}$.

The values for $\lambda, \lambda_{\mathrm{MN}}, \varrho_n$ are fixed according to Ref. 23. While calculating the values of μ_{MN} for the different schemes, we had to first calculate the average size of messages in the Re-INVITE scheme and Prior-INVITE scheme. We have selected a server with a fixed processing rate ($\mu = 1\,\text{MBPS}$) with respect to bytes per second;, on the other hand, to calculate the queuing delay, we need the processing rate in packets per second.[22] Assuming the average message size in each scheme as an average packet size in that scheme, we can get the processing rates in packets per second for the corresponding calculation. The following formula is used:

$$\mu(\text{packets per second}) = \frac{\mu\,(\text{in bytes per second for server})}{(\text{average message size in bytes})}\,. \tag{16}$$

5.2. *Cost of signaling analysis*

In this section, a signaling cost model is developed. The signaling cost can be used to measure the transmission of a signal. The total signaling cost can be calculated as[5,24,25]

$$\begin{aligned}\text{signal cost} = {} & \left\{G_m \times \left[\sum Sz_{(\text{INVITE})} - I \times K_{(a-b)}\right]\right\} \\ & + \left\{G_s \times \left[Sz_{(\text{REINVITE})} - I \times K(a-b)\right]\right\},\end{aligned} \tag{17}$$

where

$$\begin{aligned} G_s &= \text{average call arrival rate}, \\ G_m &= \text{average network mobility rate}, \\ Sz_{\text{INVITE}} - I &= \text{size of the message in the IMS–MIP sequence}, \\ Sz_{\text{REINVITE}} - I &= \text{size of the message in the SIP reinivite sequence}, \\ K_{(a-b)} &= \text{Number of hops}, \\ N_m &= \text{Number of handovers}. \end{aligned}$$

For the cost of signaling analysis with respect to the number of handovers, we have fixed the number of hops (five), and multiplied the whole expression by the number of handovers.

$$\begin{aligned}\text{signal cost} = {} & \left\{G_m \times \left[\sum Sz_{(\text{INVITE})} - I \times K_{(a-b)}\right]\right\} \\ & + \left\{G_s \times \left[Sz_{(\text{REINVITE})} - I \times K(a-b)\right]\right\} \times N_m\,,\end{aligned} \tag{18}$$

where N_m is the number of handovers and $K_{(a-b)}$ is the number of hops. Equation (18) is modified to obtain the relationship between signaling cost with call to mobility rate (CMR) and utilization factor as given in "Eqs. (19) and (20)".

$$\begin{aligned}\text{signal cost} = {} & \left\{G_m \times \left[\sum Sz_{(\text{REINVITE})} - I \times K_{(a-b)}\right]\right\} \times G_s/G_m \\ & + \left\{G_m \times \left[Sz_{(\text{INVITE})} - I \times K(a-b)\right]\right\},\end{aligned} \tag{19}$$

where CMR is the rate G_S/G_m.

$$\begin{aligned}\text{signal cost} = {} & \left\{\mu_h \times \left[\sum Sz_{(\text{REINVITE})} - I \times K_{(a-b)}\right]\right\} \times G_s/\mu_h \\ & + \left\{G_m \times \left[Sz_{(\text{INVITE})} - I \times K(a-b)\right]\right\},\end{aligned} \tag{20}$$

where μ_h is an average call completion rate and G_s/μ_h is the utilization.

5.3. *Packet loss rate analysis*

Packet loss is the total sum of packets lost during the vertical handoff while MN is receiving the downlink data packets. The following expression gives the relationship between the packet loss rate and the number of hops or packet loss rate and number of handovers.[4,24]

$$\text{Packet Loss} = [2 \times T_{\text{ad}} + \text{DL}] \times G \times N_m\,, \tag{21}$$

where

T_{ad} = Time for the agent advertisement signal,
DL = vertical handover delay,
G = downlink packet transmission rate,
N_m = average number of handovers.

When we plot against the number of handovers, we fix the number of hops, and then include the calculated delay. While plotting against the number of hops, the number of hops are variable and the number of handovers are kept fixed and the delay is calculated. This is delay is included in the above expression, which then gives us the results for the packet loss rate.

6. Results

The simulations have been done using MATLAB 2013. The important parameters used for all the three schemes are tabulated in the Table 4. The performance of the proposed scheme is evaluated through the handover delay, packet loss rate and signaling cost. Initially, the handover delay of the proposed scheme is compared against the existing schemes.

6.1. *Handover delay*

Going by the ITU Recommendations, the handover latency must be under 200 ms. From the above graph we can clearly see that for the noncross-layer, cross-layer and

Table 4. Parameter values used for simulation.

Messages	Size (Bytes)	Parameter	Value
mih_ho_comm.req	225	$K_{(a-b)}$	5
mih_ho_comm.res	225	G_s	80 packets/s
SIP-register (regis)	225	G_m	80 packets/s
SIP-200 OK	100	G	70 Mbps
SIP-Preinv	810	T_{ad}	1 s
SIP-ACK	60		

Table 5. Delay analysis of various schemes.

Schemes	No. of hops	Delay (sec)	Packet loss in bytes
Re-INVITE noncross-layer	10	0.1807	7.633×10^8
Re-INVITE cross-layer	10	0.1509	7.528×10^8
Prior-INVITE	10	0.06467	7.228×10^8

Prior-INVITE scheme, the number of hops before passing the specified limit is 12, 14, 37, respectively. Thus, the Prior-INVITE scheme clearly outperforms the other two methods shown in the Table 5.

Since the bandwidth is very large with respect to the size of the message, there is very small change in delay, which is negligible.

6.2. *Cost of signaling*

As expected, the signaling cost shows linear behavior with respect to the number of handovers in all the three cases as shown in Fig. 9. The cross-layer technique shows

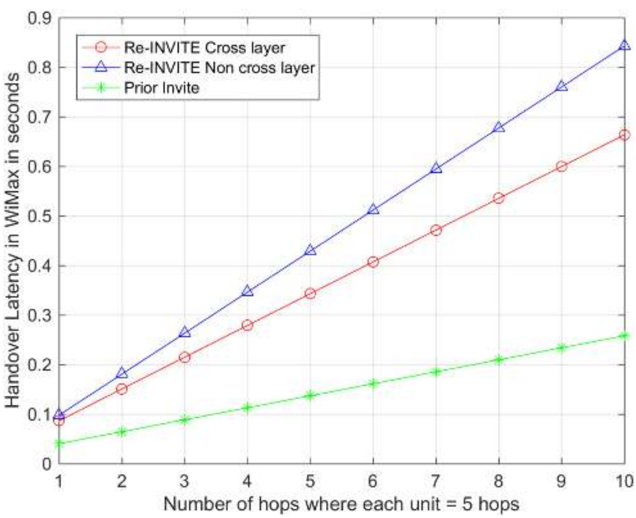

Fig. 7. Graph of handover delay in WiMAX versus number of hops.

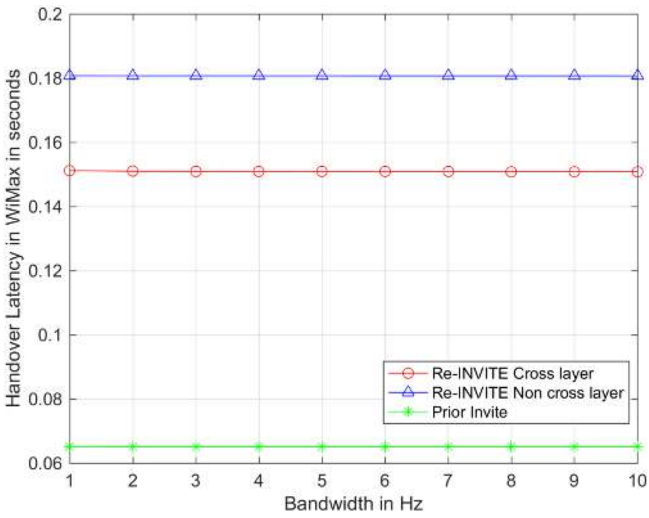

Fig. 8. Graph of handover delay in Wimax versus bandwidth.

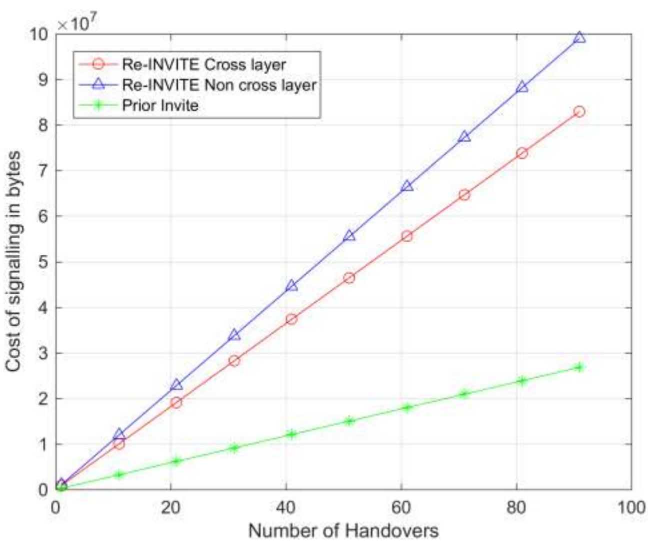

Fig. 9. Graph of signaling cost versus number of handovers.

an improvement of 18.2% in signaling cost and the Prior-INVITE method shows an improvement of over 70% in signaling cost as compared to the noncross-layer technique.

The cross-layer technique shows an improvement of over 14% in signaling cost and the Prior-INVITE method shows an improvement of over 85% in signaling cost as compared to the noncross-layer technique with respect to the number of hops, shown in Fig. 10.

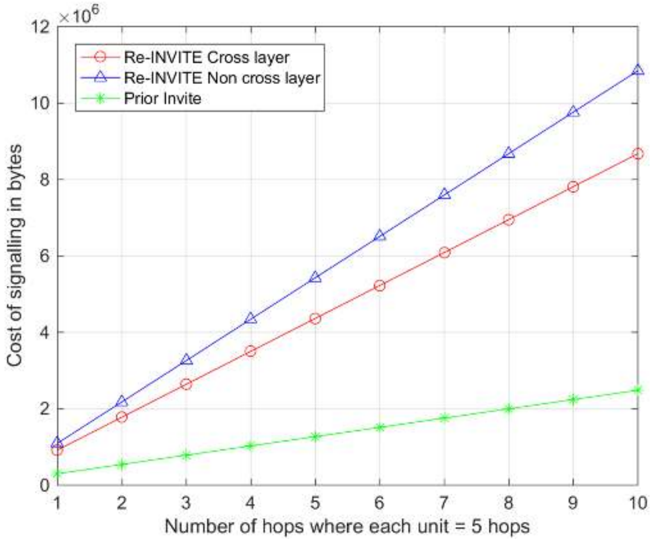

Fig. 10. Graph of signaling cost versus number of hops.

6.3. *Packet loss rate*

The cross-layer and Prior-INVITE methods show an improvement of 2.7% and 10.8% in the packet loss rate with respect to the number of handovers. As number of handover increases the number of packets lost, transferring delay, cost of signaling at

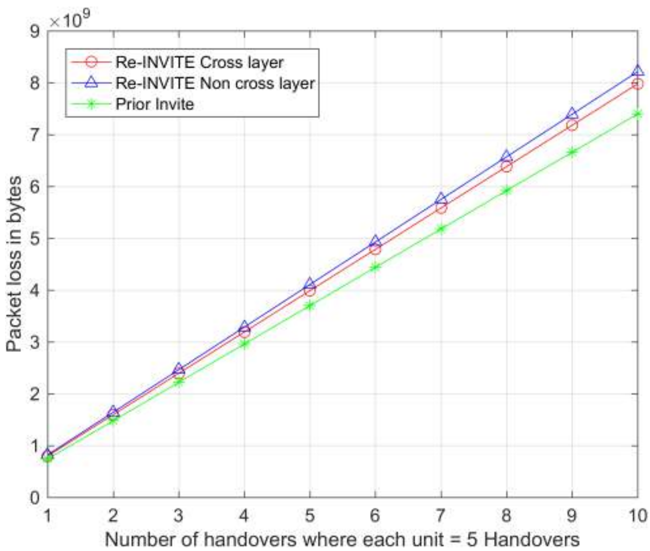

Fig. 11. Graph of packet loss versus number of handovers.

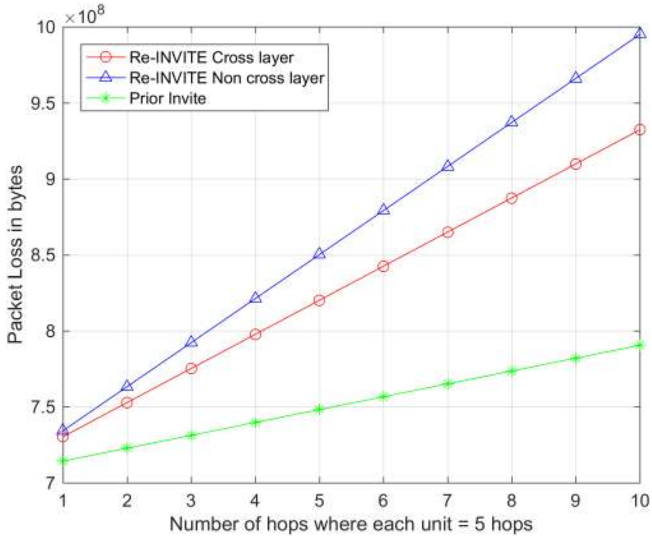

Fig. 12. Graph of packet loss versus number of hops.

each handover are added which results in more vertical handover latency, packet loss as shown in Fig. 11.

The cross-layer and Prior-INVITE methods shows an improvement of 6.2% and 24.4%, respectively, in the packet loss rate versus number of hops as shown in Fig. 12. Figure 13 shows the relationship between signaling cost and CMR. It can be seen

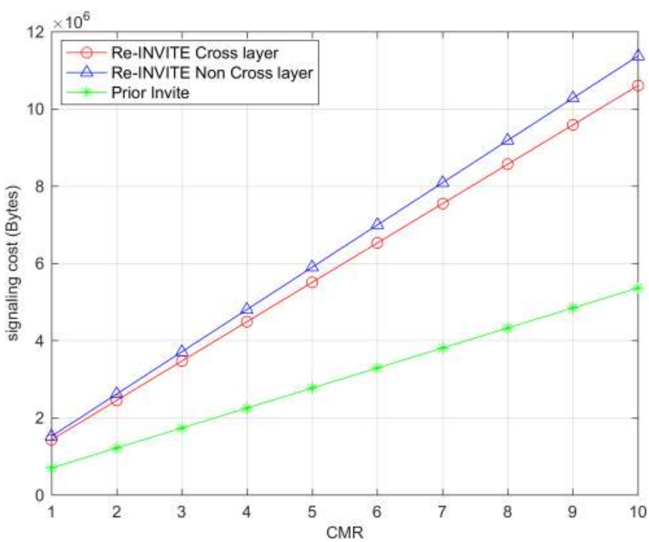

Fig. 13. Graph of signaling cost versus CMR.

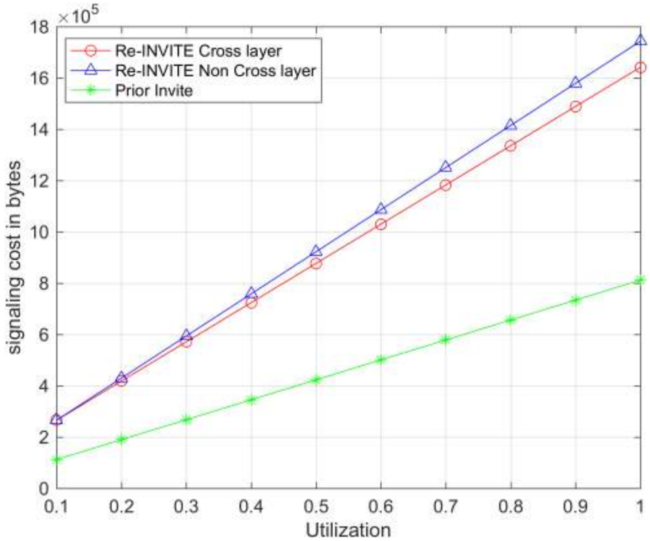

Fig. 14. Graph of signaling cost versus utilization.

that signaling over head is reduced in the proposed Prior-INVITE scheme when compared to other schemes. Figure 14 illustrates that the signaling cost increases linearly with utilization and the proposed scheme outperforms the other schemes in terms of signaling cost.

7. Conclusion

As is evident from the simulation results we obtained, the performance of the SIP Prior-INVITE scheme is far better than that of the other schemes which use the SIP Re-INVITE scheme. The proposed algorithm uses make before handoff which reduces the handoff delay. Secondly, signaling overhead is reduced since the messages need not pass through all the layers. In addition, the SIP messages need not originate at the MN, which reduces the number of hops a message needs to cross. Thirdly, the P-DAD procedure used in the proposed approach helps in bringing down the signaling cost and delay. The total number of message exchanges during a handover session for the existing SIP Re-INVITE scheme compared with the proposed scheme reduces to 34%. The presented mathematical approach helps to achieve this very high handover efficiency. This clearly proves the effectiveness of the proposed scheme and the derived mathematical equations. This approach will simplify work for future pioneers and researchers in the heterogeneous wireless network environment.

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