A Simulation Study on the Performance of Mobile IPv6 in a WLAN-Based Cellular Network

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Abstract— **We performed a simulative evaluation of Mobile IPv6 via *ns-2* for a scenario comprising up to three access routers and 30 mobile nodes that communicate in accordance with the IEEE 802.11 wireless LAN standard. We compare basic MIPv6 to a fast handoff procedure and analyze route optimization issues. We present results with respect to handoff latency, packet loss, end-to-end delay, signaling load, channel utilization, and bandwidth per station, and show how various traffic types like UDP CBR, VoIP, and TCP are affected by the handoffs. While some simulation results corroborate the intention of the protocol specification, other results give insights not easily gained without performing simulations. E.g., we learned that *i)* the fast handoff mechanism almost fully eliminates packet losses but only improves handoff latency in specific cases, *ii)* the signaling load introduced by MIPv6 does not severely affect the performance, and *iii)* in a shared environment like IEEE 802.11, traffic type can affect the handoff rate since in high load situations router advertisements might get lost more frequently thereby 'irritating' the handoff decision mechanism. As a challenge for future simulations we have identified the design of a random mobility model that allows good control of the handoff rate.**

Keywords—**Mobile IP, IPv6, handoffs, simulations, performance evaluation, ns-2**

I. Introduction

The Internet goes mobile and cellular systems are embracing IP technology. This trend has stimulated work on protocol design for IP-based mobility management during the last few years. The IETF working group on Mobile IP is proposing Mobile IPv4 [1] and Mobile IPv6 [2] as the main protocols for supporting IP mobility. In order to improve the 'QoS experience' of a mobile user who hands over from one access router to another one as well as to reduce signaling load on the network, a large number of extensions to Mobile IP for localized mobility management and fast/seamless handoffs have been proposed, see for example [3], and discussed heavily on the IETF Mobile IP and Seamoby mailing lists. Main discussion items are scalability, handoff latency and signaling overhead of a protocol. However, it is not an easy task to predict a protocol's performance only studying the protocol specification. For a deeper understanding as well as for assessing more general system aspects, simulation is the most important tool.

In this paper we present a simulation study on the effects of handoffs and on route optimization in Mobile IPv6. Our main interest has been to assess latency and packet losses introduced by 'standard' as well as 'improved' MIPv6 handoff procedures, the effect of route optimization on end-to-end delay, and associated signaling costs. Moreover, the impact on channel utilization and bandwidth per station has been studied. Clearly, simulation results depend on the scenario used and, thus, the question of 'what is the right scenario' arises immediately. A trade-off has to be made between how general or system-specific the simulation results should be. We have chosen a basic scenario consisting of up to three access routers and 30 mobile nodes that communicate in accordance with the IEEE 802.11 wireless LAN standard. We focus not solely on MIPv6 protocol performance but on system performance including the impact of a shared link on the performance parameters. The scenario can also be used to form more complex, e.g., hierarchical, settings. As mobility pattern for the mobile node, the random waypoint mobility model [4] is used. As data traffic types we studied UDP CBR (constant bit rate), VoIP, and TCP. The simulation code used for the experiments was designed on top of Thierry Ernst's MIPv6 [5] for *ns-2* [6] implementation.

Previous work on simulative evaluations of Mobile IPv6 exclusively dealt with the case of IPv4 networks. Perkins and Wang [7] have used *ns-1* to analyze the effects of route optimization and buffering ('smooth handoff'). Caneel and Lamprecht [8] have designed *ns-2* modules for Cellular IP, HAWAII, and THEMA. The Columbia IP Micro-Mobility Suite [9] comprises *ns-2* modules for Cellular IP, HAWAII, and Hierarchical IP. An evaluation of Cellular IP is provided by Campbell et al. in [10].

The paper is structured as follows. Section II recalls the basics of Mobile IP. In Section III we describe the simulation scenario in detail. The performance aspects subject of interest are given in Section IV. Simulation results are provided in Section V. Finally, Section VI presents some hints for future work and the summary.

II. Mobile IP

In this section we will briefly describe the features of Mobile IPv6 in comparison to Mobile IPv4. For Mobile IPv4, the reader is referred to [1]. Route optimization is an integral part of Mobile IPv6: when a correspondent node (CN) is aware of the current care-of address (coa) of a mobile node, the CN can send packets directly to this care-of address, thus, triangle routing is eliminated. Only if the CN is not aware of a current coa, packets are first sent to the home address of the mobile node and then tunneled by a home agent (HA). Since both, CN and HA, maintain lists of coas (binding cache), a common message format called 'binding updates' is used to inform CN and HA about changes in the point of attachment. Binding updates (BU) can be acknowledged by BU ACKs. In contrast to MIPv4, all signaling is done in extension headers that can also be piggybacked on 'regular' traffic. In order to avoid the ingress-filtering problem, the mobile node uses its coa as source address and attaches its home address in a 'home address option'. To acquire a coa, a mobile node can build on IPv6 stateless and stateful auto-configuration methods. For more details see [2].

III. Simulation Scenario

In order to study the performance of Mobile IPv6, a basic scenario is necessary in order to analyze various aspects separately. The studied scenario (see Figure 1) is composed of a group of correspondent nodes, one for each mobile node, connected to one central router (CR) through the 'Internet', the access routers (AR) –each one representing a different IP subnet– connected also to the CR, and ten mobile nodes (MN) per AR in the initial setup. The home agent of each mobile node is located on the access router closest to the mobile node when the simulation starts.

The distance between the access routers is 450 meters and the transmission range 250 meters. Thus, the coverage areas of the access routers overlap and the mobile nodes always move randomly within the total coverage area. As mobility pattern we use the random waypoint mobility model [4]. This model has been previously used mainly for ad hoc network simulations. With this model, node movements are piecewise linear movements. As parameters one can set the average speed of the nodes as well as the number of changes of directions per minute. As explained later, these parameters have to be tuned in order to force specific handoff rates.

A 2Mbps Wireless LAN 802.11 [11] with DCF is simulated in the wireless medium. The number of mobile nodes has been chosen with the aim of working close to the saturation throughput assuming sources sending about 75kbps[1].

The default link delay (ld) between the CR and the ARs is 1.8ms. The link delay between the CR and CNs is fixed to 10.8ms for all simulations. In the simulations we have varied the ld value to model various 'distances' between the ARs and between CNs and MNs.

In this study when differentiation between the kind of handoff (home link to foreign link, foreign to foreign link) is needed three access routers have been simulated, otherwise two access routers are considered.

While moving within the overlapping area, the mobile nodes are able to send/receive data only via the access router that corresponds to their current care of address.[2]

In each access router five mobile nodes receive data from five correspondent nodes and the other five mobile nodes

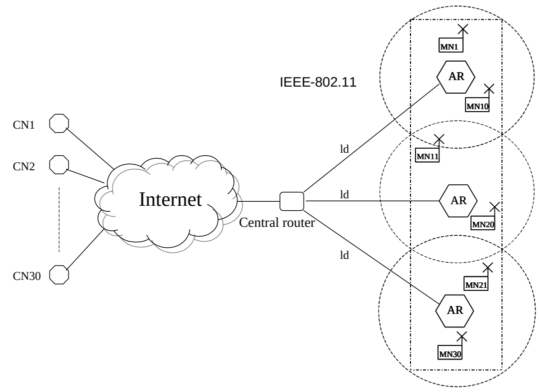

Fig. 1. Simulations scenario

send data to another five correspondent nodes. The scenario has been chosen to simulate a realistic case where a mobile node will receive packets from the shared access router queue and where a mobile node will also compete with other mobile nodes and with an access router to access the channel. The correspondent nodes sending to the mobile nodes introduce delay in the access router queue and the mobile nodes sending to the correspondent nodes introduce delay in the wireless link.

It is important to note the following fact that results of using a shared access: whenever we work close to the maximum throughput of the channel, the stations that will experience first a reduction in their throughput will be the ones *receiving* from the correspondent nodes. The reason is that these stations receive all the packets from the same station, the access router, sharing the access queue to the wireless channel, while the other mobile nodes sending to the correspondent nodes do not share their access queue to the channel.

In our simulations different types of traffic will be simulated. UDP CBR sources provide constant traffic where no acknowledgments are required. This kind of traffic is usually generated by real-time applications. In the study we will simulate traffic sources sending at the following rates: 40, 50 and 75kbps. In the case of 40 or 50kbps the load in the channel is below the saturation throughput while in the case of 75kbps the load in the channel is close to the saturation throughput in a scenario with two access routers.

TCP is the most widely used transport protocol. We will simulate endless FTP sources to understand the impact of IP mobility on congestion control mechanism of TCP.

One of the applications expected to be used with MIPv6 is VoIP. We have implemented a VoIP model based on [12]. The model assumes silence suppression and models each voice source as an on-off Markov process. The alternating active *on* and silence *off* periods are exponentially distributed with average durations of 1.004 and 1.587 s. As recommended by the ITU-T specification for conversational speech [13], an average talk spurt of 38.57% and an average silence period of 61.47% is considered. A rate of 88 kbps[3]

[1] The transmission range in the simulator is 250m and the interference range is 550m. Therefore, when two access routers are used, 20 mobile nodes sending or receiving 75kbps via the wireless medium, the saturation throughput for both access routers is around 1.5 Mbps since their transmission and interference range overlap

[2] Technologies like IEEE802.11 allow the mobile nodes gathering information about the neighboring access routers but do not allow to receive IP flows with different destination addresses simultaneously from two access routers except of particular cases like having an additional wireless interface.

[3] Assume 8KHz 8 bits/sample PCM codec was used with 20 s frame

in *on* periods and 0 kbps in *off* periods is assumed for a voice source that generates CBR traffic.

IV. Aspects to be studied

We analyze handoff performance of Mobile IPv6 with respect to handoff latency, packet losses during handoffs, signaling load, channel utilization and obtained bandwidth per station. These parameters are explained in more detail below. Furthermore, we investigate how a fast handoff mechanism improves above mentioned measurements. We have implemented a fast handoff procedure inspired by [3]. We use stateless address auto-configuration and send the BU (to HA in the case of a home link to foreign link handoff and to HA, previous AR and CN in the case of a foreign to foreign link handoff) with the newly formed care of address via the old access router instead of via the new one. By doing this, redirection of the traffic to the MN will be processed directly at the receiving access router. The MN then switches to the new access router and waits for the redirected packets to come.

In addition, we study route optimization, which is embedded in MIPv6 while it is only an extension in MIPv4. The avoidance of triangle routing by route optimization is a clear advantage of using route optimization with respect to end-to-end delay. However, the increase in the signaling load produced by route optimization as well as the increase in the performance only of some parameters under specific circumstances let the question arise whether the price in signaling load is worth to pay.

In the following we define the parameters investigated in the simulations:

Handoff Latency

Handoff latency is defined for a receiving MN as the time that elapses between the last packet received via the old route and the arrival of the first packet along the new route after a handoff. Latency is an important parameter for delay sensitive applications like VoIP that could suffer from a period with a higher rate of packet drops due to a long latency time. This packet drop period would result in a noticeable disruption in the voice transmission. We study handoff latency for various values of link delays ld as well as for home link to foreign link (HL-FL) and foreign to foreign link (FL-FL) handoffs. A differentiation between the latency due to a handoff between the home link and a foreign link and the handoff between two foreign links is needed. When performing a handoff between the home link and the foreign link the mobile node does not send a BU to the CN since in our implementation the entry in the BU list for it will be introduced when an encapsulated packet is received. The first packet received from a CN to a mobile node after a home link to foreign link handoff will produce a latency proportional to the round-trip time necessary by a BU to arrive from the foreign link to the home link. The foreign link to foreign link handoff case differs in that the latency is proportional to the minimum time required by a BU to arrive either at the previous access router, HA or the CN.

per packet. With 12 byte RTP header, 8 byte UDP header and 40 byte IP header, the size of each voice packet is 220 bytes. The bandwidth required will be (220 x 8)/20=88kbps

Packet Loss

Packet loss is defined for a receiving MN as the number of packets lost during the handoff. While usually one assumes that packet losses are directly proportional to latency, it will be made clear that this is generally not true, particularly not for the fast handoff procedure. We study handoff latency for various values of link delays ld as well as for home link to foreign link and foreign to foreign link handoffs.

Signaling Load

We study the signaling load due to sending of BUs and BACKs. The number of BUs and BACKs sent will be the same for MIPv6 and the fast handoff procedure and different for MIPv6 with/without route optimization. We study the signaling load for various handoff rates (number of handoffs per minute).

Channel Utilization

We measure the channel utilization in order to see the effects of mobility, i.e., the handoff signaling, on the overall throughput. We study the impact of mobility on channel utilization for different traffic source types under increasing mobility of the mobile nodes.

Bandwidth per Station

We study the probability to obtain the required bandwidth and the corresponding expected variance for an increasing number of handoffs, various traffic types, and by using/not using route optimization or the fast handoff procedure.

End-to-End Delay

To assess the effect of route optimization, we measure the end-to-end delay between CNs and MNs for MIPv6 with and without route optimization.

V. Performance evaluation and discussion of results

Based on the general scenario outlined in Section III we build two sub-scenarios, one for measuring aspects from a single node's perspective and another one for measurements concerning the overall system.

Sub-Scenario 1 ($SS1$): The MN for which measurements are taken follows a deterministic path through the coverage area while all other nodes follow the random waypoint mobility model.

Sub-Scenario 2 ($SS2$): All MNs follow the random waypoint mobility model.

Both sub-scenarios, SS1 and SS2, might be used with either 2 or 3 ARs and 20 or 30 MNs, respectively.

Three access routers are only simulated when differentiation between the type of handoff is needed otherwise two access routers are simulated for computing time reasons.

The mobile nodes move according to our modified *ns-2* random way-point model. The *ns-2* random way-point model has been modified in two ways: *i)* the mobile nodes do not perform a change of direction and speed all at the same time after a fixed amount of seconds but according to a uniform random value distributed between 1 to 60 seconds, *ii)* this value is not the same during the whole simulation but recomputed for each change of direction and speed. This modification has been introduced in order to de-correlate the changes of direction and speed between the mobile nodes. When no other value is indicated all the simulations have been performed with a speed limit of 5m/s.

All simulations have a duration of 125 seconds with a 5 seconds warm-up phase and the option of sending a BU to the previous access router is always set.

Averages in the following graphs represent the average of 100 simulations. In the case of bandwidth histogram per mobile node graphs, they have been obtained after performing 1000 simulations.

A. Basic MIPv6

A.1 Latency

For handoff latency evaluation we have chosen sub-scenario SS1. The single MN following a deterministic path performs a handoff from home link to a foreign link and from a foreign link to another foreign link. In total, we employ 3 ARs and 30 MNs. Moreover, the data traffic in this simulation has to be continuous in order to obtain the whole latency time. The data rate of the UDP CBR sources has been chosen taking in account the interference range of 550m. The access router in the center interferes and is interfered by two access routers. Thus, to obtain a significant traffic load value but below the saturation throughput we have chosen a data rate of 40kbps.

In Figures 2 and 3 the instantaneous bandwidth for two different link delay values (1.8ms and 1000.8ms) between the ARs and the CR are represented comparing the results for the above mentioned scenario with the one for a static ‘mobile node’. The integration time to compute the instantaneous bandwidth was set to 200ms. Note that a smaller integration time would increase the ‘variance’ of the measurements while a larger value would make the curve smooth such that details (like the spike in Figure 2) would not be seen anymore.

When the transmission time of a BU between the MN via the new AR to HA and/or CN has a lower value than the inter-arrival packet time, the latency is nearly not perceptible as it can be seen in Figure 2. On the other hand, when the transmission time is above the inter-arrival packet time then the gap is noticeable (see Figure 3). The first latency is produced due to a home to foreign link handoff and the second one due to a foreign to foreign link handoff. The difference between the instantaneous bandwidth received while the mobile node moves in the home link and the one after performing a handoff is due to an encapsulated packet or the routing header.

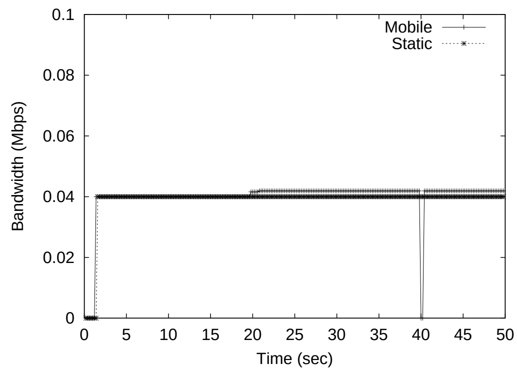

Fig. 2. UDP CBR latency,1.8ms link delay

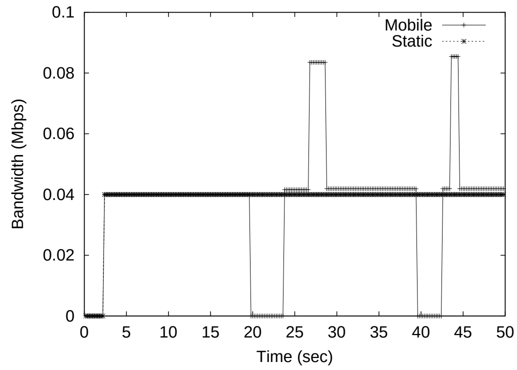

Fig. 3. UDP CBR latency,1000.8ms link delay

In Figure 4 we can observe the variation of the latency depending on the link delay between the AR and the router for both kinds of handoffs.

From the graph we can deduct for our scenario that the HL-FL handoff will always result in a higher or similar latency value compared to the FL-FL case. This is because in the case of a HL-FL handoff the mobile node will only send a BU to the HA and not to the CN. This will result in a higher latency time than in the case of FL-FL because from MN’s perspective at the foreign AR, the HA is a ’link delay’ farther away than the CN. The consequence is clear observing the last points in Figure 4. What we can observe from the graph as well is that the latency time

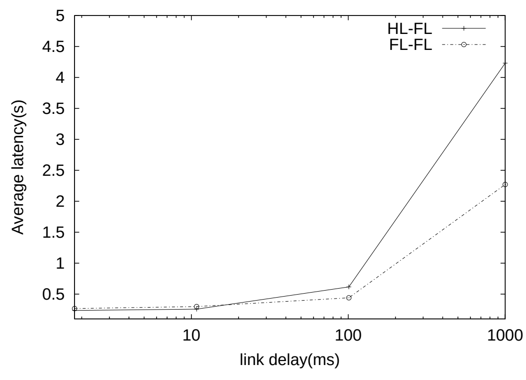

Fig. 4. UDP CBR latency

remains almost constant until the link delay reaches the value of 10ms. This means that the wireless delay is much bigger compared to the wired one until this point, so any improvement that we could do to decrease the wired delay will nearly not affect in scenarios where the wired delay is below this value.

A.2 Packet loss

When we consider the experiment for handoff latency evaluation (Section V-A.1), what will be the corresponding packet loss? With the 40kbps UDP CBR traffic, a packet is sent every 200ms. Figure 5 presents the number of packets lost because of the handoffs. The result shows that for a link delay value below 100ms we can expect an average number of packet losses per handoff below or around 1 mainly depending on the difficulty in accessing the wireless medium. For higher link delay values the packet losses increase significantly due to the increase in latency. We see that there are nearly no differences with respect to packet losses for the HL-FL and FL-FL handoff cases despite the differences in latency. In both cases, the packet losses are proportional to $2 \cdot ld$. This becomes clear when one notes that an amount of packets proportional to one link delay is lost while the new BU travels from the new AR to the CR. When it arrives there, there is another amount of packet proportional to the link delay on the link between CR and old AR. All these packets will be lost as well.

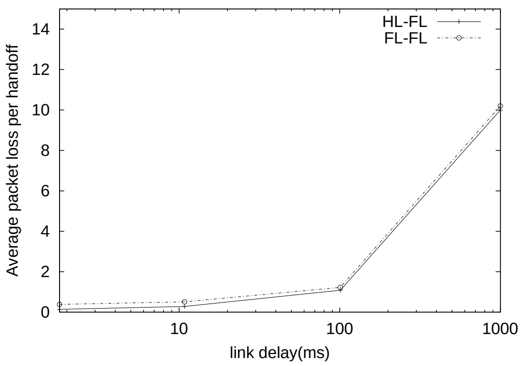

Fig. 5. Packet loss per handoff

A.3 Channel utilization

For the analysis of channel utilization we use subscenario SS2 with 2 ARs and 20 MNs. We study the impact of the number of handoffs on channel utilization with different traffic sources and under saturation or no saturation conditions of the wireless channel. UDP CBR sources are studied in order to determine the impact of handoffs over a continuous data flow with no required acknowledgment. The behavior of bursty sources under an increasing number of handoffs is examined. Endless TCP FTP sources are simulated to investigate the result of handoffs over the congestion control mechanism. UDP CBR sources sending at a rate of 75kbps and endless TCP FTP sources represent the saturation case. The no saturation case is represented by the 50kbps UDP CBR and VoIP bursty sources.

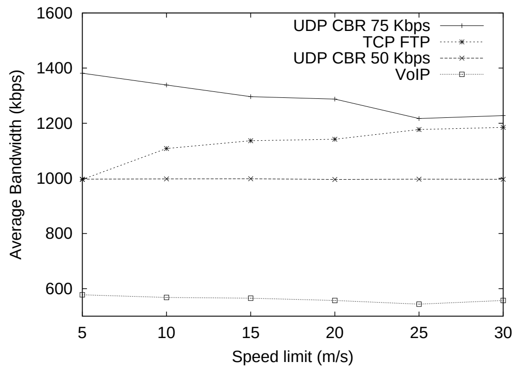

Fig. 6. Channel utilization

In order to vary the average number of handoffs per minute we vary the maximum speed of the nodes. A discussion on the relation between maximum speed and average number of handoffs for the various traffic types is presented below. In Figure 6 we observe that channel utilization is almost not affected by an increasing number of handoffs in the case where the traffic load of the channel is below the saturation throughput. On the other hand, in the saturation case the impact of the handoffs on the experienced throughput is quite significant since from an expected overall throughput of 1500kbps, for the case of UDP CBR 75kbps, we experience a maximum around 1400kbps. Thus, the increase in mobility and signaling on top of an already saturated channel leads to significant performance penalties.

Endless TCP FTP sources always have a packet to transmit. For a small number of handoffs per minute the congestion control suffers the well-known problem of TCP not being adapted to the wireless medium and achieves a channel utilization quite below the saturation throughput. On the other hand, when the number of handoffs increases, TCP increases the achieved bandwidth as well. Probably TCP takes advantage of the higher leftover bandwidth due to the increase of mobile nodes having latency time reducing the impact of the number of handoffs over the channel utilization thanks to the congestion control that adapts the rate to the channel conditions.

To better understand the above mentioned results, we have to show the dependencies between maximal speed of the nodes and the handoff rate for the various traffic types. Figure 7 shows the relationship between the speed limit and the average number of handoffs per minute per simulation and considering different traffic sources. We can observe that the number of handoffs per minute increases according to an increase in the speed limit as expected.

The average number of handoffs per minute per simulation changes depending on the traffic considered. A priori we would expect minor differences but in Figure 8 we can see that the differences are not negligible. The reason for this difference is that loss probabilities of a router advertisement (RAds) depend on the simulated traffic and rate. When a mobile node enters into the overlapping coverage area of two access routers, in some point of time it re-

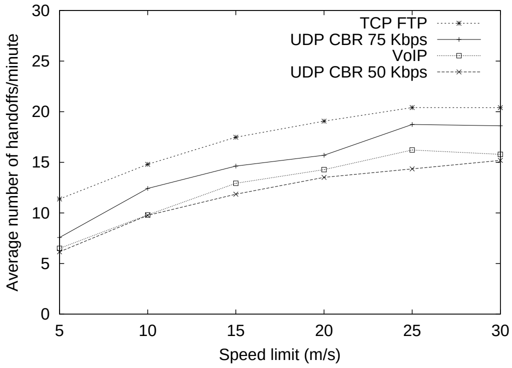

Fig. 7. Relation between maximal speed and handoff rate

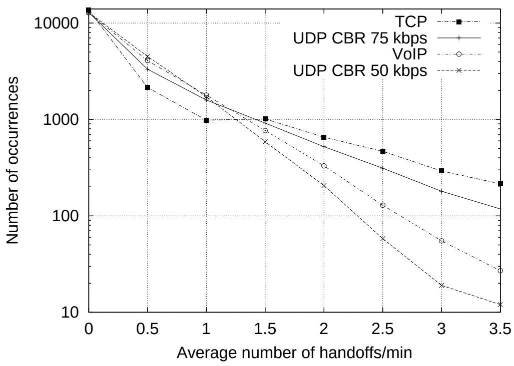

Fig. 8. Average number of handoffs depending on the speed limit

ceives a router advertisement from the new access router. The mobile nodes maintain an AR list with the ARs from which they received router advertisements. When a mobile node enters or stays in an overlapping coverage area if it receives a router advertisement from an access router that it is not in the access router list the mobile node starts the process to attach to the new access router. If the router advertisement received is from an access router that is already in the access router list then the lifetime entry for this access router is refreshed in the access router list. In this way a handoff per each received router advertisement from a different access router while the mobile node moves within an overlapping coverage area is avoided. The problem arises when due to congestion in the wireless channel some router advertisements are dropped. In this case the loss of some router advertisements could lead to the situation of unnecessary handoffs due to the lack of refreshment of the access router list. Figure 9 shows that a higher data rate or bursty traffic produces a higher average number of dropped router advertisements.

A.4 Bandwidth per station

The previous section has shown the handoff impact on the channel utilization. In this section we study the handoff impact over the achieved bandwidth per station which is the impact that the user perceives. Using the same scenario as in V-A.3 we have performed an experiment computing the bandwidth achieved per station. The result of 1000 simulations are represented in Figures 10 and 11. The kind

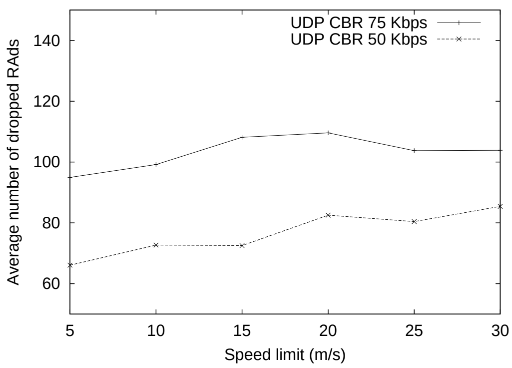

Fig. 9. Average number of dropped RAds

of representation chosen has been an histogram in order to obtain an idea of the probability to obtain the desired bandwidth. The right part of each histogram corresponds to a higher number of handoffs compared to the left part.

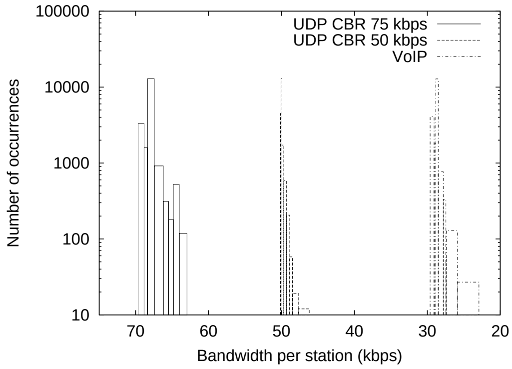

Fig. 10. UDP CBR and VoIP sources

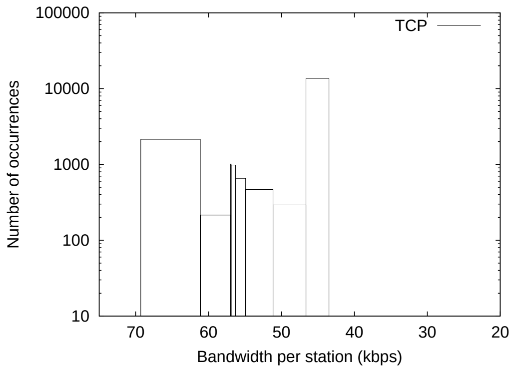

Fig. 11. TCP sources

In Figures 10 and 11 we can observe that UDP CBR 75 kbps, TCP/FTP and VoIP sources result in a wider histogram than UDP CBR 50kbps sources and therefore, a more variable service. UDP CBR 75 kbps and VoIP sources have a wider histogram due to their higher probability to suffer collisions. This higher probability to suffer collisions in the case of UDP CBR 75kbps sources is due to a global data rate close to the saturation throughput and for the VoIP sources due to the produced bursty traffic.

TCP/FTP sources present a wider histogram due to the rate adaptation to the channel conditions.

A.5 Signaling load

An increasing number of handoffs produces a higher signaling load. Considering the same scenario as in section V-A.3 the experiment shows the slope of this increment and the relevance of the signaling load compared to the channel capacity. In Figure 12 we have plotted the amount of signaling data sent over the air. The points in the graph represent the total signaling load in the wireless channel during the simulation. The computed signaling are binding updates and binding acknowledgments. As we can see in the figure, the signaling load over the air for our scenario is lower than 10kbps for UDP CBR and bursty sources. The TCP FTP case shows a signaling load below 20kbps. The signaling load is higher due to the TCP acknowledgments. Thus, mobile nodes sending to the CNs now receive also TCP acks and therefore these mobile nodes have to send BU/BACKs as well while in the case of UDP traffic this is not necessary. However, even for a high mobility of the mobile nodes and a large number of stations the signaling load due to binding updates and acknowledgments remains below 1% of the channel capacity in the worst case.

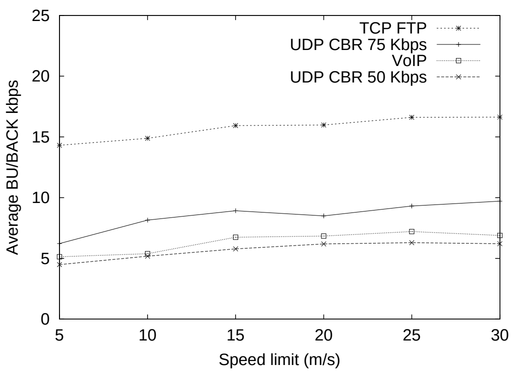

Fig. 12. Signaling load

B. *Route optimization*

An important feature of Mobile IPv6 is route optimization. We analyze the possible pros and cons of using route optimization by comparing measurements for the case of no route optimization with the basic MIPv6 case.

B.1 Latency

When we perform the experiments of Section V-A.1 now *without* route optimization, how will handoff latency be affected?

The results for the scenario used in V-A.1 but without route optimization is given in Figure 13. First of all, for the HL-FL handoff there will be no difference since even in the case of route optimization the MN does not maintain a BU list for CNs while at home. However, we can observe a difference in the Figures 3 and 13 that is due to the fact that in a home to foreign link handoff, with route optimization, there is a period of time where the MN receives the packets

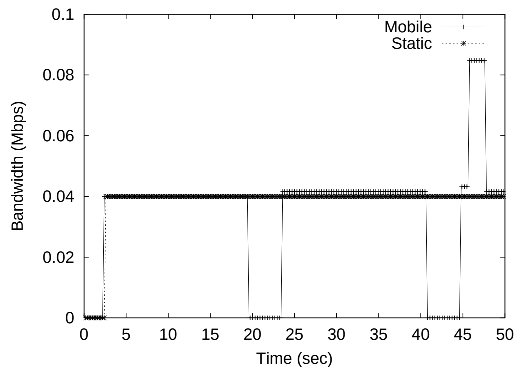

Fig. 13. UDP CBR latency, no RO,1000.8ms link delay

'on the fly' already sent to the HA and the new ones sent by the CN directly. When the delay between the CN and the HA is small this effect is not noticeable (Figure 2). On the other hand, for the FL-FL link handoff we have in both the same effect of receiving packets from two entities during a short period of time. This is because even in the case of not using route optimization the mobile node performing this kind of handoff sends a BU to the previous access router since previous access router forwarding is allowed.

Latency times for route and no route optimization for various link delay values ld are presented in Figure 14. For values of ld below 10ms there is no significant difference between route optimization and no route optimization. However, for larger values of ld route optimization outperforms no route optimization in the FL-FL case. With route optimization, the latency is proportional to the minimum round-trip time between the mobile node and its home agent or its previous access router or its correspondent node while without route optimization it is proportional to the round-trip time between the mobile node and its home agent.

Our simulations on packet losses directly correspond to the HL-FL packet losses investigation of section V-A.2, so we do not include the results here.

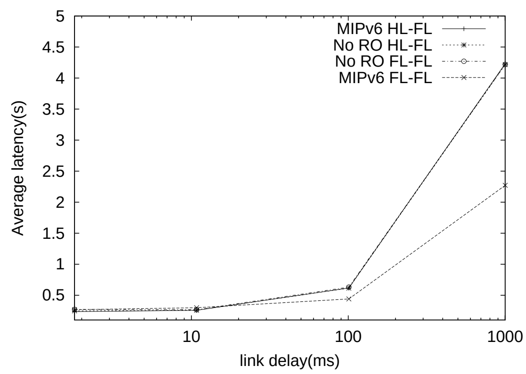

Fig. 14. UDP CBR latency

B.2 End-to-end delay

The main purpose of route optimization is to reduce end-to-end delay by avoiding triangle routing. With a SS2 scenario with 2 ARs and 20 MNs we measured the end-to-end delay between CN and MN for MNs that are attached to a foreign link and analyze the effect of route optimization. The results for various link delays ld are presented in Figure 15.

In the case of no route optimization, the end-to-end delay is composed of the wireless delay plus the delay from the CN to the HA and from the HA to the current AR. In the graph we can observe that for delay values under 10ms there is no difference between route and no route optimization since the wireless delay is the dominating factor. When the wired delay is much bigger compared to the wireless one, we have a direct relation between the link delay and the end-to-end delay of three times the link delay for the case of no route optimization and one link delay for the case of route optimization.

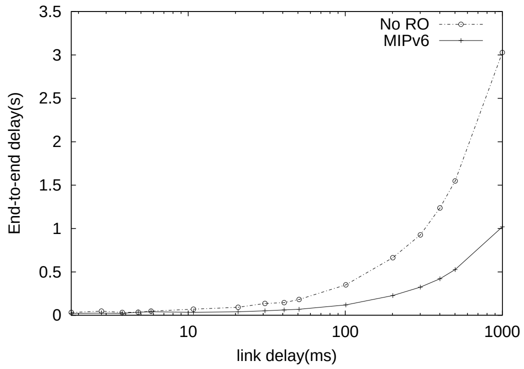

Fig. 15. RO Impact over the end-to-end delay

B.3 Bandwidth per station

The usage of route optimization introduces a small improvement in channel utilization terms especially in saturation conditions as we have seen in the previous section. In Figure 16 the enhancement can be observed with the fact that both histograms have the same shape but the no route optimization one is displaced to lower values of bandwidth and therefore in average a user using route optimization will experience a higher bandwidth. Again, the histogram was obtained from 1000 simulations with a SS2 scenario with 2 ARs and 20 MNs.

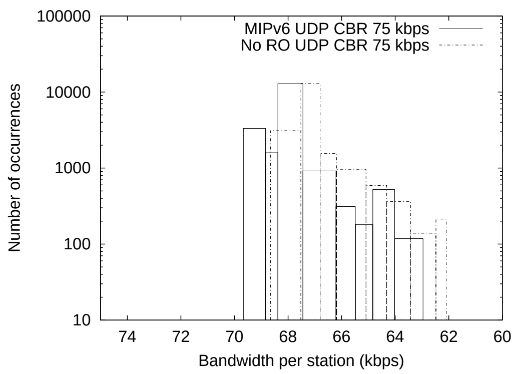

Fig. 16. Bandwidth per station histogram

B.4 Signaling load

For the case of signaling load we can observe in Figure 17 that usually there is a higher signaling load for the case of route optimization, as expected. The measurements were taken from simulations with a SS2 scenario with 2 ARs and 20 MNs. However, the differences are not so significant. As a side note we like to point out that access efficiency will suffer when the number of signaling messages per data packet is increased unless piggybacking is used.

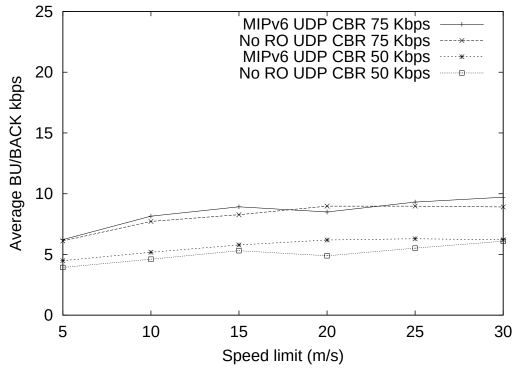

Fig. 17. UDP CBR Signaling load

C. *Fast Handoff procedure*

In this section we will evaluate the fast handoff procedure explained in section IV by comparing it with basic MIPv6.

C.1 Latency

In Figure 18 we compare the latency obtained via the fast handoff procedure(FHO) with the one obtained with basic MIPv6 as explained in V-A.1 The result shows some advantage of using the fast handoff procedure. In the extreme case of one second of link delay we save around two seconds of latency for the HL-FL handoff. When the value of the link delay increases, the importance of saving in the HL-FL case two link delay in the round-trip time of the binding update to the previous access router increases, too.

When we compare Figure 19 with 3 relative to the instantaneous bandwidth we can observe the smaller latency time.

C.2 Packet loss

With the fast handoff procedure, latency and packet losses are fully 'separated'. Since packet redirection takes place when a BU sent by the MN reaches the MNs current AR, there is almost no time to loose a packet (see Figure 20).

In this way two wired 'link delays' are saved in order to keep up to date either the HA or the previous access router with the corresponding saving in packet losses due to a forward to an outdated care of address. When the wired delay increases, the packet losses decrease because the transmission time necessary for a data packet to arrive to the new access router is bigger and when the packets

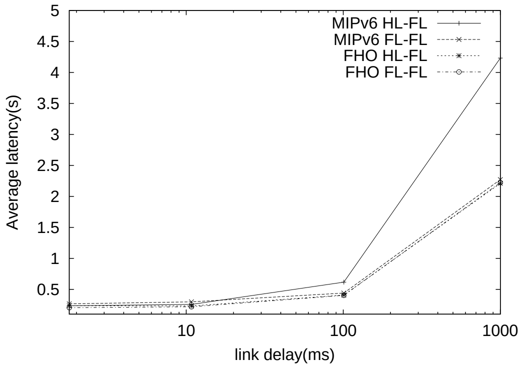

Fig. 18. UDP CBR latency

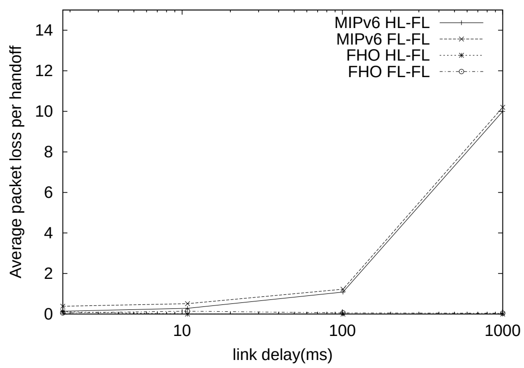

Fig. 20. Packet loss per handoff

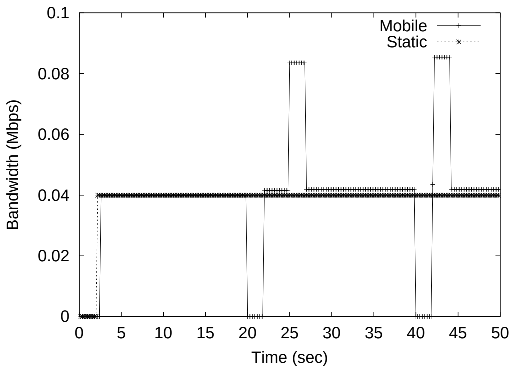

Fig. 19. UDP CBR latency,1000.8ms link delay

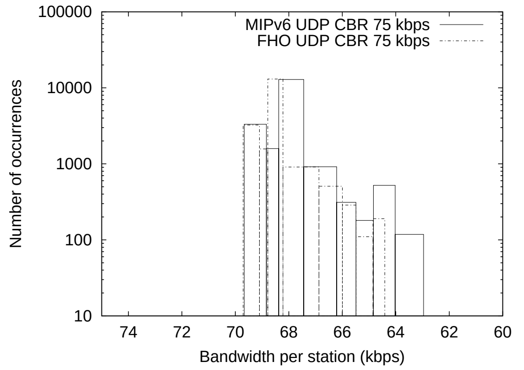

Fig. 21. Bandwidth per station histogram

arrive the mobile node has had more time to complete the handoff process.

C.3 Bandwidth per station

The introduction of the fast handoff procedure in basic MIPv6 reduces the variance of the bandwidth achieved per station. Figure 21 shows a narrower bandwidth per station histogram compared to basic MIPv6 this results in a higher degree of confidence in the expected service.

VI. Summary and Future work

In this paper we presented an evaluation of the performance of Mobile IPv6 in a WLAN-based cellular network by means of simulation with *ns-2*. The employed simulation scenario corresponds to the 'real world' case where IEEE 802.11 access points are attached to various existing IP subnets in a way that the WLAN cells overlap but the various IP subnets might not be close to each other with respect to 'number of hops'. These distances have been modeled as a single 'link delay' whose value is varied for this study.

Main focus was on the evaluation of the handoff performance of basic Mobile IPv6 in comparison with a fast handoff mechanism as well as investigating route optimization. In particular, we showed that the fast handoff mechanism clearly eliminates packet losses. In addition, we checked other QoS-relevant parameters of the system (e.g. bandwidth per station) and how they were affected by mobility.

In addition to the quantitative results provided, the simulations taught us following insights: *i)* differences with respect to latency between HL-FL and FL-FL handoff can occur, *ii)* handoff latency and packet losses might not be directly related to each other, in particular *iii)* the fast handoff mechanism eliminates packet loss but does not improve latency for the FL-FL handoff case, *iv)* traffic type can have impact on handoff rate due to shared access, and *v)* signaling load even with route optimization only contributes a marginal part to the overall rate.

For future work, the study showed the need for a better (e.g. analytical) understanding of a mobility model's parameter's relation with the resulting handoff rate. Furthermore, the fast handoff procedure can clearly be improved by packet bicasting and an enhanced handoff decision algorithm: for example, the MN can estimate the point in time when the first packet will arrive at the new AR and –when permitted by the MN's position– stay with its current AR until this point in time.

VII. Acknowledgments

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