# Wireless Inter-System Quality-of-Service: A Practical Network Performance Analysis of 3G and Beyond

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# Wireless Inter-System Quality-of-Service: A Practical Network Performance Analysis of 3G and Beyond

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**Abstract.** The provisioning of Quality-of-Service and the interworking between heterogeneous mobile environments will be of vital importance for the future economic success of mobile telecommunications. This paper presents a strictly practical performance evaluation of live and lab UMTS/GPRS/WLAN networks as performed within the project WISQY – Wireless Inter-System Quality-of-Service. We report on experiments using Mobilkom Austria's UMTS/GPRS live network and compare the results to lab trials results. Moreover, a novel IPv6-based mobility management approach is evaluated through extensive measurements in a dedicated WLAN testbed. Our performance evaluation results provide quantitative information which might be especially useful for the correct calibration of simulation scenarios for 3G and beyond networks.

## 1 Introduction and Overview

During the next few years, the successful deployment of 3G networks like UMTS and their seamless interplay with Wireless LAN (WLAN) hotspot solutions will be of vital importance for the future economic success of mobile telecommunications. Hence, the provision of Quality-of-Service (QoS) as well as the interworking between heterogeneous mobile environments provides an important step towards All-IP mobile networks and will be one of the crucial prerequisites for launching new services in 3G and beyond networks.

In contrast to the huge variety of related work and projects investigating this research area more from a conceptual point-of-view, the project WISQY – "Wireless Inter-System Quality-of-Service" [11] is focused on a strictly practical performance evaluation of live UMTS/GPRS/WLAN networks. The results presented in this paper concentrate on two key aspects, i.e. (1) the quantitative behaviour of live and lab UMTS/GPRS networks for various measurement scenarios, and (2) optimized IP-based mobility management in a dedicated IEEE 802.11b WLAN testbed.

Therefore, our first main focus concerns the analysis of the overall user experience of QoS in live UMTS networks compared to the (already wide-spread) GPRS networks. Employing an application end-to-end perspective, we focus on the network impact on the application layer, measuring round-trip time (RTT), HTTP, SMTP, POP3, FTP upload and download (including DNS lookup) for connection establishment, data transfer etc. The experiments have been performed using the A1 network, i.e. the live network of Mobilkom Austria, Austria's market leader in mobile communications. Note that already in August 2000, Mobilkom Austria launched the world's first full-coverage GPRS network in Europe, and in September 2002, it became the first European network operator to launch a national UMTS network, with the customer launch in April 2003. Based on this rich experience, the A1 network has turned out to be an ideal testbed for our live measurements.

The GPRS/UMTS live measurements have been repeated in a dedicated lab environment in order to compare the live results with an undisturbed environment. To this end we used the GPRS/UMTS Lab of Kapsch CarrierCom (KCC), a system synnovator of communication technology solutions for fixed, mobile and data network operators. Due to the long-term system integration relationship between Mobilkom Austria and KCC, hard- and software of the GPRS/UMTS Lab is almost identical to the A1 network. The KCC test system is a simple but complete and fully functional GSM/GPRS and UMTS Network (conforming to 3GPP R99 specifications) applicable for end-to-end tests, which can be performed with real GSM/GPRS or UMTS equipment.

Our evaluation results are especially relevant for existing and future simulator developments, as simulation tools for wireless networks require careful calibration with real-world network performance measurements, involving the transfer of huge amounts of data and the associated costs for live network tests. The results also emphasize the need for optimization of highly interactive protocols to handle the high delay of wireless networks in a way that is acceptable to the end user.

Secondly, as far as the optimization of IP-based mobility management is concerned, the main contribution of this paper concerns the performance evaluation of Mobile IPv6 (MIPv6) mechanisms in a multi-operator WLAN environment. The increasing popularity of IEEE 802.11-based WLANs available through hotspot ISP providers, combined with the upcoming 3G cellular network technology, has created the urgent need to coordinate the utilization of these two complementary wireless IP access technologies in order to open their respective advantages for mobile customers. While the link-layer handles the horizontal handover (intra-WLAN or intra-3G) with no need to change the configuration at the IP layer, vertical (inter-technological) handovers between WLAN and 3G require more sophisticated solutions, e.g. the application of MIPv6. In this paper we deal with the reduction of WLAN link-layer Handover (HO) latency, and present a performance comparison between baseline MIPv6 and enhanced Fast MIPv6 approaches.

The remainder of the paper is structured as follows: After Section 2 has introduced the testbed architecture and measurement scenarios, in Section 3 we will take a quick look at the employed measurement tools. Section 4 presents the UMTS/GPRS live and lab performance evaluation results, followed by an overview on our Fast MIPv6 approach for WLAN-based networks in Section 5. Section 6 concludes the paper with a couple of summarizing remarks.

## 2 Testbed Architecture and Measurement Scenarios

### 2.1 GPRS/UMTS Experiments

In our GPRS/UMTS experiments, the mobile client consists of a WindowsXP notebook connected via serial (for GPRS) or USB (for UMTS) cable to UMTS/GPRS mobiles, thus preventing bandwidth bottlenecks and other potential disturbances due to IrDA or Bluetooth connections. All application servers (HTTP, SMTP, POP3, FTP) are running on a SUSE Linux-based server. Our live client has been located at the Institut für Breitbandkommunikation (IBK), a department of the Technical University of Vienna situated close to the city centre of Vienna. The Linux Application Server was connected via the TU Vienna's intranet and through other ISP's to Mobilkom Austria's Gi LAN interface.

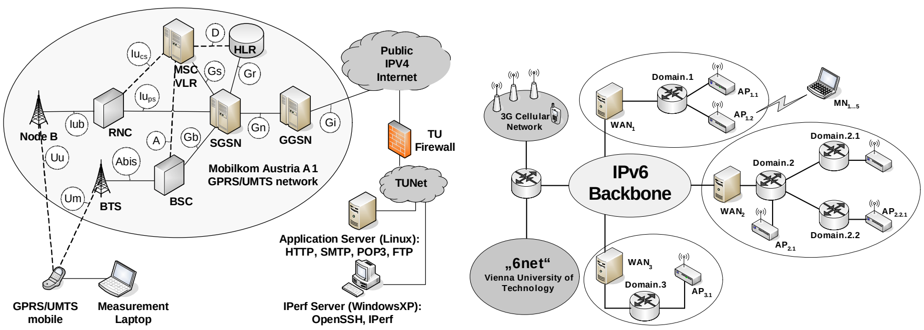


**Fig. 1:** Setup for A1 Live UMTS Network (left) and MIPv6 Wireless Testbed (right)

As already mentioned, we tested the live Mobilkom A1 UMTS network with standard APN (DCH: DL 384 kbit/s, UL 64 kbit/s). A Nokia 6650 UMTS phone was connected to the mobile client (laptop HP Omnibook 6000 with Windows XP SP1) via USB cable. Figure 1 left shows the setup used for our experiments; for a more general introduction into the basic building blocks of GPRS/UMTS networks we refer to 3GPP TS23.060 R99. RNC, SGSN, GGSN, VLR, and HLR were located in the A1 network environment (for live measurements) and the KCC UMTS test network environment (for lab measurements). In the lab environment, both ALONE and IPerf servers were connected directly to the Gi LAN. The DNS server required for the ALONE test suite was installed on the Suse Linux application server. For our measurements, we used GPRS mobile phones like Siemens S55 or TelMe T919 which support GPRS class 10 (total of 5 timeslots, dynamically 4+1 or 3+2).
All tests at KCC were run as exclusive single-user tests, i.e. no other mobile phones could allocate any resources within the mobile network and/or originate any interfering traffic. Additionally, the mobiles were put into an RF shield box to guarantee undisturbed radio conditions. Moreover, the Gi LAN was fully dedicated to test server operation and free of external traffic influences.

### 2.2 Mobile IPv6 Studies

For the MIPv6 performance evaluation, we have implemented an enhanced IPv6 test-bed connected to the worldwide native "6net" infrastructure. As sketched in Figure 1 right, a couple of subnetworks are attached to a central backbone network. Between each pair of different network providers we implemented WAN emulators thwarting all IPv6 packets transmitted. Thus we are able to tune the link-delay individually, depending on the analyzed scenarios. Both IEEE 802.11b and IEEE 802.11g technologies are deployed in the overall wireless hotspot infrastructure.
Three independent network operator domains are deployed, one of them including the Home Agent for the mobile node experiments. Another network operator domain includes some kind of a hierarchical structure in order to allow a performance comparison with the alternative Hierarchical MIPv6 approach. All hosts have RedHat Linux 8.0 with Kernel 2.4.22 installed. For the MIPv6 basic functionality we utilized MIPL 1.0, provided by Helsinki University of Technology (HUT) [3]. The Linux driver for all WLAN activities is based on the HostAP project [4], a very flexible tool for fast link-layer triggering.

## 3 Evaluation Tools

**ALONE.** For measuring the end-to-end application-layer performance for TCP/IP-based application protocols over UMTS and GPRS networks we used the ALONE tool suite. ALONE (Application Oriented Network Evaluation) is a NSPR-library based measurement tool, which implements RFC-compliant clients for HTTP/1.0, SMTP, POP3 and FTP. The ALONE client connects to RFC-compliant application servers, opens a protocol session and exchanges data with these servers according to the selected protocol. It records the success or failure of the protocol session along with the duration of DNS lookup, end-to-end connection establishment, data transfer and total session duration. ALONE's data exchange is file-based, i.e., the ALONE client uploads or downloads files according to the selected protocol. Our tests transfer zip-compressed files to exclude server-side optimizations (data compression) in the mobile operator's network.
For SMTP, POP3, and FTP ALONE supports multiple data transfers within one protocol session. E.g., the ALONE FTP client opens a connection to an FTP server, authenticates itself once and then transfers the user-specified data file n times. This 1:n data transfer model extends the 1:1 model commonly used for performance tests.

**IPerf (TCP/UDP).** The ALONE-measured application-level network performance is compared against the raw UMTS/GPRS TCP/IP network throughput as reported by IPerf 1.7.0 for TCP. We also measured UMTS network streaming performance using IPerf 1.7.0 in UDP mode and ICMP packet round-trip times using the standard ping utility with varying payload size.
ALONE, IPerf and Ping have been invoked by means of Perl wrapper scripts. The scripts automate our tests by starting clients and servers locally and remotely with the pre-configured range of parameters like tested protocol, file name, file size, payload size, timeout values, etc. Moreover, they implement automated dial-in and disconnect

procedures, log the duration of these commands, and parse, process and log the output of third-party tools like IPerf. Local and remote systems are synchronized via SSH client/server communication.

**WebSim.** For more detailed HTTP measurements we used the WebSim traffic generator tool developed at FTW [2] for simulating HTTP/1.0 traffic according to the SURGE model [1]. WebSim consists of a server simulating an HTTP server, and a client simulating user behavior. The client requests typical pages (consisting of several files) by opening a new TCP session for each requested file, thus allowing for multiple simultaneous (parallel) TCP sessions, and simulates user think times between page requests and parsing times of the browser. Pages typically consist of an HTML document with several embedded objects. The server responds with files whose file size distribution corresponds to the SURGE model. For further information on the WebSim traffic generator we refer to [2].

**i-Motion.** i-Motion is our new tool for reducing the link-layer disruption in WLAN environments [6]. It aims at minimizing the considerable service interruption period between the interface card disconnecting from the previous link and the Mobile Node arriving and reestablishing at the new Access Point (AP). Initial analysis of the IEEE 802.11 link-layer handover process has already been performed in [8], [9] and [10], suggesting how to reduce its duration. There, it is demonstrated that the HO latency can vary between 500 and 1500 ms, depending on the deployed equipment. The major latency problem with HOs is caused by the firmware inside the interface-card detecting the lack of radio connectivity only after several unsuccessful frame transmissions. Therefore, in our WLAN testbed we implemented i-Motion as an enhanced mobility management tool that continuously monitors the radio signal quality of the attached AP and starts searching for alternative APs already before any frame has been dropped. In this way the overall HO time can already be reduced by 50 % compared to traditional implementations [6].

## 4 Results

As pointed out before, our mobile network performance analysis reflects the current status of GPRS and UMTS networks from an end-user perspective, both in live networks and in test laboratory environments. We measured raw TCP/IP throughput, application-layer throughput for HTTP, SMTP, POP3 and FTP, UDP bandwidth, packet loss and jitter, and ICMP packet round-trip-time. All measurements in the live network were evenly distributed over at least one week to reflect a representative load in the live GPRS/UMTS network.

According to the technical specifications, UMTS networks should outperform GPRS networks by at least one order of magnitude both in terms of throughput and round-trip-time. On the other hand, GPRS networks are already highly optimized while UMTS networks have just started to operate. Our measurements analyze the benefit end-users can expect from a high-performance infrastructure like UMTS compared to GPRS. We point out potential performance limitations due to the design of application

protocols under specific traffic situations which may have a negative impact on the user acceptance of UMTS and other wireless network technologies.
Note finally that all our tests rely on a strict end-user view of the mobile network. We regard the GPRS/UMTS network as a black box with no user-configurable parameters. Optimization was restricted to selection and configuration of mobile devices and to the fine-tuning of operating system parameters.

### 4.1 Raw Network Performance: TCP and UDP

Starting with TCP, we tested 30-second TCP/IP throughput bursts with IPerf 1.7.0 in the UMTS and GPRS network as a reference for our application-level measurements. According to our measurement results the TCP window size has visible but minor impact on the throughput.

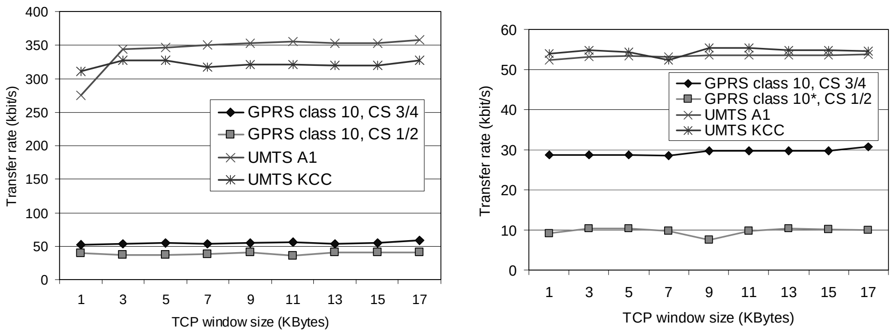


**Fig. 2:** IPerf TCP/IP: Download Transfer Rate (left) and Upload Transfer Rate (right)

Figure 2 shows that the TCP/IP throughput of GPRS networks depends on the number of GPRS timeslots allocated for upload and for download and the GPRS coding scheme, but not on channel conditions, user equipment etc. The number of timeslots available to the user for data transfer depends on the mobile devices' GPRS class, on the maximum number of GPRS timeslots that the operator grants to one mobile device and on the current mobile network load. The A1 GPRS network fully supports mobile phones up to GPRS class 10. This enables allocation of a total of 5 timeslots, either as 4 download + 1 upload or as 3 download + 2 upload. The allocation changes dynamically depending on the predominant data transfer direction. From Figure 2 we can see that the A1 network uses the (4+1) scheme for TCP/IP download and switches to (3+2) for TCP/IP-based upload. Combined with GPRS coding scheme CS 3/4 the A1 network transfers an average of up to 60 kbit/s.
We compared these results against measurements within the KCC GPRS test network. The KCC network was configured to support CS 1/2 (in order to allow a comparison to CS 3/4 in the live network), a maximum of 4 timeslots for download and 1 timeslot for upload. Figure 2 left shows the difference in download throughput over GPRS networks for 4 timeslots using CS 3/4 and 4 timeslots using CS 1/2. The performance

gain of GPRS CS 3/4 compared to CS 1/2 is remarkable both in upload and download direction.
UMTS outperforms GPRS both in TCP/IP upload and download direction. UMTS upload throughput peaks at 55 kbit/s, which is about twice the throughput of a GPRS mobile with CS 3/4 and full operator support (two timeslots in upload direction). Finally, our live network UMTS download measurements report a peak of 357 kbit/s goodput, which is close to the theoretical feasible throughput for UMTS network micro cells. Note that the KCC UMTS Lab network configuration was not optimized for maximum throughput leading to slightly lower download results. The UMTS download throughput amounts to six times the maximum GPRS download performance.

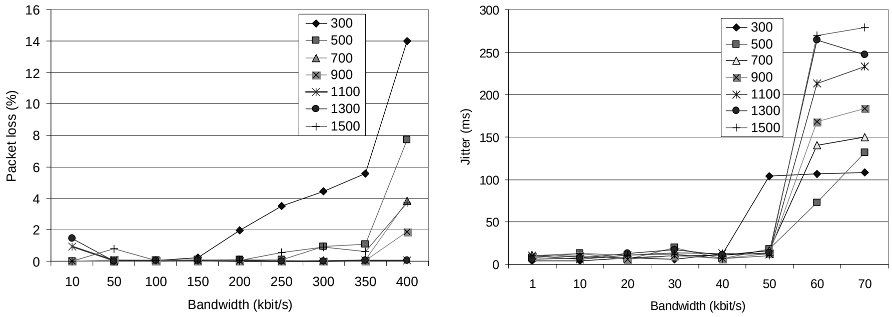


**Fig. 3:** IPerf UDP: Download Packet Loss (left) and Upload Jitter (right) in UMTS networks

Turning now to the case of UDP, IPerf sends UDP packet streams with user-configurable bandwidth and UDP packet size from clients to servers and reports jitter and packet loss for the stream. We tested all possible combinations of packet lengths between 300 and 1500 bytes (step 200 bytes) and stream bandwidths between 1-70 kbit/s (step 10K) for upload and 10-400 kbit/s (step 50K) for download. During one UDP stream test of 30 sec the bandwidth and packet parameters were kept constant.
Figure 3 left displays the packet loss for UDP stream downloads over UMTS networks. The packet loss increases with increasing bandwidth; above 300 kbit/s the loss becomes significant for all UDP packet sizes. Small UDP packet streams are more likely to suffer from packet loss than streams transfering large UDP packets because of the significant UDP/IP overhead associated with small UDP packets. 300-byte payload UDP packet streams waste roughly 10 % of the available bandwidth for overhead, whereas for 1500-byte payload streams, overhead becomes less than 2%.
The jitter in upload direction increases to values above 150ms when the upload bandwidth exceeds 50 kbit/s as shown in Figure 3 right. Like in download direction streams consisting of small UDP packets are more severely affected by jitter.

## 4.2 Ping ICMP Round-Trip Time

In the next experiment, we measured average ICMP packet round-trip times (RTTs) for varying packet payloads. The payload was verified to be identical for sent and

returned packets. Figure 4 left shows the average ICMP RTT for GPRS and UMTS networks. We executed the complete ICMP test 65 times for any payload size between 100 and 1450 bytes, distributed over a one-week interval. Any of these measurement points averages the result of 20 ping packets. The results are a good approximation of RTTs to be expected for UDP packets over GPRS and UMTS networks.
The round-trip performance that we measured for UMTS networks is excellent compared to GPRS. UMTS ranges from less than 150 ms for 32 byte and 160 ms for 100 byte packets to 537 ms for 1450 byte packets while GPRS round-trip-time peaks at 2142 ms for 1450 byte packets. Typical SIP/IMS messages carrying a payload of around 800 bytes will incur an UMTS RTT of approximately 360 ms.

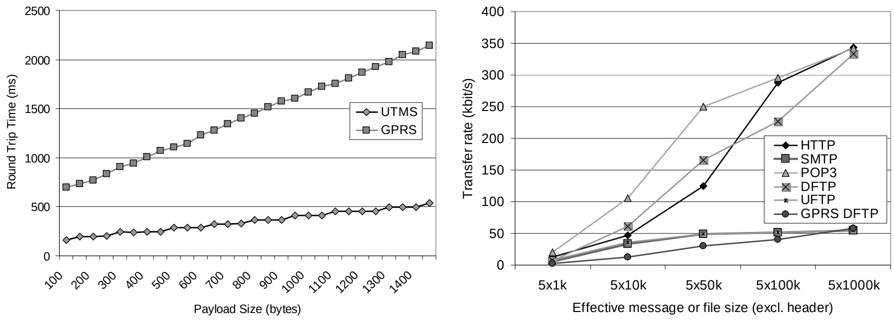


**Fig. 4:** ICMP Ping RTT (left) and Application-Level Transfer Rate (right) for UMTS

### 4.3 Application-level Performance

The IPerf measurements approximate the optimum transfer rate and round-trip-time for the tested GPRS and UMTS networks. Mechanisms implemented in the layers positioned between network layer and end-user require part of this available network resource. Examples include application protocol headers, application server delays, handshakes, round-trip delays, etc. and optimization techniques like TCP slow start and Nagle algorithm – to name just a few. End-users do not gain the full benefit of the network performance except for some huge file transfers or streams.
We measured application-layer performance using ALONE and WebSim and compared the results against network performance measured with IPerf. We evaluated the impact of specific application protocols on the network performance offered to the end-user. Our conclusion is that existing protocols waste a huge part of the available network resources and must be explicitly optimized for the use in wireless networks.
Figure 4 right depicts the application-level, end-to-end transfer rate of the application protocols HTTP, SMTP, POP3, Download FTP and Upload FTP. The diagram shows that the user gains full benefit of the UMTS network download performance only for files exceeding 1MByte of size. FTP download is outperformed by HTTP and POP3 because of the round-trip times that are required to start and end a FTP file transfer. The chart is based on the true payload that is transferred by HTTP, SMTP and POP3

and ignores protocol-specific headers added to the message or file. It must be stated that POP3 and SMTP are both not capable of transferring binary files while FTP copes well with both, text and binary files. The transfer of binary files using POP3 or SMTP results in part of the available bandwidth being wasted for encoding.
Figure 5 shows more detailed results for HTTP/1.0, i.e. the transfer rate over web page size (including embedded objects) for the UMTS (left) and GPRS (right) live network. Due to the delay incurred at every connection establishment the user perceived transfer rate varies according to file size. Also, because HTTP/1.0 opens a new TCP session for each embedded file, pages made up of several small files suffer a significant decrease in transfer rate. On the other hand we observed performance near the theoretical maximum of UMTS for files of 1 MB and above (approx. 335 kbit/s). For GPRS we measured a maximum of around 60 kbit/s. This results in min. page download times of 0.3 sec for UMTS and 1.3 sec for GPRS which shows the improvement UTMS brings for user perceived QoS of interactive web applications since responses times below 1 sec are typically perceived as "fast" and below 300 ms as "instantaneous".

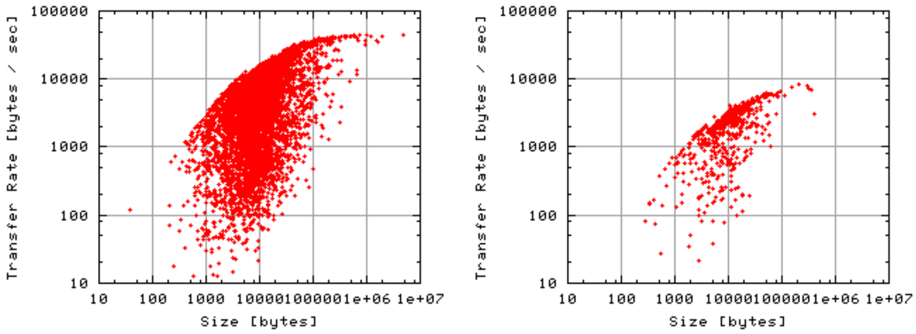


**Fig. 5:** User Perceived Transfer Rates for HTTP/1.0: UMTS (left) and GPRS (right)

Our measurements indicate that TCP/IP connection establishment and synchronization handshakes are major penalty factors. POP3 uses one single TCP/IP connection for the transfer of all 5 messages and performs best of all tested protocols, especially for small files. HTTP 1.1 should perform significantly better than HTTP 1.0 for pages made up of several small files because of TCP/IP connection re-use.

## 5 Mobility Management for All-IP Networks

### 5.1 Basics of Fast MIPv6

Using Mobile IPv6, a mobile node can effectively maintain its IP-layer connectivity to the Internet while changing its point-of-attachment. During the accomplishment of the handover, the mobile node is unable to send or receive IPv6 packets because of its L2 and L3 handover operations. This high handover latency is unacceptable to real-time applications or delay sensitive traffic. Each time  a mobile client moves, it is necessary to perform movement detection by its current point-of-attachment. In Mobile IPv6 [5], the movement detection algorithm relies on the periodic sending of router advertisements in order to enable the mobile node to determine its current location. The only way to improve the detection performance is to broadcast router advertisements at a faster rate, which may result in a poor link utilization. For that reason, a fast handover protocol is designed to achieve a seamless handoff when mobile nodes switch from one subnet to another.

The Fast Mobile IPv6 approach enables a Mobile Node to quickly detect at IP-layer that it has moved to a new subnet by receiving link-related information from the link-layer and furthermore gathering anticipative information about the new Access Point and the associated subnet prefix when the Mobile Node is still connected to its previous subnet. It is also possible to initiate vertical handovers between different wireless access technologies. In the mobile-initiated and anticipated fast-handover scenario described in [7], the mobile node first sends a Router Solicitation for Proxy (RtSolPr) message to the current access router containing any Access Point specific identifiers.

The trigger for sending the initiating RtSolPr message can be derived directly from a link-layer specific event, e.g. an imminent movement to an AP providing better signal-quality. Consequently, the MIPv6 protocol assumes that the link-layer protocol is capable of delivering the L2 identifier of the new access point to the mobile node. For seamless handover initiation it is more important that the current AR must be able to map the new L2 identifier into the IP address of the target AR. In the remainder of this section we will present first testbed results, which demonstrate the beneficial behavior of the Fast Handover approach for Mobile IPv6 in Wireless LAN based networks.

### 5.2 Performance Evaluation of Link-Layer HO

Figure 6 left presents the results for link-layer HO with and without our i-Motion monitoring tool for a client with maximum throughput at each AP and demonstrates the reduced HO latency behavior in comparison to conventional WLAN switching.

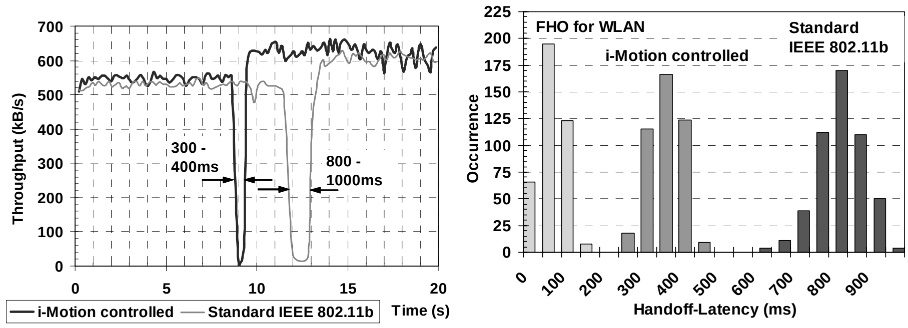


**Fig. 6:** IEEE 802.11b Handover Latency

For further optimization of the HO latency, we remove the firmware-based Active-Scanning procedure during the execution-phase. An Active-Scan is a procedure in which a WLAN station searches for all APs (on all available channels) in range. Each

AP can be addressed by a specific Service-Set ID (SSID). As the i-Motion monitoring tool in our scenario already knows the exact SSID for the selected AP, it does not make sense to restart an Active-Scan once more. Scanning all channels takes time and is not desirable when the execution of movement to another AP is going on. For that reason we re-developed the WLAN hardware drivers to achieve switching without active-scanning. Figure 6 right demonstrates the result for a WLAN station moving in between two APs: we observe that the delay without any optimization is about 800ms, the handoff delay is about 350ms for predictive mode of i-Motion, whereas for the same experiment with enabled i-Motion monitoring tool and skipped active-scanning phase, the overall delay is only about 80ms. This is an excellent outcome for these initial optimization approaches. However it should be noticed that the presented HO results for IEEE 802.11 are obtained through measurements in our testbed and carried out for only 4 WLAN station at the respective APs. It is expected that the HO delay might slightly increase for scenarios with additional user activity.

### 5.3 Performance Evaluation

This section presents our evaluation results for the implemented basic Mobile IPv6 wireless testbed. Furthermore, interesting results will be demonstrated for the application of enhanced fast HO mechanisms for MIPv6. All measurements consist of roughly 1000 samples averaged for each individual point.
Figure 7 left demonstrates the functionality of the IPv6 Router Discovery mechanism. It shows the dependence of the overall handoff latency (L2 and L3) on the frequence of sent Router Advertisements (RA). Obviously, the HO latency decreases as RA messages are sent more frequently.

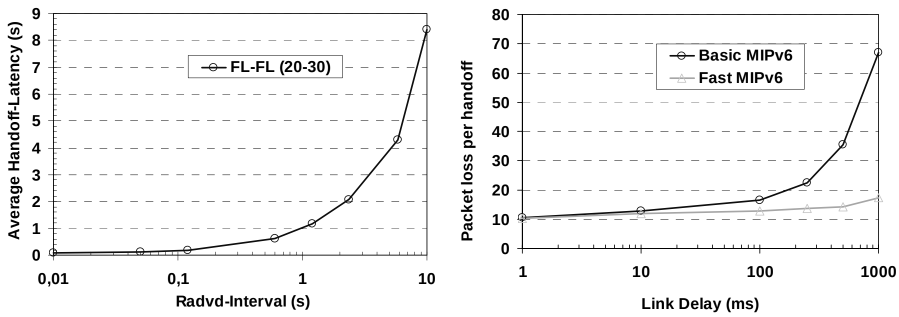


**Fig. 7:** HO Latency for Varying RA Intervals (left) and for Basic Mobile IPv6 (right)

Figure 7 right illustrates the packet loss results for an IPerf-generated UDP-data stream of 160 kbit/s in between the Mobile Node and its correspondent node. As already mentioned, we deployed WAN emulators in order to be able to delay packets in between each provider subnet and the core network.
As can be observed, the packet loss during the HO between different network operators highly increases for basic MIPv6. In contrast, the Fast Handover approach shows

a lower increase for the whole range of wired link-delays. These results show that the Fast Handover approach for WLAN based networks may provide acceptable performance in a realtime multimedia infrastructure (e.g., VoIP over WLANs).

## 6 Summary and Conclusions

This paper is dedicated to a couple of central results achieved within the application-oriented project WISQY. We have investigated two important topics in the general context of performance analysis for UMTS/GPRS/WLAN networks: first, we provide a quantitative evaluation of live and lab UMTS/GPRS networks with respect to several important parameters which are extremely relevant e.g. for the correct calibration of future simulation tools for heterogeneous wireless environments. Secondly, a novel IP-based WLAN mobility management approach has been introduced and evaluated with respect to optimized IEEE 802.11b HO latencies.

Future work in this area includes QoS and performance evaluations of IP Internet applications for mobile networks, especially investigating the planned introduction of the 3GPP IMS (IP Multimedia Subsystem) architecture.

**Acknowledgements**

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