

Performance Evaluation of Multimedia Services in Heterogenous Wireless Environment

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Abstract— Emerging multimedia services are very attractive for next-generation wireless networks. However, both limited quality of service (QoS) support and lack of bandwidth at the air interface restrain their wide deployment in current evolution of third generation (3G) wireless systems. Mp3-based entertainment applications, such as file downloading and streaming which are common nowadays in the Internet, are supposed to be very challenging and demanding services for next-generation wireless networks. In this paper we consider mp3 traffic delivery using two wireless access technologies, which are claimed to be used in next generation wireless networks, wireless local area network (WLAN) and general packet radio service (GPRS). We compare the QoS of these services provided through wireless access with local area network (LAN) access technology. Additionally, we practically investigate peculiarities and specific issues arising in QoS assurance for both mp3 streaming and downloading services.

Keywords— WLAN, GPRS, phase2+, wireless access, testbed.

I. INTRODUCTION

NO wadays, a considerable part of research activities in telecommunications are switching towards development of third and fourth generation (3G/4G) IP-based mobile networks. The major motivation behind that is to introduce a common service platform and transport facilities for future composite Internet mobile network.

In addition to broadband wireless access to the Internet, next-generation wireless networks should be able to provide quality of service (QoS) to various applications. In order to achieve that four traffic classes have been defined by 3G partnership project (3GPP). These are conversational, streaming, interactive and background traffic classes. The conversational and streaming traffic classes are used to carry real-time traffic flows like audio and video. The interactive and background classes are used for non real-time traditional Internet applications (web browsing, telnet, e-mail, FTP).

New multimedia services attracted for next-generation wireless networks, such as downloading and streaming, are striving hard towards commercial market. Both limited QoS support and scarce bandwidth of wireless environment can restrain their wide deployment in 3G/4G.

The most critical layers for QoS support in next-generation wireless networks are physical, data link and transport. In this paper we restrict our attention to transport layer only. The reason behind that is

to check whether the particular multimedia services can already be smoothly implemented on commercial basis over currently available wireless access networks or not.

In this paper we consider two access networks which are claimed to be used in next generation wireless mobile systems. These are wireless local area network (WLAN, IEEE 802.11b) and current state (2.5G) of 3G evolution - general packet radio service (GPRS), which is already implemented in most European GSM networks. In 2.5G networks there is a clear separation between wireless part Radio Access Network (RAN) and fixed one - Core Network (CN). RAN is used to hide all access specific peculiarities from the CN. Because of that, CN has a little impact on introduction of new RANs, and therefore, can evolve independently. Additionally, it is necessary to mention here that WLAN and GPRS are not competitors, but associates mostly because in next-generation wireless networks the different roles are assigned to different access networks.

A clear separation between RAN and CN has already led the cellular network to multi-access environment. This environment introduces an additional conceptual notion of next-generation wireless networks - an Always Best Connected (ABC) concept [1]. Implementation of ABC will allow users of next-generation wireless networks to choose the most suitable RAN at any instant of time during the whole duration of call. Particularly, this feature is claimed to be very attractive for users with complex and variable mobility patterns. Technical implementation of ABC is to be based on dynamic intersystem (vertical) handover that should be implemented in a seamless way between any types of access networks [2].

In our study we have implemented series of tests of both wireless access networks and compare obtained parameters with fixed local area network (LAN, IEEE Ethernet) access, which is currently de-facto standard in multimedia networking. Indeed, it is highly anticipated [3] that testing of real implementations can bring better understanding and new knowledge in the area.

We concentrate our attention on tracing of real traffic, but not simulated one. We consider two mp3-based entertainment applications: mp3 file downloading and streaming as far as they are supposed to be very challenging and demanding services for next-generation wireless networks.

From traffic transmission point of view, mp3 downloading service has essential peculiarity like relatively small time of connection/flow (transmission phase)

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duration. Indeed, traffic load simulators allow us to generate bulk traffic and therefore obtain some type of stationary traffic processes. However, taking into account average sizes of mp3 files, MPEG-based compression techniques and unpredictable nature of music sources we can state that mp3-based applications are intended to produce traffic, which cannot be stochastically described by stationary mathematical processes. Even in those cases when, based on certain available information about codecs and music sources, we can expect some type of stationary behavior of mp3 traffic, we are still not able to *prove* it statistically because of short duration of mp3 sessions.

Second type of mp3 applications we considered here is a streaming-based one. Internet Live Radio stations are widely spread nowadays in the global network and QoS perceived by end-user varies in average from good up to excellent. For example, some of stations are already available in 128kbps stereo format and can be streamed continuously for several days without brake in connection via public Internet.

The rest of the paper is organized as follows. Some general prerequisites are considered in Section II. Testbed configuration is given in Section III. Carried measurements are outlined in Section IV. In Section V we discuss obtained results. Conclusions are drawn in last section.

II. TESTING PREREQUISITES

A. GPRS and WLAN

GPRS phones' data exchange is asymmetrical. Data uploading (e.g. sending an e-mail message) is slower than downloading (e.g. reading an e-mail message). It denotes by couple (U,D) where the first number indicates the phone's ability to employ channels for downloading while the second one indicates the ability to employ channels for uploading. Thus, GPRS (3+1) phone can send data by one and receive it by three channels, data transmission rates are distributed between all employed channels. However, downloading data by three channels does not mean that the data transmission rate grows three times since in order to better utilize the channel's capacity, four different coding methods are used (correspondingly CS-1 9.05, CS-2 13.4, CS-3 15.6 and CS-4 21.4 kbps per one channel). Currently, most commercial networks support only two coding schemes (CS-1 and CS-2) and, therefore, theoretical GPRS' maximum rate cannot be achieved.

Initially, GPRS supports up to 8 channels on each bearer frequency. It may be the case that all channels will never be used for GPRS data transfers since GSM voice calls employ the same channels and the probability of voice call blocking can increase substantially. Additionally, it is supposed that the phones will not be given eight time slots in near future. The other reason is that it would require greater computing efficiency (and, therefore, result in greater power consumption) that phones lack.

It is known that WLANs were initially developed to match requirements of non real-time services like file transfers, while cellular networks, including GPRS, are targeted on constant-rate voice communications. In order to enable IP-based multimedia services for nomadic users, it is necessary to study a system architecture that combines short-range broadband wireless access (IEEE 802.11b) with cellular access technologies (GPRS). The main reason to consider such heterogeneous environment is that it is almost impossible to define a RAN that combines all advantages of different access technologies [4].

B. Multimedia traffic

Multimedia applications are continuously growing in popularity and the availability of high-speed access networks is the primary reason behind that. Indeed, nowadays it is strategically necessary to support these services over wireless networks.

Basically, multimedia traffic consists of one or more media streams and can be characterized by strict delay requirements while it can tolerate some losses. It is supposed that applications emerging from Internet will become capable of defining the required QoS level soon. However, currently in almost all networks multimedia traffic is treated similar to ordinary best effort data traffic, which do not often require strict delay bounds. Therefore, it is crucial to predict the QoS degradation that may be experienced by multimedia applications over wireless access networks.

In our paper we consider two multimedia entertainment applications: mp3 file transfers and mp3-based Internet Live Radio. Depending on bandwidth requirements, these applications fall into two major categories: limited bulk transfer (mp3 downloading) and controlled transfer (Internet Live Radio). Note that from the user point of view both services behave quite similar and can be described by two phases: prefetching phase and playback phase. While in prefetching phase the application stores data and then turns into playback phase. When both applications are in prefetching phase they use all available bandwidth to prefetch data. However, when playing back, mp3 player continues downloading at maximum available bandwidth while Internet Live Radio restricts itself to the certain necessary rate.

We also would like to note that it is almost impossible to characterize mp3 traffic analytically. It stems from both relatively short duration of mp3 session and unpredictable nature of mp3 content. Because of these reasons we cannot model mp3 bulk traffic as a stationary process which is often used for traffic modelling, since it is not possible to judge whether the traffic is stationary or not. It is also too complicated to model Internet Live Radio traffic mathematically since we are dealing with source-controlled service. Therefore, real measurements are the only way to characterize these services.

C. TCP-related issues

Logically, streaming services have to be implemented over UDP/RTP(RTSP) combination, however we found that all Internet Live Radio stations are currently based on TCP protocol. One of possible reason behind this fact is that the overall Internet quality has increased significantly each year in the past. Indeed, using ordinary LAN access we have traced short prefetching and very rare re-prefetching periods. It gives us the possibility to confirm the thesis from [5] that the overall quality in public Internet is growing in terms of both average delay and loss. Moreover, TCP stream much easier than UDP one can traverse through end user's firewall.

As far as TCP is a transport layer protocol used in implementations of downloading and streaming services of mp3-based applications, we cannot neglect wireless TCP issues in our study.

There are plenty of studies devoted to TCP performance evaluation in wireless environment. For example, it is highly anticipated that isolation of wireless link from the rest of the network is promised solution [6]. The other approaches propose some improvements on end-to-end basis. However, most of proposed approaches has been verified on simple simulation studies and do not have software implementations yet. Therefore, currently wireless TCP implementation cannot be easily included into testbed of real environment.

Additionally, we note that in practice we already use ordinary '*fixed*' TCP implementations over wireless links(for example, Internet Live Radio service over WLAN). Note that in this scenario the perceived QoS and performance are good in non-nomadic and limited nomadic cases. Based on these arguments in our study we admit the usage of ordinary '*fixed*' TCP implementation provided by operating systems of computers we have used in testbed.

III. TESTBED

A. Configuration

Our testbed has been built in such way that we were able to test and compare performance of different access networks for different workloads and mobility patterns. We have used three access networks: WLAN, GPRS and fixed Ethernet LAN. Note that the Ethernet LANs are currently seen as a de-facto standard in multimedia networking.

We have chosen two types of traffic workloads, which are basically assumed to be demanding in next-generation wireless networks. These are limited bulk (mp3 file downloading) and controlled (mp3 streaming) transfers.

Testbed configuration is presented in Fig.1. In our testbed environment we have used several computers equipped with different operating systems (OS) and different access network devices. The mobile node called '*mobila*' was Mac PowerBook G4 under Mac OS X Jaguar v.10.2. It was equipped

with WLAN 802.11b (interior) and Bluetooth facilities (used to connect computer with GPRS adapter namely GPRS phone). To ensure mobile node performance against OS-specific issues we have validated all our tests with different mobile node which was based on IBM ThinkPad PIII laptop under Win2000 OS and equipped with exterior Cisco's WLAN 802.11b card and same Bluetooth adapter. To emulate the mp3 file server we have used desktop PC PIII under Win2000 OS connected to 100 Mbps Ethernet LAN. This node is called '*mp3-source*' in Fig. 1. Again, to ensure fixed node performance against OS-specific issues we have validated our tests with a different fixed node, which was based on a PC PIII with Linux OS. This was done because the end-user perceived QoS may vary significantly from one OS to other. Additionally, it was stimulated by the fact that different operating systems give slightly different performance results. The detailed analysis of this fact is out of scope of this paper, but in our tests we have decided to use a mobile-fixed couple PowerBook Mac OS X - PC Linux.

To evaluate the performance of live mp3 streaming service we have chosen one of the stable Internet Live Radio. This machine is called '*mp3-live*' in our testbed (Fig. 1). It is clear that paths between the end user and both mp3 file server and Internet Live Radio node are very different. To overcome this obstacle we performed traffic tests of each destination over relatively large time periods. We found that the both paths are stable, and therefore, can be used in our testbed.

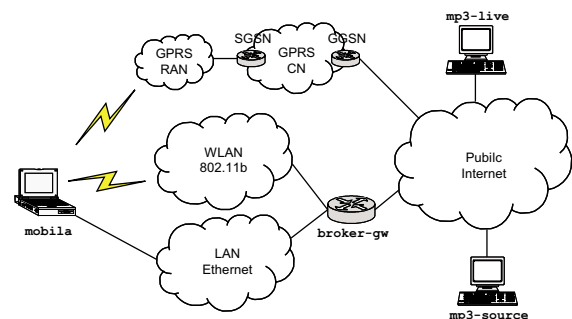


Fig. 1. Testbed configuration

We also have to note that in this paper we present results only when locations of nomadic users were fixed. The access networks we used in our tests provide connectivity for user with different models of behavior. WLAN is oriented to user with hot-spot areas and therefore considerable number of subscribers will have fixed or limited nomadic behavior. GPRS is oriented to nomadic users, but fixed ones will be served just with better QoS parameters. In our testbed environment we consider that user's behavior will be characterized by at most as limited nomadic. To get a real example we can guess the situation when nomadic user is coming with live GPRS connection from outside into airport WLAN hot spot area.

B. Tools and environment

WLAN tests were carried out on a base of running implementation of WLAN 802.11b in campus area of Tampere University of Technology.

In order to enable GPRS access we used tandem configuration of Nokia 7650 terminal and Mac PowerBook G4 connected via Bluetooth. We consider that Bluetooth access is not a bottleneck in our testbed, because both devices were located near by each other, i.e. signal strength and channel error rate were in acceptable bounds. In our testbed we used GPRS connection via one of the commercial Finnish GPRS networks, which is currently offering GPRS (3+1) time slots allocation with CS-2 coding scheme. Therefore, theoretically, our GPRS is capable to achieve 36 kbps downlink and 12 kbps uplink transfer rates.

To enable LAN access we used 100 Mbps Ethernet. Both WLAN and LAN are connected to campus network via 'broker-gw' edge router (Fig.1). In our tests this point is used as reference one.

To capture generated traffic and to obtain statistics we used Ethereal software [7] package in conjunction with own post processing Perl scripts.

C. End-to-end performance testing

Before performance evaluation of mp3-based services we had to explore performance characteristics of each access network. Several advanced UNIX-based utilities [8] have been used. We obtained end-to-end performance parameters via three different access networks, i.e. for 'mobila'-'mp3-source' route via WLAN, GPRS and LAN. The values for these parameters are presented in Table I, where 'T - E.R.' denotes the path between terminal and 'broker-gw' edge router. The following parameters are of particular interest: maximum throughput of access network, end-to-end round trip time (minimum, average, maximum), loss probability, jitter, and round trip time (RTT) between nomadic node 'mobila' and 'broker-gw' edge router. Since behind the edge router the traffic generated by end node traverses through fixed part of campus Internet, which gives quite low delays and error probability it is possible to characterize the particular access network using the latter parameter. In order to characterize GPRS access network we have used GGSN as an edge router. As we can see the differences between end-to-end RTT and RTT between terminal and edge router are relatively small. Note the fact that in the case of GPRS such approach gives us an estimation of joint access network and core network parameters. However, since the GPRS core network is built on the basis of high-speed wired technology [9] for which the values of delays and probability of loss are much smaller than in GPRS access network, we can just neglect it.

To obtain stable characteristics of each access network we had to measure it on a wide time scale. To get values for each column of the Table I we performed 120 minutes of testing.

Estimation of loss probability for transport layer of WLAN and GPRS is non-trivial task because both wireless access technologies use extensive error recovery algorithms. We also would like to note that low WLAN maximum throughput is explained by fact that we have been stressed to conduct tests under the realistic conditions and involved WLAN was in public use. Additionally, nomadic node was located in range of average signal volume.

TABLE I
PARAMETERS OF ACCESS NETWORKS

Parameters	Access		
	LAN	WLAN	GPRS
Max thrpt., Mbps	95.6	3.96	21.2E-3
Min RTT, ms	0.215	2.39	582.39
Avg. RTT, ms	0.43	6.76	705.97
Max RTT, ms	4.56	63.45	1174.29
Loss pr.	10E-7	0.0071	0.07
Jitter, ms	0.02	15	200
Min RTT, ms	0.265	1.17	573.36
Avg. RTT, ms	0.40	5.28	690.029
Max RTT, ms	0.76	65.38	1018.47

IV. MEASUREMENTS

A. Limited bulk transfer

The first type of traffic load for our testbed is limited bulk transfer. If one examine mp3 downloading service user's behavior, it can be seen as sudden transfers with relatively long periods of silence. To be fair in user behavior estimation we would also have to take into account those traffic produced by user while surfing mp3 repository (it could be HTTP or WAP based service). However, such traffic is out of scope of this paper. To evaluate the QoS for HTTP and WAP based traffic a different study has to be solicited.

In our testbed this type of traffic was generated by user of 'mobila' node by posing the requests on mp3 file downloads from 'mp3-source' machine located in the fixed part of Internet. To download files we used FTP over TCP service. The summary of statistics for fixed user is presented in Table II.

TABLE II
SUMMARY ON STATISTICS FOR BULK TRANSFER TESTS

Parameters	Access		
	LAN	WLAN	GPRS
Handshake, ms	41.18	16.35	854.63
Packet size, bytes	1191	1080	767
Packets/s.	274.75	88.41	5.78
Avg. thrpt., Mbps	2.619	0.764	0.036

For each case we derived three types of graphs: TCP Stevens' graph, RTT graph and throughput graph.

Unfortunately due to space limitation we cannot provide it in this paper. We, refer to extended version of this paper available in the Internet [10].

We observed that the perceived QoS of limited bulk transfer in wired LAN environment was excellent. Despite this it should be noted that during the test a (relatively) long collision occurred. However, it has not affected the perceived QoS significantly since most collisions in current 100Mbps Ethernet are almost not noticeable for users. Because of that, the perceived QoS obtained with LAN access can serve as reference one for wireless access technologies. We note that most RTT values are almost the same (equal to each other) and significantly lower than 0.05s.

We observed that the perceived QoS in WLAN environment was very good and comparable with LAN. The throughput is significantly less with WLAN since the both theoretical and actual bandwidth of 802.11b WLAN are considerably lower. Additionally, several collisions on WLAN have also occurred. However, it also has not affected the perceived QoS. Note that RTT values deviate from its mean noticeably more compared to LAN environment.

The perceived QoS in GPRS environment was unacceptably bad compared to both LAN and WLAN environment. It took approximately 820 seconds to download entire mp3 file of 3 MB size for fixed user and there was approximately 720 seconds long prefetching phase (actually, such long prefetching time is due to the software implementation, we believe it can be significantly reduced). The throughput was constant at approximately 4900 bytes per second. However, we observed that the goodput varied substantially.

Additionally, we noticed the other phenomena with GPRS access: the RTT values are spread uniformly over the constant interval that allows us to assume that they were mostly contributed by specific implementation of GPRS network.

B. Controlled transfer

As we mentioned before, most of Internet Live Radio stations are currently using TCP protocol instead of proposed UDP/RTP(RTSP) combination. From our point of view such interesting fact can be explained by those results obtained in paper [5] dedicated to VoIP performance testing throughout several years. Among the other conclusions drawn by the authors one should notice essential fact that the overall quality of fixed public Internet network is becoming significantly better every year. The lack of session control protocol for UDP is also one of the reasons behind TCP usage. Also, as we have emphasized above, TCP stream much easier than UDP one can traverse through firewall.

To obtain results we have chosen one of the stable Internet Live Radio. In our testbed we called that node 'mp3-live' (Fig. 1). Because of GPRS bandwidth limitations we have chosen 24 Kbits mp3 stream. We would like to emphasize that we have checked the stability of Internet Live Radio by sur-

TABLE III

SUMMARY ON STATISTICS FOR CONTROLLED TRANSFER TESTS

Parameters	Access		
	LAN	WLAN	GPRS
Handshake, ms	113.09	113.82	998.99
Packet size, bytes	601	614	706
Packets/s.	6.49	6.40	5.14
Avg. thrpt., Mbps	0.031	0.032	0.029

vival tests, i.e. how many hours the service will be provided without break. For all three available levels of quality, 24 Kbps, 56 Kbps and 128 Kbps, the chosen station gave very strong results - no break in service within 24 hours.

As we discussed above, prefetching phase allows receiver software to buffer some certain amount of streaming data to smooth end-to-end packets' delay variation (jitter). For controlled transfer tests we chosen application level buffer size of 240 kbits which is equal to 10 seconds of playback.

It is known that three-way handshake procedure performed by every TCP connection is considered as a drawback for real-time streaming applications [11]. We found that in average this procedure takes quite small time, not exceed one second even for GPRS and obviously can be neglected. We have also to point out that for certain packets the PUSH flag in TCP segment is set. This tries to assure timely delivery of TCP segment to application.

The summary of statistics for fixed user under controlled transfer is presented in Table III.

As one can expect, the perceived QoS in LAN environment was excellent. Prefetching phase took the least time and then the service has been operated stably. One can notice that there was prefetching phase and due to high bandwidth there was no re-prefetching during the whole time of connection. RTT has remarkable structure, the values oscillate. We consider that the reason behind is that the service is low bandwidth one, and therefore, at some time instants the server does not have information to transmit. The same type of periodicity in RTT values is found in WLAN environment for both fixed and walking user. We observed the same high quality perceived QoS in WLAN environment for both fixed and walking user.

V. RESULTS OVERVIEW

Based on our tests we found out that the overall performance of mp3-based entertainment applications in WLAN environment is roughly similar to that one obtained with LAN. In the case of GPRS the only limiting factor is the throughput of the wireless link. Moreover, at least in certain cases, there is no need for both specific QoS assurance methods and specific transmission facilities at the wireless interface of both access technologies.

From the user point of view WLAN performs as well

as LAN does for mp3 file downloading. The similar conclusion can be made for Internet Live Radio for any rate currently available in the Internet (starting from 24Kbps up to 128Kbps). Since in our testbed we have used an ordinary 'fixed' TCP implementation available with Mac OS X, Win2000 and Linux operating systems, which do not incorporate any wireless-specific features, it is not extremely crucial to use more complicated TCP versions at least for current testbed scenarios.

Running over GPRS the prefetching phase of mp3 file downloading was significantly longer compared to both WLAN and LAN. It is caused by current limitations of cellular access technologies based on TDMA. Those Internet Live Radio stations which use 24Kbps streaming service performs well over GPRS network. However, when Internet Live Radio's rate exceeds the current available throughput of the GPRS access network (3+1 time slots) the service is not stable and buffering always occurs.

Based on carried measurements we can state that both WLAN and GPRS are ready for mp3-based entertainment application like low-rate Internet Live Radio. Mp3 file downloading can also be supported for GPRS users, however, the bandwidth still remain the scarce resource at the 'last-mile'.

VI. CONCLUSION

Summarizing, we can conclude that it is already possible to implement on a commercial basis some low-bandwidth multimedia services like low-rate Internet Live Radio for nomadic users in current phase of 3G evolution towards All-IP network. Moreover, mp3-based entertainment multimedia services are also ready for wide implementation in hot-spot areas. However, the correct provisioning of these wireless access technologies is of paramount importance.

The only drawback of current evolution of 3G systems that restricts implementation of certain multimedia service available in Internet is in the *lack of bandwidth* on wireless interface. All other issues like the absence of specific wireless-oriented TCP implementations are not of vital importance. Therefore, we can expect that with deployment of WCDMA technology at the air interface at the final stage of 3G systems evolution will immediately open the possibilities for wide use of multimedia services currently available in the Internet.

REFERENCES

- [1] M. Droma, I. Ganchev, G. Morabito, R. Narcisi, N Passas, S. Paskalis, V. Friderikos, A. Jahan, E. Tsontsis, C. Bader, J. Rotrou, and H. Chaouchi, "Always Best Connected" enabled 4G wireless world," in *Proc. of IST Mobile and Wireless Communications Summit*, 2003.
- [2] F. Fitzek, A. Kopsel, M. Krishnam, and M. Reisslein, "Providing application-level QoS in 3G/4G wireless systems: A comprehensive framework based on multi-rate CDMA," *IEEE Wireless Communications*, vol. 9, no. 2, pp. 42-47, April 2002.
- [3] "National science foundation workshop on network research testbeds," Tech. Rep., NSF, 2002.
- [4] G. Leijonhufvud, "Multi-access networks and Always Best

- Connected (ABC)," in *Proc. of MMC workshop*, Berlin, Germany, November 2001.
- [5] I. Marsh and F. Li, "VoIP measurement infra-structure," in *Proc. of NTS16*, Espoo, Finland, August 2002.
- [6] A. Serhrouchni, A. Andaloussi, and A. Obaid, "A solution for improving TCP performance over wireless links," in *Proc. of Net-Con*, Paris, France, October 2002, pp. 457-468.
- [7] Etherial software, "Etherial software," available at: <http://www.ethereal.com/>.
- [8] NLANR, "Network performance and measurements tools," available at: <http://dast.nlanr.net/npmt/>.
- [9] Y. Koucheryavy, A. Krendzel, S. Lopatin, and J. Harju, "Performance estimation of UMTS PS CN domain elements," in *Proc. of MWCN*, Stockholm, Sweden, September 2002.
- [10] Y. Koucheryavy, D. Moltchanov, and J. Harju, "Performance evaluation of multimedia services in heterogenous wireless environment under different mobility patterns," available at: <http://www.cs.tut.fi/yk/pubs/report.html>, Institute of Communication Engineering, Tampere University of Technology, 2003.
- [11] J. Kurose and K. Ross, "Computer Networking: A top-down approach featuring the Internet," Addison Wesley, 2nd edition, 2003.