# Performance analysis of TCP traffic over UTRA-FDD channels

Eugenio Costamagna, Lorenzo Favalli, Pietro Savazzi and Francesco Tarantola
Dipartimento di Elettronica,
Università di Pavia, Pavia, Italy
e-mail {name.surname}@unipv.it

***Abstract*** **-- One of the targets of third generatio mobile radio systems and among these of the Universal Mobile Telecommunication System (UMTS) is the provision of Internet applications to mobile users. Extending IP and TCP control is severely affected by the channel conditions in the radio part and by the retransmission procedures used to improve reliability. In this paper we consider a radio access network based on the FDD scheme of the UMTS system as the final leg of an IP based packet network. To have the ability to make a complete simulation, the firs step has been the implementation of the radio transmission chain at the physical level. The second has been the extraction of proper channel models using Hidden Markov Models. These models have been determined both at the bit level and at the packet level and then used to evaluate the performance of the system for different channel and load conditions. Load has been provided by means of real traffic traces determined by means of measurements conducted on the campus network of the University of Pavia.**

***Keywords-UMTS, TCP, performance, WWW traffic.***

## I. Introduction

The 3GPP partnership project has developped an evolution path for the new UMTS system that through different releases leads to a full integration with the world of IP-based networks. These changes may well be tracked in the many documents provided by the 3GPP itself [1], the UMTS forum [2], and a large number of papers and books (see [3],…,[9]) for example. On its side, this approach makes a unique platform for the delivery of high spped data and multimedia services. The odds are related to the fact that Internet protocols have not been built to tackle the two main issues of wide area wireless networks: mobility and information errors due to adverse channel conditions. For what the radio link is concerned, a lot of research has been made in these years to determine how the performance of the TCP protocol, resposible for quality of service in the Internet, are affected by the higher latencies of a radio channel [6],[9].

The aim of the UMTS system is to extend the features of fixed telecommunication system to mobile users. As such, a framework to define the objective and subjective quality of service (QoS) must be clearly defined.

Generally speaking, the concept of QoS involves several quality measures including service accessibility, integrity, and other 'system level' figures. Given that a system is properly designed, what mostly affect the user's perception of QoS is related to the delay/error peformance of the system for the required throughput. The expectations from the user are different for different services, for which, the usual classification divides the services into four different classes with respect to their quality of service requirements: Conversational, Streaming, Interactive Data and Background.

The focus of the paper is the determination of some quality of service (QoS) parameters in a IP+wireless system. The wireless system is an implementation of the ETSI standard UTRA-FDD. The simulator is described in section 3: transmission is supposed to be carried out on a Dedicated channel (DTCH). The IP traffic has been determined by means of measurements on the University campus network as an aggregate and for different connections/protocols: as outlined in section 2 our attention has been focused on http and pop3 packets as these applications appear to be the more likely to attract users. Discussion of the results is performed in section 4.

## II. Sources characterization

The campus network at the University of Pavia comprises several fast Ethernet branches connected by switches to a single gateway that routes the outgoing traffic to the backbone national infrastructure. Our monitoring station running tcpdump was inserted just prior the gateway so that it could monitor all traffic over the network there is also a connection carrying H323 service between the campus in Pavia and a remote campus in Mantova: the two premises are about 100 Km apart and are connected via a 2Mb HDSL link. The analysis has been performed at different times of the day and the large amount of data has been filtered to extract data subsets relative to different parameters at the aggregate level. and to classify the data according to protocol and service, and to identify specific performance measures. Among all the possible traffic characteristics, our work has been focused on two parameters: packet size and packet interarrival time.

Since it is more likely that the main data applications will be web browsing and e-mail rather than the download of large files, we have concentrated our attention on the http and pop3 protocols that run on top of TCP. For what the http protocol is involved, distributions of the packet lenghts and interarrival times are given is given in Figure 1. while Figure 2. gives the same plots for the mail protocol. Note that pop3 packets mostly carry a whole message and they are contained in a single TCP packet.

## III. SIMULATIONS

For our purpose, a straightforward implementation of a simulator of a transport layer has been realized. No admission control is performed and consequently all the transmission parameters must be selected directly at the beginning of the simulation.. Referring to the UMTS protocol stack depicted in Figure 3. this is equivalent to say that we have a “full” though ideal implementation of the physical layer, a rough implementation of the MAC sublayer where transmission parameters are pre-set at the beginning of the simulation and a good RLC layer providing a subset of the alternatives given in the standard. To keep the resource assignement problem out of the simulator, we decided to use the structure of a dedicated transmission channel.

The delays introduced are then only due to retransmissions caused by radio channel errors, and are not connected to buffering at the packet scheduler.

The UMTS standard foresees three different possible ways to provide delivery of the user data:

- *transparent*, in which the RLC only provides segmentation of the data into properly sized PDUs without adding any control information;
- *unacknowledged*, where control information is added at the RLC level to provide concatenation between the different PDUs;
- *acknowledged*, following the usual ARQ mechanisms.

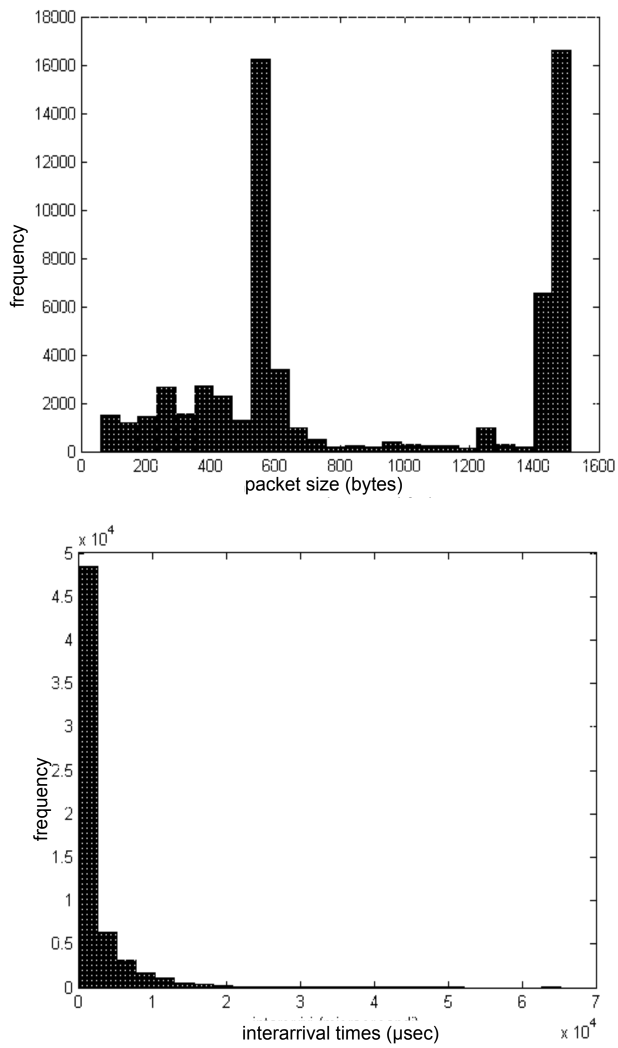


Figure 1. Measured histograms for the http protocol: packet sizes (upper) and interarrival times (below).

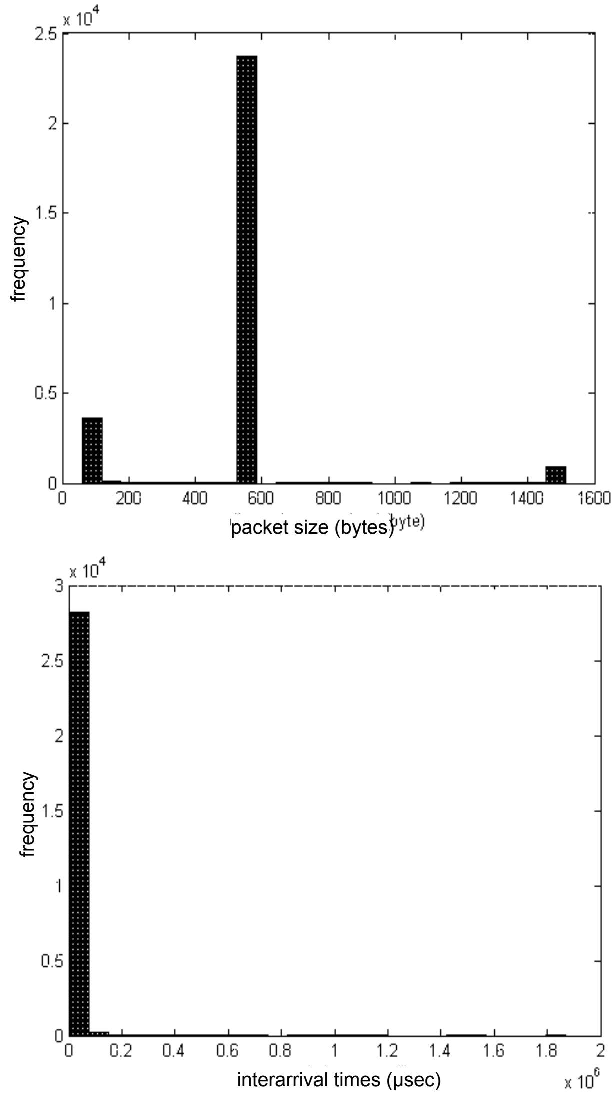


Figure 2. Measured histograms for the http protocol: packet sizes (upper) and interarrival times (below).

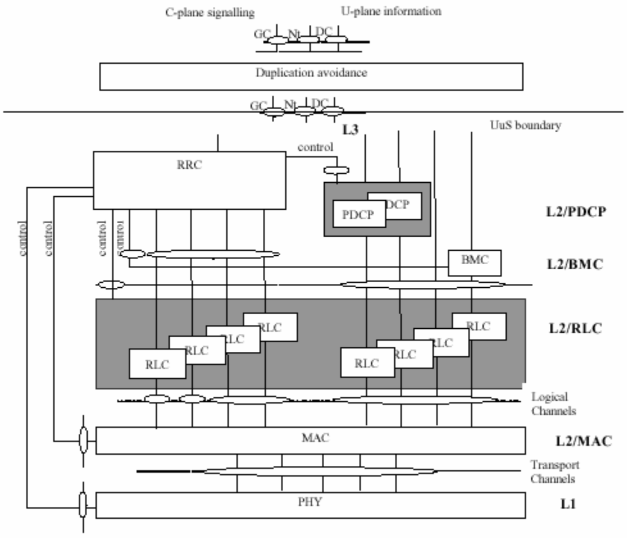


Figure 3. UTRAN protocol stack.

The structure of the PDU to be transmitted changes according to the type of channel to be used either shared or dedicated. The simulator implements and multiplexes the data structures of the DTCH/DCCH to build a DPDCH, which is then multiplexed with the DPCCH.

A rate 1/3 turbo coding (three decoding iterations) is used to protect the information which the flows along a 3GPP multipath channel where it also gets a dose of interference from other users as well as white noise. The interference from other cells is simply considered as an additional interference term. In the receiver we assume a three fingers rake receiver.

The structure of the simulator is reported in Figure 4. In accordance with the current implemetation of the UMTS-FDD standard, rate adaptation is mainly obtained by scaling the spreading factor assigned to each user connection so that a single code is used for each simulation

Three different connection speeds have been taken into account: 64, 144 and 384 kbps with fixed length PDUs of 40 bytes two of which are header bytes and the others represent the payload in agreement with the ETSI specifications when a MAC-SDU is to be sent via a dedicated channel with no multiplexing.

With these parameters we can draw TABLE I. the parameters there reported provide input to the segmentation and reassembly process of the packets to be transmitted and also to the modeling of the retransmission protocol since retransmission is performed at the radioblock level.

TABLE I. IMULATION PARAMETERS.

| Source Rate [Kbps] | PDUs per block | block size [Byte] | Transm. Time Interval | Spread Factor | Blocks in DTCH |
|---|---|---|---|---|---|
| 64 | 4 | 160 | 20 ms | 16 | 1 |
| 144 | 9 | 360 | 20 ms | 8 | 2 |
| 384 | 12 | 480 | 10 ms | 4 | 4 |

Simulation results allow the direct analysis of the raw bit and radioblock errors. These results can be easily post-processed to map them on whatever retransmission and apllication protocol desired.

The successive step has been to model the error sequences and the retransmission procedure by means of hidden Markov models (HMM). The inclusion of such models in an IP network simulator (NS 2) to simulate an end to end network is a "work in progress" issue, nonetheless as it is possible to include transition matrices of a markov model describing the error process, this step has been considered necessary to achieve good speed performance for the end to end simulations since simulation of $10^8$ bits (which only allow for an average number opackets at least two orders of magnitude lower) takes a whole night on a 2GHz, 512 MB RAM P4 computer. We must underline that no resource allocation protocol has been implemented at the UMTS interface yet, so that a single connection is considered and a transmission channel is always available when necessary.

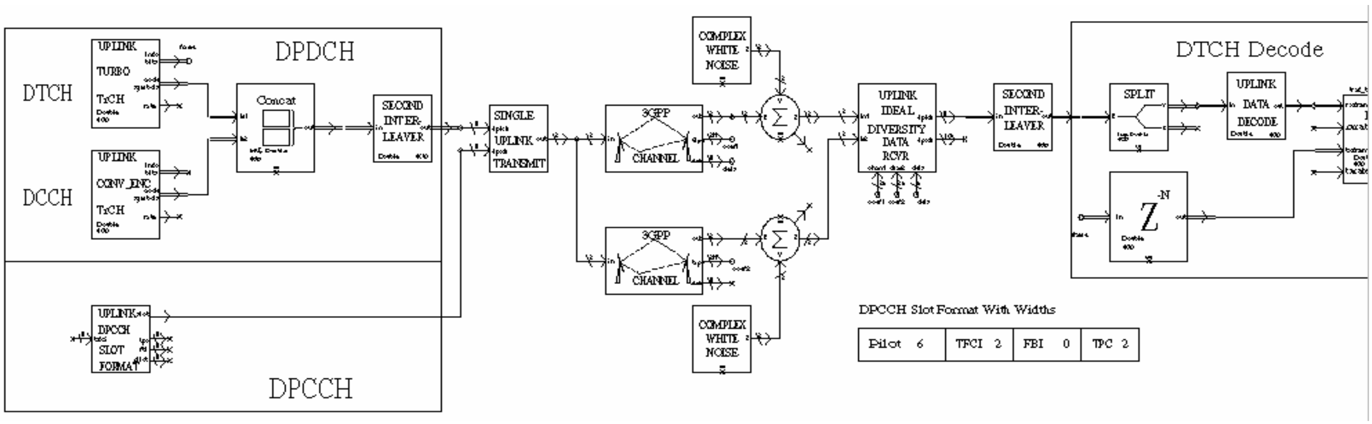


Figure 4. Simulation chain of the physical layer

## IV. RESULTS

The first set of results gives the error rates at the bit level. We may notice that the error rate becomes negligible at signal to noise ratios above 4 dBs thanks to the high redundancy intrinsic in the transmission chain as seen in TABLE II.

TABLE II. BIT ERROR RATE /BER) AT SELECTED SIGNALE TO NOISE RATIOS (SNR)

| SNR [DB] | 2 | 2.5 | 3 | 3.5 | 4 |
|---|---|---|---|---|---|
| BER | $2.99*10^{-4}$ | $6.393*10^{-5}$ | $3.44*10^{-5}$ | $1.174*10^{-6}$ | $6.523*10^{-7}$ |

The error rates at the radioblock and PDU level reported in Figure 5. and Figure 6. It can be noted that the difference at the PDU level is smaller that at the radioblock level. Furthermore it is possible to notice that there are two thresholds where the slope of the curves changes and only above 3.5 dBs the error rate differs for the different transmission speeds.

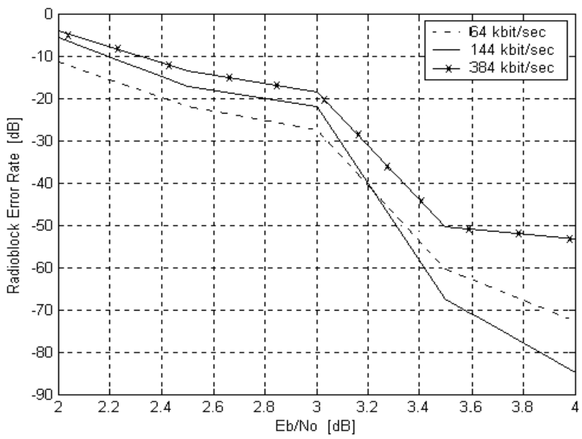


Figure 5. Fig. 4. Radioblock Error Rate

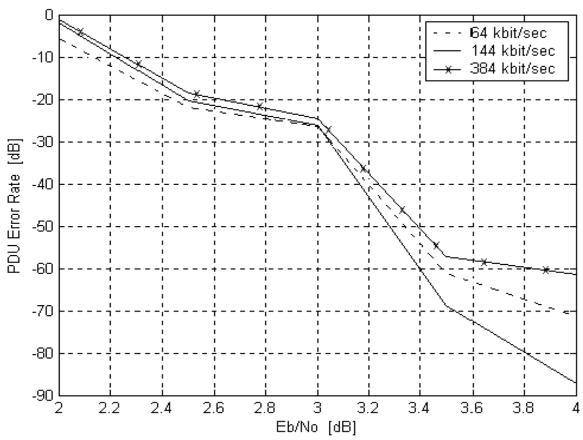


Figure 6. Fig. 5. PDU Error Rate.

The number of retransmissions is heavy only at very low signal to noise ratios as can be seen in Figure 7. and this is in agreement with the previous graphs.

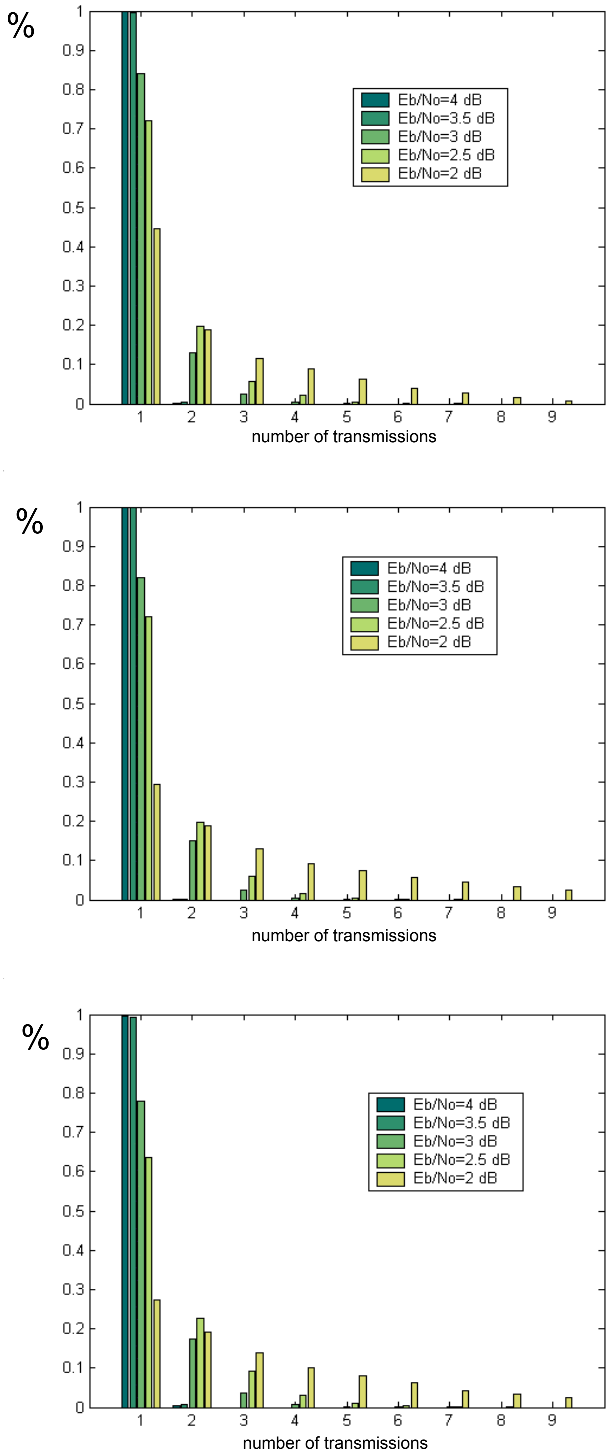


Figure 7. Radioblocks retransmissions for (top to bottoma) 64 kbps, 144 kbps and 384 kbps UMTS channels.

The retransmission process has been statistcally characterized and modelled using the well known Baum-Welch algorithm [10].

The first results in modelling the error process at the bit level using HMMs have been rather discouraging. As it can be seen in Figure 8. the cumulative distribution of the error process can be rather well described by the markov model, but the correlation between successive errors and the clustering function are rather poor. Figure 9. shows how the HMM model cannot provide enough correlation even using 8 states: above this number the complexity of the optimization algorithm tends to esplode and it takes more than two days to converge without clear improvements.

Same considerations can be derived from the observation of the clustering function in Figure 10. The K(r) function describes the variation coefficient of the current channel with respect to a Binary Symmetric Channel (BSC) which can be considered as a purely random process: the higher and steeper the curve, the more events happen in clusters. A flat line represents a renewal process.

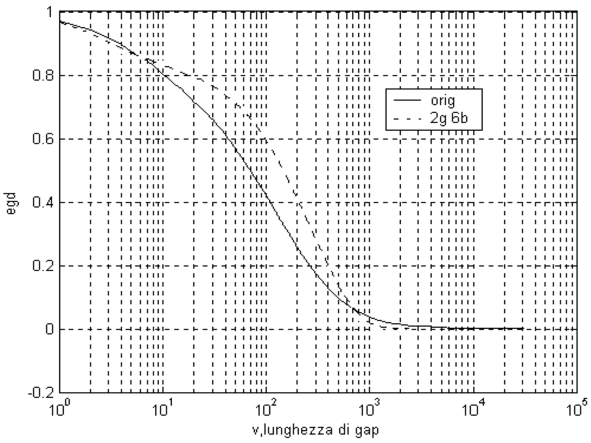


Figure 8. Error Gap distribution for the original error process and a model having “ ‘good’ states and 6 ‘bad’ states.

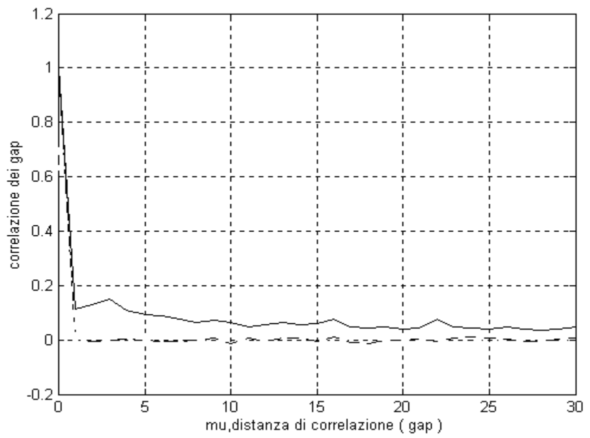


Figure 9. Correlation of original and modelled erorr sequences.

Some insight on this behavior can be derivid observing another curve that describes a “short term” feature of the error process. This function is the so called *P(m,n)* or the probabilità of m events in a window n and is plotted in Figure 11. It can be seen that there is a high probability of relatively short error bursts: this can be probably ascribed to the decoding algorithm.

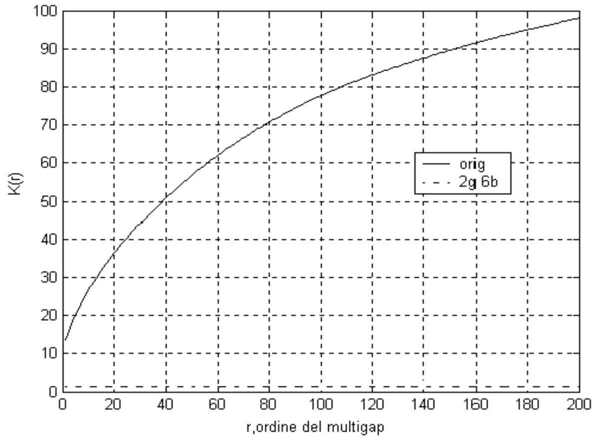


Figure 10. Clustring function of original and modelled erorr sequences.

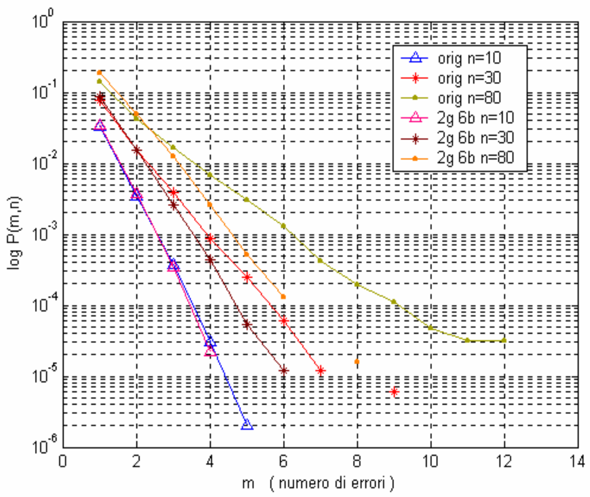


Figure 11. *P(m,n)* of original and modelled erorr sequences for different values of blocks and bursts.

The above hypothesis is confirmed if we move to consider the process of block and PDU errors. It has been found that the residual error process and the ARQ protocol may be characterized [11] using a simple two states markov model with one good state with low error probability and a bad state with higher error probability. Note that this does not hold for the error process at the bit level. Figure 12. and Figure 13. describe the correlation and clustering characteristics of the retransmission requests.

From Figure 14. to Figure 17. finally we give the performance of the two chosen protocols in terms of delay in the delivery of a packet and the total throughput. The throughput can be seen to soon flatten to a percentage close to 75% of the UMTS link capacity while the delay is more sensitive (as it should) to the number of retransmissions and the waiting time for the user becomes very large as the channel conditions fall below a threshold that may be set at a bit below 3 dBs. Above this value, in the simulation conditions described, the main bottleneck may be constituted by the ground segment of the network.

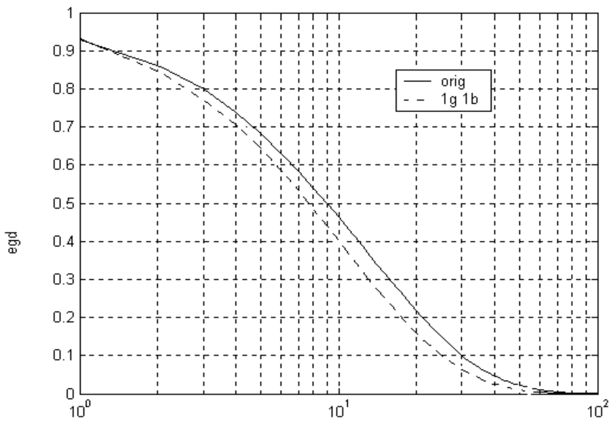


Figure 12. .Correlationn of the intervals between successive retransmissions.

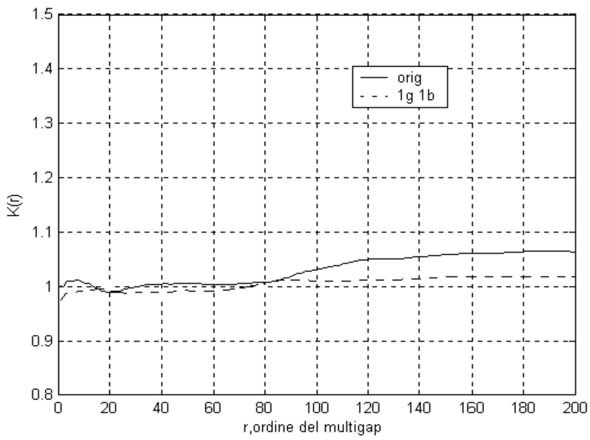


Figure 13. Long term distribution of error intervals

## V. CONCLUSIONS

We have presented a performance evaluation of Internet applications in a UMTS-FDD environment. Although more complex simulations including resource assigmnement should be required, it seems that the transmission chain is rather robust.

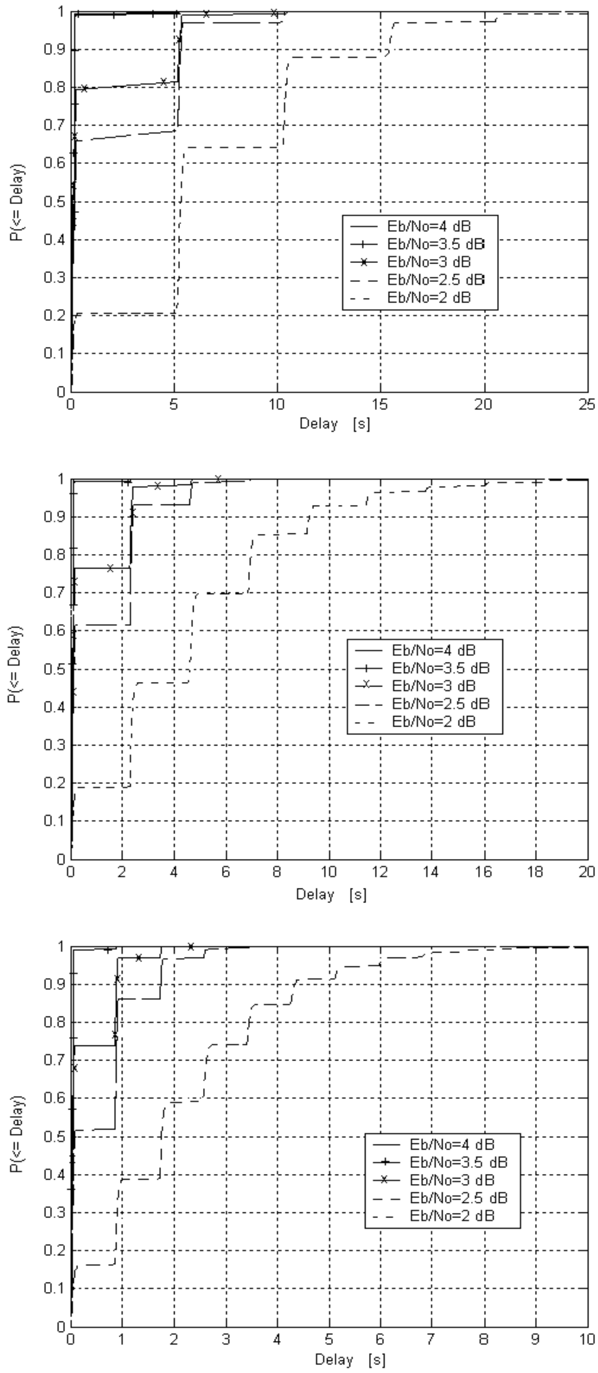


Figure 14. http packet delay with UMTS link at 64 kbps, 144 kbps and 384 kbps.

During our work, another one with similar goals have been published [9], but while the general conclusions appear to be rather similar, a direct comparison cannot be made since the simulation conditions start from opposite hypothesis.

Work is currently under way to complete the simulator to compare resource allocation algorithms. The major finding is the ability to model the retransmissions as a 2 states HMM. This allowes its inclusion in a end-to-end simulator and will allow in the near future to introduce it in a 'Markov modulated' process where 'macrostates' provide the necessary dynamicity in channel conditions tha fluctuates depending on location and users behavior. This work is currently being performed and is being extended to the inclusion of real time services handled via IP protocols such as transmission of H.263 and H.264 sequences.

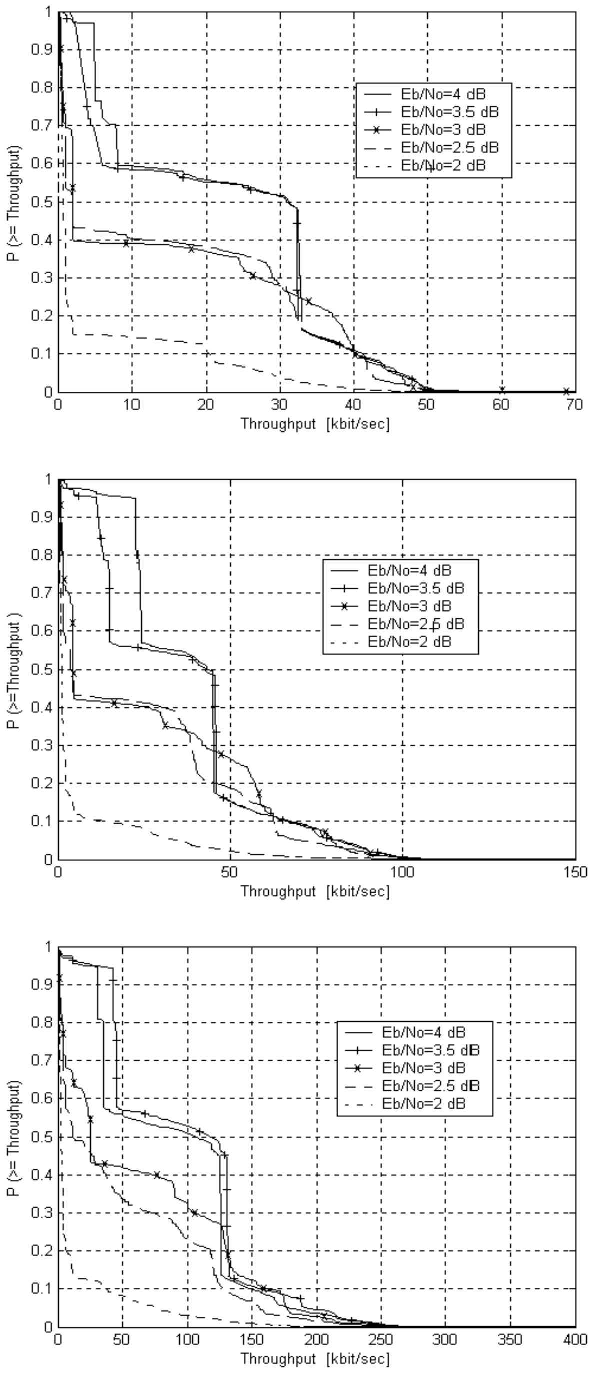


Figure 15. Throughput of IP and UMTS link at 64 kbps, 144 kbps and 384 kbps for the http protocol.

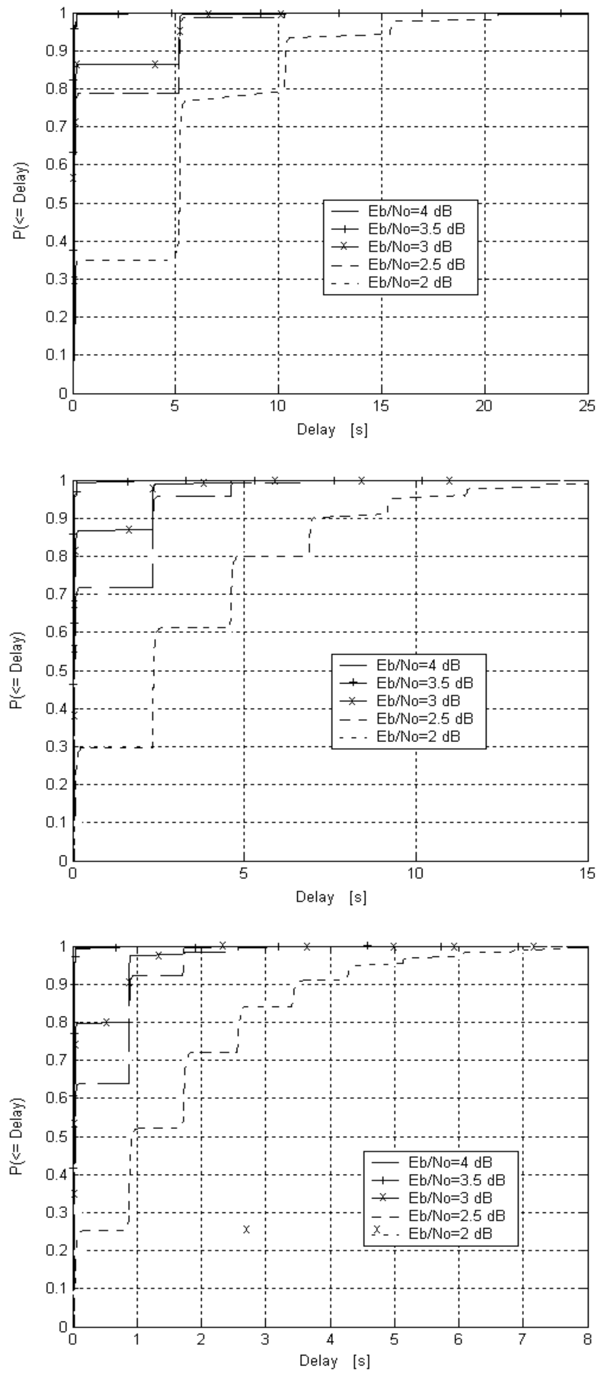


Figure 16. Pop3 packet delay with UMTS link at 64 kbps, 144 kbps and 384 kbps.

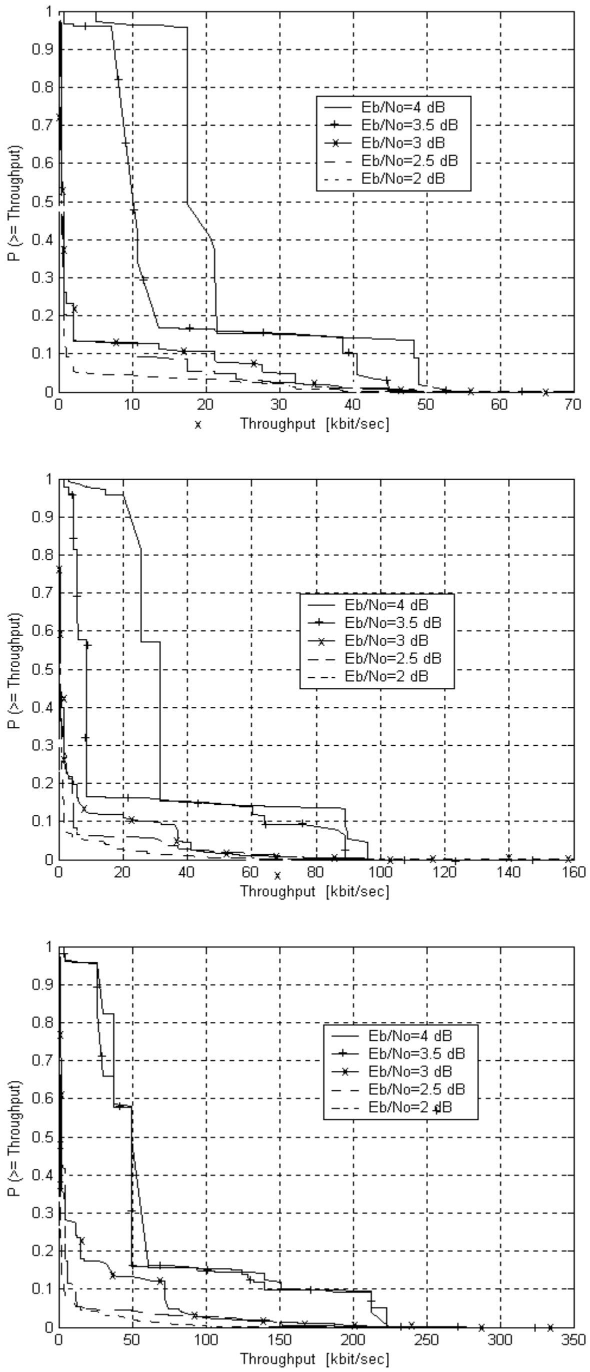


Figure 17. Throughput of IP and UMTS link at 64 kbps, 144 kbps and 384 kbps for the pop3 protocol.

## REFERENCES


[1] http://www.3GPP.org

[2] http://www.umts-forum.org

[3] B. Walke, *Mobile Radio Networks*, Second Edition, J. Wiley & Sons Ltd., Chichester, GB.

[4] H. Holma, A. Toskala, WCDMA for UMTS: radio access for third generation mobile communicatios, J. Wiley and Sons Ltd. Chichester, GB.

[5] Y.-B. Lin, A.-C- Pang, Y.-R. Haung, I. Chlamtac, "An all-IP approach for UMTS third generation mobile networks," IEEE Network, Sept./Oct. 2002, pp. 8-19.

[6] A. Chockalingam, M. Zorzi, V. Tralli, " Wireless TCP performance with Link Layer FEC/ARQ," IEEE ICC 2001,

[7] A.-F. Canton, T. Chahed, " End-to-end reliability in UMTS: TCP over ARQ," IEEE Globecom 2001, San Antonio, TX, USA, 25-29 Nov, 2001, vol 6 pp. 3473-3477.

[8] Fig. 11. Throughput of IP and UMTS link at 64 kbps, 144 kbps and 384 kbps for the pop3 protocol.

[9] S. Heier, D. Heindrichs, A. Kemper, "Performance evaluation of Internet applications over the UMTS radio interface," IEEE 55th Vehicular Technology Conference, 2002. VTC Spring 2002., Volume 4 , 2002, pp. 1834-1838.

[10] Rabiner L.R., "A tutorial on hidden Markov models and selected applications in speech recognition", Proceedings of the IEEE, Volume: 77 Issue: 2, Feb 1989, Page(s): 257 –286

[11] L. N. Kanal, A. R. K. Sastry, Models for channel with memory and their applications to error control, Proceeding of the IEEE, vol. 66, no. 7, giugno 1978.