QoS Parameters Mapping for the E-learning Traffic Mix in LTE Networks

V.H. Muntean*, M. Otesteanu* and G.-M. Muntean**
* "Politehnica" University Timisoara/Faculty of Electronics and Telecommunications, Timisoara, Romania
** Dublin City University/Department of Wireless Communications, Dublin, Ireland
*{horia.muntean, marius.otesteanu}@etc.upt.ro,**munteang@eeng.dcu.ie

Abstract—**Next Generation Networks (NGN), like Worldwide Interoperability for Microwave Access (WiMAX) and Long Term Evolution (LTE) are expected to become the "anywhere and anytime" access networks to high speed wireless communications. This paper proposes a method of mapping the LTE QoS parameters in order to improve the quality of experience of the end-user when an e-learning application that uses Dynamic Quality Oriented Adaptive Scheme (DQOAS) and generates a traffic mix, is deployed over a Long Term Evolution network. Simulation results show that the proposed mapping method offers improved results compared to the normal mapping scheme used in LTE. The best results are obtained in combination with the proportional fair scheduling algorithm, while round robin and maximum throughput schedulers' satisfaction rate does not go over 50% in our simulation setup.**

Index Terms — **content adaptation, QoS, E-learning, LTE, scheduling algorithms, wireless networks**

I. Introduction

During the last years, research efforts in the e-learning field were concentrated on developing of new delivery algorithms able to improve the quality of the learning process. This is a pressing problem, considering that an increasing number of users are accessing learning content via wireless networks and using mobile devices. E-learning process involves using multiple applications – web browsing, video and audio streaming, rich voice and ftp background traffic – that are generating a traffic mix. Managing the traffic mix flows is a difficult problem, especially when they are delivered over wireless networks, because wireless technologies are offering limited radio resources and the services in such networks are highly susceptible of being affected by environmental factors, traffic load, number of clients and their mobility pattern. The authors have proposed in [1] an adaptive multimedia delivery algorithm – DQOAS – designed to improve the end users' quality of experience (QoE) during the learning process, when this process is taking place in an IEEE 802.11 wireless LAN environment. The results are showing that a dynamic adaptation policy based on user preferences and network conditions is improving significantly the end-user perceived quality. Also, the total number of simultaneous served users is increased, as well as the link utilization. Taking into account the good results of DQOAS and the user-oriented approach, it is of further interest to analyze the behavior of this algorithm over other wireless networks, like NGNs. In this paper, LTE is the chosen technology as the next wide coverage wireless network.

This work was partially supported by the strategic grant POSDRU 2009 project ID 50783 of the Ministry of Labour, Family and Social Protection, Romania, co-financed by the European Social Fund – Investing in People and by Enterprise Ireland, Proof of Concept and Science Foundation Ireland Research Frontiers programs.

LTE is an all-IP network standardised by 3rd Generation Partnership Project (3GPP) in Release 8 which uses new multiple access schemes on the air interface. Orthogonal Frequency Division Multiple Access (OFDMA) is used in the downlink and Single Carrier Frequency Division Multiple Access (SC-FDMA) is used in the uplink to fulfil all the ambitious requirements for data rate, spectrum efficiency, latency, and capacity. Another important technique used is Multiple-Input-Multiple-Output (MIMO) that involves using multiple transmitters and receivers to achieve higher bit rates and improved coverage [2].

The paper aims to offer a mapping alternative for LTE QoS parameters in case of an e-learning traffic mix, in order to obtain an optimal end-user QoE when DQOAS algorithm is used. The proposed solution uses the same QoS class for two different traffic flows improving the overall quality of experience for the end-user. It also enables DQOAS to update the quality levels (increase/decrease) of the selected multimedia stream based on user preferences, on instantaneous channel conditions and on the resource allocation scheme, with a minimum impact on the second service (web browsing traffic in our case).

The paper is structured as follows. Section II presents the studies done by researchers regarding LTE QoS parameters and scheduling in case of traffic mix connections for both Downlink and Uplink, while Section III describes DQOAS algorithm and the QoS concepts and architecture as defined in LTE technology. Section IV presents the proposed mapping scheme followed by test results in Section V. Conclusions are offered in Section VI.

II. Scheduling Algorithms for LTE Uplink and Downlink

As Long Term Evolution technology evolves and important operators in telecom world announced their interest for LTE, researchers are starting to develop algorithms capable of improving the network delivery. Their work concerns both the uplink and downlink, considering multiple solutions for implementing scheduling algorithms in different traffic conditions, considering multiclass flows. The scheduling methods are

978-1-4244-7433-2/10/$26.00 ©2010 IEEE

looking for improving the system capacity in terms of number of QoS flows that can be supported and also for reducing the resource utilization. Reference [3] divides the work done in this area into two categories, based on the type of traffic the scheduler was designed for: scheduling for elastic (non-real-time flows) [4] and scheduling for real-time flows [5]. LTE schedulers can also be classified based on their awareness parameter(s) into channel-aware schedulers [6], queue-aware schedulers and queue- and channel-aware schedulers [3].

Regarding the Uplink (UL) schedulers, a lot of work has been done. A performance comparison on control-less scheduling policies for Voice over IP (VoIP) in LTE UL was conducted in [7] and it was proven that semi-persistent scheduling obtains better performances that group scheduling when no group interactions occurs for group scheduling. In [8], the authors suggested an opportunistic scheduling algorithm based on the gradient algorithm called Heuristic Localized Gradient Algorithm (HLGA) that allocates resource blocks to users while maintaining the allocation constraint and considers retransmissions requests. Channel-aware scheduling algorithms for SC-FDMA are proposed in [6] in local and wide area scenarios. The first two, First Maximum Expansion (FMA) and Recursive Maximum Expansion (RME), represent simple solutions for localized allocation of the resource blocks, whereas the third algorithm, Minimum Area-Difference to the Envelope (MADE), is more complex but performs closer to the optimal combinatorial solution.

The QoS aspects of the LTE OFDMA Downlink (DL) are influenced by a large number of factors – channel conditions, resource allocation policies, available resources, delay sensitive/insensitive traffic, etc – and therefore new means were needed to enhance QoS beyond what the default IP service provided. This problem is addressed in [3], where a new scheduler for LTE downlink is proposed. The performance of this scheduler is analyzed using multiclass traffic and the results are indicating that a channel- and queue-aware scheduler is a good choice for LTE DL. The work in [9] showed that strict prioritization for session initiation protocol (SIP) packets over other packets – voice and data – can lead to better overall performances. References [10] and [11] are analyzing the packet scheduling of mixed traffic in LTE DL. The results in both are showing that it is necessary to perform service differentiation and prioritization of delay-critical traffic as VoIP traffic, especially when in combination with delay-insensitive traffic like web surfing or TCP download.

III. LTE QoS Concept and DQOAS Adaptive Algorithm Description

A. LTE QoS Concept and Architecture Aspects

LTE technology evolved from UMTS/HSDPA cellular technology to meet current used demands of high data rates and increased mobility. The LTE radio access is based on OFDM technique and supports different carrier frequency bandwidths (1.4-20 MHz) in both frequency-division duplex (FDD) and time-division duplex (TDD) modes [12]. The use of SC-FDMA in the uplink reduces Peak-to-Average Power Ratio compared to OFDMA, increasing the battery life and the usage time on the User Equipments (UEs). In DL peak data rates go from 100 Mbps to 326.4 Mbps, depending on the modulation type and antenna configuration used. LTE aims at providing IP backbone services, flexible spectrum, lower power consumption and simple network architecture with open interfaces [2].

In LTE, all network services available for users, are considered end-to-end, or from a Terminal Equipment (TE) to another TE. The provided services can be classified according to their QoS level, but finally is the user who can decide if the provided QoS for a certain service is satisfactory or not. Some of the most important general requirements for QoS attributes are stating that they must have an unambiguous meaning and the mapping should provide different levels of QoS by using UMTS specific control mechanisms. The technical specifications for QoS attributes should meet a number of criteria, of which the most important are presented as follows: UMTS QoS mechanisms shall provide a mapping between application requirements and UMTS services, shall be able to interwork efficiently with existing QoS schemes, shall support efficient resource utilization, shall support asymmetric bearers and shall provide control on a peer to peer basis between UE and 3G gateway node [13].

The 3GPP QoS concept is based on traffic differentiation and prioritization of data flows, using network-initiated bearers in conjunction with simple QoS profiles based on QoS Class Identifiers (QCIs). In order to obtain a desired network QoS, a Bearer Service (BS) with defined characteristics and functionality has to be set between the two network elements involved in the data exchange). The BS includes aspects like control signaling, user plane transport and QoS management functionality in order to be able to provide the desired QoS [13]. As shown in the BS layered architecture depicted in Figure 1, the BS on a specific layer is offering its services to the bearer on the next level, using the services provided by the layer below.

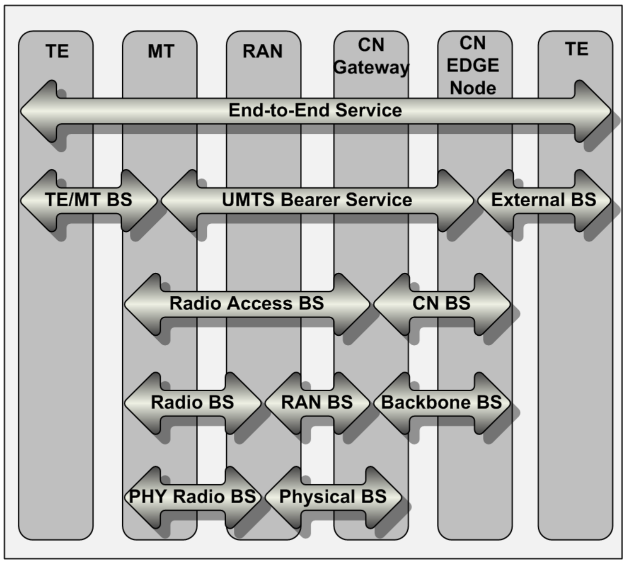

Figure 1 – UMTS QoS Architecture [13]

When two UEs are involved in a data communication, the data flow between them has to pass across different bearer services of the LTE network. The bearer service that provides in fact the QoS services offered by the operator is the UMTS BS. This bearer service is composed of two parts, the Radio Access BS, which provides confidential transport of user data and signaling, and the Core Network (CN) BS, which has the role to control and utilize the backbone network in order to provide the desired UMTS bearer service. The Radio BS is responsible with the aspects of the radio interface transport and handles the part of the user flow that belongs to one subflow. Radio Access Network (RAN) bearer service takes part in the transport between RAN and CN, together with the Physical BS. Radio bearer service and RAN bearer service are composing the Radio Access bearer.

Unlike the complex QoS mechanisms defined in fixed networks, cellular networks use simple and robust mechanisms, able to offer a good QoS resolution, considering the fact that the air interface has different error characteristics. In LTE, the concept of traffic class was implemented. A traffic class or a QoS class is defined considering the restrictions and the limitations of the radio interface. Based on the traffic sensitivity to packet delay, there are four classes defined as follows: conversational, streaming, interactive and background class. Conversational class is meant for traffic that has a high sensitivity to delay (e.g. VoIP), while background class deals with traffic that has a low sensitivity to delay (e.g. background download of files). As stated in [13], there is no strict one-to-one mapping between classes of services and the traffic classes defined above. For example, if a service is interactive by nature or if the user has strict requirements about delay, then that service can use the conversational traffic class for obtaining the desired QoS.

B. DQOAS Algorithm for E-learning Content Delivery

DQOAS algorithm was designed to perform an optimal dynamic multimedia content adaptation in wireless LAN environments, based on the end-users' profile and preferences and on the network conditions, in order to obtain high QoE during the learning process and to increase the total number of simultaneous connected users.

DQOAS extends the QOAS [14] algorithm by adding user QoE expectation levels as parameters in the adaptation process. Like this, multimedia content will be delivered to users taking into account not only the network conditions but also their quality expectations. Figure 2 presents the block design of DQOAS algorithm. On the client side, the Feedback module monitors the network conditions, registers the delivery-interest parameters (loss rate, delay and jitter) and sends short reports to the N-level Builder, on the server side. Another input for the N-level Builder is the Rules&Param list, which contains user-related information (expected QoE level) and session-specific parameters.

Dynamic level building and adaptation is done on the server side in three steps. First step, Initial level building, is needed whenever a new user requests a multimedia stream. Because this user does not have a QoE expectation level, the levels will be set statically by the QOAS algorithm. As soon as the QoE level is estimated for this new user, DQOAS algorithm will dynamically build a new set of levels, based on the minimum quality accepted level. In the second step, dynamic update of the quality levels is performed. The procedure is triggered by a significant change in network conditions, users attaching/detaching to the network or new minimum quality levels for certain users. The minimum quality level represents the minimum video bitrate accepted by the user. The N-level Builder creates a number of M levels for a user, the lowest level being the minimum accepted quality. The maximum level can be as high as the original video bitrate, if network conditions are favorable. In the last step, the dynamic delivery takes place, by tuning the QOAS module on the new adaptation levels built for a user. This is a continuous process, as rate control adaptation is performed every time a media-rich fragment

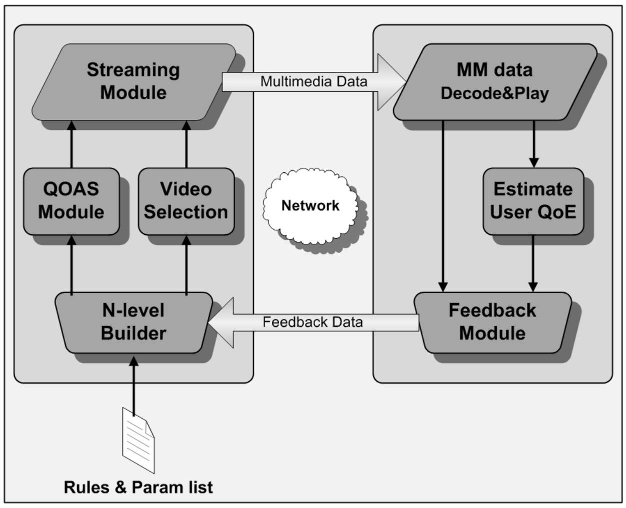

Figure 2 – DQOAS Adaptation Algorithm

needs to adapt the bitrate in order to match the network conditions and user requirements. As the e-learning process itself is highly dynamic, DQOAS algorithm is designed to keep up with the continuously changing conditions and parameters involved, improving end-user quality of experience by adjusting dynamically its adaptation policy [1].

IV. MAPPING OF QOS PARAMETERS

E-learning applications are usually generating more than one traffic type, so the data to be sent over the network to a user can be seen as a traffic mix (e.g. video streaming and web browsing traffic). Based on this fact and on the service classification in LTE by the SPI (Service Priority Information) field, there will be at least two queues corresponding to two different QoS classes, deserving the application. Considering service prioritization, the objects belonging to different queues have different probabilities of being scheduled, depending on the chosen scheduler model. The most common options for a scheduler are:

- Round Robin scheduler (RR), where all users get equal time shares tor transmission, in an order based on their last scheduled time: $M_n = t_n$

- Proportional Fair scheduler (PF), where both the instantaneous channel conditions and the users' past average throughput are considered; it offers the same average throughput for each user: $M_n = {}^{d_n}/_{r_n}$, where d_n is the instantaneous supported throughput and r_n is the past average throughput

- Maximum Throughput scheduler (MT), based only on the instantaneous channel conditions, giving an advantage to users that have the best channel conditions at the given time: $M_n = d_n$

Considering an application generating two services classified in different QoS queues and a RR scheduler, we can write the following equation, describing the i-th user satisfaction condition, according to [11]:

$$\frac{\left(f_1 + \frac{\alpha}{\rho} f_2\right) \cdot T \cdot \left[\frac{N}{n}\right] \cdot \Delta}{(T + d^{max}) \cdot \beta} \leq \frac{1}{1 - \varepsilon}, \quad (1)$$

where f_1 and f_2 represent the average packet transmission ratio, ρ is the priority of the first service over the second, T is the time interval in which the transmission takes place, N represents the maximum cell load that satisfies the quality criteria for user i, n denotes the number of scheduled users at every Transmission Time Interval (TTI), Δ is TTI length, d^{max} is the maximum scheduling delay and ε is the maximum ratio of delayed and loss packets with which the service quality perceived by the user is still satisfactory. If s_1 and s_2 are the average packet sizes of the two services, and s_i^{max} is the average amount of data that can be transmitted to user i in a single transport block, then $\alpha = s_2/s_1$ and $\beta = s_i^{max}/s_i$.

In our case, the maximum ratio of delayed and lost packets, ε, is different from one user to another, being a user-dependent parameter, not a service dependent parameter like it is in case of VoIP. Based on DQOAS description, every user has a minimum accepted quality level for the incoming video stream, representing the dynamically encoded bitrate of the stream, M_i. As this is the minimum accepted level that DQOAS can send for user i, ε as defined above has no significance because any lost or delayed packets will decrease the quality below M_i. To overcome this problem, two solutions are possible. First solution is to add a guard to the minimum expected level, equal with the maximum agreed ratio of delayed and loss packets, ε: $M_i^{new} = M_i + \varepsilon$. Second solution implies conditioning $\varepsilon = 0$ only when DQOAS module is tuned on M_i, because if the quality level is higher, then decreasing this level with the maximum ratio of delayed and loss packets, keeps the video quality above the user satisfaction limit. The first proposed solution is easier to compute, because the changes are done inside the DQOAS module, based on the current network conditions and resource allocation scheme used.

Prioritizing the services types has very good results when the traffic sources are independent. A flow with a higher priority will have a significant capacity gain with the cost of a small capacity loss of the second service. But when the traffic source is the same for the different flows, the user might achieve a higher per-application QoE if both flows have the same QoS class (ρ=1). Considering that f_1 and f_2 represent the average packet transmission ratio of two flows generated by the same application, the satisfaction equation for user i reads:

$$\frac{(f_1 + \alpha f_2) \cdot T \cdot \left[\frac{N}{n}\right] \cdot \Delta}{(T + d^{max}) \cdot \beta} \leq \frac{1}{1 - \varepsilon} \qquad (M_i^{new} = M_i(1 + \varepsilon))$$

In the case considered here, an e-learning application that generates video streaming and web-browsing traffic is used. Following the assumptions, the second traffic type (with a lower priority) will have the same QoS class as video streaming traffic – streaming class, as presented in Figure 3.

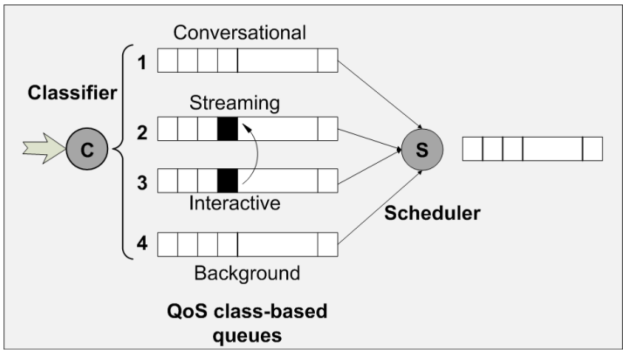

Figure 3 – Changing QoS Class for low priority E-learning traffic flow

The advantage of being in the same QoS class is that the queue specific sorting algorithm will consider both flows with the same priority (the users' priority in the queue). This way, the packets coming from the same application, even on different bearers, will have the same queuing delay, improving the QoE of the application as a whole. In these conditions, DQOAS can update the quality levels for the multimedia stream based on user preferences, on instantaneous channel conditions and on the resource allocation scheme, with a minimum impact on the second service (web browsing traffic in our case).

V. TEST RESULTS

The proposed solution was tested using the LTE System Level Simulator [15], capable of simulating LTE SISO (Single Input Single Output) and MIMO networks using TxD (Transmission Diversity) or OLSM (Open Loop Spatial Multiplexing) transmit modes.

TABLE I.
PARAMETERS USED FOR RUNNING SIMULATION SCENARIOS

Parameter	*Value*
Frequency	2.0 GHz
Bandwidth	5 MHz
Thermal noise density	-174 dBm/Hz
Receiver noise figure	9 dB
nTX x nRX	2 x 2
TTI length	1e-3 s
Simulation length	1000 TTIs
Subcarrier averaging algorithm	EESM
UE speed	5 Km/h

PHY layer model is based on the post-equalization SINR, offering pre-calculated fading parameters and so reducing computational complexity at run-time. In the conducted tests three schedulers were used (round robin, proportional fair and maximum throughput), with two parallel streams for every UE. A number of 7 eNodeBs with 10 UEs attached on each represents the LTE network used for testing. Table 1 presents the parameters used.

Figure 4 presents the LTE network used for simulation, highlighting the position of UE 4, attached to eNodeB 1. The two streams per user are considered to have the same priority, with the same weight in the schedulers' queue. The throughput of the two streams for UE 4 is presented in figures 5, 6 and 7 when proportional fair, round robin and maximum throughput schedulers are used. For TCP download, a user is considered satisfied if the experienced throughput is at least 300kbps [11], and for the video streaming the user i is satisfied if the stream quality level is kept above M_i^{new}. For user 4, it is observed that only the proportional fair scheduler is providing enough radio resources to be fully satisfied. If round robin scheduler is used, the user is at the satisfaction limit (throughput ~ 0.2 Mbps for both streams). If maximum throughput scheduler is used, user 4 is not able to initiate an e-learning session because of the poor quality conditions due to its relative position to eNodeB 1. Considering the other users, their behavior is presented in Table 2.

TABLE II.
USER SATISFACTION UNDER DIFFERENT SCHEDULING ALGORITHMS

Scheduler used	*Satisfied users*	*Unsatisfied users*	*Maximum Throughput*
Proportional Fair	60%	40%	1.5 Mbps
Round Robin	50% (20% of them are at the limit)	50%	1.8 Mbps
Maximum Throughput	30%	70%	5.5 Mbps

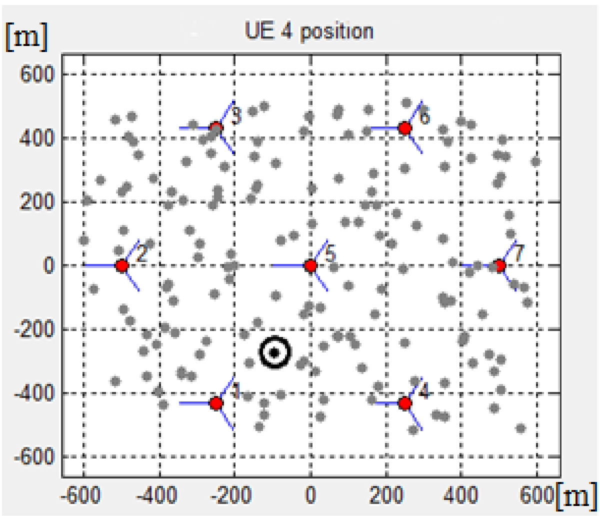

Figure 4 - LTE network map; 7 eNBs with 10 UEs attached to each

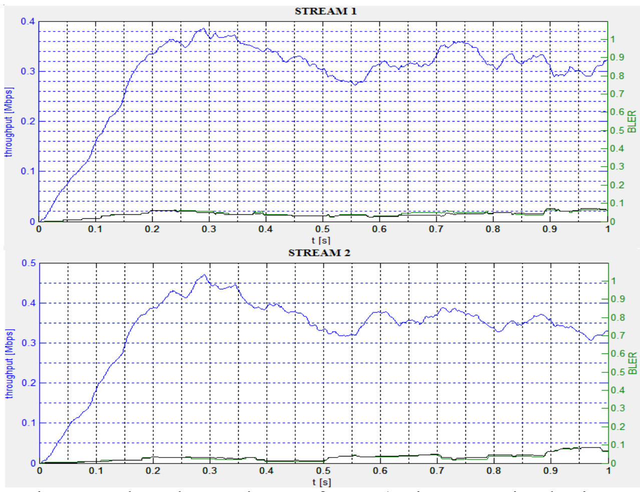

Figure 5 - Throughput and BLER for UE 4 using Proportional Fair scheduler

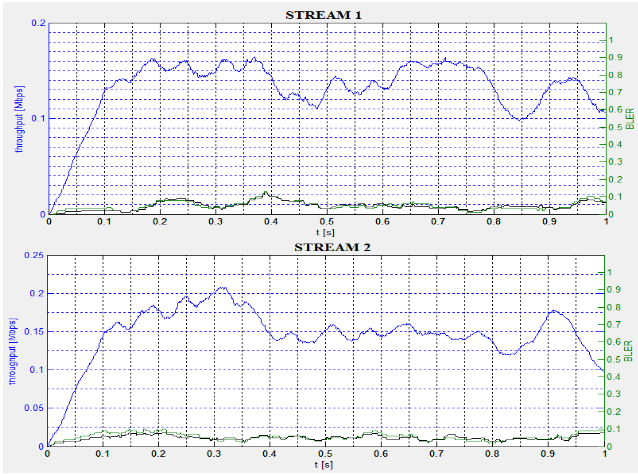

Figure 6 - Throughput and BLER for UE 4 using Round Robin scheduler

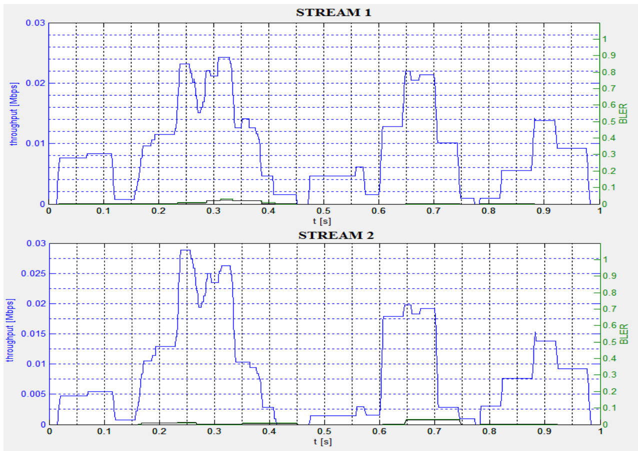

Figure 7 - Throughput and BLER using Maximum Throughput scheduler

It is noticed that in case of a maximum throughput scheduler, the number of users that can be served using the proposed algorithm decreases drastically. Still, the three satisfied users are experiencing very high data rates in comparison with the maximum data rates obtained when PF or RR schedulers were used.

VI. Conclusions and Further Work

This paper proposes a new method of mapping the QoS parameters in order to improve the quality of experience of the end-user when an e-learning application generating a traffic mix is deployed over a Long Term Evolution network. Two different streams coming from the same application were considered for each user in the network, while some scheduling algorithms were tested. The results show that proportional fair algorithm in combination with the proposed mapping scheme offers improved performances compared to the normal mapping scheme used in LTE. If maximum throughput algorithm is used, only 30% of the users are satisfied with the offered quality, but these users are experiencing a very high throughput, as it was expected.

Further work implies extensive testing using different propagation models available with the simulator and also extending the mapping scheme in order to increase the total number of satisfied users. This can be achieved by using different weights in the same queue based on the traffic type and by utilizing other schedulers developed by researchers in the field.

References

[1] V. H. Muntean, G.-M. Muntean, “A Novel Adaptive Multimedia Delivery Algorithm for Increasing User Quality of Experience during Wireless and Mobile Learning” in *IEEE International Symposium on Broadband Multimedia Systems and Broadcasting*, Bilbao, Spain, May 2009.

[2] B. Furht and S. Ahson, Long Term Evolution – 3GPP LTE Radio and Cellular Technology. CRC Press, 2009.

[3] B. Sadiq, R. Madan and A. Sampath, “Downlink Scheduling for Multiclass Traffic in LTE” in *EURASIP Journal on Wireless Communications and Networking*, vol. 2009, Article ID 510617, 18 pages, July 2009.

[4] K. Seong, M. Mohseni and J. M. Cioffi, “Optimal resource allocation for OFDMA downlink systems” in *Proceedings of the IEEE International Symposium on Information Theory (ISIT ’06)*, Seattle, USA, July 2006.

[5] N. Chen and S. Jordan, “Downlink scheduling with probabilistic guarantees on short-term average throughputs” in *Proceedings of the IEEE Wireless Communications and Networking Conference (WCNC ’08)*, Las Vegas, USA, April 2008.

[6] L. A. de Temino, G. Berardinelli et co., “Channel-Aware Scheduling algorithms for SC-FDMA in LTE Uplink” in *IEEE 19th International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC 2008)*, Cannes, France, September 2008.

[7] H. Wang and D. Jiang, “Performance Comparison of Control-less Scheduling Policies for VoIP in LTE UL” in *IEEE Wireless Communications and Networking Conference (WCNC 2008)*, Las Vegas, USA, April 2008.

[8] M. Al-Rawi, R. Jantti, J. Torsner and M. Sagfors, “On the Performance of Heuristic Opportunistic Scheduling in the Uplink of 3G LTE Networks” in *IEEE 19th International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC 2008)*, Cannes, France, September 2008.

[9] M. Wernersson, S. Wanstedt and P. Synnergren, “Effects of QoS scheduling strategies on performance of mixed services over LTE” in *Proceedings of the 18th IEEE International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC ’07)*, Athens, Greece, September 2007.

[10] J. Puttonen, N. Kolehmainen et co., “Mixed Traffic Packet Scheduling in UTRAN Long Term Evolution Downlink” in *IEEE 19th International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC 2008)*, Cannes, France, September 2008.

[11] I. Siomina and S. Wanstedt, “The Impact of QoS Support on the End User Satisfaction in LTE Networks with Mixed Traffic” in *IEEE 19th International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC 2008)*, Cannes, France, September 2008

[12] K. Bogineni, R. Ludwig et co. “LTE Part I: Core Network – Guest Editorial” in *IEEE Communications Magazine*, vol. 47, No. 2, pp. 40-43, February 2009.

[13] 3GPP TS 23.107 v9.0.0, “Technical Specification Group Services and System Aspects; Quality of Service (QOS) concept and architecture,” Rel. 9, December 2009.

[14] G.-M. Muntean, “Efficient Delivery of Multimedia Streams Over Broadband Networks using QOAS” in *IEEE Trans. on Broadcasting*, vol.52, No.2, pp. 230-235, June 2006.

[15] J. C. Ikuno, M. Wrulich and M. Rupp, “System level simulator of LTE networks” in *Proceedings of IEEE 71st Vehicular Technology Conference*, Taipei, Taiwan, May 2010. Available online at http://www.nt.tuwien.ac.at/about-us/staff/josep-colom-ikuno/lte-system-level-simulator/