# Dynamic Resource Allocation for IEEE802.16e

Alberto Nascimento[1,2] and Jonathan Rodriguez[2,3]

[1] University of Madeira (UMa)
[2] Telecommunications Institute, Aveiro
[3] University of Aveiro (UA)
`ajn@uma.pt, jonathan@av.it.pt`

**Abstract.** Mobile communications has witnessed an exponential increase in the amount of users, services and applications. New high bandwidth consuming applications are targeted for B3G networks raising more stringent requirements for Dynamic Resource Allocation (DRA) architectures and packet schedulers that must be spectrum efficient and deliver QoS for heterogeneous applications and services. In this paper we propose a new cross layer-based architecture framework embedded in a newly designed DRA architecture for the Mobile WiMAX standard. System level simulation results show that the proposed architecture can be considered a viable candidate solution for supporting mixed services in a cost-effective manner in contrast to existing approaches.

**Keywords:** Mobile WiMAX, Dynamic Resource Allocation, Scheduler, Cross-Layer, Quality of Service, Fairness.

## 1 Introduction

The cost-effective delivery of Triple play applications (video, audio and data) in a ubiquitous and seamless manner are expected to be the main drivers of B3G networks, placing highly stringent constraints on both high data rate transmission and Quality of Service (QoS) demands. Therefore, it is important to investigate techniques that can maximize spectral utility by efficiently managing the radio resources for heterogeneous packet-based services. Fundamental to resource management is packet scheduling. An efficient scheduling policy must exploit cross-layer information from the PHY layer as well as from higher layers, to provide efficient mapping of data blocks onto the available transport channels within the Medium Access Control (MAC) layer, whilst providing the requested quality of service (QoS) in a heterogeneous service environment [1,2]. In order to cater for fairness, efficiency in resource usage and QoS provisioning, different variations of Opportunistic Scheduling have been proposed. The Proportional Fairness scheduler (PF) algorithm is widely used in 1xEV-DO, also named HDR networks [2]. The algorithm selects the user, among all backlogged users, which has the best feasible data rate normalized by the average throughput over a sliding window filter. Modified Largest Weighted Delay First Scheduler (M-LWDF) scheduler is a kind of modified PF, where the priority value to assign to the user is equal to that of the PF scaled by the weighted delay the user has endured. The algorithm attempts to satisfy the QoS statistically [4]. The Exponential

Scheduler (EXP) attempts to equalize the weighted delays of all queues when their differences become large [5]. Utility schedulers [6] prioritize users according to their potential revenue if serviced. Different schedulers based on the notion of utility functions have been proposed in the literature [7-11]. But most of these invariably attempt to maximize the total system utility. In [7] the authors propose a scheduling algorithm principle similar to the concept of the utility function proposed in this work. However, they do not consider the channel state in the scheduling process and cannot benefit from a fully-compliant cross-layer scheduling design.

In this paper we propose an efficient packet scheduling algorithm fully aligned with this cross-layer paradigm, and specifically designed around an adaptive DRA architecture framework for the OFDMA multiple access scheme adopted by the IEEE 802.16e, also called Mobile WiMAX [12]. A key principle of our DRA is the exploitation of the inherent system diversities in various domains through the intelligent management of the allocation and access of users to the available radio resources.

This paper is organized as follows: section 2 summarizes our utility-based scheduling principle, section 3 describes the WiMAX Dynamic Resource Allocation architecture, section 4 presents the simulation scenarios and the performance results, and finally section 5 concludes the paper.

## 2    Utility-Based Scheduling

This family of schedulers prioritizes users according to their potential revenue if serviced. The notion of cost or revenue can be quantified by the means of "utility functions", a notion derived from utility theory in economics which can answer these economic scheduling criteria [6]. Fig. 1 provides an example of diverse utility functions for traffic applications with different delay constraints.

Typical utility schedulers attempt to maximize the total system utility. The typical approach is to select packets from queues in order to maximize the following metric [9, 11]:

$$k(n) = \arg \max_i \left( \left| U^{'}(\tau_i(n)) \right| . R_i(n) \right) \tag{1}$$

Where:

- $\left| U^{'}(\tau_i(n)) \right|$ is the absolute value of the derivative of the utility function for user $i$ at the beginning of (TTI) $n$.

- $\tau_i(n)$ is the HOL packet delay of user $i$ at the beginning of transmission time interval (TTI) $n$.

- $R_i(n)$ is the transmitted data rate offered by the channel of user $i$ if it is scheduled for transmission.

Although conceptually simple, this solution may degenerate into a greedy algorithm such as the Max C/I because it only concerns a single user. Therefore, the transmission of data for a given user translates in a benefit to this particular user, but at the same time there is also a cost in deferring transmissions for other users. However, they do not consider the channel state in the scheduling process and cannot benefit from a fully-compliant cross-layer scheduling design.
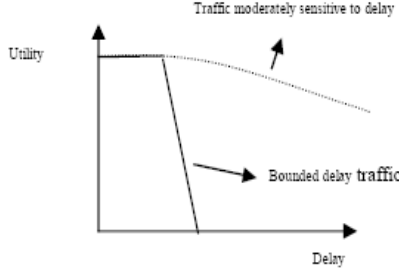
**Fig. 1.** Different types of Utility Functions

## 2.1 Proposed Algorithm

The scheduler should maximize the total utility of the system expressed as the sum of utilities of each user. More precisely the basic idea behind the proposed utility based scheduler is described herein:

1. At the beginning of TTI $n$ estimate the total amount of potential utility: $U_p(n)$

2. For a given user $j$, estimate the amount of utility that will be transferred by the network if the user is scheduled for transmission:

$$U_j\big(R_j(n),\mathbf{Q}_j(n)\big)= \sum_{k=1}^{L_j}U\big(\tau_k^j\big)(1-PER_j)\tag{2}$$

Where:

- $R_j(n)$ is capacity of the channel for user $j$ during TTI $n$. It represents the number of packets $(L_j)$ that can be transferred if this user is scheduled for transmission in this TTI.
- $\mathbf{Q}_j(n)$ is the vector representing the delay of each packet from the buffer of user $j$.
- $PER_j$ is the packet error rate associated with the channel for user $j$ during TTI $n$.
- 3. Assuming user $j$ is scheduled for transmission, the packets not transmitted loose their utility (they have their delays increased by one TTI). This translates to a remaining potential utility: $U_p(n+1\mid j)$.
- 4. The selected user is the one that satisfies the following scheduling decision:

$$k(n) = \arg\max_j\big(U_j\big(R_j(n),\mathbf{Q}_j(n)\big)-L_j(n)\big)\tag{3}$$

Where:

$$L_j(n) = U_p(n)-U_p(n+1\mid j)-U_j(R_j(n),\mathbf{Q}_j(n))\tag{4}$$

is the loss of utility incurred by choosing user $j$. This strategy means that one wants to maximize the utility transferred to a given user by subtracting from the transferred utility the losses incurred.

## 2.2 Utility Function Definition

The utility function of packet delay $U(\tau)$ must satisfy the following properties:

4. $U(\tau):[0,\tau_{\max}[\to[0,1]]$. If the network offers a delay higher than the maximum delay tolerated its utility is equal to zero in order to express the total user dissatisfaction.

- $\lim_{\tau\to0^+} U(\tau)=1$. This property reflects a maximum user satisfaction (100%) when the network offered delay tends to zero.
- $U(\tau):[0,\tau_{\max}[\to[0,1]]$. If the network offers a delay higher than the maximum delay tolerated its utility is equal to zero in order to express the total user dissatisfaction.
- $\lim_{\tau\to0^+} U(\tau)=1$. This property reflects a maximum user satisfaction (100%) when the network offered delay tends to zero.
- if $\tau_i > \tau_j \Rightarrow U(\tau_i) \le U(\tau_j))$. The utility function is monotonically decreasing with delay.
- $U'(\tau) \le 0$ and $U''(\tau) > 0$. The first order derivative is negative to reflect the decrease in the level of satisfaction of the user if the offered delay increases.
- The second order derivative must increase as the delay approaches the deadline.

One particular case of a function complying to these properties is the sigmoid function. Sigmoid functions have been used in the literature to approximate the user's satisfaction with respect to the quality of service provided by operators. We used the following modified sigmoid utility function:

$$U(\tau) = \begin{cases} 1 - \dfrac{2}{1+\exp(-\alpha(\tau-\beta))}, \text{if } \tau \in [0,\tau_{\max}] \\ 0 \qquad\qquad\qquad\quad, \text{ if } \tau \in [\tau_{\max},+\infty[ \end{cases} \tag{5}$$

The parameters $\alpha$ and $\beta$ determine, respectively the steepness and the center of the curve. They can be tuned to customize the function for different types of service. In our case $\beta = \tau_{\max}$ and $\alpha = 0,5$.

## 3   WiMAX DRA Architecture

A system level simulator that models the salient features of the WiMAX standard IEEE802.16e [12-14] was developed. The simulator implements the DL communication in a PMP configuration using the TDD mode of transmission. In the MAC frame, resources are available in both frequency (sub-channels) and time domains (OFDM symbols). The smallest granularity of resource allocation in the time and frequency domains is the slot. The size of the slot depends on the type of sub-channelization mode and on the direction of transmission. A burst is a rectangular allocation of a group of slots with the same modulation and coding scheme (MSC) belonging to the same or different users. In our simulations each packet is mapped into a group of contiguous slots forming a burst intended to a single user provided

the same modulation and coding scheme is followed in the transmission of all packets allocated to it. Each burst contains only one MAC PDU (to which a single packet is mapped into). The MAC PDU is segment into forward error correction (FEC) blocks that are coded and interleaved within the burst.

In the implemented DRA, Partial Usage sub-channelization Scheme (PUSC) was used. In PUSC sub-channels are realized using a distributed sub-carrier permutation method that pseudo-randomly draws sub-carriers to form a sub-channel and used to achieve frequency diversity in cases where the mobile speed makes it difficult or inefficient to track frequency-selective channel variations. For further details regarding the SINR modeling, Link Level Interface and Link Adaptation schemes please refer to [14].
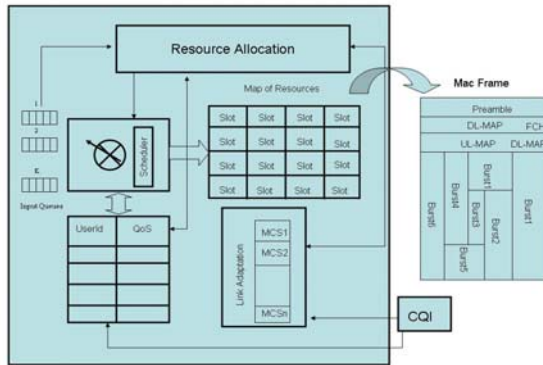


**Fig. 2.** Dynamic Resource Allocation Architecture

Figure 2 is a schematic representation of the proposed DRA scheme for the Wi-MAX standard. The packet scheduler creates a list sorted by the decreasing order of the priority metric. Packets achieving the maximum delay bound for the service are dropped in the BS. The Resource Allocator assigns slots for the packets remaining in the buffer of the selected user. The process continues until there are no packets for transmission or no slots available in the map of resources. Each burst is assigned a free HARQ channel that uses type II Chase Combining . The whole transmission process elapses along a cycle equivalent to two frames. Feedback channels (HARQ and CQICH) are assumed to be transmitted in the uplink sub-frame of the same frame in which the corresponding downlink transmission occurred.

In the simulations all users experiencing a bad channel condition, i.e., those users reporting a CQI value which do not allow the selection of the most robust MCS scheme, are not considered in the scheduling process. The idea is to limit the amount of packets received in error due to bad channel quality.

### 3.1 Channel Models

Fast fading is generated by a modified Jakes model where the carrier frequency and the mobile speed are used for fast generation of independent Rayleigh faders [15]. In the simulation a wideband SISO channel model is implemented by a six tapped delay

model, according to the Pedestrian-B 3Km/h channel model for the serving sectors of the central BS. A flat fading channel is assumed for neighboring cells. Bi-dimensional log-normal shadowing is generated at the beginning of each run $i$ [16]. The shadowing $SH(x, y, j)$ in DB between one $MS(x, y)$ and one BS $j$ is the sum of two variables:

$$SH(x, y, j) = \sqrt{0.5(F_0(x, y) + F_j(x, y))} \tag{6}$$

Where $F_0(.)$ and $F_j(.)$ are spatial functions generated using the method described in [16]. They have a Gaussian distribution with zero mean and $\sigma_{shad}$ standard deviation and a spatial correlation given by:

$$R(d) = e^{-\ln(2)d / DecorrLength} \tag{7}$$

Where $d$ is the distance between two points and "DecorrLength" is the shadowing de-correlation length.

The path loss model is the modified COST231 Hata for the urban macrocell at a carrier frequency $f$ [GHz] between 2 and 6 GHz, assuming the BS and MS heights of 32m and 1.5m respectively, and is given by:

$$PL(d) = 35.2 + 35\log_{10}(d) + 26\log_{10}(f / 2) \tag{8}$$

Using the aforementioned channel models, the SINR (Signal-to-Interference-plus-Noise Ratio) of each OFDM sub-carrier is computed according to the approach:

$$\gamma_k = \frac{I_{or}}{I_{oc} + N_o} \cdot \frac{N_{used}}{N_d + PDR.N_p} \cdot H_k \tag{9}$$

Where $N_{used}$ is the total number of sub-carriers, PDR is the Pilot-to-Data sub-carriers power ratio and $N_d$ is the number of data sub-carriers per OFDM symbol, for the PUSC channel mode. $N_o$ is the receiver thermal noise power and $I_{oc}$ is the other-cell interference power density, assumed spatially and temporarily uncorrelated. It is assumed that neighboring cells transmit with maximum power, i.e. with full load.

The gain of the $k^{th}$ sub-carrier is computed according to the recommendations of as follows:

$$H_k = \left| \sum_{p=1}^{N_{paths}} M_p A_p e^{j\theta_p} e^{-2\pi f_k T_p} \right|^2 \tag{10}$$

Where $p$ represents the multi-path index for the Ped. B channel model $A_k$ is the amplitude value corresponding to the long-term average power for the $p^{th}$ path of this same channel model $f_k$ is the relative frequency offset of the $k^{th}$ sub-carrier of the specific FUSC sub-channel and $T_p$ is the relative time delay of the $p^{th}$ path.

In the receiver, post-processing of the signals received from the serving and interfering BSs is performed. For a Single-Input-Single-Output (SISO) architecture and a matched filter in the receiver, the received signal at the $k^{th}$ sub-carrier for the target user is computed according to:

The transmission of a coded block over different sets of sub-carriers results in a number of SINR measures that equals the number of sub-carriers sets, which can be quite high. Hence, data compression is mandatory. The coded symbol SINR in sub-carrier $k$ is given by:

$$SINR^{(0)}[n] = \frac{P^{(0)}_{slot} P^{(0)}_{loss} \left| H^{(0)}[k] \right|^2}{\sum_{j=1}^{N} P^{(j)}_{slot} P^{(j)}_{loss} \left| H^{(j)}[k] \right|^2 + \sigma_n^2} \tag{11}$$

Where $N$ is the number of interferers, $P^{(j)}_{slot}$ is the transmit power per slot for the $j^{th}$ cell, $P^{(j)}_{loss}$ is the distance dependent path loss, including shadowing and antenna gains/losses, $H^{(j)}[k]$ is the channel gain for the desired MS from the $j^{th}$ cell and for the or the $k^{th}$ sub-carrier, $X^{(j)}[k]$ is the transmitted symbol by the $j^{th}$ cell at the $k^{th}$ sub-carrier, $N^{(0)}[k]$ is the thermal noise at the received, modeled as AWGN with zero mean and variance $\sigma_n^2$.

The set of coded symbols SINRs are mapped onto a single value named the Effective SINR value. This value can be used to match AWGN Look-Up Tables (LUTs). The EESM expression determines how the effective SINR is obtained from the multiple SINR's on the different subcarriers:

$$SINR_{eff} = -\beta \ln\left( \frac{1}{P} \sum_{p=1}^{P} e^{-\frac{SINR_p}{\beta}} \right) \tag{12}$$

Where the parameter $\beta$ is to be optimized for every link mode (MCS) based on link level simulation results.

## 3.2  User Quality Tracking

Periodically the SIR measurement performed by each MS, $y_n(MS_i)$, using the symbol of the preamble is reported and is available at the BS at $nT_{CQI}$ ($T_{CQI}$ is the reporting period). Every $T_{CQI}$ the quality of the channel is updated combining the past information and the new value $w_n(MS_i)$ provided by these measurements, according to the time-smoothing filter:

$$CQI_n(MS_i) = 0.7 * w_n(MS_i) + 0.3 * CQI_{n-1}(MS_i) \tag{13}$$

### 3.3  Traffic Models

Simulations were carried on using the NRTV64 traffic model [17]. This traffic model periodically generates packets of variable sizes which compose a frame of video data arriving at a regular interval of $T$ seconds. The NRTV traffic model is a bursty traffic model with an average source bits rate of 64Kbps. Table I lists the parameters used in the NRTV traffic model.

**Table 1.**

| Characteristics | Distribution | Parameters |
|---|---|---|
| Inter-arrival time between frames ($T$) | Deterministic | 100ms |
| Number of packets/frame | Deterministic | 8 |
| Packet size | Truncated Pareto (Mean: 50, Max: 125 (bytes)) | K=20 bytes, $\alpha=1.2$ |
| Inter-arrival time between packets | Truncated Pareto (mean: 6, Max: 12.5 (ms)) | K=2.5ms, $\alpha=1.2$ |

### 3.4  Simulation Scenario

The system level performance evaluation methodology is based on the draft evaluation proposal for IEEE802.16. Simulations are carried out using a combined snapshot-dynamic mode, where in each simulation run (of a total of $N_{run}=5$ independent dynamic runs) $N_{user}$ users are randomly uniformly distributed over a hexagonal network of tri-sectored cells composed of three tiers of 19 BSs with 3 sectors each). Each simulation run lasts for $T_{run}=75000$ frame periods. User positions were updated on every run and for each MS-BS pair, a random shadowing value is drawn whilst mobile positions, shadowing and path loss values are kept constant for the whole simulation duration. A full load scenario is assumed where all BSs around the central cluster are assumed as transmitting with maximum power all the time. They are considered for DL interference only on the central BS, which is the only one simulated.

In each frame interval the following events are performed: packets are generated according to the NRTV64 traffic model; the fast fading channel is updated; DRA executed and packet quality detection is performed. During the packet quality detection, the received block SIR of the packet is computed and mapped to the corresponding BLER using the link interface. A random variable $u$ uniformly distributed between 0 and 1 is drawn. If $u<BLER$ the packet is assumed as erroneous and a NACK message is feedback to the serving BS on the CQICH of the uplink subframe, otherwise the packet is assumed as correct and an ACK message is sent. The ACK or NACK message is assumed to be received within the uplink portion of the same TDD frame (uplink sub-frame) in which data is initially transmitted or retransmitted

### 3.5  Performance Metrics

The following metrics were used to quantify system level performance:

**Average Service Throughput per MS:** $R_{Serv}^i = \dfrac{b_i}{T}$

Where $b_i$ the total amount of bits is correctly received by the MS $i$ over the whole simulated time and $T$ is the simulation elapsed time.

**Average Transfer Delay per MS:** $T_D^i = \dfrac{1}{N} \sum_{n=1}^{N} T_D^i(n)$

Where $T_D^i(n)$ is the transferred delay associated to the $n^{th}$ packet successfully received by user $i$. This is the average packet transfer delay for all packets received with success by the mobile.

**95-percentile of the CDF of the transfer delay per user**
This corresponds to the 95-percentile of the delay from the CDF of all packets transferred delays for each user in the system.

**Average Packet Error Rate per user:** $PER^i = \dfrac{N_{Correct}^i}{N}$

Where $N_{Correct}^i(n)$ is the total amount of packets received with success by the user.

### 3.6  Simulation Results

We performed simulations for 50, 60, 70, 80 and 90 users for the UTIL, CI, PF, M-LWDF and EXP schedulers.

Figure 3 show the evolution of the average packet delay vs. the number of users in the system. In the overloaded scenario the M-LWDF scheduler does not perform well because it prioritizes packets based on the HOL packet delay, without considering how close a packet's delay can be to its delay threshold. The EXP scheduler has a better performance because it attempts to equalize the HOL packet delays for all
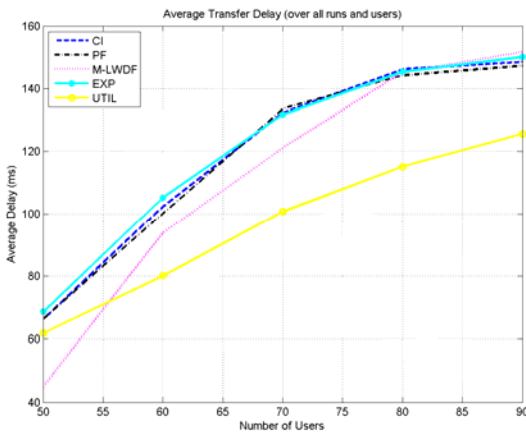


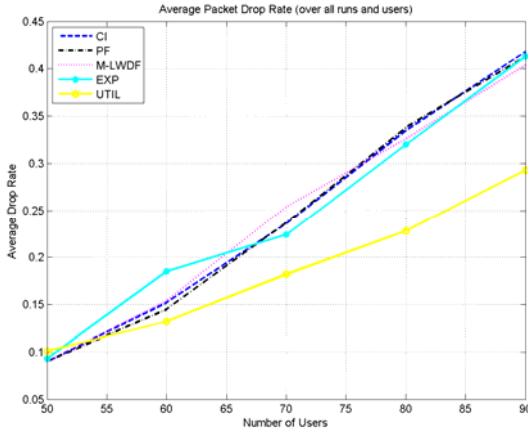**Fig. 3.** Average Transfer Delay vs. total number of users

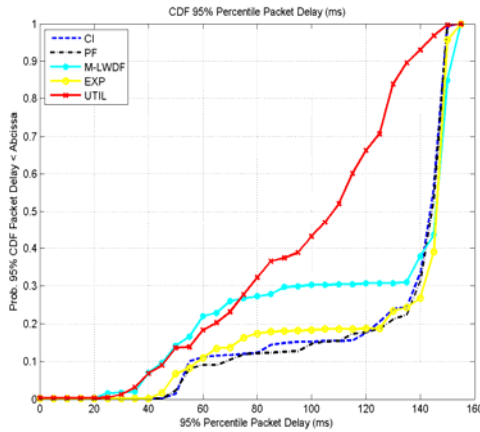**Fig. 4.** Average Packet Drop Error Rate vs. total number of users



**Fig. 5.** CDF of the 95 Percentile of the Packet Delay

users. However, the highest gain can be observed with the UTIL scheduler. This is due to the sensitivity of the algorithm as the delay approaches its delay bound, which is determined by the according to the type of utility function used.

Figure 3 show the evolution of the average packet delay vs. the number of users in the system. In the overloaded scenario the M-LWDF scheduler does not perform well because it prioritizes packets based on the HOL packet delay, without considering how close a packet's delay can be to its delay threshold. The EXP scheduler has a better performance because it attempts to equalize the HOL packet delays for all users. However, the highest gain can be observed with the UTIL scheduler. This is due to the sensitivity of the algorithm as the delay approaches its delay bound, which is determined by the according to the type of utility function used.

Figure 4 is the plot of the average packet error rate vs. no. of users. It can be noticed that the gain achieved with the UTIL algorithm for a total amount of users is higher

than 70. The better performance achieved with the MLWDF for a low a number of users smaller number of users is due to the higher percentage of packets dropped.

Figures 5 and 6 shows the CDF of the 95th percentile of the CDF of all packets delays and the average packet delay for all users respectively, for 70 users (for 70 users the average PER is below 20% and the average service throughput is around 50kbits. Note that the offered service data rate is 64 kbps). The performance gains of the UTIL algorithm are evident from both the graphs in Fig. 3 and 4.
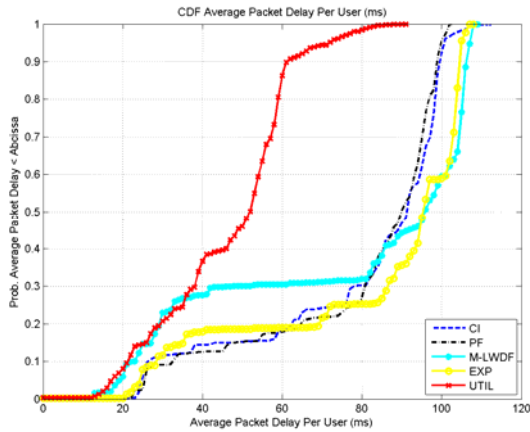


**Fig. 6.** CDF of the Average Packet Delay
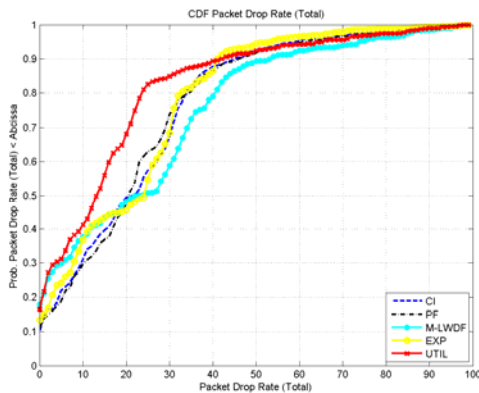


**Fig. 7.** CDF of the Average Packet Drop Rate

Figures 7 corroborate the gains achieved with the UTIL scheduler. The packet drop rate (including both packets dropped for quality reasons and those dropped due to delay bound violation) has a better performance for the UTIL scheduler.

Figure 8 shows the average user throughput versus the average sector throughput for the utility-based scheduler and for the max C/I scheduler as a benchmark for
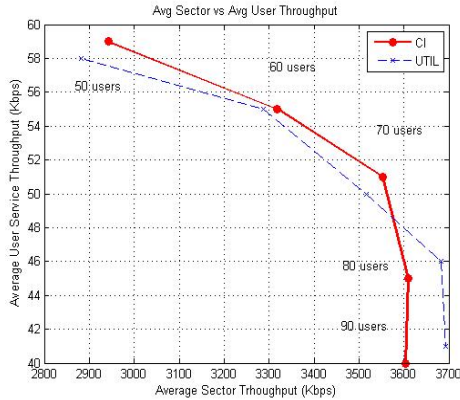
**Fig. 8.** Avg. User Throughput vs. avg Sector Throughput

different loads. It can be seen than although the Max C/I scheduler is more efficient in terms of resource utilization, the performance achieved with utility-based scheduling is quite close to that one achieved with the max C/I. Indeed, for higher amount of users, overloading results in the utility-based scheduler achieving even higher user throughputs for the same amount of sector throughput as for the Max C/I scheduler.

## 4  Conclusions

In this paper we propose an innovative cross-layer DRA architecture for the Mobile WIMAX standard using a packet based scheduler based on the notion of utility functions. Both the channel state as well as the packet delay are considered in the scheduling process. The simulation results show that the DRA scheme together with the utility-based scheduling algorithm has the capacity to increase the delivered QoS within WiMAX whilst utilizing spectrum efficiently.

## Acknowledgement

## References

1. Gomez, J., Alonso, J., Verikoukis, Ch., Pérez-Neira, A., Alonso, L.: Cooperation on Demand Protocols for Wireless Networks. In: IEEE PIMRC 2007, Athens (September 2007)
2. Shakkottai, S., Rappaport, T.S., Karlson, P.: Cross-Layer Design for Wireless Networks. IEEE Comm. Magazine 4(10), 74–80 (2003)
3. Kopp, R., Humblet, P.: Information Capacity and Power Control in Single-Cell Multiuser Communications. In: Proc. Of ICC (June 1995)

4. Viswanath, P., Tse, D.N.C., Laroia, R.: Opportunistic beamforming using dumb antennas. IEEE Trans. Inform. Theory 48(6), 1277–1294 (2002)
5. Andrews, M., Kumaran, K., Ramanan, R., Stolyar, A., Vijayakumar, R., Whiting, P.: CDMA data QoS scheduling on the forward link with variable channel conditions, Bell Labs Technical Report preprint (April 2000)
6. Shakkottai, S., Stolyar, A.: Scheduling algorithms for a mixture of real-time and non-real time data in HDR. In: Proc. of the 17th ITC, Salvador, Brazil (2001)
7. Shenker, S.: Fundamental design issues for the future Internet. IEEE JSAC 13(7), 1176–1188 (1995)
8. Choi, Y.J.: Delay-Sensitive Packet Scheduling for a Wireless Access Link. IEEE Trans. on Mob. Computing 5(10), 1374–1382 (2008)
9. Choi, Y.J., Choi, J.G., Bahk, S.: Upper-level scheduling supporting multimedia traffic in cellular data networks. Computer Networks 51(3), 621–631 (2007)
10. Ryu, S., Ryu, B.-H., Seo, H., Dhin, M., Park, S.: Wireless Packet Scheduling Algorithm for OFDMA System Based on Time-Utility and Channel State. ETRI Journal 27(6), 777–787 (2005)
11. Sang, A., Wang, X., Madihian, M., Gitlin, R.: A Flexible Downlink Scheduling Scheme in Cellular Packet Data Systems. IEEE Trans. On Wireless Comm. 5(3) (March 2006)
12. Lei, H., Zhang, L., Yang, D.: A Packet Scheduling Algorithm Using Utility Function for Mixed Services in the Downlink of OFDMA Systems. In: IEEE VTC-2007 Fall, September 2007, pp. 1664–1668 (2007)
13. IEEE Std 802.16e, Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems (December 2005)
14. WiMAX forum, http://www.wimaxforum.org
15. Nascimento, A., Rodriguez, J., Gameiro, A.: Dynamic Resource Allocation Architecture for IEEE802.16e: Design and Performance Analysis. ICT Mobile Summit, Stockholm (June 2008)
16. Li, Y., Huang, X.: The generation of independent Rayleigh faders. In: Proc. ICC200 – International Conference on Communications, vol. 1, pp. 41–45
17. Cai, X., Giannakis, G.B.: A two-dimensional channel simulation model for shadowing processes. IEEE Transactions on Vehicular Technology 52(6)
18. UMTS, Selection procedures for the choice of radio transmission technologies of the UMTS, ETSI, UMTS TR 101.112 v.3.2.0. Tech. Rep. (April 1998)