A Channel Aware Scheduling Algorithm for HSDPA System

Giorgio Manfredi*†, Pierfrancesco Annese*, Umberto Spagnolini†
*Siemens Mobile Communications, Milano (Italy)
† Dip. Elettronica e Informazione, Politecnico di Milano (Italy)

Abstract— **High Speed Downlink Packet Access (HSDPA) is an evolutionary step to boost the WCDMA performance for downlink packet traffic as it increases spectral efficiency by supporting high user data rates.**

Because of the time varying channel fluctuations of wireless links, efficient use of the large available bandwidth cannot be achieved without adopting channel aware packet scheduling strategies. The HSDPA scheduling algorithm is based on splitting the scheduling decision into user selection and resource allocation. In this paper we propose two user selection algorithms that can be adjusted (or pre-configured) in terms of cell throughput and/or user throughput fairness. Finally, we discuss a channel aware resource allocation algorithm that aims to maximize the number of served users among the selected ones, without any loss of the overall cell throughput with respect to the traditional greedy resource assignment.

I. INTRODUCTION

High-speed Downlink Packet Access (HSDPA) for WCDMA has been included in the 3GPP Release 5 UMTS specifications [1] to better support data communications that, compared to voice services, are characterized by lower real-time constraints, higher data rates and asymmetric traffic (i.e., higher traffic volume for downlink rather than uplink). The technological enhancements incorporated by the HSDPA upgrade the WCDMA performance either in term of spectral efficiency and user data rates (see e.g. [2]).

To implement the HSDPA service the 3GPP specifications introduced a new downlink transport channel, the High Speed Downlink Shared Control Channel (HS-DSCH), that carries data to a selected user (or User Equipment - UE) within the transmission time interval (TTI) of 2 ms. Bits from HS-DSCHs are mapped onto up to 15 physical downlink shared channels. The associated High Speed Shared Control Channel (HS-SCCH, there are up to 4 configurable HS-SCCH in a cell) is used to address the selected UEs and give them the information needed to decode HS–DSCHs. In the uplink direction, a user dedicated physical control channel brings *Channel Quality Indicator* (CQI) bits and positive or negative acknowledgements related to the received HS-DSCH frames. The packet data coming from Radio Network Controller (RNC) into base station (or Node B) are characterized by different scheduling priority weights. At Node B side, these packets are stored in the UE queue buffer associated to the indicated priority level (i.e., priority queue buffer). To resume, adapt or stop the incoming traffic carrying user data from RNC, a new handshake protocol has been introduced (i.e., capacity request/allocation mechanism). The key idea of the HSDPA is to increase packet data throughput by *adaptive modulation and coding, H-ARQ and fast scheduling* [3].

The principle of adaptive modulation and coding is to change the modulation and coding in accordance with variations in the channel conditions for each user, estimated by CQI and power measurements on the associated dedicated channel (DPCH). CQI is sent in the uplink by the UE, and provides implicit information about the instantaneous signal quality received by the user. A turbo code is used for data protection. The encoder is based on the Release 99 turbo encoder with rate of 1/3 even though other effective coding rates can be achieved by means of rate matching (i.e., puncturing and repetition). Hybrid ARQ allows the reduction of the number of retransmissions due to erroneous received data packets by combining the information from successive retransmissions prior to decoding.

Fast scheduling is the mechanism that determines which UEs to transmit to within a given transmission interval. Moving the scheduling to Node B enables a more efficient implementation of scheduler by allowing it to work with the instantaneous channel information. More specifically, the scheduler can select the modulation and coding scheme to better match the current channel conditions and fading environment.

This paper is focused on the Packet Scheduler as central entity of the HSDPA design [4]. It controls the allocation of the resources to users and, to a large extent, it strongly influences system performance in terms of system capacity, throughput and fairness among users. The scheduler exploits the multi-user diversity and strives to transmit to users when radio conditions permit high data-rates.

The paper is organized as follows: the definition of the scheduling problem for packet system and the proposed sub-optimal solution are in Section II. In Section III we propose two different user selection algorithms that exploit the information available at Node B; these are flexible enough to be adapted to match the needs of the operator (i.e., fairness, cell throughput, etc.). In Section IV we introduce an iterative resource allocation strategy to share power and codes among users by using a metric based on fading prediction and users' queue state. Finally, in Section V, we validate the benefits of the proposed scheduling method by HSDPA link level simulations.

II. Definition of the Scheduling Problem

The aim of the scheduling algorithm is to select a subset $\mathcal{K}$ composed of $K = |\mathcal{K}|$ UEs out of the R configured ($K \leq$

R) and, for each of the selected users (say i-th user, with $i \in \mathcal{K}$), to choose the transmission power (p_i) and number of codes (n_i). The set of parameters

$$\{n_i, p_i\} \qquad i \in \mathcal{K} \tag{1}$$

have to optimize an objective function based on the reduction of the packet losses, the maximization of cell throughput still maintaining a degree of fairness. The codes $\mathbf{n}=\{n_i\}_{i\in\mathcal{K}}$ and power $\mathbf{p}=\{p_i\}_{i\in\mathcal{K}}$ distribution have to take into account both system-wide and individual resource constraints

$$\sum_{i\in\mathcal{K}} n_i \leq N, \qquad \sum_{i\in\mathcal{K}} p_i \leq P, \qquad n_i \leq N_i. \tag{2}$$

Here N and P denote system constraints on the total number of spreading codes and the total system power, respectively. N_i is an individual constraint on the maximum number of codes that can be supported in a delivered HS-DSCH. This last parameter is assigned according to the user category described in [5].

According to the formulation of a scheduling problem provided by Kelly [6], each UE is represented by an increasing concave utility function $U_i(\mathbf{s}_{i,t})$ of a set of user conditions $\mathbf{s}_{i,t}$ at time t (e.g., channel conditions, pending data, etc.). We model the downlink channel conditions by a time-varying channel vector $\mathbf{h}_t = [h_{1,t}, ..., h_{R,t}]$, where $h_{i,t}$ represents the user's received signal to noise ratio (SNR) per unit of transmission power at time t. Assuming that all spreading codes are mutually orthogonal, the SNR per code for i-th UE is:

$$\mathrm{SNR}_{i,t} = \frac{p_{i,t} h_{i,t}}{n_{i,t}}. \tag{3}$$

For a given channel state vector there is a rate-region $\mathcal{R}(\mathbf{h_t}) \subset \mathbb{R}_+^{\mathrm{R}}$ that indicates the set of feasible[1] transmission rates $\mathbf{r}_t = [r_{1,t}, ..., r_{R,t}]$ where:

$$r_{i,t} = n_{i,t} \log_2 (1 + \frac{p_{i,t} h_{i,t}}{n_{i,t}}). \tag{4}$$

At each scheduling time t the vector of code allocations $\mathbf{n}_t$ and the vector of power allocations $\mathbf{p}_t$ have to maximize the weighted sum rate:

$$(\mathbf{n}_t^{(\mathrm{opt})}, \mathbf{p}_t^{(\mathrm{opt})}) = \arg \max_{(\mathbf{n}_t, \mathbf{p_t}) \in \chi} \sum_i n_{i,t} w_{i,t} \log_2 (1 + \frac{p_{i,t} h_{i,t}}{n_{i,t}}) \tag{5}$$

constrained to:

$$\chi \triangleq \{(\mathbf{n},\mathbf{p}) \geq 0\colon\ n_{i,t} \leq N_i, \sum_i n_{i,t} \leq N, \sum_i p_{i,t} \leq P \ \forall i\}. \tag{6}$$

The weights are:

$$w_{i,t} = \frac{\partial U_i(\mathbf{s}_{i,t})}{\partial \mathbf{s}_{i,t}}. \tag{7}$$

[1] A set of rates is said to be feasible, if it respects all the constraints imposed from network and user capabilities.

In principle, the solution of this optimization problem can be found by maximizing the corresponding Lagrangian form. However, it is not be feasible to pursue the above solution in a real scheduling scenario for its computational complexity. To cope with the optimization (5), we use a simple greedy algorithm based on splitting the scheduling into user selection (Sect. III) and resource allocation (Sect. IV).

More specifically, first a scheduling order for the users is found by ordering them according to the time-varying weights $\{w_{i,t}\}$ assigned to the UEs at time t. Given the scheduling order, the resources are allocated by taking each UE in order and by choosing a set of resources that maximizes the transmission rate the UE can receive choosing the available power and codes (i.e., the resources unassigned to the previous users are free to be assigned).

III. Adaptive User Selection Algorithms

Here we propose two algorithms for user selection that classify all configured UEs according to the time-varying weights $\{w_{i,t}\}$. The weight $w_{i,t}$ represents user condition at time t according to predicted channel (or fading) $\widehat{\alpha}_{i,t}$ or estimated current bit rate $d_{i,t}$, queue state $Q_{i,t}$ and timing of pending retransmission $T_{NACK_{i,t}}$. Current bit rate $d_{i,t}$ can be calculated from the predicted fading value by assuming to assign to the UE all the available physical resources:

$$d_{i,t} = n_i \log_2 (1 + \frac{P \cdot \widehat{\alpha}_{i,t}}{n_i \cdot N_0}), \tag{8}$$

with $n_i = \min\{N_i, N\}$. Notice that $d_{i,t}$ must be evaluated according to the available amount of data in user's queues and N_0 is receiver noise level.

Queue state $Q_{i,t}$ is defined here as

$$Q_{i,t} = \max_j \{[q_M - (q_M - q_{i,t_o}(j)) e^{-(\frac{t-t_0}{\Delta T})}]\}, \quad j = 1, .., L_i(t) \tag{9}$$

where $q_{i,t}(j)$ is the j-th queue size (i.e., amount of pending data in j-th queue) at time t, q_M is the maximum queue size, t_o is the last time $q_{i,t}(j)$ has changed, ΔT defines the saturation velocity of $Q_{i,t}$ and $L_i(t)$ is the number of queues for i-th user.

The timing of pending re-transmission is

$$T_{NACK_{i,t}} = \max_j \{\frac{t_{nack_{i,t}}(j)}{T_{1,j}}\} \qquad j = 1, .., H_i \tag{10}$$

where $t_{nack_{i,t}}(j)$ is the time elapsed from the arrival at Node B of a NACK, associated to the j-th stopped HARQ process, H is the number of configured HARQ processes for the i-th UE and $T_{1,j}$ represents the value of timer T_1 associated to the j-th HARQ process[2].

From the CQI values periodically reported by the UEs, we can infer fading condition. More specifically, this is carried out by linear prediction from the last two fading

[2] Timer T1 is related to the queue from which data to be sent are taken, and hence it can be associated to the j-th HARQ process pertaining these data.

samples in a conventional way as from autoregressive spectral analysis for fading process. For each configured UEs, we assume to have a new available sample $\alpha_i(nT)$ every T ms, according to CQI feedback Cycle [7]. All R configured users are sorted according to their values of $w_{i,t}$ and the leading set $\mathcal{K}$, with $K = |\mathcal{K}|$ equal to the number of configured HS-SCCHs, is considered for further resource allocation.

A. Fading Queue Time (FQT) User Selection

We represent each of the R configured UE by the time-varying weight $w_{i,t}$ assigned to the $i-$th user at time t as a linear combination of three terms:

$$w_{i,t}(\widehat{\alpha}_{i,t}, Q_{i,t}, T_{NACK_{i,t}}) = \\ = \delta \cdot \frac{\widehat{\alpha}_{i,t}}{\sum\limits_{j=1}^{R} \widehat{\alpha}_{j,t}} + \gamma \cdot \frac{Q_{i,t}}{\sum\limits_{j=1}^{R} Q_{j,t}} + \eta \cdot \frac{T_{NACK_{i,t}}}{\sum\limits_{j=1}^{R} T_{NACK_{j,t}}} \tag{11}$$

where $\delta+\gamma+\eta = 1$. The parameters δ, γ, η are used to control the behavior of the metric (11) between MaxC/I (see [8]) obtained for $\delta = 1, \eta = \gamma = 0$, and MaxQueue with $\delta = \eta = 0, \gamma = 1$ (see [9], [1]). Notice that $\delta = \gamma = 0, \eta = 1$ is not considered as a valid parameter configuration because, in this way, no information about channel or data condition is brought, thus producing a "blind user selection" if there are no UEs with pending retransmission at time t.

In (11) the term that depends on $\widehat{\alpha}_{i,t}$ favours users with best channel condition thus giving a degree of unfairness to the scheduling policy, while $Q_{i,t}$ restore the fairness among users leading towards selection of the UEs with larger amount of data in their queues when these have not been served because they have experienced a deep fading condition. Therefore queue state $Q_{i,t}$ prevents the selection of a UE with a good channel condition, but few data to send, that could potentially waste resources by sending only a small data packet.

The last term $T_{NACK_{i,t}}$ favours the scheduling decision towards UEs who need retransmission, thus reducing the number of lost packet for T_1 expiration, and minimizing the average time for a correct decoding.

If i-th user cannot be served in current transmission time interval because of other constraints (e.g., the setting based on minimum inter transmission interval [5]), then it is assigned $w_{i,t}(\widehat{\alpha}_{i,t}, Q_{i,t}, T_{NACK_{i,t}})$=0.

B. Rate Queue Time (RQT) User Selection

Since the MaxC/I algorithm ($\delta = 1, \eta = \gamma = 0$ in (11)) has the drawback of selecting a UE in constructive fading even if it has only few data to send, we propose to replace MaxC/I contribution in (11) with MaxRate.

Alike FQT user selection, we represent each of the R configured UEs by a time-varying weight $w_{i,t}$ assigned to the i-th user at time t as:

$$w_{i,t}(d_{i,t}, Q_{i,t}, T_{NACK_{i,t}}) = \\ \delta \cdot \frac{d_{i,t}}{\sum\limits_{j=1}^{R} d_{j,t}} + \gamma \cdot \frac{Q_{j,t}}{\sum\limits_{j=1}^{R} Q_{j,t}} + \eta \cdot \frac{T_{NACK_{i,t}}}{\sum\limits_{j=1}^{R} T_{NACK_{j,t}}} \tag{12}$$

for $\delta + \gamma + \eta = 1$. $Q_{i,t}$ and $T_{NACK_{i,t}}$ are defined as in (9) and (10), respectively.

The parameters δ, γ, η in (12) control the behavior of the metric from MaxRate (see [8]), obtained with $\delta = 1, \eta = \gamma = 0$, and MaxQueue (see [9], [1]), with $\delta = \eta = 0, \gamma = 1$. Notice that $\delta = \gamma = 0, \eta = 1$ is not considered a valid parameter configuration as for FQT strategy. We experienced that it is appropriate to set $\eta \neq 0$ so that $T_{NACK_{i,t}}$ will force scheduling decision towards UEs who need retransmission, thus reducing the number of lost packet and limiting the average time for decoding.

IV. Iterative Power and Code Allocation (IPCA)

The resource assignment is based on the separation of power allocation and codes allocation in an iterative algorithm. For each iteration, it is allocated the power to each UE belonging to a subset $\overline{\mathcal{K}} \subseteq \mathcal{K}$ in a decremental way (initialization is for $\overline{\mathcal{K}} = \mathcal{K}$).

Here we propose to allocate the power according to the following metric:

$$M_{i,t} \triangleq \varphi \cdot \frac{\widehat{\alpha}_{i,t}}{\sum\limits_{j\in\overline{\mathcal{K}}} \widehat{\alpha}_{j,t}} + \vartheta \cdot \frac{Q_{i,t}}{\sum\limits_{j\in\overline{\mathcal{K}}} Q_{j,t}} \qquad i \in \overline{\mathcal{K}}, \tag{13}$$

based on the predicted channel fading $\widehat{\alpha}_{i,t}$ and the queue state $Q_{i,t}$, with $\varphi + \vartheta = 1$. The transmission power for i-th pre-selected user can be thus reduced to

$$p_{i,t} = M_{i,t} \cdot P \tag{14}$$

as $\sum_{i=1}^{K} M_{i,t} = 1$. Metric (13) prevents the assignment of resources to a UE with the most favourable channel conditions but few data to send. Furthermore, if a UE has a large amount of data in its queues or data pending since a long time, it would be allocated resources even if the other UEs would have better channel conditions. This let us improve the fairness of the proposed scheduling algorithms (i.e., user selection combined with resource allocation) without reducing the performances in terms of overall cell throughput. The parameters φ, ϑ are used to choose if metric $M_{i,t}$ will be throughput fairness ($\vartheta = 1$) or maximum throughput oriented ($\varphi = 1$).

Once the user transmission powers $p_{i,t}$ have been assigned, the codes allocation is done by taking each ordered UE belonging to $\overline{\mathcal{K}}$ and choosing a number of available codes that maximize the transmission rate the UE can receive with a detection error no higher than 10%. This is performed by a Reference Table T($TrBk_i$, n_i, ms_i, TSNR$_i$) derived from Transport Block Size Table in [3].

In the Reference Table we find a target SNR (TSNR) on physical channel bits, corresponding to each set of Transport block size ($TrBk_i$), number of codes (n_i) and modulation scheme (ms_i). Recall that TSNR is based on link level simulations made under the assumption of additive white Gaussian noise.

Let $n_{k,i}$ be the number of physical channels assigned to i-th UE of the subset $\overline{\mathcal{K}}$. Starting from the first UE of the subset, we estimate the SNR values, at receiver side, corresponding to a different number of possible codes assignments:

$$\mathrm{SNR}(n_{k,i}) = \frac{p_{i,t} \cdot \widehat{\alpha}_{i,t}}{n_{k,i} \cdot N_0} \qquad n_{k,i} = 1, .., N_{k,i} \tag{15}$$

with

$$N_{k,i} = \min(N_{res}, N_i). \tag{16}$$

Here N_{res} is the residual number of codes (i.e., initialized to N) and N_0 is the receiver noise power level.

If a pre-selected UE has *no pending retransmission* (i.e., $T_{NACK_{i,t}}$=0 in (11)), for each value of $n_{k,i}$, we compare the target SNR with SNR($n_{k,i}$) defined in (15). We choose $\overline{n}_{k,i}$ for which $SNR(n_{k,i})$ is closer to $TSNR(n_{k,i})$ (as SNR($n_{k,i}$)$\leq$TSNR($n_{k,i}$)) but maximizes the Transport Block Size that the UE can receive.

If a UE needs *retransmission* the Transport Block Size is fixed, so we choose $\overline{n}_{k,i}$ that minimize the number of codes allocated for this UE and for which SNR($n_{k,i}$) is closer to TSNR($n_{k,i}$). After this the residual number of codes is updated $N_{res} = N - \overline{n}_{k,i}$ and this is iterated until all available codes have been assigned ($N_{res} = 0$) or all the UEs belonging to $\overline{\mathcal{K}}$ have been configured for transmission. If there is at least one UE of the subset with power but no codes assigned resource allocation algorithm must be iterated, to avoid waste of power, by removing those UEs from $\overline{\mathcal{K}}$.

Scheduled UEs that will be served are finally represented by the following parameters:

$$UE_i = \{p_i, \overline{n}_{k,i}, ms_i, TrBk_i\} \qquad i \in \overline{\mathcal{K}}. \tag{17}$$

This resource allocation algorithm let us guarantees to serve a larger number of users than greedy algorithm because available resources are not assigned only to a single user (i.e., first classified according to user selection) but are shared among different UEs. In this way, the fairness of the scheduling algorithm is improved together with user throughput because, in assigning resources, the channel conditions are accounted for. This is validated by numerical analysis in the next section.

V. Performance Analysis

In this section we will present a performance analysis that will show the main advantages of the previous described user selection and resource allocation algorithms. The results are obtained with an HSDPA link level simulator (see Tab. 1). The UEs are made slowly varying their geographical location with time so that the corresponding path loss varies within two values (i.e., UEs are assumed to move around within a short range). Fading is modelled as Rayleigh processes. The power delay profile is described by the indoor to outdoor and pedestrian model [10]. A speed dependent Doppler effect is included in every tap of the power delay profile, and the user default speed is 3 km/h. As we consider a microcell environment and the UEs are assumed to move within a short range, with a low speed, the effect of shadowing can be considered as negligible. We consider 32 simultaneously configured users belonging to UE categories 1,2,3,4,5,6,11,12 (see [5] for specification).

PARAMETER	VALUE
Path loss	-24 dB to -26 dB
Shadowing	Not present
Power delay profile	Pedestrian A
Mobile speed	3 km/h
Number of HS-DSCH codes	15
Max cell TX power	20 Watts
HS-DSCH power	16 Watts
HARQ-SAW channels	8
Simulation Step	2 ms

TABLE 1: SIMULATION PARAMETERS

In order to model the capacity request/allocation mechanism, users' queues are assumed of finite dimensions filled at a constant bit rate, maximum ($\mathrm{Q_{max}}$ = 60 kbits) and minimum ($\mathrm{Q_{min}}$ = 12 kbits) filling threshold are defined. Using Capacity Allocation, Node B stops the incoming traffic if the maximum threshold ($\mathrm{Q_{max}}$) is reached and reactivate the filling process only when the queue is emptied below the minimum threshold ($\mathrm{Q_{min}}$).

Fig.1 shows performance of FQT/RQT-IPCA algorithms in terms of users average throughput and users throughput standard deviation, this latter term is used here as a fairness indicator. The FQT scheduler performance has been obtained by varying only the user selection coefficients. As expected, the tuning of the FQT parameters let the scheduler change performance from MaxQueue (i.e., high fairness, low throughput, $\gamma = 1$, $\delta = \eta = 0$) to MaxC/I (i.e., reduced fairness, high cell throughput, $\delta = 1$, $\gamma = \eta = 0$). Same considerations can be made about RQT: by selective changing user selection coefficients scheduler performances vary from MaxQueue ($\gamma = 1$, $\delta = \eta = 0$) to MaxRate ($\delta = 1$, $\gamma = \eta = 0$). Comparing the performance of the scheduler proposed in this paper with a greedy MaxRate there is a relevant fairness enhancement. Moreover, by taking into account the MaxC/I (MaxRate for RQT) settings of the proposed algorithm it can be stated that this improvement is due to IPCA resource allocation algorithm that increases overall cell throughput. The main cause of this enhancement is that the IPCA represents a better sub-optimal solution to (5) than greedy approach. In fact, instead of assigning all available power to the first preselected UE, here it is proposed to share the power among users. In this way, the UEs would be assigned the higher possible rate they can support according to their channel state and amount of data ready to be sent, without wasting resources.

Results in Fig.1 confirm that the proposed class of schedulers have better performance when compared to tunable proportional fair (PF) strategy [11]. This is namely for all the settings on the right hand side of the intersection of PF with FQT-IPCA or RQT-IPCA , as the lower standard deviation guarantees a better throughput fairness for the same user average throughput. The IPCA can also reach an user average throughput higher than the maximum reached by MaxRate (greedy).

The clusters of markers in Fig.1 (i.e., FQT[1/4, 3/4, 0], RQT[1/3, 1/3, 1/3]) shows the performance of IPCA resource allocation for different parameters settings (i.e., different values of φ, ϑ in (13)). Since there are no relevant differences in performance among the various possible settings, we propose a balanced selection $\varphi = \vartheta = 1/2$.

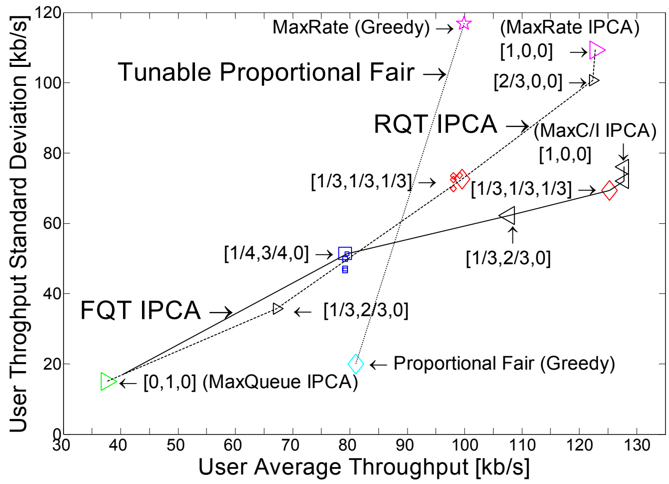

Fig.1: FQT/RQT-IPCA Scheduler Performance according to $[\delta, \gamma, \eta]$.

To better illustrate the performance, in Fig.2 it is compared the cumulative distribution function (CDF) of users throughput for the algorithm settings that has been experienced to provide the better trade-off in term of fairness/throughput (i.e., the balanced metric with $\delta = \gamma = \eta = 1/3$). It can be noticed that, compared to conventional scheduling methods (PF and MaxRate (greedy)) the proposed FQT/RQT-IPCA improve both fairness and users throughput respect to the traditional MaxRate and enhance throughput at the expense of a low reduction of fairness if compared to PF.

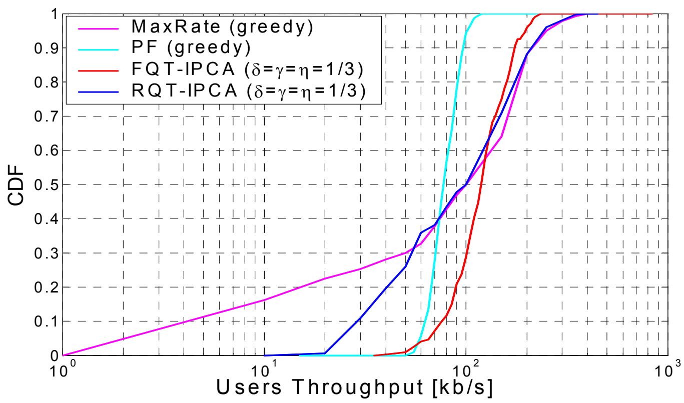

Fig.2: CDF of Users Throughput

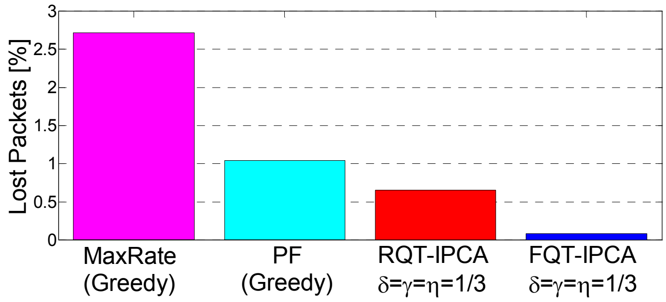

Fig.3: Effects of T_{NACK}.

Fig.3 shows the effect of the parameter T_{NACK} introduced in user selection weights. When comparing FQT-IPCA and RQT-IPCA (for $\eta \neq 0$) with PF or MaxRate (greedy) we can notice that the percentage of lost packets is decreased by means of T_{NACK} that forces the scheduling decision towards those users that need retransmission thus reducing the probability of losing packets for T_1 expiration.

VI. Conclusions

The scheduling strategy investigated in this paper splits the scheduling into user selection and resource allocation. It is proposed two class of scheduling algorithm that are suitable for implementation within the context of HSDPA. Both algorithms take into account channel conditions experienced by the UEs combined with all available information about queue state. The advantage is an increased cell throughput at the cost of a moderate reduction of throughput fairness among UEs when compared to proprtional fair schemes. The reduction of packets losses and the flexibility to operators' needs are other relevant aspects.

References

[1] H. Holma and A. Toskala, "WCDMA for UMTS: radio access for third generation mobile communications", J. Wiley and Sons, Ltd, Third Edition, 2003.

[2] T. Kolding, "Performance Aspects of WCDMA Systems with High Speed Downlink Packet Access (HSDPA)", IEEE, VTC 2002 Fall. Vol. 1. pp. 477-481.

[3] 3GPP TS 25.321 v5.5.0 (2003-06), "Medium Access Control (MAC) protocol specification", (Release 5).

[4] R.C Elliott and W.A. Kraymièn, "Scheduling Algorithms for the CDMA2000 Packet Data Evolution", IEEE, VTC 2002 Fall. Vol. 1, pp. 304-310.

[5] 3GPP TS 25.306 V5.8.0 (2004-03), "UE Radio Access capabilities" (Release 5).

[6] F. Kelly, "Charging and rate control for elastic traffic", Eur. Trans. Telecomm., Vol.8, pp. 33–37, Jan. 1997.

[7] 3GPP TS 25.214 V5.8.0 (2004-03),"Physical Layer Procedure (FDD)", (Release 5).

[8] TSG RAN WG1 Rel 5 adhoc (TSGR1(01)0725), "HSDPA scheduling and ARQ algorithm", Philips, Espoo, Finland, 26th-28th June 2001.

[9] J. H. Rhee, T. H. Kim and D. K. Kim, "A Wireless Fair Scheduling Algorithm for 1xEV-DO System", IEEE, VTC 2001 Fall, pp. 743-746.

[10] TR 101 112 V3.2.0 "Universal Mobile Telecommunications System (UMTS); Selection procedures for the choice of radio transmission technologies of the UMTS", UMTS 30.03 version 3.2.0,(1998-04), ETSI.

[11] H. Kim, K. Kim, Y. Han and J. Lee, "An Efficient Algorithm for Qos in Wireless Packet Data Transmission", Proc. Personal, Indoor and Mobile Radio Communications (PIMRC'2002), vol. 5, pp. 2244–2248, Sept. 2002.