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Full Length Research Paper

Supporting intelligent terminals by cross-layer adaptation over HSDPA network

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Next generation of mobile networks will support intelligent terminals having different categories and generating traffic with various qualities of service (QoS) requirements. To provide an end-to-end QoS, application terminals need to be adaptive and deal with network changing conditions. Knowledge has to be shared between network layers to obtain the highest possible adaptivity. In this context, we propose and evaluate an adaptive proportional fair algorithm with user multiplexing. Performance results show that the proposed algorithm is effective in reducing sojourn delays and in enhancing fairness among terminal users. The study investigates as well the impact of terminal categories with different levels of performance on our scheduling algorithm.

Key words: Adaptation, cross-layer, scheduling, quality of service, fairness.

INTRODUCTION

The "Anywhere and Anytime" concept in wireless networks requires efficient resource allocation techniques for quality of service (QoS) provisioning. Although the available bandwidth is much larger in Third Generation (3G) and beyond networks, it is still critical to efficiently use radio resources due to the fast growth of wireless subscribers, the increasing demand for new multimedia services and to wireless specific challenges.

Wireless channel characteristics affect all traditional OSI layers. Exploiting dependencies and interactions between layers increases performance in wireless networking. Providing knowledge about channel conditions to routing, transport and application layers allows designing more sophisticated allocation and optimization algorithms.

Cross-layer design takes into account wireless networks conditions to fulfill quality of service demands of applications by exploiting dependencies of protocol layers. To this end, the cross-layer design considers adaptive quality of service, adaptive resource allocation, rate of adaptation and network signaling. In fact, there is a growing consensus that adaptive quality of service enhances wireless network performance. In this context, application will be adapted to different levels of QoS based on underlying channel and network characteristics. On the other hand, networks will be adapted to application requirements.

Cross-layer network design has recently been applied to many applications and in many wireless networks (Jiang et al., 2005). It has been used to investigate performance of video over wireless (Yoo et al., 2004), sensor networks with energy constraints (Shuguang et al., 2005) and ad hoc networks (ElBatt and Ephremeide, 2004; Shakkottai et al., 2003).

In this paper, we investigate the cross-layer design in high speed downlink packet access (HSDPA) networks. One of the promising wireless access networks in next generation of networks is the HSDPA network. HSDPA applies scheduling and control for link adaptation closer to the air interface, more specifically at the Node B (Toskala et al., 2002; 3GPP-TR 25.848, 2001) (Figure 1).

The Node B estimates the channel quality of each active HSDPA user on the basis of ACK/NACK ratio, quality of service and HSDPA-specific user feedback. Scheduling and link adaptation are then conducted at a fast pace depending on the active scheduling algorithm and the user prioritization scheme.

In HSDPA, bandwidth adaptation is applied via the adaptive modulation and coding (AMC) that raises the overall system capacity. With AMC, modulation and coding

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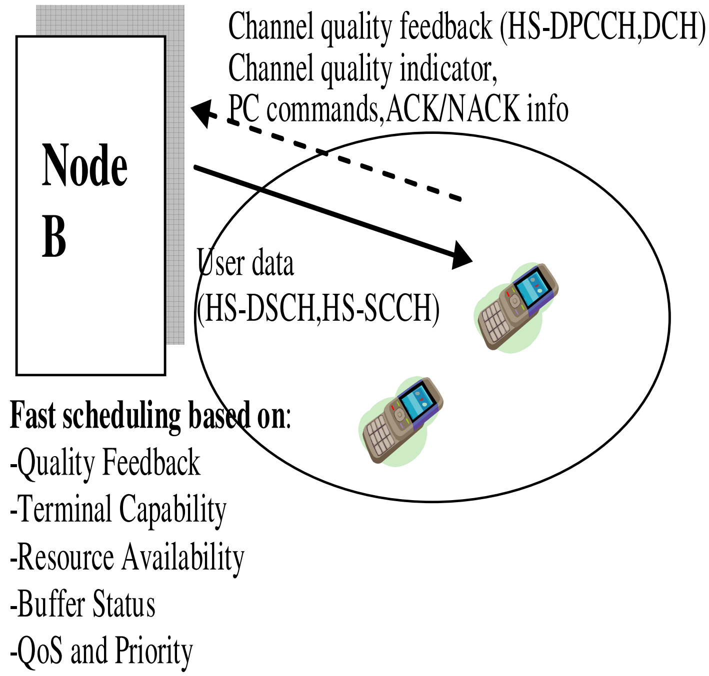

Figure 1. HSDPA architecture changes.

format are changed to match current received signal quality or channel conditions. Consequently, users close to the Node B are typically assigned higher order modulation with higher code rates than users far from the Node B.

To enable a dynamic range of the HSDPA link adaptation and to maintain a good spectral efficiency, a user may simultaneously use up to 15 codes in parallel.

HSDPA link is expected to support multimedia applications that generate traffic having diverse QoS requirements. Scheduling has attracted much attention of wireless data networks. In addition to throughput optimization, service delay and fairness are two additional important aspects that need to be considered in designing a good scheduling algorithm. Maximum Carrier-to-Interface Ratio (Max CIR) (Borst, 2003; Al-Manthari et al., 2007) tends to maximize the system throughput by serving, in every time transmission interval (TTI), the user with the best channel quality. It can be seen that this algorithm provides high system throughput since only those with high current supportable data rates get served. However, this algorithm has an obvious drawback in that it ignores those users with bad channel conditions, which may lead to starvation.

The unfairness issue in this algorithm has led to many proposals for scheduling algorithms that try to distribute resources evenly among users. One such proposal is Proportional Fairness (PF) (Jalali et al., 2000). The PF algorithm tries to increase the degree of fairness among users by selecting those with the largest relative channel quality. Relative channel quality is the instantaneous channel quality condition of the user divided by its current average throughput. Therefore, this algorithm considers not only those users with good channel conditions but also those with low average throughputs.

Bonald proposed the Score-Based (SB) algorithm in order to overcome this problem (Bonald, 2002). The SB algorithm computes the rank of each user's current channel condition among the past channel conditions observed over a window of size W. Then, it selects for transmission the user with the lowest rank. Therefore, this algorithm selects the user whose current channel condition is high relative to its own rate statistics instead Of selecting the one whose channel condition is high relative to its average throughput, as in the PF algorithm. SB is slightly more complex in terms of implementation than the PF algorithm. In addition, choosing the size of W might be another problem. Small values of W might not be appropriate to track the distribution of user channel conditions, while large values of W increase the time it takes to find the rank.

Users with Round-Robin (RR) algorithm are served in a cyclic order. This algorithm does not make use of information about the channel quality of users and therefore may offer lower system; it is fair in that it ensures that all users in the system get equal opportunity for transmission regardless of their channel quality conditions (Jose, 2003).

Another scheduling algorithm is Fair Throughput (FT). The goal of this algorithm is to ensure that an equal number of bits is received by each user in the system regardless of their channel quality conditions (Al-Manthari et al., 2007). This algorithm is fair in terms of the distribution of user throughput since each user gets the same amount of throughput regardless of its channel condition. Similar to RR, it suffers from lower system throughout than fast scheduling algorithms.

In this paper, we propose a new algorithm that provides network adaptability and fairness. We strongly believe that in wireless mobile networks, the major issue to be addressed is the high level of fluctuation in resource availability due mainly to mobility. There is a growing consensus that adaptive QoS presents a viable approach to this issue.

Applications need to be adaptive, renegotiate the service and deal with changing conditions by accepting different QoS levels imposed by the network. End systems should be network aware as they take the network status into account and adapt application accordingly. Network must provide application aware-services and handle the adaptive QoS required by these applications.

As a result, the end-to-end QoS provisioning is no longer the sole responsibility of the application. Now, this responsibility is shared between the application and the network. They should together choose the QoS level to deliver multimedia content to a mobile terminal in the most acceptable form given the available network resources.

In this paper, we propose to apply the cross-layer design in HSDPA by integrating a new scheduling algorithm “Adaptive Proportional Fair with User Multiplexing” (APFUM). This algorithm takes into account network conditions (available bandwidth, available codes, and bandwidth requests) and adapts the assigned bandwidth of accepted users. APFUM enhances Proportional Fair algorithm with the introduction of adaptability concept. Users which are rejected with PF due to the lack of codes can be scheduled with APFUM if adaptation is performed.

Our approach consists of the following design principles: 1) Support for soft QoS: We think that the network should provide at least a minimum level of QoS to the users’ applications. As for the applications, they should be able to adapt to the changing network conditions, and 2) Serve terminals fairly: Terminals with bad channel conditions should be considered for scheduling when their average throughput significantly decreases.

In a first step, we evaluate the performance of our algorithm and show that APFUM reduces queue access delay and enhances user fairness in the case of exhaustive traffic in a saturated cell. In a second step, we extend our study and investigate the impact of adaptability on different categories of terminals downloading Web traffic.

This paper is organized thus; terminal categories are presented in the next section; our proposed packet scheduler model and a simulation which validates the study are presented in the subsequent sections before concluding the paper.

TERMINAL CATEGORIES IN HSDPA

In HSDPA, the scheduler implemented in the Node B is affected by network conditions (bandwidth available, codes available, etc), terminal conditions (traffic backlogged in user queues, channel quality) as well as terminal categories. The 3GPP standards state that user terminals have various levels of performance. Twenty different categories of HSDPA capable mobile devices are defined (3GPP TS 25.306, 2004).

Terminal categories affect the coding and modulation scheme, peak data rates as well as the minimum inter-TTI interval. The minimum inter-TTI interval defines the distance from the beginning of a TTI to the beginning of the next TTI assigned to the terminal. For instance, terminals belonging to category 8 have a minimum inter-TTI of 1. This will give these category terminals the opportunity to send data each TTI.

Terminals with higher-performance receivers will receive higher data rates on average than lower-performing terminals. This is a fundamental change from the operation of the release 99 channels, which are power controlled to achieve a constant data rate.

The Node B varies the data rate sent to a terminal to achieve a constant Block Error Rate (BLER) of 10%. Thus, a higher-performance terminal will command a higher data rate than a lower performance terminal operating in the same conditions.

The relationship between terminal performance and user data throughput allows terminal manufacturers and chip set vendors to differentiate their products. Higher-performance terminals will deliver higher average data rates on any given HSDPA network and are therefore more desirable to both network operators and users.

It is noteworthy that terminals that profit from a high level of category performance could get a QoS level less than expected. This issue depends on fairness envisioned by the operator and on scheduling algorithms as shown by our simulation results. Next section is devoted to scheduling algorithms proposed in the literature for HSDPA.

HSDPA PACKET SCHEDULER MODEL

In this section, we briefly describe our packet scheduler model and how it operates in HSDPA. The packet scheduler for HSDPA is implemented at the MAC-hs layer of Node B. We assume that each user has one connection request. Thus, a Node B maintains one queue for every user, as shown in Figure 2. Upon call arrival, the radio link controller (RLC) layer receives traffic in the form of packets from higher layers, which are segmented into fixed-size protocol data units (PDUs) or transport blocks (TBs). These TBs are stored in the transmission queue of the corresponding user in a first-in first-out fashion. Subsequently, the TBs are transmitted to the

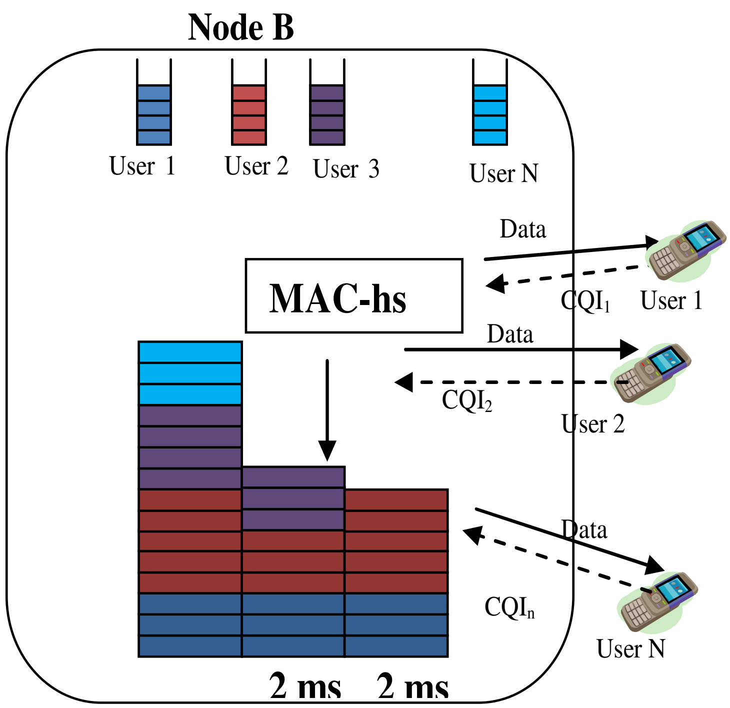

Figure 2. HSDPA packet scheduler model.

appropriate mobile user according to the adopted scheduling discipline. Next, we describe the implemented scheduling algorithm.

PF scheduling algorithm

As our proposed algorithm APFUM is based on proportional fairness algorithm, we start first by describing the PF strategy. Then, we exhibit APFUM algorithm.

Proportional fairness algorithm: To achieve a trade-off between fairness and efficiency, the PF strategy has been proposed. The user with the highest relative channel quality indicator (RCQI) is selected for transmission. Relative channel quality is the instantaneous data rate of the user depending on the channel quality condition divided by its current average throughput. The latter is evaluated through an exponential weighted low-pass filter.

The relative channel quality indicator and average throughput are computed as follows:

$$RCQIi(t) = \frac{Ri(t)}{Ti(t)}$$

$$Ti(t) = (1 - \frac{1}{Tc}) * Ti(t-1) + xi * \frac{1}{Tc} * R'i(t) \quad (1)$$

Where:

Ri = the instantaneous rate
Ti = the average throughput; get updated every time transmission interval for all active users.
R'i = the actual served rate when the user is scheduled.
xi = equal to 1 if user is scheduled and 0 otherwise.
Tc = a parameter varying between 800 and 1000.

According to this scheduling scheme, if the average throughput of a user is low, the RCQI could be high and it might be granted the right of transmission even if its current channel condition is not the best.

Algorithms outlined in the next section perform one-by-one scheduling, that is, only one user is allowed to transmit in any given TTI, even though HSDPA allows the code multiplexing of several users in the same TTI (3GPP-TR 25.848, 2001). The primary reasons why many authors have thus far only considered one-by-one packet scheduling schemes are probably their simplicity and their assumption of negligible self-interference. Golaup et al. (2005) and Cho and Hong (2004), argued that self-interference can be as high as multi-user interference in a multipath fading environment, and based on this argument the authors show that the throughput with code division multiplexing can be equal to that with Time Division Multiplexing in a fully loaded system without data rate limitation.

Consequently, we combine the PF algorithm with the multi-user code multiplexing; we consider that the Node B can schedule at most four users given that the total number of codes does not exceed 15.

AFPUM scheduling algorithm

With our proposed APFUM algorithm, HSDPA operation goes through the following steps:

a) The scheduler in the Node B evaluates for different users the relative channel quality indicator.
b) The Node B determines then a list of four elected users (called elected_list), at most, to be served in a particular TTI according to the Proportional Fairness with user multiplexing.
c) The Node B then proceeds to check the eligibility of each user; It checks whether the total number of assigned codes does not exceed 15 and whether the minimum inter-TTI interval is respected. This parameter depends upon the terminal category (3GPP TS 25.306, 2004). Rejected users due to lack of codes are inserted in a list, rejected_list.
d) Once a user is selected for transmission, the Node B identifies the necessary parameters (3GPP-TS 25.214, 2003): modulation type, number of codes (nb_assigned_ codes), transport block size (TBS_assigned). These parameters depend upon the channel conditions of the terminal and its capability limitations. Users with good channel conditions will enjoy potentially higher supportable data rates by using higher modulation and coding rates, whereas users with bad channel conditions will experience lower data rates.
e) If the number of scheduled users is less than 4 and the total assigned codes less than 15, then the Node B tries to adapt bandwidth of elected users in order to schedule other users rejected for lack of codes. Adaptation proceeds as follows:

i) The Node B computes the number of codes requested by the users in the rejected_list. It then tries to adapt the assigned TBS of the users in the elected_list, starting from the user with the smallest value of RCQI.
ii) Adaptation consists to find the greatest value of TBS, adapted_TBS, less than the TBS_assigned with number of codes, adapted_codes, and strictly less than actual value of nb_assigned_codes. Consequently, adaptation permits to reduce the TBS of elected users and to release codes in order to schedule rejected users. This process reduces the delay for access to the HSDPA link.
iii) When the maximum number of permitted users per TTI is reached or when the available number of remaining codes cannot be assigned to any user, upgrade operation is performed. If the available number of remaining codes is not zero, then users with highest value of RCQI are upgraded and allocated the initial value of TBS.

f) The terminal sends in the uplink direction an ACK/NACK indicator, depending on the outcome of the CRC check conducted on the HS-DSCH data as well as the monitored channel quality indicator (CQI). The latter is computed as stated in next section.

CQI reporting

On every 7.5 slots, each user informs the Node B of its channel quality condition by sending CQI on the high-speed Dedicated Physical Control Channel (HS-DPCCH). Based on 3GPP standard (3GPP-TS 25.214, 2003), with a BLER equal to 0.1, CQI is related to signal-to-interference ratio (SIR) according to the following relations:

$$CQI = \begin{pmatrix} 0 & if\ SIR \le -16dB \\ E[\frac{SIR}{1.02} + 16.62] & if\ -16dB < SIR < 14dB \\ 30 & if\ 14dB \le SIR \end{pmatrix} \quad (2)$$

The SIR of a mobile, located at a distance r from the Node B, is computed as follows:

$$SIR(r) = \frac{P_{HSDSCH}}{I(r).q(r)} SF \quad (3)$$

Where;

PHSDSCH is the power allocated by the Node B on HS-DSCH channel;
SF is the spreading factor equal to 16 in HSDPA.

The interference, I(r), is computed as follows:

$$I\ (r) = \alpha I_{\text{int}\,ra}(r) + I_{\text{int}\,er}(r) + N_0 \quad (4)$$

Where;

Iintra (r) is the intra-cell interference taking into account the orthogonal factor α. Iinter(r) is the inter-cell interference.
N_0 is the background noise.

The path loss between Node B and the receiver, q(r), is defined as

$$q(r) = r^{\gamma} 10^{\frac{\xi}{10}} \quad (5)$$

Where;
ξ is a Gaussian variable with zero mean and standard deviation σ represented for shadowing effects. We assume that packets are transmitted without errors and neglect multipath fading.
γ is the path loss factor.

PERFORMANCE EVALUATION

In order to compare the different scenarios in HSDPA, the

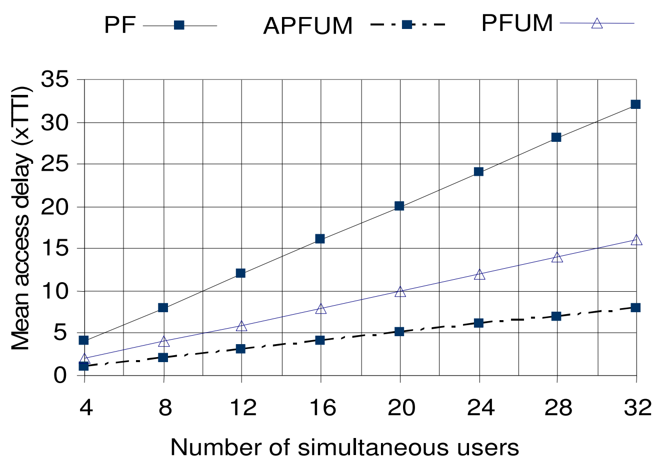

Figure 3. Mean access delay for PF, PFUM and APFUM.

the network depicted in Figure 2 was simulated with C++. The cell radius is 500 m. The Node B is located at the cell center. Users are connected to the Node B on the downlink by an HS-PDSCH, which is the actual physical channel for HSDPA and on the uplink by a HS-DPCCH channel, which is used to send the users' current estimates of their channel conditions. The simulation time step is one TTI, which is 2 ms, and the simulation time is 50000 TTI.

Users of category 8 and of minimum inter-TTI interval 1 are located on rings with radius 10, 50, 150 and 500 m. The orthogonal factor and the path loss factor are assumed to be 0.1 and 2.7 respectively. The maximum number of codes and the maximum number of users per TTI are 15 and 4. For PF scheduling, the parameter Tc is taken to be equal to 1000.

Performance parameters are depicted in the following figures. The mean access delay refers to the delay experienced by users before being scheduled. The average throughput is computed according to equation 1. Cell effective throughput is the mean useful throughput per cell. The adaptation probability refers to the rate of adaptation. The rejection probability is the probability of not scheduling users when the number of parallel multiplexed users is less than four.

The performance evaluation of our algorithm is elaborated in case of users generating exhaustive traffic (presented in the sub section below) and in case of users receiving web traffic (in subsequent subsection).

Performance evaluation of APFUM with exhaustive traffic

In a first step, we assume that users are receiving exhaustive traffic in a saturated cell. We consider the worst case in which there is always backlogged data ready to be transmitted due to the exhaustive nature of application traffic.

We conducted simulation runs in order to compare three schemes: the Proportional Fairness Scheduling PF, the PF with user multiplexing PFUM and our adaptive PF with user multiplexing APFUM.

Figure 3 illustrates the mean access delay obtained with the three schemes. The mean access delay retrieved with the Proportional Fair and with the PFUM is higher than that with our proposed scheme for all active users. In fact, the APFUM permits to adapt bandwidth of elected users in order to schedule users which were rejected with PF and with PFUM; This leads to enhance the fairness and to lower the mean access delay with APFUM.

The rejection and adaptation probability are depicted in Figure 4. It can be seen that the APFUM keeps the probability of rejection Prej equal to zero and less than Prej with PF. With the PF algorithm, the elected list will contain strictly one user, leading to a rejection probability of 75%. The rejection probability is 0.5 with the PFUM.
In fact, given the users' locations and the category 8, the CQI reported by terminals will be 26, 22 or 10. These CQI values correspond to either 10 or 3 codes. Thus, only two users can be scheduled per TTI. With our scheme, we were concerned about scheduling the maximum number of users per TTI; elected users were adapted in order to satisfy other users. This result is obtained at the expense of generating the adaptation probability Pada.

Figure 5 depicts the cell average throughput for each of the three schemes. As expected, APFUM achieves better throughput than the PF. However the PFUM provides higher throughput than with APFUM. In fact APFUM accepts more users at the expense of increasing intra-cell interference. Thus, the throughput will be reduced when compared to the PFUM. As a conclusion, the adaptive proportional fair scheduling enhances fairness in serving users and reduces the mean access delays. This achievement is done at the expense of reducing the average cell throughput. One can see that the throughput achieved with APFUM is better than with PF. Thus there is a trade-off between fairness and cell throughput.

Performance evaluation of APFUM with web traffic

Since the bursty-ness is a characteristic feature of transmission in wireless network, we consider at this stage that users are generating Web traffic. A typical WWW browsing session consists of a sequence of packet calls (ETSI TR 101 112). The session arrival process is modeled as a Poisson process. The number of packet calls per session (Npc), the reading time (Dpc), the number of packets within a packet call (Nd), the time interval between consecutive packets (Dd) are geometrically distributed random variables respectively with means µNpc, µDpc, µNd and µDd. The size of a

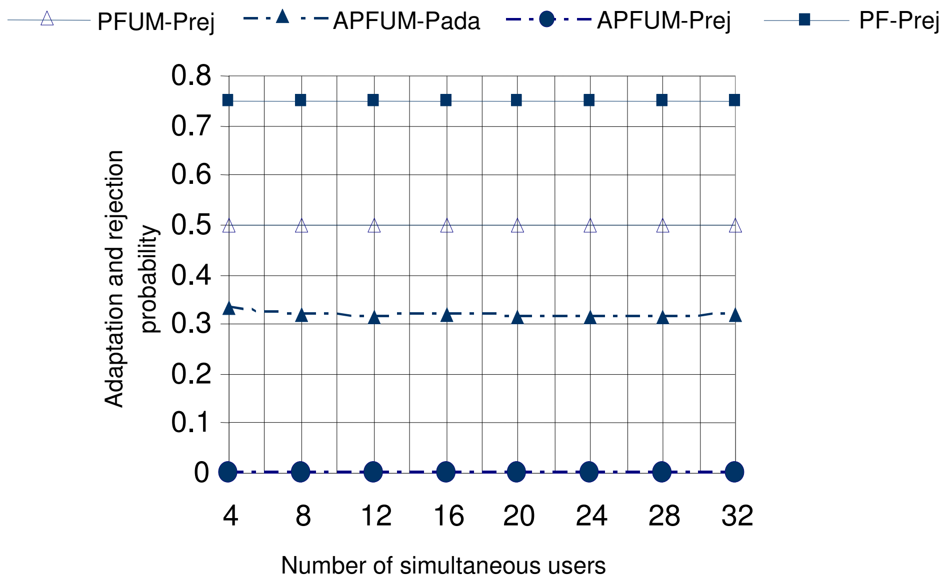

Figure 4. Rejection and adaptation probability for PF and APFUM.

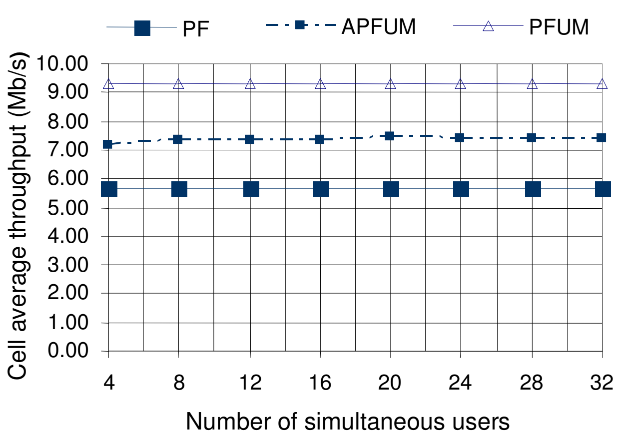

Figure 5. Mean effective cell throughput for PF and APFUM.

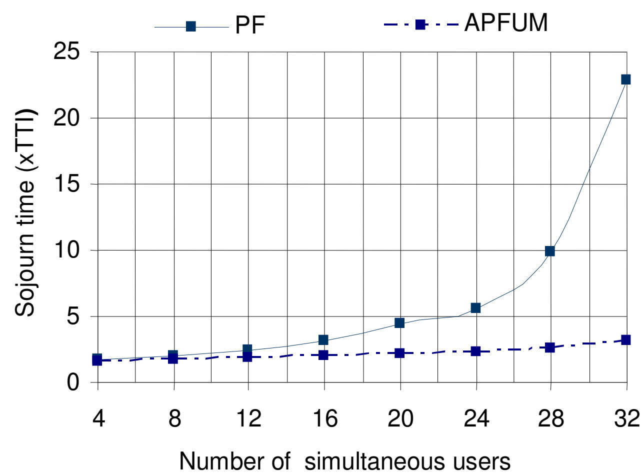

Figure 6. Average sojourn delay.

datagram Sd is a random variable with a Pareto distribution with cut-off. In our simulation we consider a WWW service having a rate of 144Kb/s, µNpc of 5, µDpc of 412, µNd of 25 and µDd of 0.016.

The performance parameters relative to Web traffic are depicted in the following figures. Figure 6 illustrates the mean sojourn delay obtained with the PF and APUM schemes. The mean sojourn time is the average time elapsed before sending a packet stored in a waiting queue. The mean sojourn delay retrieved with the PF algorithm is higher than that with our proposed scheme for all active users. In fact, the APFUM permits to adapt bandwidth of elected users in order to schedule users which were rejected with PF. This leads to enhance the fairness and to lower the mean sojourn delay with APFUM.

This is achieved at the expense of a slight reduction of the effective throughput as shown in Figure 7. Adapting some calls lead to reducing their assigned TBS and thus reducing the effective cell throughput.

The APFUM rejection and adaptation probability are depicted in Figure 8 categories.

The rejection probability increases as the number of simultaneous users increases: Number of packets waiting in queues increase and the probability to reject packets due to lack of codes increases. Adaptation occurs more frequently with load increase as it can be seen. At high loads, the APFUM keeps the probability of adaptation bounded by the maximum 0.15. This is related to the users' locations, terminals categories, reported CQI and maximum number of simultaneous users.

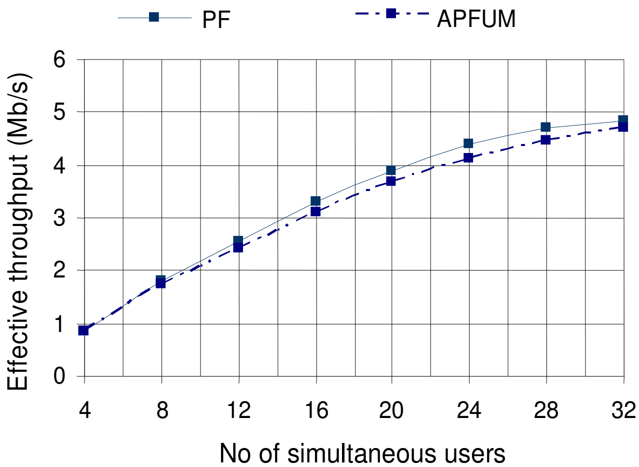

Figure 7. Mean effective cell throughput.

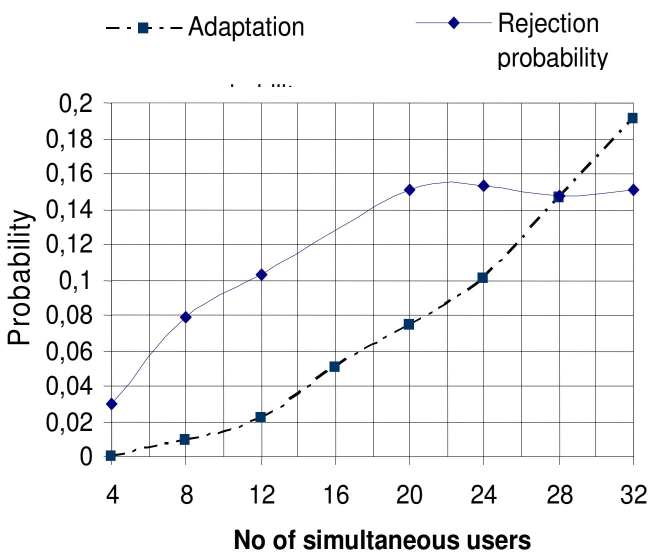

Figure 8. Adaptation and rejection probability.

As a conclusion, the adaptive proportional fair scheduling reduces the mean sojourn delays. This achievement is done at the expense of slightly reducing the average cell throughput.

Impact of APFUM on terminal categories

Other simulation runs were conducted with terminals, located at distance R, having different sets of categories (Table 1). The objective of these runs is to investigate the impact of user categories on our adaptive algorithm. Categories 8 and 18 benefit from a minimum inter-TTI of 1, whereas category 11 has a minimum inter-TTI of 2. Each category permits to have a different TBS allocation.

The highest effective throughput is achieved with Set 3 (Figure 9). This result is expected since this set includes terminals of high performance. This will have an impact on the overall effective cell throughput.

Contrarily to Set 2, Set 1 assumes that terminals of low performance category (11) are located near the Node B, and users of high performance category (18) are far from the Node B. Consequently, terminals of category 11 will not benefit from the Node B proximity and will be allocated a maximum TBS equal to 3319 (3GPP TS 25.306, 2004). This issue will have a bad impact on the cell effective throughput especially that category 11 has a minimum inter-TTI of 2. On the other hand, category 18 terminals will suffer from interference and path loss and thus will be degraded. The opposite situation occurs with Set 2 which will benefit from a higher throughput than that of Set 1.

The lowest effective throughput is obtained with Set 4 which assumes that terminals located at different cell positions have low performance. Thus, the cell effective throughput will be affected by the terminal categories.

Figure 10 illustrates the average sojourn time achieved by each of the terminal categories sets. At low loads, the mean sojourn time obtained with Set 4 is the highest since terminals of this set are allocated the lowest TBS with the highest minimum inter-TTI. Thus data packets will be fragmented into small TBS and consequently will be penalized and will get high waiting time.

When the cell load increases, average sojourn time of all sets increase. With Set 2, terminals of category 18 located at the Node B proximity will be allocated large TBS and high number of codes. Even if these terminals try to adapt their TBS, they will get high number of codes and will not easily free codes to other terminals categories. This explains why the adaptation probability with Set 2 is less than that of Set 1. Consequently, the average sojourn time with Set 2 is higher than that of Set 1. Besides, terminals of low performance category with Set 2 will be penalized by the path loss and will be allocated low values of TBS. This will drastically higher the overall sojourn time.

With Set 1, the proximity of low-performing terminals to the Node B will compensate their low level of performance. Furthermore, adaptation of high-performing terminals (which will get reduced TBS due to path loss and interference) will free codes and help to schedule a higher number of low-performing users per TTI. This will help to lower the packet waiting time.

When comparing Set 4 and Set 3, we find that the sojourn time of Set 3 is less than Set 4. The fact that terminals with Set 4 wait two TTIs before sending their TBs penalize the average cell sojourn time. Besides, packets with Set 4 are fragmented into TBs of small size. Thus the number of TBs waiting in user queues will increase. This will contribute to increasing packet waiting time. However when the cell load increases, adaptation occurs less frequently with Set 3. This will higher the sojourn time of Set 3 terminals.

Table 1. Terminal categories.

Set	Category at R = 40	Category at R = 140	Category at R = 170	Category at R = 400
Set 1	11	11	18	18
Set 2	18	18	11	11
Set 3	18	18	18	18
Set 4	11	11	11	11

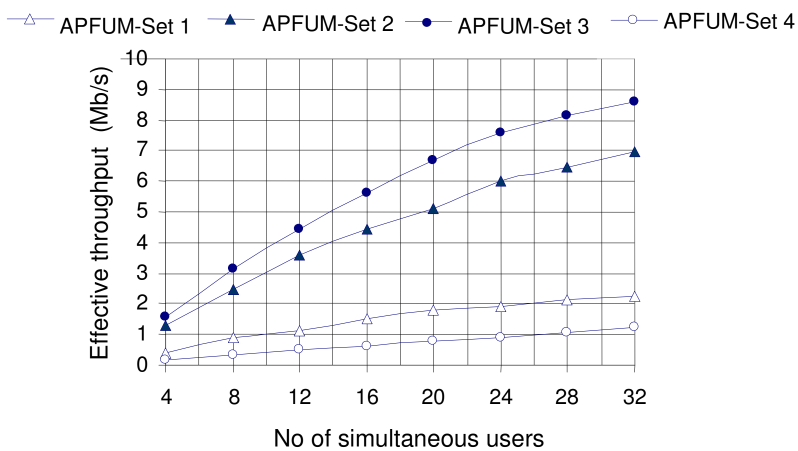

Figure 9. Mean effective throughput with different categories.

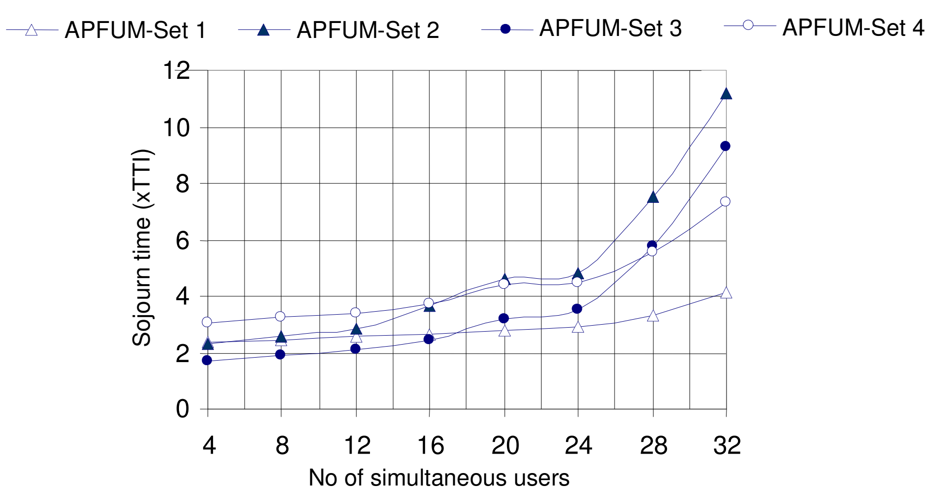

Figure 10. Average sojourn time with different categories.

Conclusion

To fully optimize wireless networks, both the challenges from the physical medium and the QoS-demands from the applications have to be taken into account. Rate and coding at the physical layer can be adapted to meet applications requirements given current channel and network conditions.

In this paper, we investigated the cross-layer in HSDPA network and proposed a scheduling algorithm APFUM that enhances QoS performance in HSDPA networks. We proceeded into two steps. In a first step, we considered an exhaustive traffic and compared the performance of APFUM with PF and PFUM algorithms. Simulations runs were conducted for different sets of active users. We showed that APFUM scheme lowers the mean access delay and the rejection probability and increases the average throughput when compared to the PF scheme.

APFUM succeeds to enhance the system fairness by scheduling the maximum number of users at the expense of lowering the cell throughput when compared to the PFUM scheme. This leads us to conclude that APFUM outperforms PF scheme, and that there is a com-promise between fairness and throughput achievement.

In a second step, we investigated the impact of APFUM on different terminal categories. Highest throughput is obtained when all terminals are high-performing. The lowest sojourn time is achieved when low-performing terminals are located at the Node B proximity and when high-performing terminals are located far from the Node B. In fact, with this set of terminals, adaptation of bandwidth occurs frequently and thus good performance is achieved. Moreover, the proximity of low-performing terminals to the Node B will compensate their low level of performance. Finally, we can conclude that categories repartition over the HSDPA cell and terminals position affect adaptation process and has a significant impact on the cell performance parameters.

REFERENCES

3GPP-TR 25.848 (2001). Physical Layer Aspects of UTRA High Speed Downlink Packet Access. Release 4.

3GPP-TS 25.214 (2003). Physical Layers Procedures (FDD). Release 8.

3GPP TS 25.306 (2004). UE Radio Access capabilities. Release 8.

Al-Manthari B, Hassanein H, Nasser N (2007). Packet Scheduling in 3.5G High-Speed Downlink Packet Access Networks: Breadth Depth IEEE Network. 1: 41-46.

ElBatt T, Ephremeide A (2004). Joint Scheduling and Power Control for Wireless Ad-Hoc Networks. IEEE. Trans. Wireless Commun. 74-85.

Bonald T (2002). A Score-Based Opportunistic Scheduler for Fading Radio Channels. Proc. Euro. Wireless. pp. 2244-2248.

Borst S (2003). User-Level Performance of Channel-Aware Scheduling Algorithms in Wireless Data Networks. Proc. IEEE INFOCOM.1:321–31.

Cho J, Hong D (2004).Throughput Comparison of TDM and CDM for Downlink Packet Transmission in CDMA Systems with a Limited Data Rate Set. IEEE Trans. Wireless Commun., pp. 600-611.

ETSI TR 101 112. Universal Mobile Telecommunications System (UMTS) ; Selection procedures for the choice of radio transmission technologies of the UMTS.

Golaup A , Holland O, Aghvami A (2005). A Packet Scheduling Algorithm Supporting Multimedia Traffic over the HSDPA Link Based on Early Delay Notification. Proc. Int Conf. Multimedia Services Access Networks (MSAN '05), pp. 78-82.

Jalali A, Padovani R, Pankaj R (2000). Data Throughput of CDMA-HDR a High Efficiency-High Date Rate Personal Communication Wireless System. Proc. IEEE VTC. pp. 1854-1858.

Jiang H, Zhuang W, Shen W (2005). Cross-Layer Design for Resource Allocation in 3G Wireless Networks and Beyond. IEEE Commun. Mag.

Jose P(2003). Packet Scheduling and Quality of Service in HSDPA. Ph.D. dissertation.

Shakkottai S, Rappaport TS, Karlsson PC (2003). Cross-layer Design for Wireless Networks. IEEE Commun. Mag., 41: 74-80.

Shuguang C, Madan R, Goldsmith A, Lall S. Energy-delay tradeoffs for data collection in TDMA-based sensor networks (2005). Proc. IEEE Int. Conf. Commun. (ICC). 5:3278 – 3284.

Toskala A, Holma H, Kolding T, Frederiksen F , Mogensen P (2002). WCDMA for UMTS-Radio Access For Third Generation Mobile Communications. John Wiley and Sons.

Yoo T, Setton E, Xiaoqing Z, Goldsmith A, Girod B (2004). Cross-layer Design for Video Streaming over Wireless `Ad hoc Networks. Proc. IEEE Int. Workshop Multimedia Signal Process.. pp.99-102.