ADAPTIVE SCHEDULING WITH QoS CONSTRAINTS IN HSDPA SYSTEM

A Thesis

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By

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List of Abbreviations

3GPP	Third Generation Partnership Project
AMC	Adaptive Modulation and Coding
APF	Adaptive Proportional Fairness
ARQ	Automatic Repeat request
BLER	Block Error Rate
CC	Chase Combing
CQI	Channel Quality Indicator
DL	Downlink
DPCH	Dedicated Physical Channel
DRC	Data Rate Control
DS-CDMA	Direct Sequence CDMA
H-ARQ	Hybrid ARQ
HSDPA	High-Speed Downlink Packet Access
HS-DPCCH	High-Speed Dedicated Physical Control Channel
HS-DSCH	High-Speed Dedicated Physical Control Channel
HS-PDSCH	High-Speed Physical Downlink Shared Channel
HS-SCCH	High-Speed Shared Control Channel
IR	Incremental Redundancy
LDD	Low Delay Data bearer service
LCD	Long Constrained Delay data bearer service
MAC	Medium Access Control

List of Abbreviations

PF	Proportional Fair
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
RB	Relatively Best
RF	Radio Frequency
RLC	Radio Link Control
RNC	Radio Network Controller
RR	Round Robin
RRM	Radio Resource Management
SAS	Soft and Safe
SAW	Stop And Wait
SCDMA	Scheduled CDMA
SDMA	Spatial Division Multiple Access
SIR	Signal-to-Interference Ratio
SNR	Signal-to-Noise Ratio
SPC	Selective Power Control
TCP	Transmission Control Protocol
TDMA	Time Division Multiple Access
TPC	Transmit Power Control
TTI	Transmission Time Interval

List of Abbreviations

UDD	Unconstrained Delay Data bearer service
UE	User Equipment
UL	Uplink
UTRA-FDD	Frequency Division Duplex
UMTS	Universal Mobile Telecommunications System
WCDMA	Wideband CDMA

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Abstract

With an increased request for wireless data services, methods for managing the scarce radio resources become imperative. Notably for applications characterized by heterogeneous quality of service (QoS) requirements, radio resource allocation becomes an extensive task. Due to the necessity of sharing the radio spectrum, mutual interference limits system capacity. Transmitter power control is a well-known method for upholding required signal quality and reducing the energy consumption. In addition, Proportional Fairness (PF) is a well known method for upholding fairness among users and throughput trade off between users.

In this thesis, transmission schemes based on adaptive Proportional Fairness are developed and analyzed for HSDPA systems. The schemes are designed to provide insurance for various QoS. The proposed research aims to provide the technical framework which will enable HSDPA systems to provide the assured network service while being in communication range with each other.

The Index Terms—HSDPA System, Proportional Fairness, QoS

Chapter 1

Introduction

1.1 Overview of Cellular Communication Systems

In recent Years tremendous success and sky-rocketing growth of wireless personal communications have necessitated careful management of common radio resources. As the demand for wireless speech and wireless data services is no doubt, continue to grow, developing transmission schemes that can efficiently utilize the available radio resources is of major importance. In comparison to a wire-line channel, the radio channel is time variant and transmissions should preferably be adapted to the channel quality as well as the service requirements. With user mobility, this imposes several challenges in providing tetherless communications. These challenges are mainly due to the scarcity of available frequency spectrum. With this some form of resource sharing must be considered. In practice, all sharing methods introduce some form of interference, impairing the ability to communicate. To ward off a complete breakdown of the system, interference control is a necessity.

Current cellular networks provide satisfactory voice services at a reasonable cost. However, this success story does not easily extend to data services, such as wireless web surfing. Whether wireless channels can provide high data rate service and whether such a service is affordable are two major concerns. To provide high data rate service, wide-band transmission is necessary. In a wide-band single-carrier system, the problems of frequency-selective-fading and inter-symbol-interference are faced. Furthermore, to make high data rate service a fordable, a higher spectrum efficiency (compared with current systems) has to be achieved.

The introduction of a new technology such as HSDPA in the Release 5 of the 3GPP [2] specifications arises the question about its performance capabilities: is HSDPA a radio access technology that can facilitate the evolution of the wireless communication market by enabling the support of high data rate services and by increasing the system capacity? The system level performance of HSDPA and the network capacity promotes that this technology can provide cellular systems have already been addressed in the literature [3], [4]. However, those studies have only considered best effort traffic without taking into consideration the concept of Quality of Service (QoS).

In order to be able to facilitate the evolution of the wireless communication market, the HSDPA technology should be able to increase the system capacity over other computing technologies (e.g. WCDMA) not only for best effort traffic. The HSDPA technology should also be able to meet the QoS demands of third generation mobile data services. Firstly, data services impose minimum QoS requirements on the communication (e.g. delay jitter, guaranteed bit rate, etc) that must be satisfied in order to ensure an acceptable degree of quality to the end user. Secondly, the provision of the service with a degree of quality beyond the minimum acceptable level imposes on the communication more stringent demands (e.g. a bit rates higher than the minimum guaranteed), which typically can be satisfied at the expenses of a reduced network throughput.

The concept of QoS is a key factor in the HSDPA technology evaluation because there exists a fundamental trade-off between the QoS provided to the end user and the achievable network throughput. The main objective of this thesis is to investigate the trade-off between the QoS provided to the end user and the achievable network throughput by this technology for different traffic types.

The packet scheduling for HSDPA is located in the medium access layer, MAC-hs. The MAC-hs is located in the Node-B, which means that packet scheduling decisions are executed almost instantaneously. In addition to this, the TTI length has been shortened to 2ms. Typically, the Round-Robin is the first time scheduler considered packet scheduling strategy, where users are served in sequential order so they all get the same average allocation time [1].

However, the high scheduling rate combined with the large AMC dynamic range available within the HSDPA concept, also facilitates advanced scheduling methods where channel allocation is conducted according to the current radio conditions. A popular packet scheduling method is the Proportional Fair (PF) packet scheduler. With this type of scheduler, the serve order is determined by the highest instantaneous relative channel quality; i.e. it attempts to track the fast fading behavior of the radio channel. Since the selection is based on relative conditions, every user still gets approximately the same amount of allocation time but the raise in system capacity easily exceeds 50%.

As wireless data services require different quality of service (QoS) as compared to pure speech services, a larger freedom in allocating the system resources arise. Nevertheless, the problem of supporting the respective QoS while maintaining low energy consumption and high capacity becomes complex and resource allocation in this new context needs to be readdressed. PF provides a reasonable trade off between throughput and Users fairness, in term of channel using, and proportional throughput depending on the user channel condition is achieved. But PF does not grant the QoS among users.

The resource allocation problem arises in the context of wireless communication. Any system design, and in particular, wireless system design is a challenging task, owing to the severe constraints on resources like bandwidth, power, fairness and time-slots. The temporal and spatial variations induced by physical phenomena and user mobility only makes matters worse. Given a certain set of resources, the onus is on system designers to design efficient allocation schemes to ensure maximal user happiness. This leads to the philosophy of QoS fairness system design, which has been an active area of interest to the wireless community in recent years. In this thesis, the performance of Adaptive Proportional Fair (APF) packet scheduling algorithm over Rayleigh fading channels while assuring the QoS is investigated. An example is the universal mobile telecommunications system (UMTS) standard, where a layered architecture for QoS support is outlined. The QoS support structure relies on layered bearer services, which are involved to compose the end-to-end bearer service. To each bearer service, QoS attributes are defined.

These attributes serve to map the end-to-end QoS requirements to appropriate requirements for each bearer service. The attributes typically describe requirements on bit rates, delays and priorities. To classify the services, the traffic in UMTS is divided into the classes; conversational, streaming, interactive and background [1]. Problems in multirate schemes are both related to the bearer mapping, i.e., how to choose rates and schedule the transmission for obtaining the QoS and more physical layer type of issues, such as; how to map the bit rates into the given bandwidth and how to inform the receiver about the characteristics of the signal. For a service where the delay requirement is loose, longer delay allows for longer interleaving, more retransmissions and therefore lowers the signal-to-interference ratio (SIR) that, in turn, could increase capacity.

Chapter 2

QoS In HSDPA

2.1 QoS Requirements

This chapter gives a brief overview of the current quality of Serves (QoS) architecture used. Guaranteeing quality of service in any QoS architecture is based on using an appropriate scheduling algorithm. It motivates the development of schedulers for QoS/link sharing and gives “the big picture” in which such an algorithm is used. Only the basic ideas of each concept will be outlined.

The term Quality of Service (QoS) "experienced" by the end user of a mobile communication system refers to the collective effect of service performances to the end user. This determines the degree of satisfaction of this end user of the service. The demands imposed by a certain service on the system to provide a satisfactory user QoS completely depend on the intrinsic characteristics of the service itself as illustrated in Figure 2.1.

Typically, the service response time is one the key factor with the most influence on the user’s perceived QoS for Non Real Time services. The service response time of such service is directly determined by the user’s data rate. The most crucial constraints of Real Time traffic are the packet transfer delay (average and/or variance) and the user’s data rate. Fundamentally, the more demanding these two constraints are, the lower spectral efficiency can be achieved by the mobile communication system.

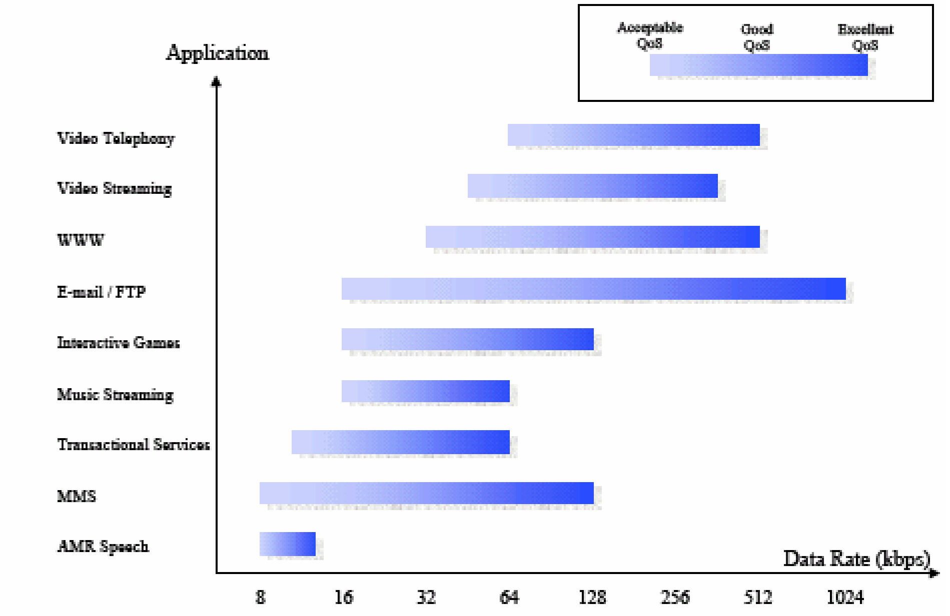


Figure 2.1: Data Rate Requirements of 3G Applications.

The combination of the level of quality provided by all parts of the system will determine the overall end-to-end QoS, and the customer acceptance of B3G will solely depend on this factor. Therefore it is important for the operator to identify the following terminology: user service requirements, the user perceived QoS, offered QoS, and actual QoS [5]:

- Customer service requirements are based on the notion that a customer can have the availability of the service at any time, and at any place.
- The user perceived QoS describes if the user is happy with the QoS of services that are being offered, and this is very much dependent on the user perception or expectations of QoS.
- Offered QoS is expressed in the operator's language, and indicated on a per service basis. It should match the user’s expectations of QoS, and should be mandated on a

service level agreement.

• Delivered QoS measures the effectiveness of the service provided by the B3G operator. Suitable QoS performance indicators are required to monitor this parameter. The QoS parameter has meaning at all levels, and an accurate mapping must exist between layers so that the correct QoS is being delivered to the end user and the system resources are being used efficiently.

2.2 Wireless QoS Packet Scheduling

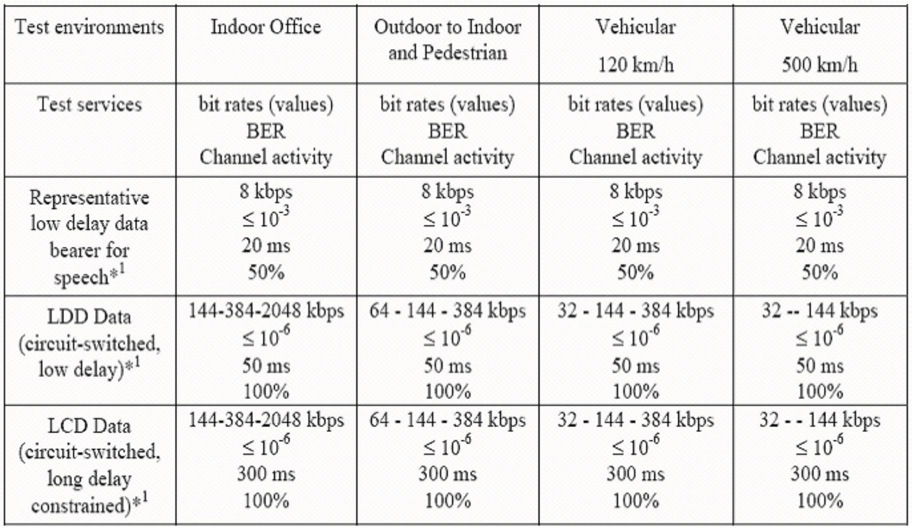
Even if PF scheduler efficiently provides data services by exploiting multi-user diversity, it cannot guarantee real-time users QoS such as the delay bound or the minimum throughput. In order to solve this problem, the control parameters inspired from EXP rules are proposed on [6, 7], which introduce a supplementary adaptation term into PFS. The improvement of the delay performance is maximized when the adaptation control parameter term is added and the Target delay constrains as depicted in chapter 5.

The issue of packet scheduling within a wireless network is more complicated than in the wireline case since the wireless channel is influenced by additional factors which a wireless scheduling mechanism needs to take into account [8]:

1. Location dependent errors (e.g. because of multipath propagation)
2. Higher probability of transmission errors/error bursts (cause by noise and interference on the radio channel)
3. Dynamically increasing /decreasing number of stations

Most wireless scheduling algorithms assume that they schedule the downstream traffic in access point and have full control over the wireless medium. Perfect knowledge on the states the uplink queues of the mobile stations and the channel conditions are also a precondition for the majority of the presented algorithms.

Most quality parameters which are dealt with in other wireless Recommendations [9] are the minimum requirements which must be met and are not to be treated in the evaluation process. Radio Transmission Technology will be evaluated on the impact of transmission processing delay on the end-to-end delay, expected average bit error ratio (BER) under the stated test conditions, on their maximum supportable bit rate under specified conditions and their overall ability to minimize circuit disruption. In addition, they will be evaluated on their ability to sustain quality under certain extreme conditions such as system overload, hardware failures, interference, etc. The Table I depicts List of test data rates for evaluation purposes.



Test environments	Indoor Office	Outdoor to Indoor and Pedestrian	Vehicular 120 km/h	Vehicular 500 km/h
Test services	bit rates (values) BER Channel activity	bit rates (values) BER Channel activity	bit rates (values) BER Channel activity	bit rates (values) BER Channel activity
Representative low delay data bearer for speech*[1]	8 kbps $\leq 10^{-3}$ 20 ms 50%	8 kbps $\leq 10^{-3}$ 20 ms 50%	8 kbps $\leq 10^{-3}$ 20 ms 50%	8 kbps $\leq 10^{-3}$ 20 ms 50%
LDD Data (circuit-switched, low delay)*[1]	144-384-2048 kbps $\leq 10^{-6}$ 50 ms 100%	64 - 144 - 384 kbps $\leq 10^{-6}$ 50 ms 100%	32 - 144 - 384 kbps $\leq 10^{-6}$ 50 ms 100%	32 -- 144 kbps $\leq 10^{-6}$ 50 ms 100%
LCD Data (circuit-switched, long delay constrained)*[1]	144-384-2048 kbps $\leq 10^{-6}$ 300 ms 100%	64 - 144 - 384 kbps $\leq 10^{-6}$ 300 ms 100%	32 - 144 - 384 kbps $\leq 10^{-6}$ 300 ms 100%	32 - - 144 kbps $\leq 10^{-6}$ 300 ms 100%

TableI List of test data rates for evaluation purposes.

According to [10], “an application is defined as a task that requires communication of one or more information streams, between two or more parties that are geographically separated”, while a set of applications with similar characteristics is classified as a service. There are many ways to categorize services depending on the classification criterion, such as the directionality (unidirectional or bi-directional), symmetry of communications (symmetric or asymmetric), etc [11]. The 3GPP created a traffic classification attending to their general QoS requirements. The QoS refers to the collective effect of service performances that determine the degree of satisfaction of the end-user of the service. The four classes are [12]

- *Conversational class.*
- *Streaming class.*
- *Interactive class.*
- *Background class.*

The main distinguishing factor between the classes is how delay sensitive the traffic is. Conversational class is meant for traffic which is very delay sensitive, whereas the background class is the most insensitive to delay. The traffic corresponding to the conversational class refers to real time conversation where the time relation between information entities of the stream must be preserved. The conversational pattern of this type of communication requires a low end-to-end delay to satisfy the stringent requirements of human perception. A service example is telephone speech, voice over IP or video conferencing. According to [13], HSDPA focuses on streaming, interactive, and background traffic classes but not on conversational traffic.

2.3 Real And Non Real Time Traffic

Traditionally, when classifying services attending to their delivery requirements, they have been primarily divided between Real and Non Real Time services. Usually, Real Time (RT) services have been considered to impose a strict delay requirement on the end-to-end communication. With this delay the network nodes involved in the Real Time connection are to transfer the packets of the RT flow within a maximum tolerable delay. Due to these severe delay constraints, the error correction possibilities of this type of communication are very limited, and, while link level retransmissions have shown difficultly in being feasible for Real Time services, the end-to-end recovery mechanisms have been completely dismissed. This implies that the Real Time flow delivered at the terminating entity is subject to transmission errors, thus, the final application must be able to cope with the errors.

On the other hand, Non Real Time traffic has been commonly considered as error sensitive, though with less demanding delay constraints than RT traffic. These characteristics of NRT traffic allow for link and end-to-end level recovery mechanisms that enable error free delivery of the payload. Another typical difference between RT and NRT services is that the receiving end of RT applications usually consumes the information before the end of the data transaction.

The 3GPP QoS specification [14] also follows the RT/NRT classification by identifying the conversational and streaming traffic with Real Time services, whereas the interactive and background services are claimed to respond to a Non Real Time traffic pattern.

However, as soon as certain delay delivery requirements are imposed on the background and mainly interactive users, their Real Time services becomes diffused [15]. Network gaming is usually considered as a good example of this situation [3]. While this service responds to a typical interactive pattern, it may have stringent delay requirements that could only be satisfied by a UMTS bearer of the conversational class.

Chapter 3

High Speed Downlink Packet Access

3.1 HSDPA Overview

As commented in Chapter 1, the evolution of the mobile communication market brings demands for both larger system capacity and higher data rates. To boost the support for the packet switched services, the 3GPP has standardized in the Release 5 a new technology denominated High Speed Downlink Packet Access (HSDPA) that represents an evolution of the WCDMA radio interface. HSDPA appears as an umbrella of features whose combination improves the network capacity, and increases the peak data rates up to a theoretical 10 Mbps for downlink packet traffic. These technological enhancements can allow operators to enable new high data rate services, improve the QoS of existing services, and achieve a lower cost per delivered data bit.

The HSDPA concept relies on a new transport channel, the High Speed Downlink Shared Channel (HS-DSCH), where a large amount of power and code resources are assigned to a single user at a certain TTI in a time and/or code multiplex manner. The time-shared nature of the HS-DSCH provides significant trunking benefits over DCH for surges of high data rate traffic [16]. In addition, HSDPA uses Adaptive Modulation and Coding, fast Physical Layer Hybrid ARQ, and fast Packet Scheduling. These features are tightly coupled and permit a per- TTI adaptation of the transmission parameters to the instantaneous variations of the radio channel quality.

The objective of the present chapter is not to give a complete description of the HSDPA concept and its performance. Rather, it will provide a general overview of HSDPA that will be required to achieve a full comprehension for the analysis/design of the Packet Scheduler functionality.

The Packet Scheduler is a key element of the overall HSDPA concept and the central scope of the following chapters and therefore, will be treated in detail there. This way, this chapter will not cover specific HSDPA aspects that are not the subject of analysis in the rest of this thesis like for example the UE capabilities. In the rest of the thesis it is assumed that the penetration of HSDPA capable terminals in the network is 100%. Other overviews of the HSDPA concept are given in [17], [18], [19].

A general description of the specific features included and excluded in HSDPA compared to basic WCDMA technology, an overview of the major architectural modifications introduced by HSDPA will be presented in the following section.

3.2 HSDPA Architecture

Figure 3.1 plots the fundamental features to be included and excluded in the HS-DSCH of HSDPA. For this new transport channel, two of the most main features of the WCDMA technology such as closed loop power control and variable spreading factor have been deactivated.

In WCDMA, fast power control stabilizes the received signal quality (EsNo) by increasing the transmission power during the fades of the received signal level. This causes peaks in the transmission power and a subsequent power rise, which reduces the total network capacity. However, delay tolerant traffic may be served only under favorable radio channel conditions, avoiding the transmission during the inefficient signal fading periods. Moreover, the operation of power control imposes the need of certain headroom in the total Node B transmission power to accommodate its variations. The elimination of power control avoids the aforementioned power rise as well as the cell transmission power headroom. But due to the exclusion of power control, HSDPA requires other link adaptation mechanisms to adapt the transmitted signal parameters to the continuously varying channel conditions.

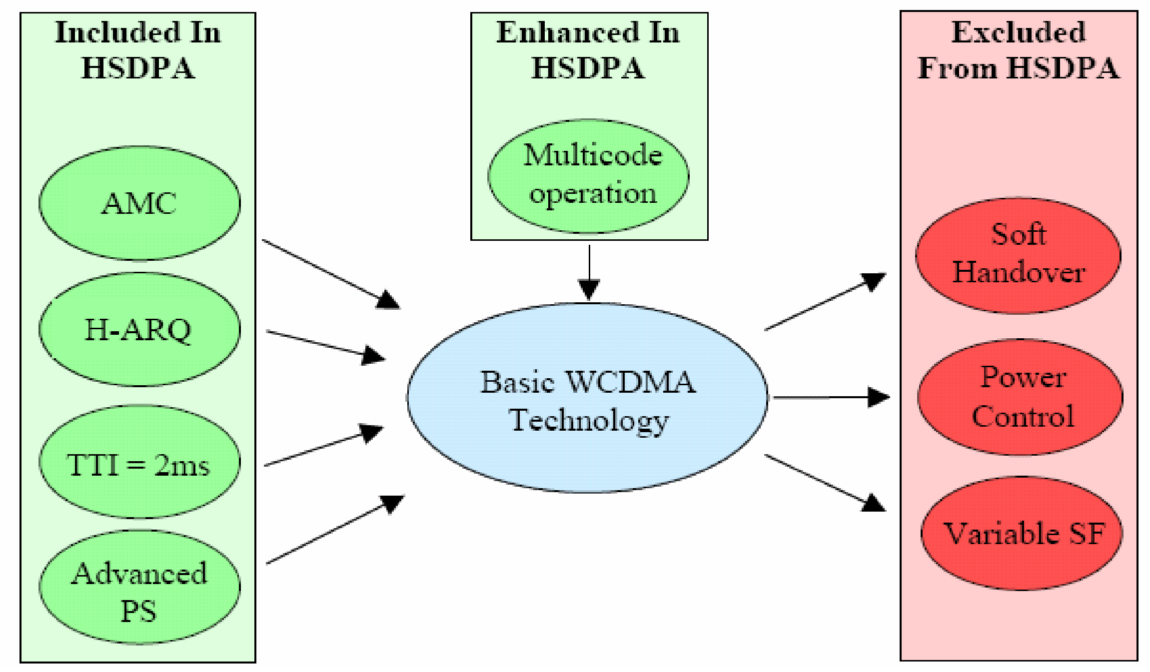


Figure 3.1: Fundamental Features To Be Included And Excluded in the HS-DSCH of HSDPA.

One of these techniques is denominated Adaptive Modulation and Coding (AMC). With it, the modulation and the coding rate are adapted to the instantaneous channel quality instead of adjusting the power. The transmission of multiple Walsh codes is also used in the link adaptation process. Since the combination of these two mechanisms already plays the link adaptation role in HSDPA, the variable spreading factor is deactivated as its long-term adjustment to the average propagation conditions is not required anymore.

As a closed power control is not present, the channel quality variations must be minimized across the TTI, which it is accomplished by reducing its duration from the minimum 10 ms in WCDMA down to 2 ms. The fast Hybrid ARQ technique is added, which rapidly retransmits the missing transport blocks and combines the soft information from the original transmission with any subsequent retransmission before the decoding process. The network may include additional redundant information that is incrementally transmitted in subsequent retransmissions (i.e. Incremental Redundancy).

To obtain recent channel quality information that permits the link adaptation and the Packet Scheduling entities to track the user’s instantaneous radio conditions, the MAC functionality in charge of the HS-DSCH channel is moved from the RNC to the Node B. The fast channel quality information allows the Packet Scheduler to serve the user only when his conditions are favourable. This fast Packet Scheduling and the time-shared nature of the HS-DSCH enable a form of multiuser selection diversity with important benefits for the cell throughput. The move of the scheduler to the Node B is a major architecture modification compared to the Release 99 architecture.

Unlike all the transport channels belonging to the Release 99 architecture, which are terminated at the RNC, the HS-DSCH is directly terminated at the Node B. With the purpose of controlling this channel, the MAC layer controlling the resources of this channel (so called MAC-hs) is directly located in the Node B (see Figure 3.2, Figure 3.3), thereby allowing the acquisition of recent channel quality reports that enable the tracking of the instantaneous signal quality for low speed mobiles. This location of the MAC-hs in the Node B also enables to execution of the HARQ protocol from the physical layer, which permits faster retransmissions.

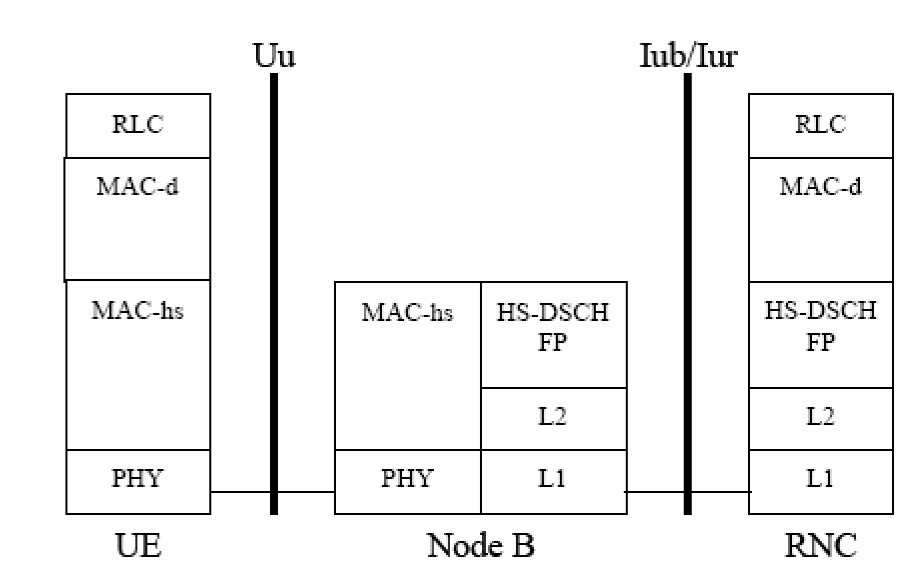


Figure 3.2: Radio Interface Protocol Architecture of the HS-DSCH Transport Channel [20].

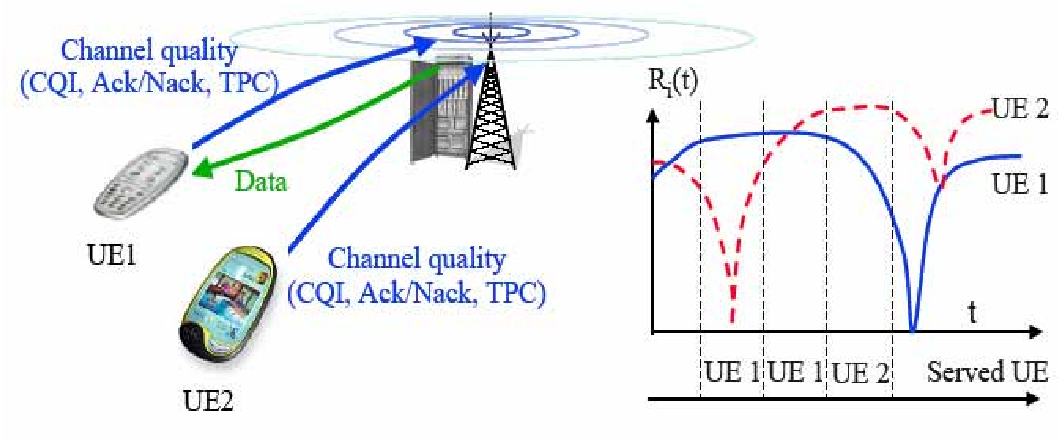


Figure 3.3: Node B Packet Scheduler Operation Procedure.

More specifically, the MAC-hs layer [21] is in charge of handling the HARQ functionality of every HSDPA user, distributing the HS-DSCH resources between all the MAC-d flows according to their priority (i.e. Packet Scheduling), and selecting the appropriate transport format for every TTI (i.e. link adaptation). The radio interface layers above the MAC are not modified from the Release 99 architecture because HSDPA is intended for transport of logical channels. Nonetheless, the RLC can only operate in either acknowledged or unacknowledged mode, but not in transparent mode due to ciphering [17]. This is because in transparent mode the ciphering is done in the MAC-d7, not in the RLC layer, and MAC-c/sh and MAC-hs do not support ciphering [21].

The MAC-hs also stores the user data to be transmitted across the air interface, which imposes some constraints on the minimum buffering capabilities of the Node B. The move of the data queues to the Node B creates the need of a flow control mechanism (HS-DSCH Frame Protocol) that aims at keeping the buffers full. The HS-DSCH FP handles the data transport from the serving RNC to the controlling RNC (if the Iur interface is involved) and between the controlling RNC and the Node B. The design of such flow control is a non-trivial task, because this functionality in cooperation with the Packet Scheduler is to ultimately regulate the user's perceived service, which must fulfill the QoS attributes according to the user's subscription (e.g. the guaranteed bit rate or the transfer delay for streaming bearers or the traffic handling priority and the allocation/ retention priority for interactive users).

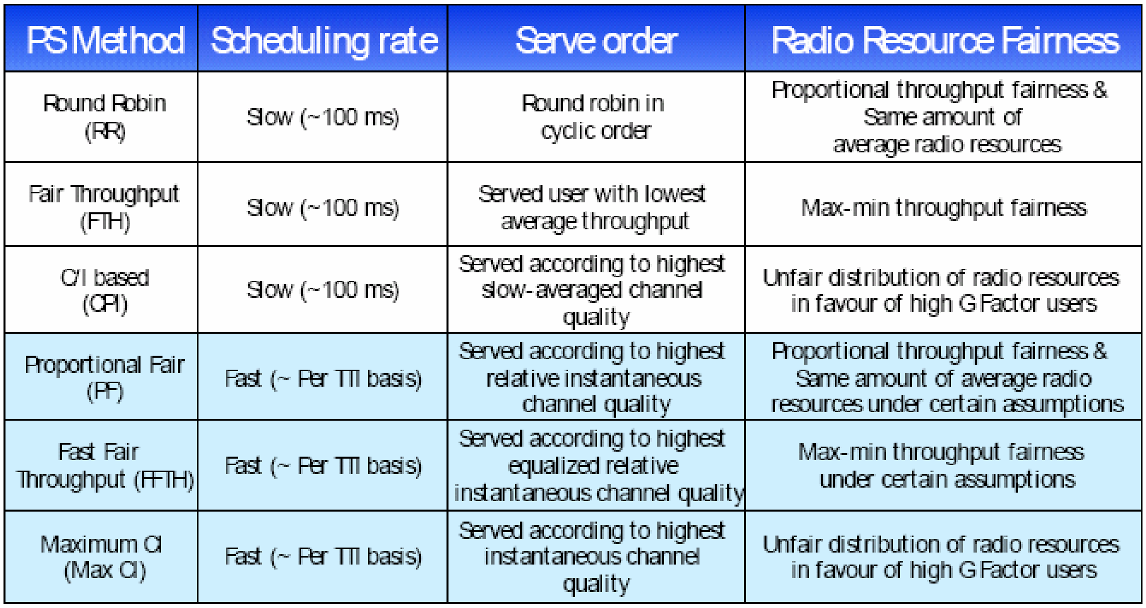
Chapter 4

Models of Scheduling Algorithms

4.1 Scheduling Overview

Wireless channels are characterized by their fast and random variations in time, making it a challenge to ensure fairness among users with different channel conditions and quality of service (QoS) requirements. Fast packet scheduling is the main component of High Speed Downlink Packet Access (HSDPA) [23] which aims at tracking the variations of the channels. In each Transmission Time Interval (TTI), the function of the scheduler is to select the user or users for which packets will be transmitted and determine their rates from a finite set of values depending on the Modulation and channel Coding Schemes (MCS), for the purpose of increasing the system's performance both in terms of throughput and fairness. One of the known algorithms that attempt to achieve a reasonable throughput- fairness tradeoff is the Proportional Fairness (PF) algorithm [24], which is implemented in HDR (High Data Rate) networks [25]. This is introduced to compromise between a fair data rate, and the total data rate for each user. The PF algorithm was shown to achieve better performance than the maximum carrier-to-interference power ratio (CIR) method in the presence of a high number of users [26]. However, with PF scheduling, the user whose channel exhibits the highest variance gets privilege over the other users, which yields unfairness in the service of communicating users, especially in the case of HSDPA where it has a finite set of discrete rate values. Table II Summaries the common Packet Scheduling Methods.

One of the proposed solutions for the above-mentioned problem is the modified PF algorithm which is also called the Data Rate Control (DRC) Exponent rule [27]. However, even with this algorithm, fairness is not guaranteed at all times but rather in specific cases only. In this thesis, a new scheduling algorithm is proposed, called *Adaptive Proportional Fairness* (APF) that resolves the PF unfairness problem in a more general way. In this algorithm, an updating module is introduced to track the data rate allocated to each user and update the exponent parameters in order to achieve fairness among the different users based on their required QoS. Unlike the DRC Exponent rule [27], herein for each user, an exponent parameter is used, which is possible of changing the proportional data rate for each user without changing those of other users.



PS Method	Scheduling rate	Serve order	Radio Resource Fairness
Round Robin (RR)	Slow (~100 ms)	Round robin in cyclic order	Proportional throughput fairness & Same amount of average radio resources
Fair Throughput (FTH)	Slow (~100 ms)	Served user with lowest average throughput	Max-min throughput fairness
C/I based (CPI)	Slow (~100 ms)	Served according to highest slow-averaged channel quality	Unfair distribution of radio resources in favour of high G Factor users
Proportional Fair (PF)	Fast (~ Per TTI basis)	Served according to highest relative instantaneous channel quality	Proportional throughput fairness & Same amount of average radio resources under certain assumptions
Fast Fair Throughput (FFTH)	Fast (~ Per TTI basis)	Served according to highest equalized relative instantaneous channel quality	Max-min throughput fairness under certain assumptions
Maximum CI (Max CI)	Fast (~ Per TTI basis)	Served according to highest instantaneous channel quality	Unfair distribution of radio resources in favour of high G Factor users

Table II. Summary of the common used Packet Scheduling Methods [1].

4.2 Proportional Fairness Scheduling Method

The user selection criterion according to the Proportional Fairness (PF) method is given by [24]

$$j = \arg \max_{1 \le i \le N} \frac{r_i}{R_i} \qquad (1)$$

where j is the index of the selected user for the next TTI, N is the total number of users, r_i is the instantaneous data rate the UE i can support under its current channel conditions, and R_i is the average achieved rate defined as the average data rate effectively received by user i. These rates are updated, at each TTI, according to the following rule:

$$\begin{cases} R_j = (1-\alpha) R_j + \alpha r_j \\ R_i = (1-\alpha) R_i & \text{if } i \ne j \end{cases} \qquad (2)$$

where α is a smoothing parameter with $0 < \alpha < 1$.

The PF method gives fair allocation of data rates between users when they experience the same channel conditions. In real systems, due to the different fading properties the users experience, channels are heterogeneous. In this case, the PF algorithm fails in allocating to users fair data rate that is proportional to their mean rate. Indeed, it was shown in [27], [29] that the UE with more channel variability gets a higher data rate. In order to provide fairness between users depending only on their average data channel rate (r), a solution to the problem was proposed in [28] where by adding an exponent term to r_i, which indicates the channel condition in the PF policy (1), the so-called DRC Exponent rule allows the allocated time control to the users with better channel quality compared to the ones with bad conditions. In the DRC Exponent method, the selection criteria is hence given by

$$j = \arg \max_{1 \le \mathrm{i} \le N} \frac{r_i^{\,n}}{R_i} \tag{3}$$

where n is the exponent parameter, introduced to manage the relationship between the average data rates (R_i) of the users with different channel conditions. However, the control parameter n takes a fixed value for all users, two problems arise with this approach. First, the control parameter being fixed in time, it does not adapt to the time-varying radio conditions of each UE. Second, as this parameter takes a unique value for all users, it is not possible to fix a value for n that ensures fairness between all users at the same time.

Next these characteristics are explained in more detail and why they are important to the design of wireless scheduling policies. In wireless networks, the channel conditions of mobile users are time-varying. Radio propagation can be roughly characterized by three nearly independent phenomena: path-loss variation, slow log-normal shadowing, and fast multipath-fading. Path losses vary with the movement of mobile stations. Slow log-normal shadowing and fast multipath-fading are time-varying with different time-scales.

Furthermore, a user receives interference from other transmissions, which is time-varying; and background noise is also constantly varying. Hence, mobile users perceive time-varying channel conditions. SINR (signal to interference plus noise ratio) is a commonly used measure of channel conditions. Fig. 4.1 shows the time varying SINR of a mobile user. Other measures include BER (Bit Error Rate) and FER (Frame Error Rate). As channel conditions are time-varying, users experience time-varying service quality and/or quantity. For voice users, better channel conditions may result in better voice quality. For packet data service, better channel conditions (or higher SINR) can be used to provide higher data rates using adaptation techniques.

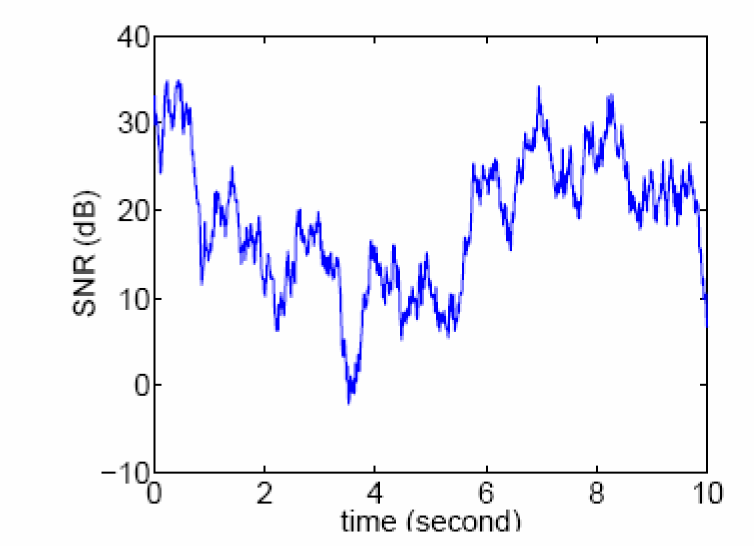


Fig.4.1. User's time-varying SINR

Research shows that cellular spectral efficiency (in terms of b/s/Hz/sector) can be increased by a factor of two or more if users with better links are served at higher data rates [32]. Procedures to exploit this are already in place for the entire major cellular standards: adaptive modulation and coding schemes are implemented in the TDMA standards, and variable spreading and coding are implemented in the CDMA standards. In general, a user is served with better quality and/or a higher data rate when the channel condition is better. Hence, good scheduling schemes should be able to exploit the variability of channel conditions to achieve higher utilization of wireless resources.

The performance (e.g., throughput) of a user depends on the channel condition it experiences; hence, different performance is expected when the same resource (e.g., radio frequency) is assigned to different users. For example, consider a cell with two users. Suppose that user 1 has a good channel, e.g., it is close to the base station. User 2 is at the edge of the cell, where the path-loss is significant and the user experiences large interference from adjacent cells. If the same amount of resource (power, time-slots, etc.) is assigned, it is likely that the throughput of user 1 will be much larger than that of user 2.

Different assignments of the wireless resource will affect the system performance, hence, resource allocation and scheduling policies are critical in wireless networks. In this Thesis, A study Adaptive Proportional Fairness scheduling is presented. Earlier, the unique features of wireless networks are described. Then an important question is: under such conditions, what should be the basic features of a scheduling policy? Consider a few users that share the same resource. The users have a constantly varying channel condition, which implies constant varying performance. The scheduling policy decides which user should transmit during a given time interval. Intuitively, assign resource is wanted to users experiencing "good" channel conditions so that the resources can be used efficiently.

At the same time, also provide some form of fairness or QoS guarantees to all users are needed. For example, allowing only users close to the base station to transmit with high transmission power may result in very high system throughput, but may starve other users. This basic dilemma motivates this work:

To improve wireless resource efficiency by exploiting time-varying channel conditions while at the same time controlling the level of fairness/QoS among users. Fairness criteria may have different implications in wireless networks.

In wireline networks, when a certain amount of resource is assigned to a user, it is equivalent to granting the user a certain amount of throughput/performance value. However, the situation is different in wireless networks, where the amount of resource and the performance values are not directly related (though correlated).

Hence, A study for two kinds of fairness is depicted: temporal vs. utilitarian. Temporal fairness means that each user gets a fair share of network resource, and utilitarian fairness means that each user gets a certain share of the overall system capacity. Further, it is considered both long-term fairness and short-term fairness in this thesis. The basic idea of previous SIR scheduling is to let users transmit in "good" channel conditions, and thus a natural question is how long a user is willing to wait for these "good" conditions. Hence, there exists tradeoff between scheduling performance gain and short-term performance. In addition to fairness, consider a long-term QoS metric: each user has a specific data-rate requirement for the system. Because the capacity of a wireless system is not fixed, it is not always an easy task to determine the feasibility of the requirements of all users. deep study for these issues in more detail is analyzed in this thesis.

Interference management is also a crucial component of efficient spectrum utilization in wireless systems because interference ultimately limits the system capacity. Power allocation is a traditional interference management mechanism. It has been well studied and widely used in wireless systems to maintain desired link quality, minimize power consumption, and alleviate interference to others [30]. Further, because users experience time-varying and location-dependent channel conditions in wireless environments, users can be scheduled so that a user can exploit more of its good channel conditions and avoid (as far as possible) bad times, at least for applications (e.g., data service) that are not time-critical. Hence, a joint scheduling and power-allocation scheme should be able to further improve the spectrum efficiency and decrease power consumption.

For example, Opportunistic scheduling exploits the variation of channel conditions, and thus provides an additional degree of freedom in the time domain. Moreover, it can be coupled with other resource management mechanisms to further increase network performance. A good example of this case is joint scheduling and power allocation as explained earlier. In the literature, opportunistic scheduling is also referred to as being multiuser diverse [33].

Occasionally, these two terms can have slightly different meanings. An example is the case where there is only one user in the system and the objective is to minimize transmission power while maintaining a certain data rate.

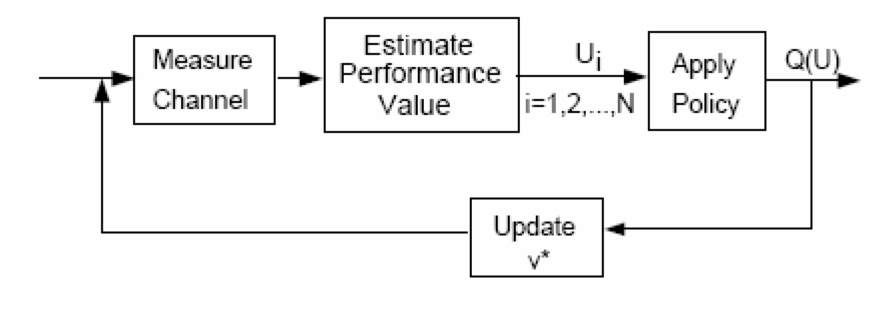


Fig.4.2. Block diagram of the scheduling policy with on-line parameter estimation [34]

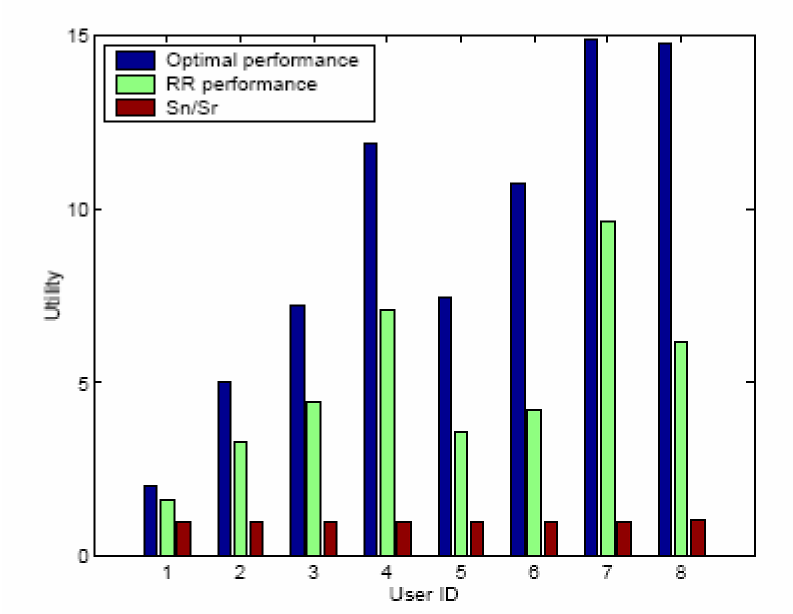


Fig.4.3. Comparison of the opportunistic scheduling policy with the round-robin scheme.

In the Figure 4.3, Sn is the number of time-slots assigned to user i in the optimal scheduling policy and Sr is the number of slots assigned to user i in the round-robin scheme.

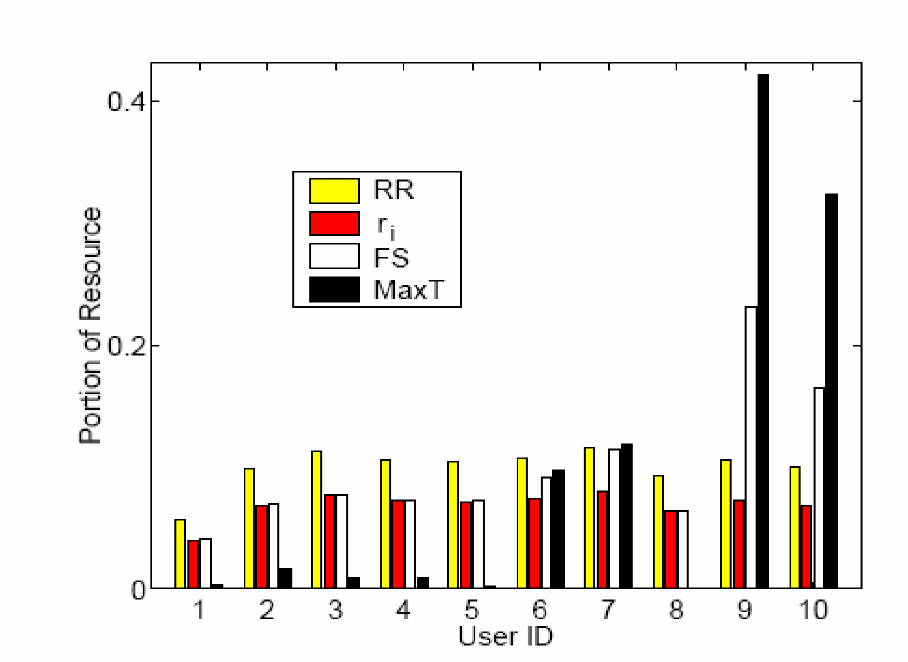


Fig.4.4. Portion of resource shared by users in the temporal fairness scheduling simulation

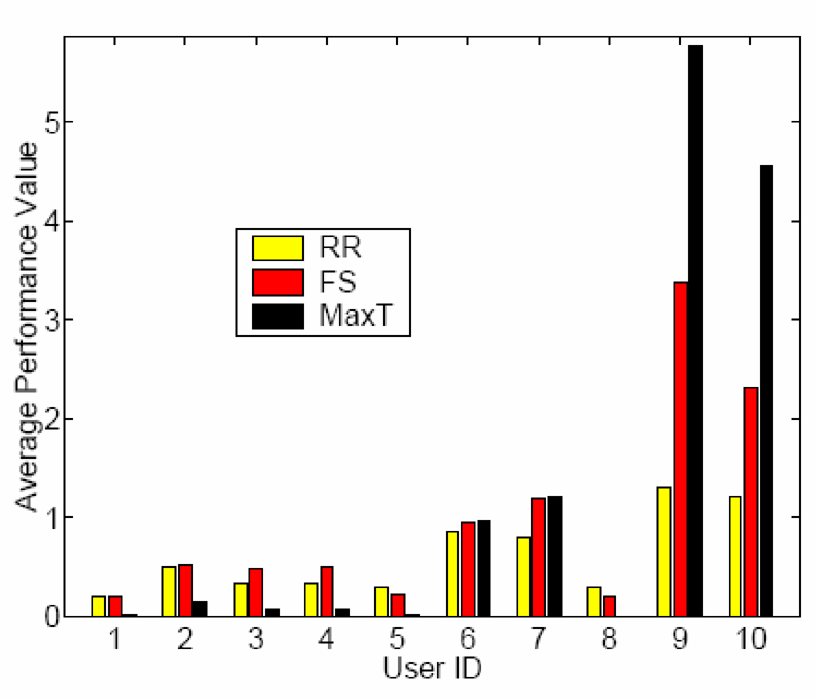


Fig.4.5 Average performance value in the temporal fairness scheduling simulation.

Figures 4.3 and 4.4 show the results of a simulation experiments. In both figures, the x-axis represents the users' IDs. In Figure 4.3, the y-axis represents the portion of resource each user gets from the different scheduling policies. The first bar is the result of round-robin, where the resource is equally shared by all active users. Note that the amounts of resource consumed by different users may not be equal because different users are active at different times. The second bar shows the minimum requirements of users, while the third bar shows the portion of resource used by users in the temporal fairness scheduling scheme [34]. The third bar is higher than the second bar for all the users, which indicates that this scheduling policy satisfies the fairness requirements of all users. Users 9 and 10 are the “fortunate" users in the system because they are most likely to be close to the base station and have large performance values. Thus, they get a much larger share of the resource than their minimum-requirements.

Scheduling of transmission attempts in time can be used to differentiate between service classes, sessions or to increase fairness among users. For CDMA, scheduling has an additional feature in interference management. In cellular system, interference can be decomposed into intracellular interference, originating from simultaneous transmissions within the cell and intercell interference caused by terminals transmitting into other cells. If properly done, a selective choice of transmissions can therefore possibly reduce the interference amount without sacrificing throughput. Some services do not require an instantaneous data rate but rather an average rate, i.e., that an amount of data is delivered over a certain time interval, allowing for more effective use of the channel. In addition to power control, utilizing the possibility of scheduling the transmissions within the cell, intracell interference can be efficiently mitigated. Scheduling is highly related to the rate control previously described and is a special case.

One lesson to be drawn from the research efforts of cellular radio systems so far is the importance of radio resource management. As mentioned earlier, when having different QoS requirements, controlling the radio resources may not be straightforward. However, in this case, suitable RRM tools can be identified that can constitute the basic foundation for. Typical methods previously suggested and successfully applied in single-rate systems are admission, congestion and power control. These methods aim to provide a benign interference level in the system. Therefore, they will be naturally connected to the case of wireless data systems. Furthermore, for wireless data, by exploiting the delay tolerance of certain services, the area of transmission scheduling becomes a possible interference management method to incorporate. It should be pointed out here, that there exists other interference suppression methods which would improve the energy efficiency and capacity, like adaptive antennas, Multiple Input Multiple Output (MIMO) systems as well as diversity schemes, but those are excluded in this work as it is tried to limit this thesis to the Packet Scheduling in HSDPA system view.

Chapter 5

System Model

In the following model, the Packet Scheduling QoS problem solution is presented and the Adaptive Proportional Fairness (APF) scheduling algorithm is described which is studied in two different scenarios. In the first, the algorithm operates in the Best-effort mode. In the second case, the algorithm operates under QoS constraints where each user is assumed to require a specific target rate [35]. Later on, modified versions of APF analysis will show grant for the required target delay.

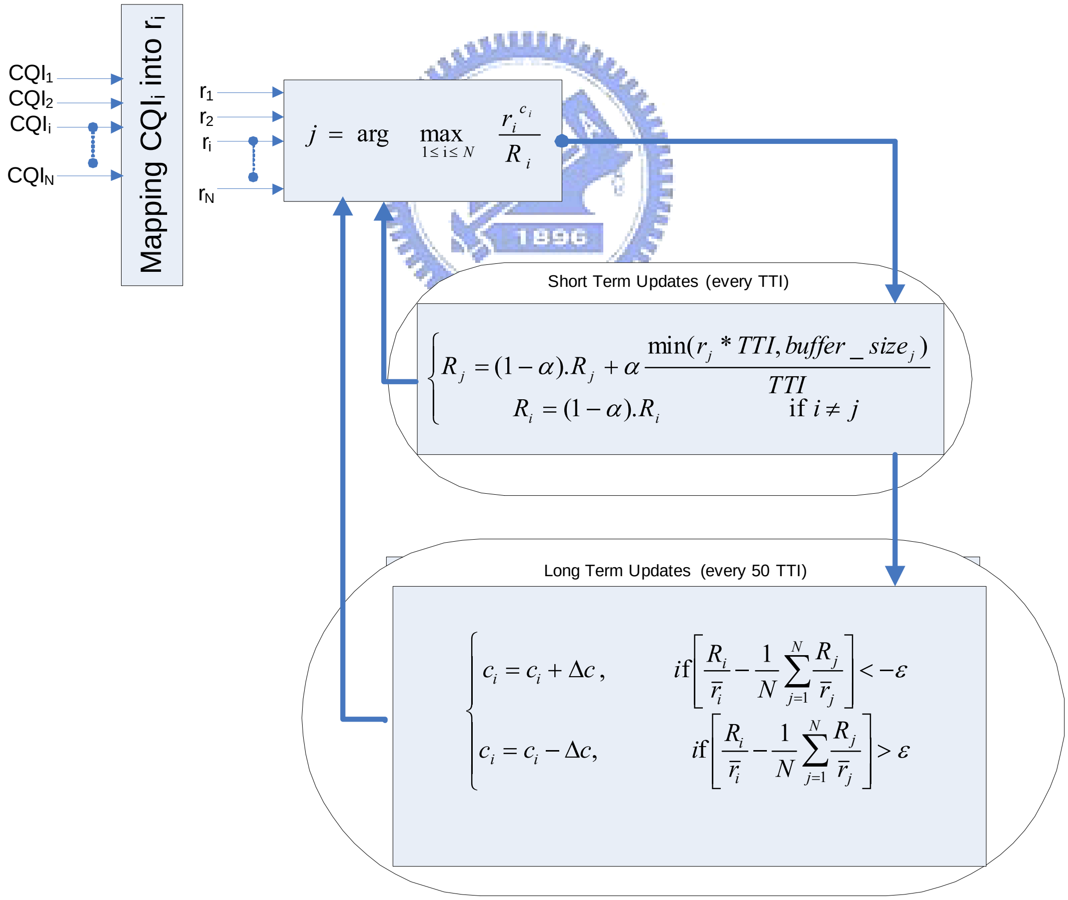


Fig.5.1. Illustration of the operation of the Adaptive Proportional Fairness algorithm under Best-effort mode

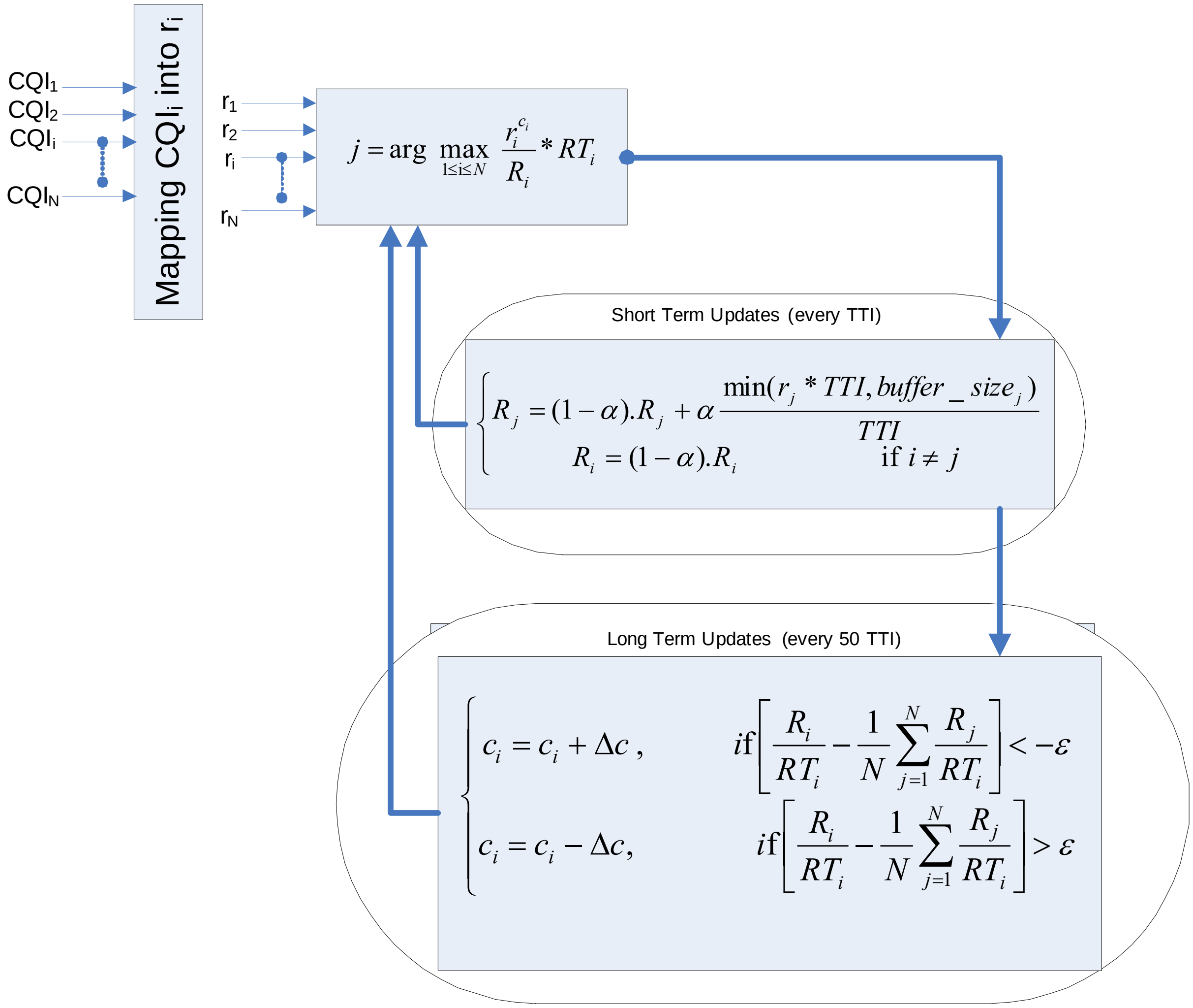


Fig.5.2. Illustration of the operation of the Adaptive Proportional Fairness algorithm under QoS constraints

5.1. Best-effort Mode

In this case, no constraint on the QoS is specified and the scheduler operates in the Best-effort mode. The issue is to maximize the total data rate while providing fairness among the communicating users. The operation of the algorithm is illustrated in Figure 5.1. The user selection criterion is given by

$$j = \arg \max_{1 \le i \le N} \frac{r_i^{c_i}}{R_i} \tag{4}$$

where c_i is the control parameter corresponding to user i. The Adaptive Proportional Fairness (APF) user selection criterion can hence be seen as an enhancement of the DRC Exponent rule where an independent control parameter is assigned to each user in order to avoid the dependency between the different users. Moreover, to be able to track the fast variations of the channels and the differences between the different users channels, a monitoring module is added which updates the values of c_i, $i = 1, 2, \ldots, N$, for the purpose of satisfying the equality between the proportional allocated data rate values ($\frac{R_i}{\bar{r}_i}$). As can be seen in Figure 5.1, the scheduling is executed at each TTI whereas the updating of the control parameters is performed at a larger time-scale. This is to ensure that the algorithm has a chance to function properly, i.e., to make good decisions about the allocated data rates (R_i), and to correct it only when necessary for the purpose of achieving proportional fairness between the users. Herein, the updating is made every 50 $* \ TTI = 0.1s$. The updating module verifies whether the difference between the proportional data rate allocated to user i, $\frac{R_i}{\bar{r}_i}$, and the average value over all users, is within acceptable limits defined by the interval $[-\varepsilon, \varepsilon]$. If the condition is not satisfied, the control parameter c_i corresponding to the user under consideration is updated, along with the parameters of the other users. The updating of c_i, $i = 1, 2, \ldots\ldots, N$, is performed according to the following rule:

$$\begin{cases} c_i = c_i + \Delta c\,, & if\left[\dfrac{R_i}{\bar{r}_i} - \dfrac{1}{N}\sum_{j=1}^{N}\dfrac{R_j}{\bar{r}_j}\right] < -\varepsilon \\ c_i = c_i - \Delta c, & if\left[\dfrac{R_i}{\bar{r}_i} - \dfrac{1}{N}\sum_{j=1}^{N}\dfrac{R_j}{\bar{r}_j}\right] > \varepsilon \end{cases} \qquad (5)$$

where the choice of Δc depends on the speed of convergence sought and the desired strength of variations around the value at convergence. By increasing the value of Δc, faster convergence to the steady state of R_i can be reached but with more oscillations around this value. In order to avoid large variations around the average effective data rate values at convergence, in this simulation, Δc is fixed at 0.01. As for the average smoothing parameter α, it is considered a value of 0.0001.

5.2. Operation under QoS Constraints

In this case, it is supposed that performs in the presence of users with specific requirements on the QoS, represented by a target data rate for each user. Under such constraints, the PF policy fails to ensure the desired QoS due to the fact that there is no consideration of the required QoS in its selection criterion. Moreover, even in its modified version, the DRC Exponent rule [27], fails to provide the desired QoS and fairness between users. When these exhibited differences in their propagating conditions, namely, differences in the variance associated with the channel of each user as described in previous Chapter. The proposed APF algorithm, on the other hand, is designed to satisfy the required QoS thanks to the features introduced in the selection and updating modules (Fig.5. 2). In this case, the user selection criterion is given by

$$j = \arg \max_{1 \le \mathrm{i} \le N} \frac{r_i^{c_i}}{R_i} * RT_i \tag{6}$$

where RT_i denotes the target rate corresponding to user i. Compared to the previously studied case, the proportional data rate is defined here as the fraction of the data rate of each user over his target data rate, i.e, $\frac{R_i}{RT_i}$. Thus, the updating of the control parameters c_i, $i = 1, 2, \ldots, N$, is performed according to the following rule:

$$\begin{cases} c_i = c_i + \Delta c\,, & \text{if} \left[\dfrac{R_i}{RT_i} - \dfrac{1}{N} \displaystyle\sum_{j=1}^{N} \dfrac{R_j}{RT_i} \right] < -\varepsilon \\ c_i = c_i - \Delta c, & \text{if} \left[\dfrac{R_i}{RT_i} - \dfrac{1}{N} \displaystyle\sum_{j=1}^{N} \dfrac{R_j}{RT_i} \right] > \varepsilon \end{cases} \tag{7}$$

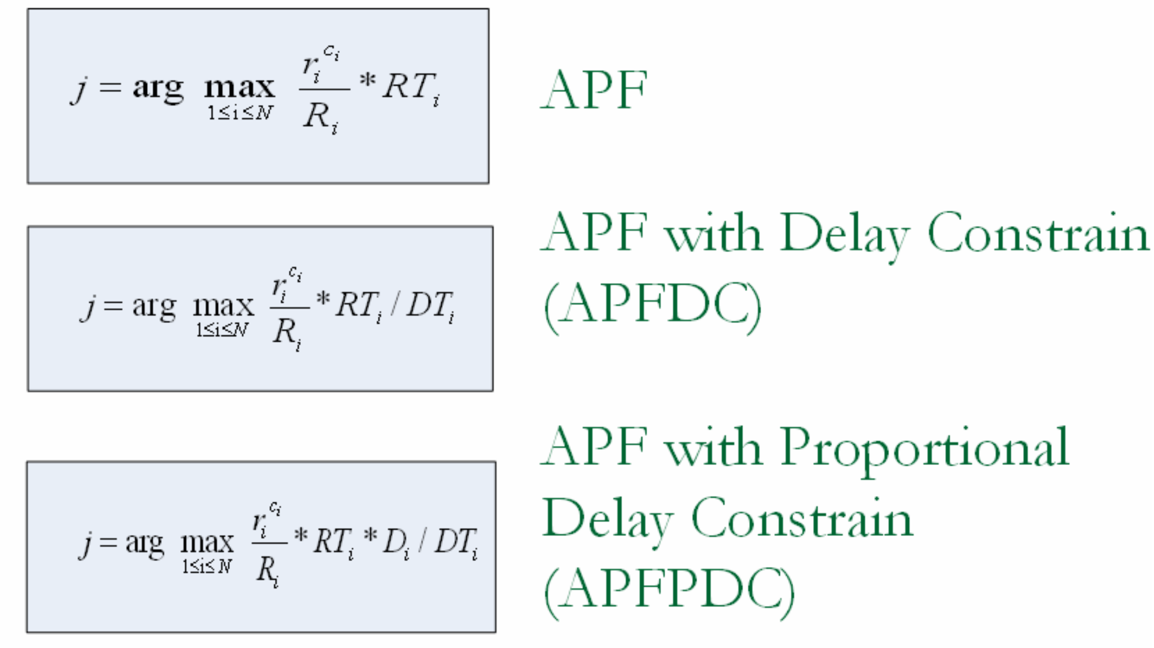
The modified versions of APF are described below, APF with Delay Constrain (APFDC) and APF with Proportional Delay Constrain (APFPDC) consequently:

$$j = \arg \max_{1 \le \mathrm{i} \le N} \frac{r_i^{c_i}}{R_i} * RT_i * DT_i \tag{8}$$

where DT_i denotes the target delay corresponding to user i.

$$j = \arg \max_{1 \le \mathrm{i} \le N} \frac{r_i^{c_i}}{R_i} * RT_i * D_i / DT_i \tag{9}$$

where D_i denotes the delay corresponding to user i.



$$j = \arg \max_{1 \le i \le N} \frac{r_i^{c_i}}{R_i} * RT_i$$

APF

$$j = \arg \max_{1 \le i \le N} \frac{r_i^{c_i}}{R_i} * RT_i / DT_i$$

APF with Delay Constrain (APFDC)

$$j = \arg \max_{1 \le i \le N} \frac{r_i^{c_i}}{R_i} * RT_i * D_i / DT_i$$

APF with Proportional Delay Constrain (APFPDC)

Fig.5.3. Illustration of the operation of the Adaptive Proportional Fairness modified algorithm under QoS constraints

Chapter 6

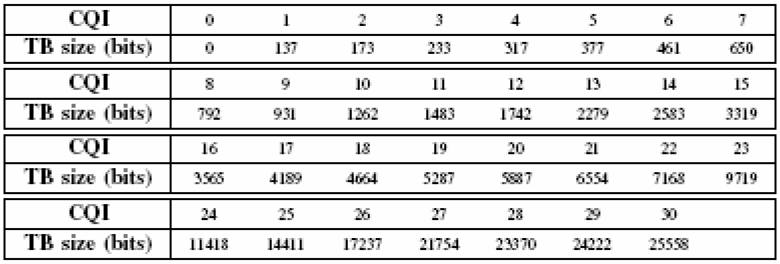
Simulations and Numerical Results

Foremost, in this Chapter, the simulation results is presented and comparisons of the Proportional Fairness method and the Adaptive Proportional Fairness algorithm. In this simulation, N active users are considered and only one user is selected in each TTI is supposed. In HSDPA, more than one user can be selected. This can be performed by this algorithm through ordering of the active users, selection of the first user, and verification of the availability of resources, namely, codes remaining out of the 16 available codes, that could be allocated to more users while satisfying the power budget constraint. In this way, time multiplexing is privileged over code multiplexing. Indeed, it is shown in [31] that time multiplexing, where it is preferable to transmit for few users but at their full available rates, yields higher performance compared to code multiplexing.

It is assumed that all users are allocated the same transmission power, the only difference being the type of variations exhibited by the channel of a user (Rayleigh, Shadowing, Pathloss) [36]. As for the mapping of a channel state, given by the signal-to-noise and interference ratio (SNIR), into a CQI, the following rule is used:

$$\mathrm{CQI} \approx \min(\max(0,[\mathrm{SNIR}+6]),30) \qquad (10)$$

where SNIR is expressed in dB and [.] denotes the integer floor operator. Each CQI value indicates the suitable Transport Block (TB) size, and the modulation/coding that should be used. The list of the thirty available TB sizes is shown in Table III [32].



CQI	0	1	2	3	4	5	6	7
TB size (bits)	0	137	173	233	317	377	461	650
CQI	8	9	10	11	12	13	14	15
TB size (bits)	792	931	1262	1483	1742	2279	2583	3319
CQI	16	17	18	19	20	21	22	23
TB size (bits)	3565	4189	4664	5287	5887	6554	7168	9719
CQI	24	25	26	27	28	29	30	
TB size (bits)	11418	14411	17237	21754	23370	24222	25558	

Table III Channel Indicator table mapping

In this simulation, it is built using Matlab programming language and its run time was 100s for a single cell environment with 5 km radius with a specific channel model for all users. It provides the corresponding parameters used mentioned in the previous chapters. The target rate RT_i(Kb/s) values are 230,68.5,631,117,68.5,230,188,631. The target DT_i(ms) rate values are 30,50,10,20,50,30,25,10.

The System Simulation Model is shown in figure 6.1. In Figure 6.2, it compares the proportional allocated data rate for the PF method and the APF algorithm. For example, it is obvious that the PF policy allocates to user5 76% of his air data rate whereas user7 gets allocated only 20% of the data rate his channel can support. On the other hand, using the APF algorithm, the allocation is between 29% and 32% for all users. However, and as expected, this improvement in fairness comes at the cost of a reduction in the throughput total data allocated as can be seen in Figure 6.3.

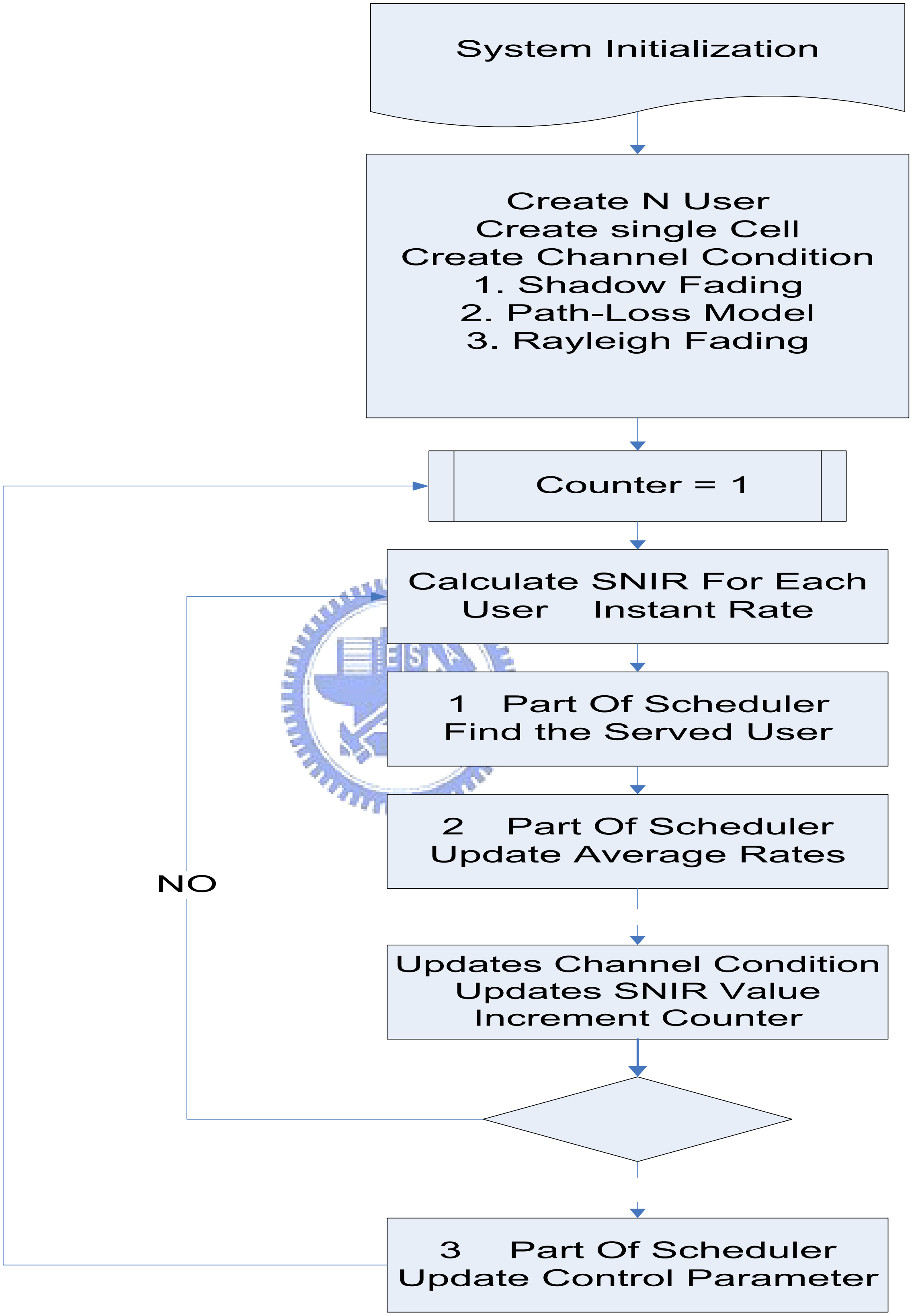


Fig.6.1. System Simulation Model

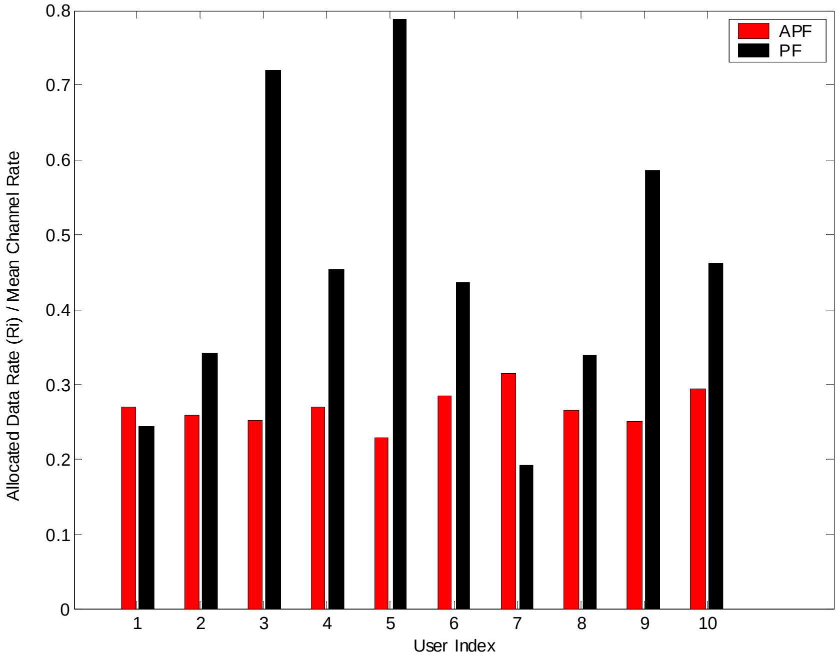


Fig.6.2. Comparison of the Proportional allocated data rate $\frac{R_i}{\bar{r}_i}$ *for the PF and APF Algorithm*

Consequently, compared to the PF policy, the APF algorithm achieves the required fairness between users with heterogeneous channels with no significant loss in total data rate. Moreover, Usage Percentage in APF and PF comparison is shown in Figure 6.4 where APF gives more fairness in channel using percentage among users than PF.

As observed, even if the users have the same transmit power, the PF method does not allocate the data rates fairly among them. To the difference in the variations exhibited by the channels of the different users, namely, the difference in the variance associated to each channel distribution. The more the channel variations are, the higher is the average data rate R_i allocated to the corresponding user. Comparing these results with the ones obtained using the APF algorithm shows how this algorithm outperforms the PF method by fairly allocating the data rates despite the differences in the channel distributions of the different users or the underlying variances.

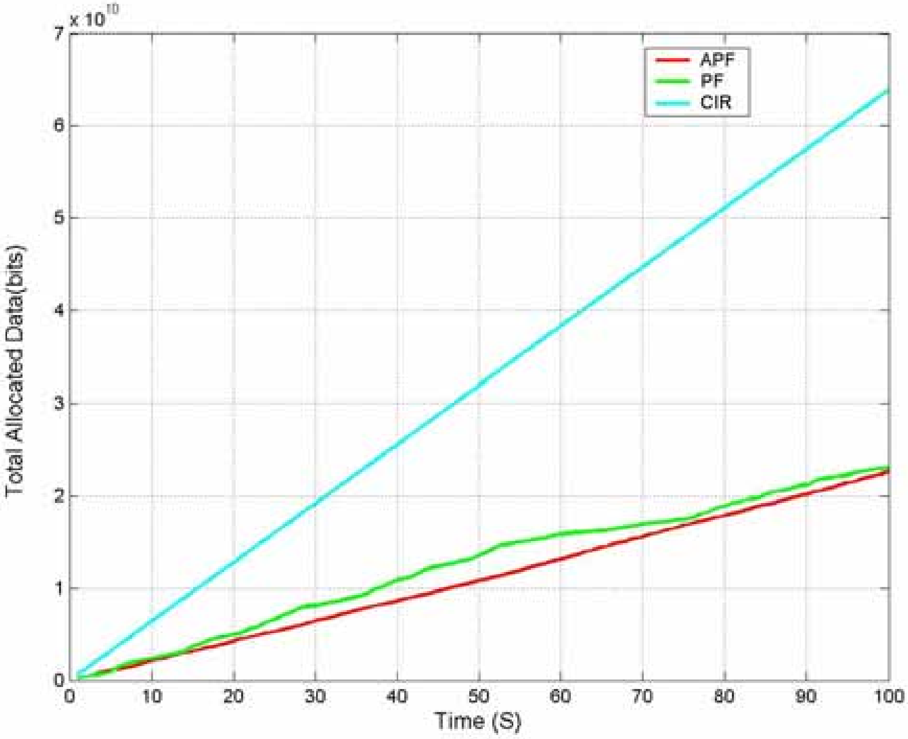


Fig.6.3. Throughput of the total allocated bit: comparison among PF, APF and CIR Algorithms.

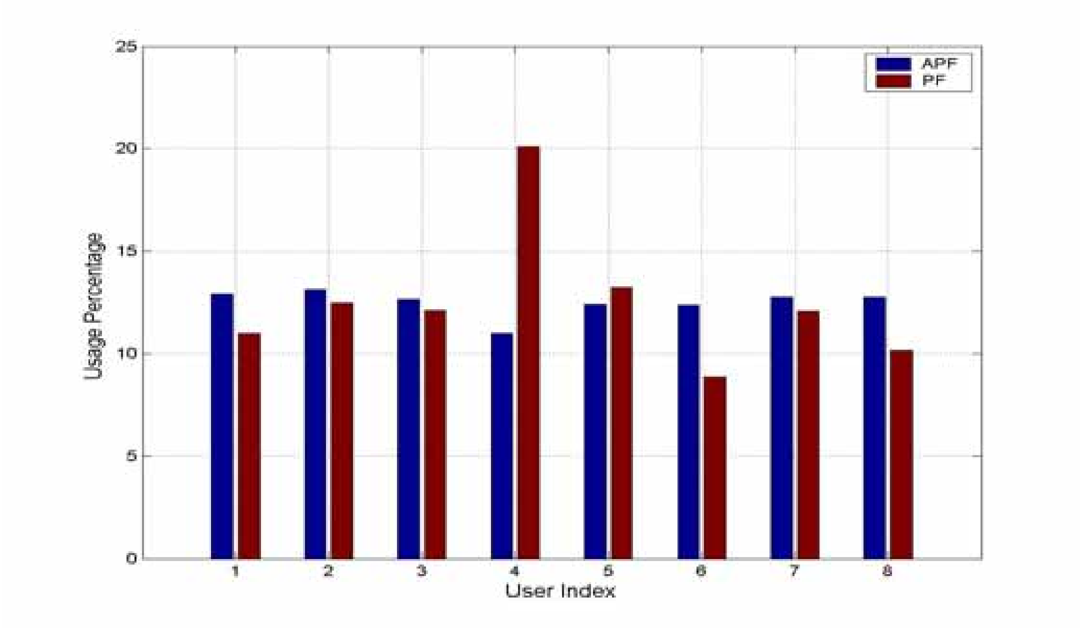


Fig.6.4. Usage Percentage in APF and PF

In the case that users Operate under constraints on the target rates, in Figure 6.5, it compares the allocated data rate to the Target Rate for PF and APF under constraints on the target rates. On one hand, the user6 the PF policy allocates twice more than the required data rate (RT_i) while it allocates to user1, user3 and user8 no more than 50 % of their target values. On the other hand, using the APF algorithm yields approximately the same allocated Ratio rate, which is at least 100% of the target data rate of any user. By using the APF algorithm, it is ensured that the requested rates are allocated to all users in a fair manner despite the differences they experience in their channel conditions and the limitation of the discrete rate values in use. This way, the available resources are used more efficiently and the inter-cell interference gets significantly reduced, which yields more resources to be available in the adjacent cells making it possible to satisfy the QoS requirements of their corresponding users.

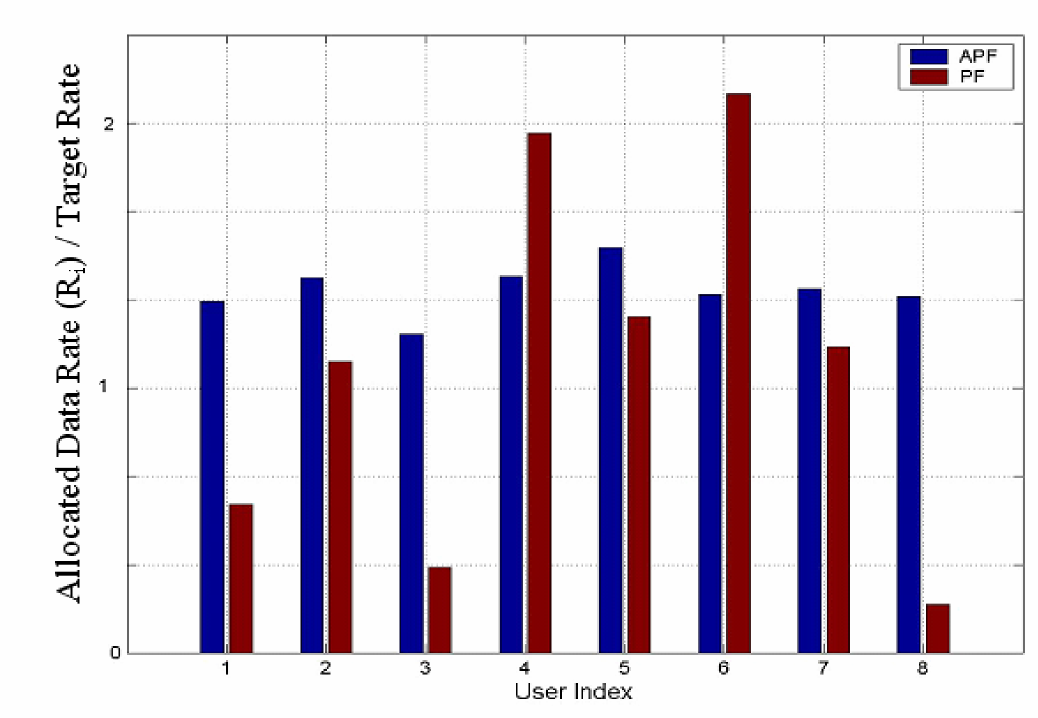


Fig.6.5. Comparison of allocated data rate to Target Rate for PF and APF under constraints of the Target Rates

Another QoS requirement is delay, the proposed methods described in the previous Chapter are implemented, where the percentage that i user do not get served (NS) while the delay (D)≥ Target delay (TD) is calculated, it is obvious that the probability for i user Pi(NS| D≥ TD) is smaller when the two modified APF methods are applied as depicted in Figure 6.6, but this come in the price of total throughput, where there is no significant loss in the total throughput as shown in Figure 6.7.

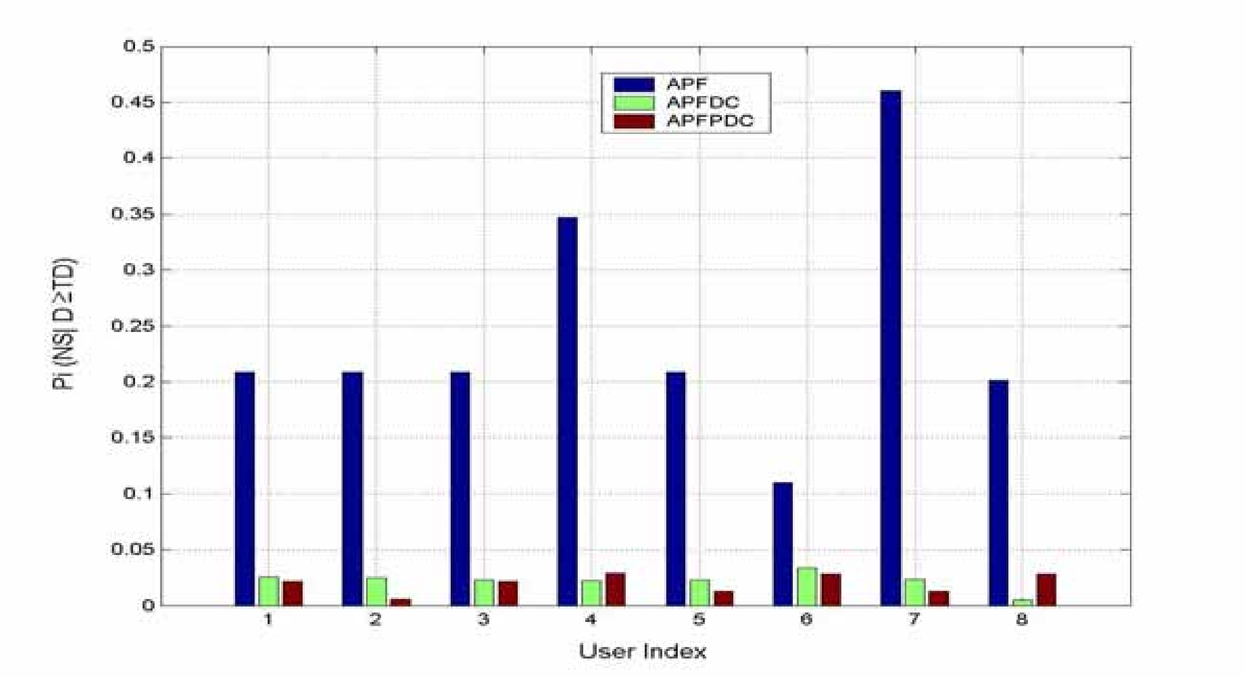


Fig.6.6. Comparison of Pi (NS| D≥TD) for different modified APF algorithms under constraints on the Target Rates

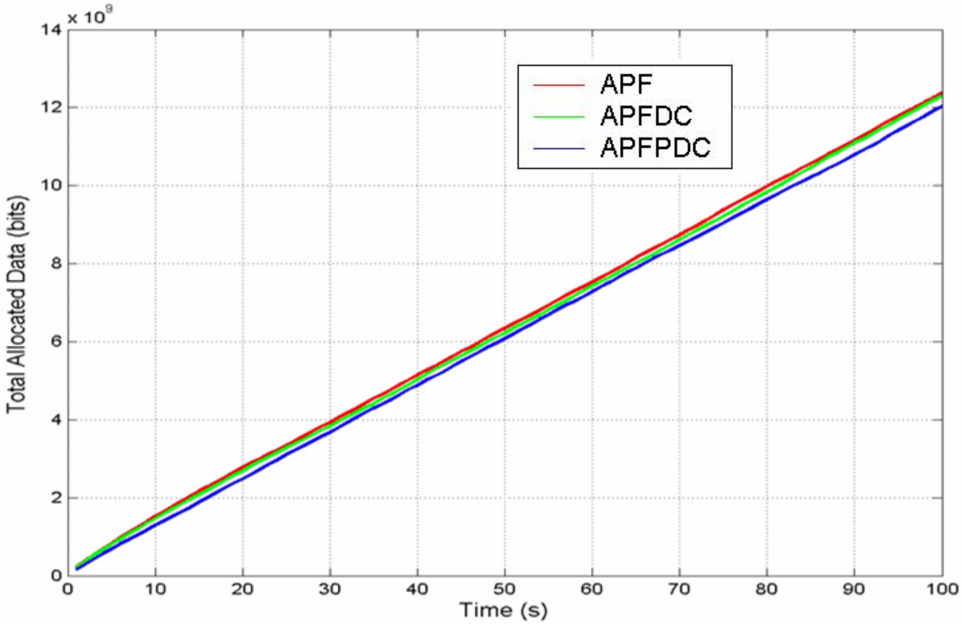


Fig.6.7. Throughput of the total allocated bit: comparison among APF, APFDC and APFPDC algorithms.

Chapter 7

Conclusion

Scheduling is a way to improve spectrum efficiency by exploiting time-varying channel conditions. In this chapter, it summarizes the present framework for APF scheduling to maximize the average system performance value by exploiting variations of the channel conditions while satisfying certain fairness/QoS constraints. The framework provides the flexibility to study a variety of QoS Constrain problems (many of the previous works by us and other researcher's goes well into this unified framework). Using this framework, two QoS scheduling schemes have been studied: to achieve required target rate and required delay target for each user. This provides optimal solutions to each scheduling problem while discussing their properties. Different scheduling schemes may be suitable for different application scenarios. Also a deep study for behavior of Adaptive scheduling schemes is presented. Further, this work extends under more general QoS conditions. Lastly, this simulation shows that the proposed scheduling schemes result in substantial system gains while maintaining users' QoS requirements.

To improve spectrum efficiency, intuitively, it is needed to assign resources for users experiencing "good" channel conditions. At the same time, it is also desirable to provide some form of fairness or QoS guarantees. Otherwise, the system performance can be trivially optimized by, for example, letting a user with the highest performance value to transmit. This may prevent "poor" users (in terms of either channel conditions or money) from accessing the network resource, and thus compromises the desirable feature of wireless networks: to provide anytime, anywhere accessibility. In this thesis, a new fast packet-scheduling algorithm is proposed, the *Adaptive Proportional Fairness* (APF) scheduling, which represents an enhancement of the DRC Exponent rule that fails to achieve fairness between users in heterogeneous channels. By adding a user control parameter in the criterion used to select the user will be served, and introducing an updating module to track the fast variations of the channels, the APF algorithm is shown to provide the required fairness between users. The heterogeneous propagating conditions experienced by the communicating users is considered and the APF performance for two scenarios is investigated, namely, the *Best-effort* service where no QoS constraints are specified, and the case where users have specific demands in terms of the rates and minimum delay they require. Taking into consideration HSDPA rate constraints, the simulation results, provided for the *Best-effort* case, show the high efficiency of the APF method in terms of fairness in that it yields between users, compared to the PF method and at no significant loss in total data rate. As for the second scenario, it also shows that the APF algorithm ensures servicing users at the required data rates while the PF policy fails to satisfy the users QoS requirements. The third scenario showed that the APFDC and APFPDC algorithm ensures servicing users with minimum delay while the APF policy fails to satisfy the performance of minimum delay. Future work can be working in multiuser scenario including more of QoS Constrain.

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