

# Evaluation of multiuser scheduling algorithm in OFDM for different services

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# Problem Description

The goal of this Master Thesis is to study shared radio resources among users with different services requirements. The analyzed properties of the wireless connection are fairness, throughput and delay for users demanding different services and QoS requirements. Four scheduling algorithms are used for allocating system resources. Two of them, Max Rate and Round Robin, are used as references to analyze throughput and fairness respectively. The other two algorithms, Proportional Fair Scheduling and Rate Craving Greedy, exploit the idea of multiuser diversity improving the throughput without comprising fairness. Different fading radio channel models are investigated, but only urban environments and pedestrian users are simulated in this report. OFDM has been the technique used to transmit signals over the wireless channel. The performance of these algorithms is analyzed and compared through MATLAB computer simulations.

Assignment given: 10. January 2006  
Supervisor: Torbjørn Ekman, IET





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Oppgavens tittel (engelsk): **Evaluation of Multiuser Scheduling Algorithms in OFDM for Different Services**

Oppgavens tekst:

The goal of this Master Thesis is to study shared radio resources among users with different service requirements. The analyzed properties of the wireless connection are fairness, throughput and delay for users demanding different services and QoS requirements. The Max Rate and Round Robin scheduling algorithms are used as references to analyze throughput and fairness respectively. The student should examine other scheduling algorithms and their tradeoffs between throughput and fairness when exploiting the idea of multiuser diversity.

The work is a simulation study of the performance of the different scheduling algorithms. Fading radio channel models have to be implemented. The performance will be simulated and evaluated for an OFDM system.

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faglærer



## Abstract

The goal of this Master Thesis is to study shared radio resources among users with different services requirements. The analyzed properties of the wireless connection are fairness, throughput and delay for users demanding different services and QoS requirements. Four scheduling algorithms are used for allocating system resources. Two of them, Max Rate and Round Robin, are used as references to analyze throughput and fairness respectively. The other two algorithms, Proportional Fair Scheduling and Rate Craving Greedy, exploit the idea of multiuser diversity improving the throughput without comprising fairness. Different fading radio channel models are investigated, but only urban environments and pedestrian users are simulated in this report. OFDM has been the technique used to transmit signals over the wireless channel. The performance of these algorithms is analyzed and compared through MATLAB computer simulations.

*Keywords: OFDM, multiuser diversity, fading, scheduler, PFS, fairness, throughput, delay.*





*“People Move, Networks don’t”*

## Foreword

This report is a Master Thesis submitted to the Department of Electronics and Telecommunications of the Norwegian University of Science and Technology (NTNU) in Trondheim, Norway and it was written thanks to the Erasmus exchange programme.

It has been a good opportunity for widening my knowledge about wireless communications. Not only I’m interested in this topic, but also the whole society because it makes communication among users easier. The worldwide growth of the wireless communications industry has been truly phenomenal, it began expanding from the business environment to include the home environment. And there is a large research group around the world working in this field in order to improve wireless performance.

At the beginning the project was not easy for me. I began with a basic knowledge about communications which was acquired during five years at ETSI de Telecomunicación in Valladolid, Spain. My target has been to simulate a wireless environment in MATLAB. So, after reading up about this topic, thanks to the good library placed in NTNU, I began programming small scripts, which were discarded in most cases because either I did not have clear ideas, or because of time limitation. But step by step I achieved smarter scripts and I got to be closer to right results. Having them as a basis, I began to build up what is showed in this report, which sometimes I’ve loved, sometimes I’ve hated.

But just the exchange experience makes the Thesis challenging enough. I feel the work has given me much valuable knowledge about wireless communications, and I am very grateful to the opportunity of staying for six months in Trondheim, where I met really good friends.



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I want to thank all of you who have given your time, assistance and patience so generously. I would like to express my sincere gratitude to my supervisor Torbjørn Ekman for his great knowledge, invaluable help, patience and support.

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Thanks to the University of Valladolid, for all the things I learnt there, and for having made possible this chance for me to study at NTNU.

Thanks to the Erasmus community in Trondheim for having made the winter warmer and the spring sunnier, special thanks given to the Spanish crew.

I wish to express my greatest thanks to my parents and my sister, thanks for all the attentions you have put on me. You started all this: now I have to finish it.

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## List of Abbreviations

<b>AWGN</b>	Additive White Gaussian Noise
<b>B</b>	Signal Bandwidth
<b>BABS</b>	Band Assignment Based on SNR
<b>BER</b>	Bit Error Rate
<b>Bps</b>	Bits Per Second
<b>BS</b>	Base Station
<b>C</b>	Channel Capacity
<b>DSL</b>	Digital Subscriber Line
<b>fc</b>	Carrier Frequency
<b>fd</b>	Doppler Frequency
<b>fo</b>	Coherence Bandwidth
<b>fs</b>	Carrier Spacing
<b>H</b>	Channel Gain
<b>ISI</b>	Inter Symbol Interference
<b>JTC</b>	Joint Technical Committee
<b>LOS</b>	Line Of Sight
<b>MS</b>	Mobile Station
<b>OFDM</b>	Orthogonal Frequency Division Multiplexing
<b>PDF</b>	Probability Density Function
<b>PDP</b>	Power Delay Profile
<b>PFS</b>	Proportional Fair Scheduling
<b>QoS</b>	Quality of Service
<b>RCG</b>	Rate Craving Greedy
<b>RF</b>	Radio Frequency
<b>Rmin</b>	The minimum rate demanded by users
<b>RR</b>	Round Robin
<b>S</b>	Transmission Power
<b>SNR</b>	Signal to Noise Ratio
<b>Tm</b>	Delay Spread
<b>To</b>	Coherence Time
<b>Tsym</b>	Symbol Time
<b>W</b>	Watt
<b>WLAN</b>	Wireless Local Area Network



## Chapter 1

# Introduction

The ability to communicate with people on the move has evolved remarkably since Guglielmo Marconi first demonstrated radio's ability to provide continuous contact with ships sailing the English Channel. That was in 1897, and since then new wireless communications methods and services has been enthusiastically adopted by people throughout the world. Particularly during the past ten years, the mobile wireless communications industry has grown by orders of magnitude, fueled by digital and RF circuit fabrication improvements, new large-scale circuit integration, and other miniaturization technologies which make portable radio equipment smaller, cheaper, and more reliable [1]. The most successful wireless networking technology this far has been 802.11.

*Orthogonal frequency division multiplexing* (OFDM) has become a popular technique for transmission of signals over wireless channels. OFDM has been adopted in several wireless standards like 802.11. OFDM converts a frequency-selective channel into a parallel collection of frequency flat sub-channels. The sub-carriers have the minimum frequency separation required to maintain orthogonality of their corresponding time domain waveforms, yet the signal spectra corresponding to the different sub-carriers overlap in frequency. Hence, the available bandwidth is used very efficiently. If knowledge of the channel is available at the transmitter, then the OFDM transmitter can adapt its signalling strategy to match the channel.

OFDM devices use one wide frequency channel by breaking it up into several components sub-carriers. Each sub-carrier is used to transmit data. All the "slow" sub-carriers are then multiplexed into one "fast" combine channel.

OFDM is not a new technique. Most of the fundamental work was done in the late 1960s. Recent DSL work and wireless data applications have rekindled interest in OFDM, especially now that better signal-processing techniques make it more practical [2].

## 1.1 Motivation

Wireless networks offer several advantages over fixed (or “wired”) networks:

### Mobility

Users move, but data is usually stored centrally. Enabling users to access data while they are in motion can lead to large productivity gains.

### Ease and speed of deployment

Many areas are difficult to wire for traditional wired LANs. Older buildings are often a problem; running cable through the walls of an older stone building to which the blueprints have been lost can be a challenge. In many places, historic preservation laws make it difficult to carry out new LAN installation in older buildings.

### Flexibility

No cables mean no recabling. Wireless networks allow users to quickly form amorphous, small group networks for meeting, and wireless networking makes moving between cubicles and office a snap. Expansion with wireless networks is easy because the network medium is already everywhere. There are no cables to pull, connect, or trip over.

### Cost

In some cases, costs can be reduced by using wireless technology. As an example, 802.11 equipment can be used to create a wireless bridge between two buildings. Setting up a wireless bridge requires some initial capital cost in terms of outdoor equipment, access points and wireless interfaces. After the initial capital expenditure, however, an 802.11-based, LOS network will have only a negligible recurring monthly operating cost. Over time, point-to-point wireless links are far cheaper than leasing capacity from the telephone company.

## 1.2 Radio Spectrum: The Key Resource

Wireless devices are constrained to operate in a certain frequency band. Each band has an associated bandwidth, which is simply the amount of frequency space in the band. Bandwidth has acquired a connotation of being a measure of the data capacity of a link [22]. A great deal of mathematics, information theory and signal processing can be used to show that higher-bandwidth slices can be used to transmit more information. The use of a radio spectrum is rigorously controlled by regulatory authorities through licensing processes. Each frequency range has a band designator and each range of frequencies behaves differently and performs different functions. The frequency spectrum is shared by civil, government, and military users of all nations according to International Telecommunications Union (ITU) radio regulations. For communications purposes, the usable frequency spectrum now extends from about 3 Hz to about 300 GHz. There are also some experiments at about 100 THz where research on laser communications is taking place but we won't discuss this now. This range

from 3 Hz to 300 GHz has been split into regions. The good thing is that once this range has been split it remained that way and became standard. And it is up to you if you want to accept this standard or not. Frequency band standard is described in International Telecommunications Union radio regulations [3], see Table 1.I.

	<b>Designation</b>	<b>Frequency</b>	<b>Wavelength</b>
<b>ELF</b>	Extremely Low Frequency	3Hz to 30Hz	100'000km to 10'000km
<b>SLF</b>	Super Low Frequency	30Hz to 300Hz	10'000km to 1'000km
<b>ULF</b>	Ultra Low Frequency	300Hz to 3000Hz	1'000km to 100km
<b>VLF</b>	Very Low Frequency	3KHz to 30KHz	100km to 10km
<b>LF</b>	Low Frequency	30KHz to 300KHz	10km to 1km
<b>MF</b>	Medium Frequency	300KHz to 3000KHz	1km to 100m
<b>HF</b>	High Frequency	3MHz to 30MHz	100m to 10m
<b>VHF</b>	Very High Frequency	30MHz to 300MHz	10m to 1m
<b>UHF</b>	Ultra High Frequency	300MHz to 3000MHz	1m to 10cm
<b>SHF</b>	Super High Frequency	3GHz to 30GHz	10cm to 1cm
<b>EHF</b>	Extremely High Frequency	30GHz to 300GHz	1cm to 1mm

**Table 1.I** Frequency bands.

### 1.3 The Limits of Wireless Networking

Wireless networks do not replace fixed networks. The main advantage of mobility is that the network user is moving. Servers and another data center equipment must access data, but the physical location of the server is irrelevant. As long as the server does not move, they may as well be connected to wires that do not move [2].

The speed of wireless networks is constrained by the available bandwidth. Unless the regulatory authorities are willing to make the unlicensed spectrum bands bigger, there is an upper limit on the speed of wireless networks.

Using radio waves as the network medium poses several challenges. Specifications for wired networks are designed so that a network will work as long as it respects the specifications. Radio waves can suffer from a number of propagation problems that may interrupt the radio link, such as multipath interference and shadows.

Security on any network is a prime concern. On wireless networks, it is often a critical concern because the network transmissions are available to anyone within range of transmitter with the appropriate antenna. This issue is not the purpose of this Thesis, but we encourage you to see [4] and [5].

## 1.4 Objectives

The subject to deal with is simulating a wireless communication between a BS and multiple MSs where radio link is shared. The Thesis results in a MATLAB simulator for the radio channel in a system with a fixed BS and moving user terminals (MSs). The simulations are based on a simplistic geometrical model for the radio environment.

In this project, the tool to be used for simulating is MATLAB 7.0.019920 (R14). With this tool we have to achieve the following two objectives the most important in the Master Thesis:

- Simulating radio link between two terminals.
- Evaluating different scheduling policies for allocating resources according to three properties, fairness, throughput and delay.

Other “collateral” objectives are:

- Getting knowledge about wireless communications.
- Getting to understand mathematical artifacts commonly implemented for simulating radio channels.
- Practising programming skills, which will be applied in MATLAB scripts.
- Promoting the student’s skills for making decisions and choosing creative solutions design which will be applied in the final solution and in the definition of a set of experiments to prove performance.

## 1.5 Report structure

The current paper is organized in eight chapters and seven appendices.

In Chapter 2, a general idea of channel characterization and modeling are presented. And the main underlying theoretical concepts about channel modeling are given.

In Chapter 3, a general idea of multiuser diversity is showed. In order to improve the throughput of multiple packet-data users sharing a wireless channel while preserving fairness, different scheduling algorithms in different scenarios are proposed.

In Chapter 4, fairness which is one of the most important properties in a wireless environment where system resources are sharing is introduced and discussed. And the different behaviour from different scheduling policies on index fairness is showed.

In Chapter 5, throughput which is one of the most important properties of an information system is introduced and discussed. In this chapter the different behaviour from different scheduling policies is evaluated in different scenarios.



In Chapter 6, delay which is one of the most important properties of current applications is introduced and discussed. In this chapter the delay of different applications (video, voice and data) is evaluated in different scenarios and with different scheme scheduling to achieve QoS.

Chapter 7 from the main ideas presented throughout the text, the main conclusions are extracted.

Chapter 8 points out possible future works to extend this Thesis.

Appendices are devoted to show the MATLAB scripts which have been used for simulating the theoretical concepts.

In Appendix A, the scripts to simulate the radio channel behaviour are showed according to two different approximations, filtered Gaussian noise process and Jakes model.

In Appendix B, the algorithms which are used by the scheduler in the BS are showed, these are Max Rate, RR, PFS and RCG.

In Appendix C, are showed the scripts which simulate the BS scheduler, the statistics resulted from the scheduler scheme and the script used to evaluate PFS behaviour with different  $t_c$  parameter values.

In Appendix D, the scripts which compute the index fairness and system throughput are showed.

In Appendix E, the script which computes the index fairness and system throughput versus users is showed.

In Appendix F, the script which computes the probability that a user is transmitting less than the demanded rate is showed.



## Chapter 2

# Channel Modeling

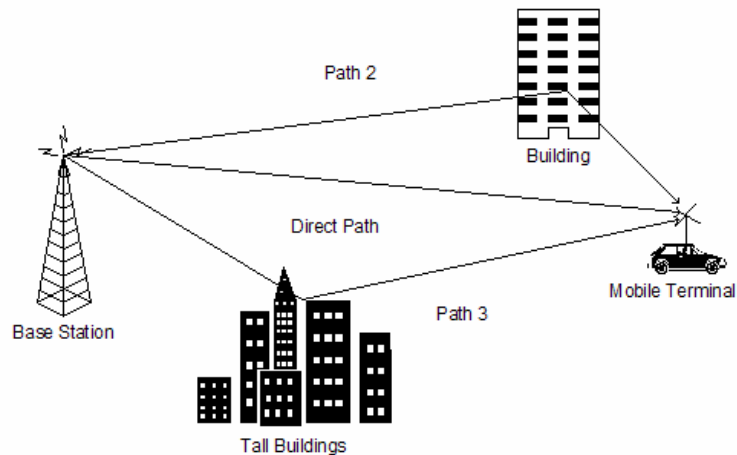
Understanding the behavior of the wireless medium is essential for appreciating the reasoning behind specific designs for wireless communications protocols. In particular, physical-layer and medium-access protocol designs are influenced heavily by the behavior of the channel, which varies substantially in different locations.

The effective design, assessment, and installation of a radio network require accurate characterization of the channel. The channel characteristics vary from one environment to another, and the particular characteristics determine the feasibility of using a proposed communication technique in a given operation environment. Having an accurate channel characterization for each frequency band and a detailed mathematical model of the channel, enables the designer or user of a wireless system to predict signal coverage, achievable data rate, and the specific performance attributes of alternative signaling and reception schemes.

The wireless network that we consider in this Thesis operates at frequency of 2.4 GHz according to IEEE 802.11- based WLANs. Frequencies in the region of few gigahertz have several attractive features for being used in wireless information networks. At these frequencies a transmitter with power of less than 1 W can provide coverage distances on the order of a few miles, as needed for cellular urban radio communications. Furthermore, at these frequencies the size of an efficient antenna can be on the order of an inch, and antenna separation as small as several inches can provide uncorrelated received signals suitable for achieving diversity in the received signal [6].

### 2.1 Fading and Multipath Channels

Most cellular wireless systems operate in built-up areas where there is no direct LOS radio path between the terminals, the transmitter and the receiver and where, due to natural and man-made obstructions (hill, trees, buildings, towers...), multidiffraction, multireflection, and multiscattering effects occurs (see Fig2.1). These cause not only additional losses (with respect to those obtained in LOS above the terrain), but also multipath fading of the signal strength observed at the receiver [9].



**Figure 2.1** Schematic presentation of each component of the multipath signal with two possible paths.

In real communications links, the field that forms the complicated interference picture of received radio waves arrives via several paths; thus, the various waves arrive with different time delays. Upon each reflection of a path from a surface, a certain fraction of the power is absorbed by the surface, and the remainder of the power in that path carries beyond the reflection. At the receiver, such waves combine vectorially at any given frequency to give an oscillating resultant signal, the character of variations of which depends on the distribution of phases among the incoming components waves. For a mobile user, the amplitude of the path changes slowly, but the phase changes rapidly at a rate of  $2\pi/\lambda$  radians per meter [6]. This means that for a mobile with a carrier frequency of 1 GHz every  $\lambda = 1/3$  m we have a  $360^\circ$  change in the phase. The signal amplitude random variations are known as the fading effect. Fading is basically a spatial phenomenon, but spatial signal variations are experienced as temporal variations by a receiver or transmitter moving through the multipath field or due to moving scatters, such as a truck passing through the area between two terminal antennas.

The urban communications channel is approximately stationary in time [7], but the spatial variations of signal level have a triple nature.

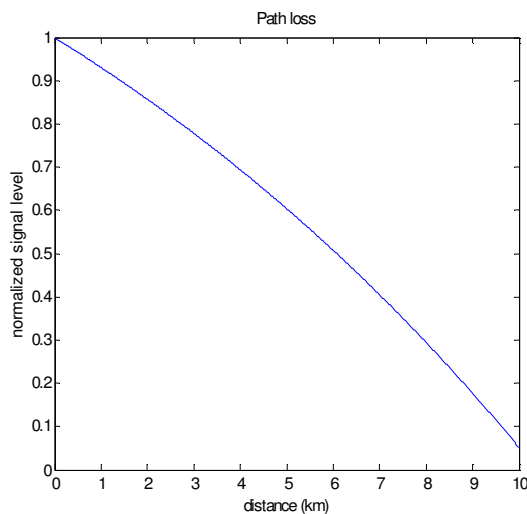
The first one is the path loss, see Fig2.2 (a), which is the simplest of all the propagation mechanisms to understand and reflects the fact that the signal drops as the distance from the transmitter increase. Theory shows that if the transmitter were in free space, then the signal would radiate in an expanding sphere from the point source of the transmitter. Since the surface area of the sphere is proportional to the radius square, the received signal power at a distance  $d$  from the transmitter is proportional to  $1/d^2$ . Free-space loss cannot occur on the Earth since one half of the expanding sphere is under the ground which has a certain reflection and transmission coefficient depending on the material making up the surface of the Earth at that particular point. Of more relevance is the fact that there will be obstructions on the ground in the form of buildings, hills, and vegetation, for example. These absorb and reflect the signal, resulting in a received signal strength that is much lower than that predicted using free-space loss. Because of the complexity of modeling every building, general guidelines are adopted as to the loss likely to be experience. Measurements

have shown that in an urban area, the environment which we will simulate, if the path loss is modeled as being proportional to  $1/d^3$ , then the results achieved best reflect real life.

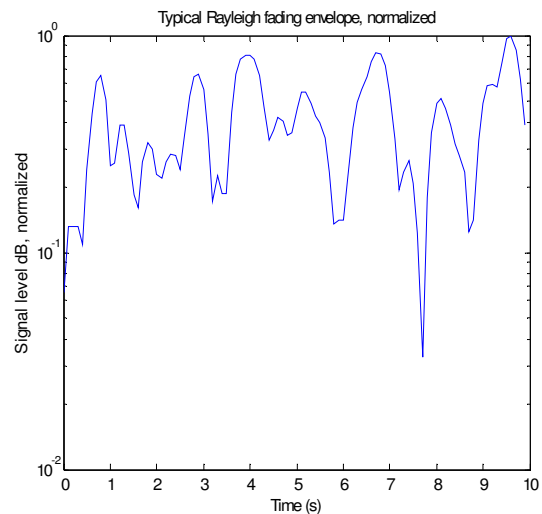
There are a few isolated cases where path loss exponents lower than the exponents of 2 predicted by free space are experienced. These typically occur in constrained spaces, often in corridors in a building. Here, the signal does not expand on the surface of a sphere because the walls of the corridor cause the signal travelling toward them to be reflected back into the corridor. Because the signal is now moving forward on a surface that is not expanding, theory would predict that no loss in signal strength will occur. In practice, some signal leaks through the corridor walls and exponents of around 1.6 to 1.8 can be realized [8].

The second one is shadowing or slow fading, see Fig2.2 (b), is the second type of fading and is caused by diffractions, scattering, and multiple reflections, which all show slow random variations in signal amplitude with the tendency to a Gaussian distribution. The spatial scale of the slow variations is up to several tens of meters, which is normal distributed [7].

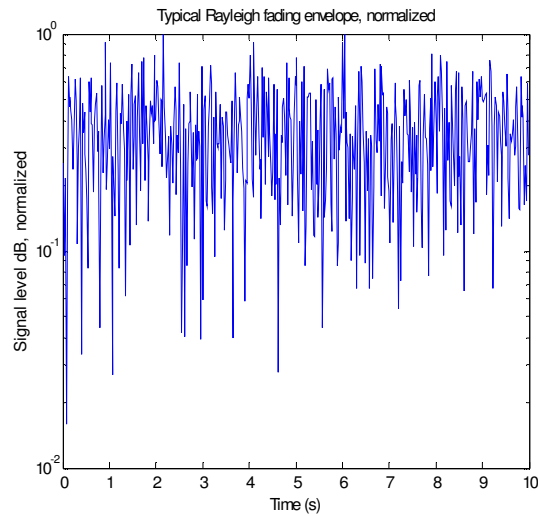
The third one is fast fading, random variations of the signal, see Fig2.2 (c), caused by the mutual interference of the wave components of the multiray field. The characteristic scale of such waves in the space domain is changed from a fraction of a wavelength to several wavelengths [7]. They are usually called small-scale or fast-fading signals in the literature.



(a)



(b)



(c)

**Figure 2.2** Multipath phenomena, (a) path loss, (b) slow fading, (c) fast fading.

## 2.2 Channel Characterization

In this section the channel will be described mathematically and we will discuss when a channel is said slow fading or fast fading in terms of symbol duration and bandwidth.

### 2.2.1 Statistical characterization of multipath channel

The transmitted signal follows many different paths before arriving at the receiving antenna, and it is the aggregate of these paths that constitutes the multipath radio propagation channel. The resulting signal strength will undergo large fluctuations, which, when the signal is small, results in a “fade”. Multipath fading manifests itself in two effects [9]:

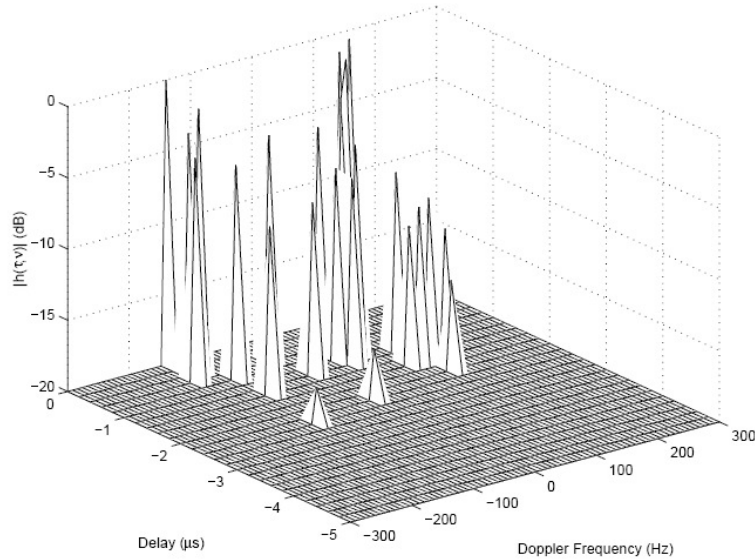
1. Time spreading (in  $\tau$ ) of the symbol duration within the signal, which is equivalent to filter and band-limit.
2. A time-variant behaviour (in  $t$ ) of the channel due to motion of received or changing environment such as movement of foliage or movement of reflectors and scatters.

The small-scale variations of a mobile radio signal can be directly related to the impulse response of the mobile radio channel. The impulse response (Fig2.3) is a wideband characterization and contains all information necessary to simulate or analyze any type of radio transmission through the channel. This comes from the fact that a mobile radio channel

may be modeled as a linear filter with a time varying impulse response, where the time variation is due to receiver motion in space. The filtering nature of the channel is caused by the summation of amplitudes and delays of the multiple arriving waves at any instant of time. The impulse response is a useful characterization of the channel, since it may be used to predict and compare the performance of many different mobile communication systems and transmission bandwidth for a particular mobile channel condition. Then, we can describe the multipath channel by a time-varying, complex, lowpass-equivalent impulse response

$$\tilde{c}(\tau, t) = \sum_n \tilde{a}(\tau_n, t) \delta(\tau - \tau_n) \quad (1)$$

where  $\tilde{a}(\tau_n, t)$  is the complex attenuation of the signal component at delay  $\tau$  and time instant  $t$ .



**Figure 2.3** Impulse response of the mobile radio channel.

The random fluctuation in the received signal due to fading can be modeled by treating  $\tilde{c}(\tau, t)$  as a random process in  $t$ . Since the components of the multipath signal arise from a large number of reflections and scattering from rough or granular surfaces, then by virtue of the central limit theorem [9],  $\tilde{c}(\tau, t)$  can be modeled as a complex Gaussian process. At any time  $t$ , the PDFs of the real and imaginary parts are Gaussian. This model implies that for each  $\tau$  the ray is composed of a large number of unresolvable components. Hence, the channel is complex Gaussian process in  $t$  and if  $\tilde{c}(\tau, t)$  has zero-mean, then the envelope  $|\tilde{c}(\tau, t)|$  has a Rayleigh PDF, eq. (2). In next sections we will describe more accurately the channel models used in this Thesis.

$$f_R(r) = \frac{r}{\sigma^2} e^{-r^2/(2\sigma^2)} \quad (2)$$

### 2.2.2 The Power Delay Profile

An example of the *power delay profile* (PDP) is showed in Fig2.5 (a). Here, the term delay, refers to excess delay. It represents the delay measured from the first perceptible signal that arrives at the receiver. The maximum excess delay, also termed maximum delay spread,  $T_m$  is the delay between the first and the last component of the signal during which the received power fall below some threshold level, e.g. 20 dB below the strongest component. The relationship between  $T_m$  and the symbol time  $T_{sym}$  can be viewed in terms of two different degradation criteria [9]:

1. A channel is said to exhibit frequency-selective fading if  $T_m > T_{sym}$ . In this condition, the multipath components extend beyond the symbol duration, which causes ISI distortion in the signal. Since the multipath is resolvable in this case, the ISI distortion can be mitigated by rake reception or equalization.
2. A channel is said to exhibit frequency-nonselective or flat fading if  $T_m < T_{sym}$ . In this case there is very little ISI, but we still have distortion in the system because the multipath signal can add destructively, reducing the SNR considerably.

The dividing line between frequency-selective and flat fading is not perfectly sharp. In channels that we call flat, frequency selective still happens, but with smaller probability.

### 2.2.3 The Spaced-Frequency Correlation Function

As we can see in Fig2.5 (b), a completely analogous characterization of signal dispersion can be done in the frequency domain. The function is defined as the Fourier transform of the PDP. Spaced-frequency correlation function represents the correlation between the channel response to two narrowband signals with the frequencies  $f_1$  and  $f_2$  as a function of the difference  $\Delta f = f_2 - f_1$ . This function can be thought of as the transfer of the channel. Therefore, time spreading can be viewed as if it was the result of a filtering process. The coherence bandwidth  $f_0$  is defined as the frequency range where all frequency component amplitudes are correlated. That is, the spectral components in that range fade together. It can be shown that  $f_0$  and  $T_m$  are reciprocally related. As a rule of thumb, it is usually assumed that



$$f_0 \approx \frac{1}{T_m} \quad (3)$$

For the case of the mobile radio, an array of radially uniformly spaced scatters, all with equal-magnitude reflection coefficients, but independent, randomly occurring phase angles, is widely accepted. This model is referred as the dense scatter channel model, and is commonly known as the Jakes model which will be described in section 2.3.2.

The two degradation criteria above for frequency-selective and flat fading can also be expressed in terms of the coherence and signal bandwidths:

1. A channel is referred as frequency-selective if  $f_0 < B$ , where  $B$  is the signal bandwidth. Since the channel coherence bandwidth is smaller than the signal bandwidth, the channel acts as a filter, hence frequency-selective fading occurs.
2. Frequency-nonselective or flat fading degradation occurs whenever  $f_0 \gg B$ . As noted before, flat fading does not introduce ISI, but performance degradation occurs due to low SNR whenever the signal is fading. It should be noted that frequency selectivity diminishes as  $f_0/B$  is increasingly larger than 1.

Flat fading is not always desirable. For example, to achieve frequency diversity for two fading signals, it is necessary that the carrier spacing  $f_s$  between the two signals is larger than the coherence bandwidth,  $f_s > f_0$ , so that the two signals are uncorrelated [9].

#### 2.2.4 The Time-Varying Channel

In this and the next section will be described the properties of the channel as they relate to its time-varying nature. For mobile radio applications, the channel is time-varying because the motion between the transmitter and the receiver results in propagation path changes. It should be noted that since the channel characteristics are dependent on the relative position of the transmitter and receiver, time variance is equivalent to space variance. The time variation of the channel is characterized by the Doppler power spectrum  $S(\nu)$ , shown in Fig2.5 (c).

$$S(\nu) = \frac{1}{\pi f_{d \max} \sqrt{1 - \left(\frac{\nu}{f_{d \max}}\right)^2}} \quad |\nu| \leq f_{d \max} \quad (4)$$

where as it will be seen in section 2.2.4.2,  $f_{d \max}$  is the maximum Doppler frequency shift.

### 2.2.4.1 The Spaced-Time Correlation Function

The spaced-time correlation function  $\rho(\Delta t)$ , Fig2.5 (d) is the inverse Fourier transform of  $S(\nu)$ , and it specifies the correlation between the channel's response to a narrowband signal sent at times  $t_1$  and  $t_2$ , where  $\Delta t = t_2 - t_1$ . The coherence time,  $T_0$ , is the expected time duration within the two signals remain correlated. If the channel is time-invariant,  $\rho(\Delta t) = 1$ . Coherence time can also be measured in distance traversed [9].

The time-variant behaviour is categorized into fast fading and slow fading:

1. A channel is said to be fast fading if  $T_0 < T_{sym}$ , where  $T_0$  is the channel coherence time and  $T_{sym}$  is the symbol time. During the fast fading the baseband symbol shapes can be severely distorted, which often results in an irreducible BER and synchronization problems.
2. A channel is referred as slow fading if  $T_0 > T_{sym}$ . The time duration that the channel remains correlated is long compared to the transmitted symbol. The primary in a slow fading channel is the loss of SNR.

### 2.2.4.2 Doppler Power Spectrum

The Doppler power spectrum has been shown to match experimental data gathered for outdoor mobile radio channels. For indoor mobile radio channels a flat spectrum is used [9].

Consider a mobile moving a constant velocity  $v$ , along a path segment having length  $d$  between points  $X$  and  $Y$ , while it receives signals from a remote source  $S$ , as illustrated in Fig2.4. The differences in path lengths travelled by the wave from source  $S$  to the mobile at points  $X$  and  $Y$  is  $\Delta l = d \cos \theta = v \Delta t \cos \theta$ , where  $\Delta t$  is the time required for the mobile to travel from  $X$  to  $Y$ , and  $\theta$  is assumed to be the same at points  $X$  and  $Y$  since the source is assumed to be very far away. The phase change in the received signal due to the difference in path lengths is therefore

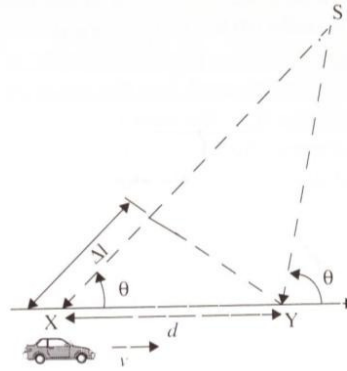
$$\Delta \varphi = \frac{2\pi \Delta l}{\lambda} = \frac{2\pi v \Delta t}{\lambda} \cos \theta \quad (5)$$

and hence the apparent change in frequency, or Doppler shift, is given by  $f_d$ , where

$$f_d = \frac{f_c v}{c} \cos \theta \quad (6)$$

the maximum Doppler frequency is

$$f_{d \max} = \frac{f_c v}{c} \quad (7)$$



**Figure 2.4** Mobile moving along a path segment, while it receives signal from  $S$ .

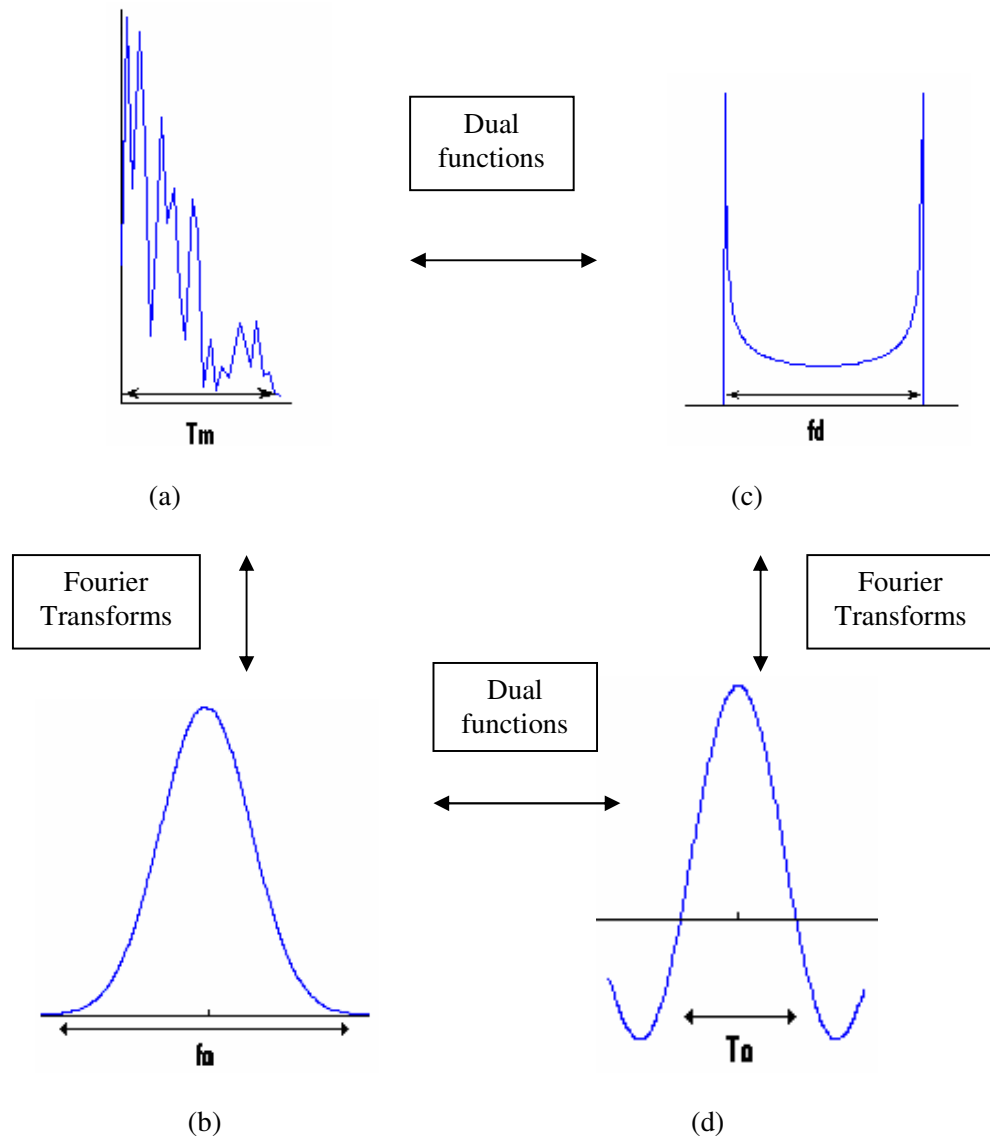
The Doppler power spectrum of the channel yields knowledge about spectral broadening of a narrowband signal (impulse in frequency) in the Doppler frequency domain. It can be regarded as a dual of the PDP, since the latter yields knowledge about the spreading of a pulse in the time domain.

$S(\nu)$  enable us to estimate the broadening imposed on the signal as a result of the channel time variations. The width of the Doppler power spectrum,  $f_{d \max}$  is referred to as the spectral broadening or Doppler spread, and is also called the fading bandwidth of the channel. Because  $S(\nu)$  and  $\rho(\Delta t)$  are related to the Fourier transform, the coherence time and the Doppler spread are inversely related as

$$\rho(\Delta t) = F^{-1}(S(\nu)) = J_0(2\pi f_d \Delta t) \quad (8)$$

The degradation criteria for fast and slow fading in terms of the signal bandwidth  $B \approx 1/T_{sym}$  and the fading rate  $f_{d \max} \approx 1/T_0$  can be stated as follows:

1. A channel is considered fast fading if  $B < f_{d \max}$ . In this case the signal is severely distorted.
2. A channel is regarded as slow fading if  $B > f_{d \max}$ . No signal distortion is present, but degradation of the SNR is possible.



**Figure 2.5** Relationships among the channel correlation functions and PDFs, (a) multipath intensity profile, (b) coherence bandwidth, (c) Doppler power spectrum, (d) spaced-time correlation function.

### 2.3 Channel Model

In the previous sections we looked at the statistical characterization of multipath channels. The characterization does not in itself provide a constructive way of emulating the channel. For this purpose we need to synthesize a generative model.

Simulation of envelope fading is very important for design and performance evaluation of wireless modems because often we cannot find closed-form solutions to compare performance of various modulation and coding techniques over wireless channels.

To simulate a narrowband channel, we need to generate a random process with a specific envelope fading density function and a specific Doppler spectrum. A wideband simulator is a group of narrowband simulators with different gains connected together through a tapped delay line. After generating a random variable with the distribution function of envelope fading, passing the random variable through a filter with a specific spectral shape, resembling the Doppler spectrum of the channel. One may instead, to generate a series of oscillators with different frequencies and add the outputs to form the specific spectrum. The first approach has been used extensively in simulations of a variety of fading channels. The second approach is often used in simulations of mobile radio channel, based on the Clarke assumption of isotropic scattering [10]. We have proposed these two approaches to simulate the envelope fading characteristics. The envelope fading considered is Rayleigh fading, so we will consider that there is no direct LOS between transmitter and receiver.

### 2.3.1 Filtered Gaussian Noise

A widely approach to simulation of fading radio channels is to constructs a fading signal from in-phase and quadrature Gaussian noise sources. Because the envelope of a complex Gaussian noise process has a Rayleigh PDF, the output of such simulator will simulate Rayleigh fading accurately. In this approach, applying the appropriate filtering to the Gaussian noise sources provides the Doppler spectrum of the channel of interest. Fig2.6 shows a block diagram of the basic technique for simulating Rayleigh fading as an RF signal using a filtered Gaussian noise process, at each tap line.

If a multipath channel is composed of a set of discrete resolvable components that originate as reflections or scattering from smaller structures, e.g., houses, small hills, etc., it is called a discrete multipath channel. The model in its most general form has, in addition to variable tap gains, variable delays and variable number of taps.

The lowpass-equivalent impulse response of a discrete mulipath channel is given as

$$\tilde{c}(\tau, t) = \sum_{k=1}^{K(t)} \tilde{a}_k(\tau_k(t), t) \delta(\tau - \tau_k(t)) \quad (11)$$

For many channels it can be assumed as a reasonable approximation that the number of discrete components is constant and the delay values vary very slowly and can also be assumed constant. These assumptions are also made for “reference” channels that are for system studies [9]. The model then simplifies to

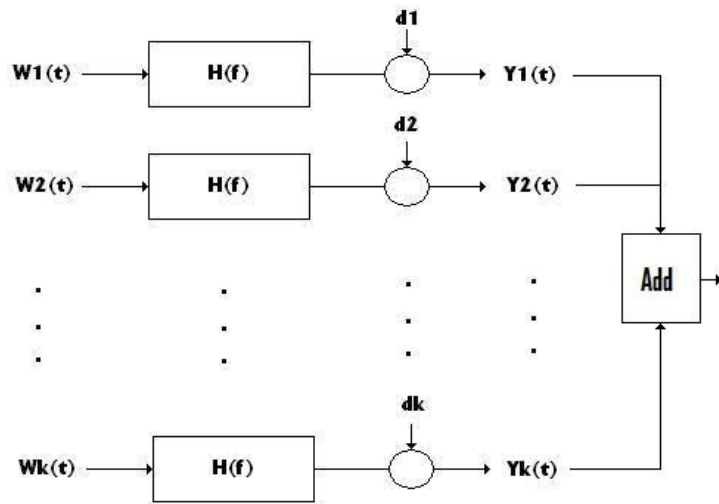
$$\tilde{c}(\tau, t) = \sum_{k=1}^K \tilde{a}_k(t) \delta(\tau - \tau_k) \quad (12)$$

where  $\tilde{a}_k(t)$  and  $\tau_k$  are the complex tap attenuation and tap delay respectively, as it will be seen in section 2.4.3, different tap attenuations will be simulated according to different scenarios.

The number of taps needed by the band-limited model is usually small. We will determine the number of taps estimating the band-limited PDP determining the maximum delay power spread  $T_m$  at which the magnitude of the delay spread is still relevant. In these simulations, the signal components which the receiver power fall below 20 dB of the strongest component will not be considered.

As we can see in Fig2.6, the generation of the tap-gain process for the discrete multipath channel model is straightforward. It starts with a set of  $K$  independent complex processes ( $W_i(t)$ ,  $i=1,2,\dots,k$ ), where the magnitude is a complex Gaussian variable with zero-mean and unit variance, and the phase is uniform distributed between zero and  $2\pi$ . This process is filtered to produce the appropriate Doppler spectrum ( $H(f)$ ), then scale them to produce the desired amplitude of the discrete channel ( $d_i$ ,  $i=1,2,\dots,k$ ). Finally all taps have to be added to model the multipath channel.

This technique was recommended by the JTC standardization committee for the simulation of channel fading [30].



**Figure 2.6** Tap delay line.

### 2.3.2 Jakes Model for Simulations of a Mobile Radio Channel

As an alternative to RF modeling with filtered complex Gaussian noise, one may instead approximate the Rayleigh fading process by summing a set of complex sinusoids. The number of sinusoids in the set must be sufficiently large that the PDF of the resulting envelope provides an acceptably accurate approximation to the Rayleigh PDF. With this modeling method, the sinusoids are weighted so as to produce an accurate approximation of the desired channel Doppler spectrum. One technique of this type is proposed by Jakes [11].

Jakes shows that the theoretical Doppler spectrum for the isotropic scattering mobile radio channel, can be well approximated by a summation of a relatively small number of sinusoids, with the frequencies and relative phases of the sinusoids set according to a specific formulation. As we said in previous section, the maximum Doppler shift frequency is  $f_m = v_m / \lambda$ , where  $v_m$  is the velocity of the mobile and  $\lambda$  is the wavelength of the carrier frequency. In the model described by Jakes, the ideal isotropic continuum of arriving scatter components is approximated by  $N$  plane waves arriving at uniformly azimuthal angles. The model restricts  $N/2$  to be an odd integer and defines another integer  $N_0 = 1/2(N/2 - 1)$ . This leads to a simulation model having one complex frequency oscillator with frequency  $\omega_m = 2\pi f_m$  plus a summation of  $N_0$  complex lower-frequency oscillators with frequencies equal to the Doppler shifts  $\omega_m \cos \theta_n$ , where  $\theta_n$  is the arrival angle for the  $n$ th plane wave and where  $n=1,2,\dots, N_0$ . Each oscillator has an initial phase, and these phases are to be chosen as a part of initializing the simulation. We can express the complex envelope  $T(t)$  of the fading signal in the form

$$T(t) = \frac{E_0}{\sqrt{2N_0 + 1}} (x_c + jx_s) \quad (9)$$

where

$$\begin{aligned} x_c(t) &= 2 \sum_{n=1}^{N_0} \cos \phi_n \cos \omega_n t + \sqrt{2} \cos \phi_N \cos \omega_m t \\ x_s(t) &= 2 \sum_{n=1}^{N_0} \sin \phi_n \cos \omega_n t + \sqrt{2} \sin \phi_N \cos \omega_m t \end{aligned} \quad (10)$$

and where  $\omega_n = \omega_m \cos(2\pi n / N)$ ,  $n=1,2,\dots,N_0$ . In the equations above,  $\phi_N$  is the initial phase of the maximum Doppler frequency sinusoid, and  $\phi_n$  is the initial phase of the  $n$ th Doppler-shifted sinusoids. The quantities  $x_c$  and  $x_s$  are the in-phase and quadrature components respectively.

In using this simulation method, one must choose the initial phases ( $\phi_n$  and  $\phi_N$ ) of the Doppler shift components in such a way that the phase of the resulting fading process exhibit a distribution as closed as possible to uniform, here we use  $\phi_N=0$  and

$\phi_n = \pi n / (N_0 + 1)$ , where  $n=1,2,\dots,N_0$  [3]. The number of Doppler-shifted sinusoids is chosen large enough that  $T(t)$  provides a good approximation to a complex Gaussian process, and therefore the envelope  $|T(t)|$  is approximately Rayleigh. Jakes suggest that  $N_0 = 8$  provides an acceptably accurate approximation to the ideal case of Rayleigh fading [11].

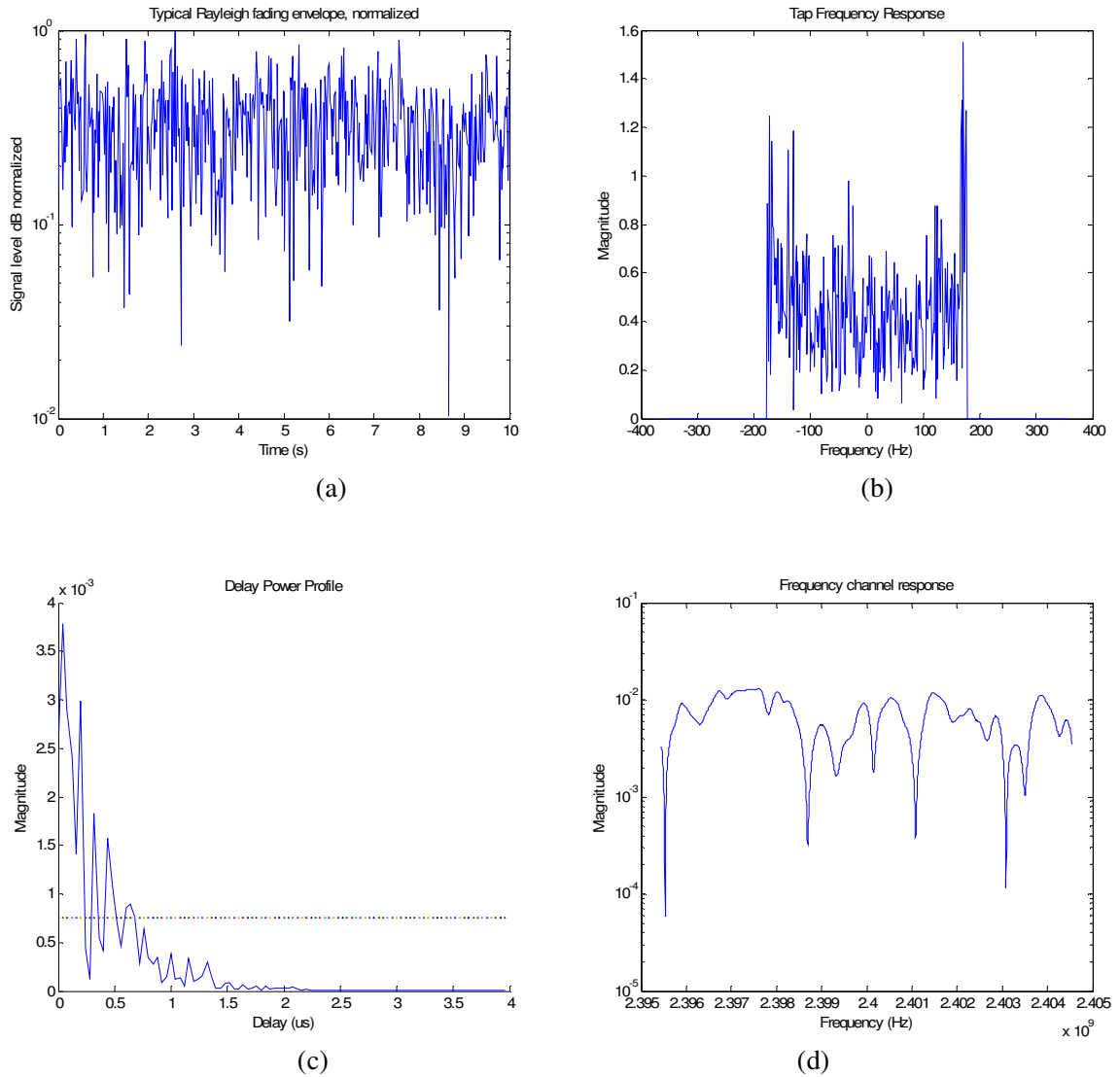
## 2.4 MATLAB Simulations

According to theoretical models explained in previous sections, two scripts have been programmed in MATLAB in order to simulate fading radio channel behaviour. The first script, *FGN\_model.m* showed in Appendix A.1, simulates fading radio channel according to section 2.3.1, and the second script, *jakes\_model.m* showed in Appendix A.2, simulates fading radio channel according section 2.3.2.

### 2.4.1 Filtered Gaussian Noise Model

As we could see in section 2.3.1, fading radio channel can be constructed from in-phase and quadrature Gaussian noise sources. In Fig2.7 (a) typical Rayleigh fading envelope of a radio channel has been plotted. As we can see, the signal has a lot of fades and peaks, each fade corresponds with a destructive interference between taps arriving from different paths and peaks correspond with constructive ones. In Fig2.7 (b) the Doppler power spectrum is plotted,  $f_{d \max}$  is about 180 Hz because of mobile speed, 80km/h. In Fig2.7 (c), an estimate PDP for a certain estimation window has been plotted, which simulates an urban environment ( $T_m$  is about 0.5 $\mu$ s), as we will see in section 2.4.3. And finally, in Fig2.7 (d) the frequency channel response has been plotted, with a bandwidth of 10 MHz, according to the delay spread.



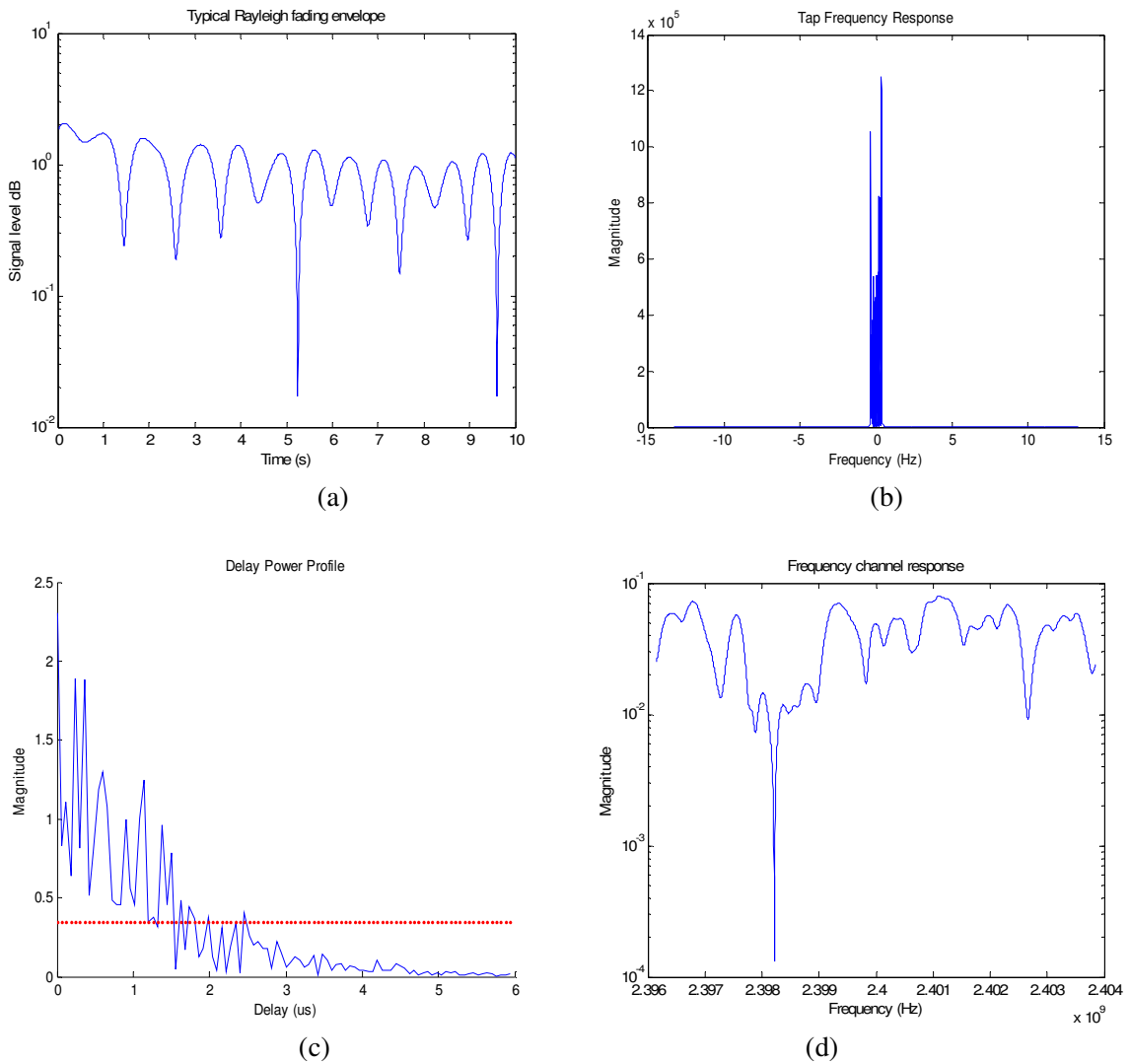


**Figure 2.7** Filtered Gaussian Noise model for simulations of a mobile radio channel, (a) fading envelope, (b) Doppler power spectrum, (c) estimate PDP, (d) frequency channel response.

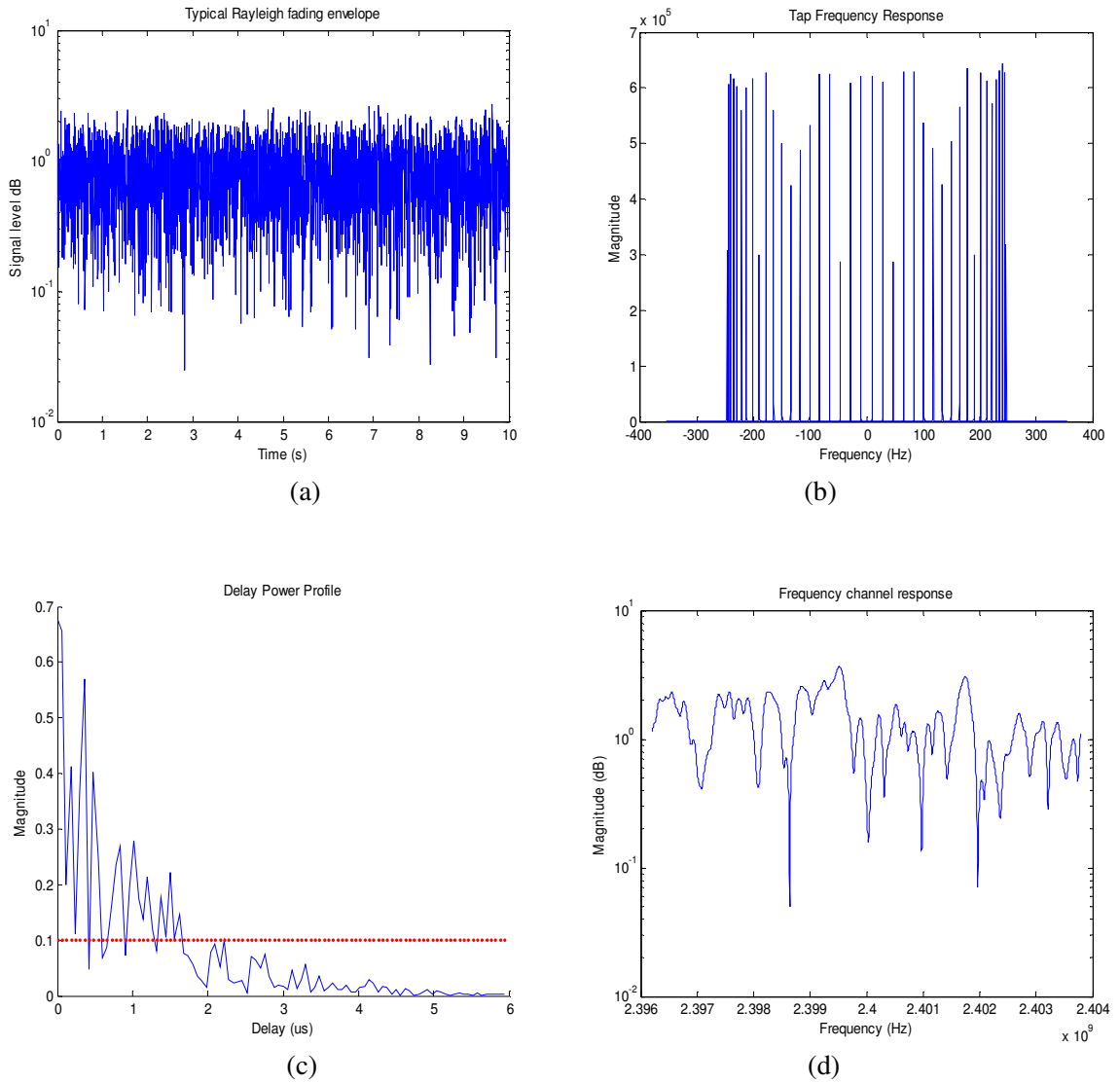
### 2.4.2 Jakes Model

As we could see in section 2.3.2, fading radio channel can be constructed by summing a set of complex sinusoids. Two different scenarios have been simulated; when the user is walking (3km/h) and when the user is driving (80km/h), see Fig2.8 (a) and Fig2.9 (a) respectively. When the user is walking, the fading radio channel is showed as slow fading but

when the user is driving, the fading radio channel looks like fast fading. The Doppler spectrum is larger when the user is driving because of mobile speed, it can be compared in Fig2.8 (b) and Fig2.9 (b). The frequency channel response, Fig2.8 (c) and Fig2.9 (c), has fewer fluctuations when the user is walking because of speed differences.



**Figure 2.8** The Jakes model for simulations of a mobile radio channel when the user is walking (3km/h), (a) fading envelope, (b) Doppler power spectrum, (c) estimate PDP, (d) frequency channel response.



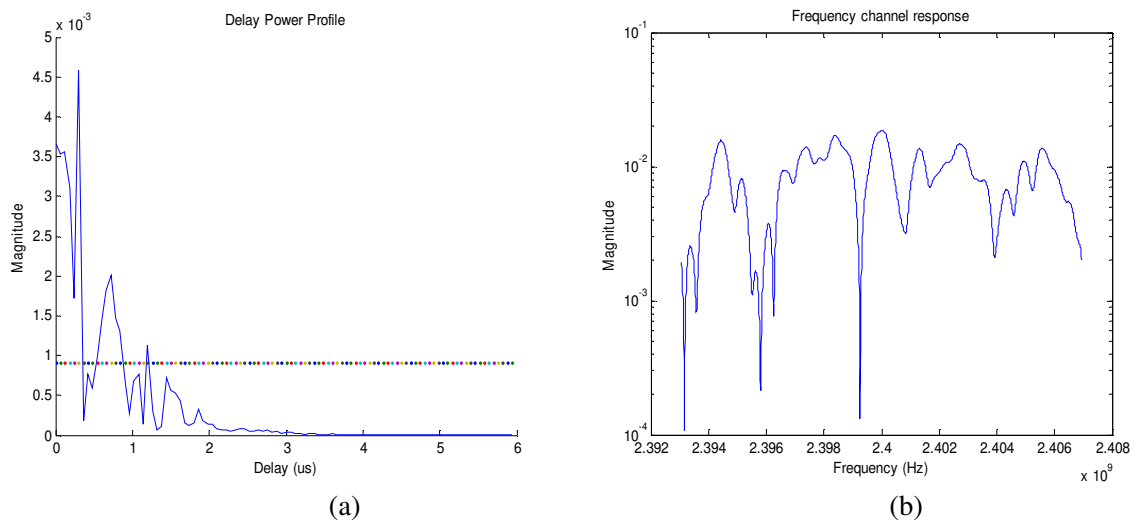
**Figure 2.9** The Jakes model for simulations of a mobile radio channel when the user is driving (80km/h), (a) fading envelope, (b) Doppler power spectrum, (c) estimate PDP, (d) frequency channel response.

### 2.4.3 Different Power Tap Delays

According to different environments exist different PDPs, so different Doppler spectra have been used for different delays. Urban area, hilly area and flat terrain have been considered as three sorts of possible environments.

### 2.4.3.1 Urban Area

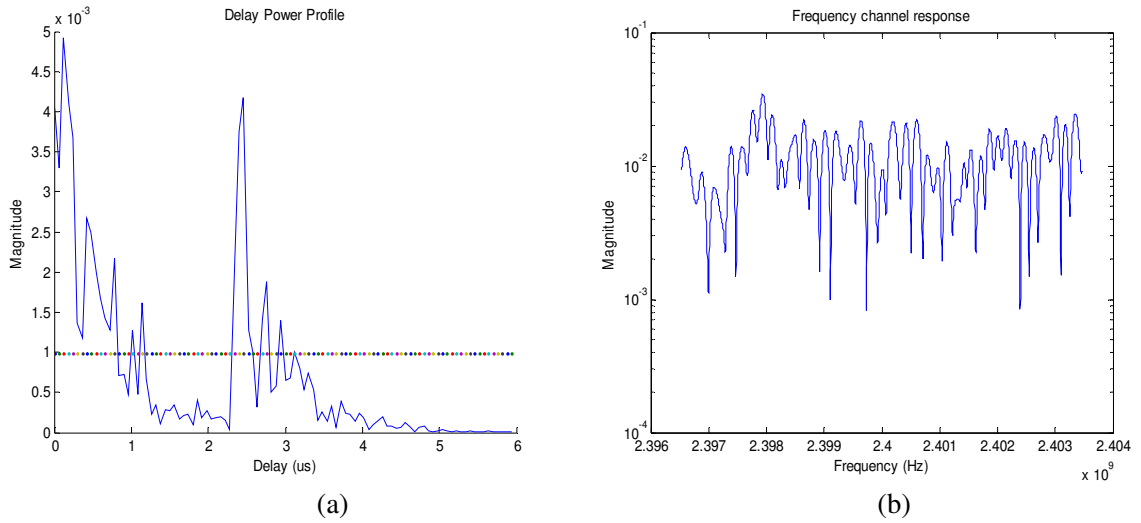
As is well known, urban areas, from the point of view of radio wave propagation, are complicated areas and the analytical description of this process cannot be presented without some simplifications for the real radio link situation [7]. For 2.4 GHz radio waves, most large city buildings are practically nontransparent with dimensions greater than the wavelength,  $\lambda$ . In such a situation, a wide spectrum of shadow zones is observed at the street level and it is very complicated to use well-known analytical models. In this Thesis, the urban environment has been simplified to an estimate PDP for a certain estimation window with a time spread  $T_m$  between 0.5 and  $3\mu\text{s}$  and with exponential decay, see Fig2.10.



**Figure 2.10** Urban Area where users are walking (3km/h) (a) estimate PDP, (b) frequency channel response.

### 2.4.3.2 Hilly Area

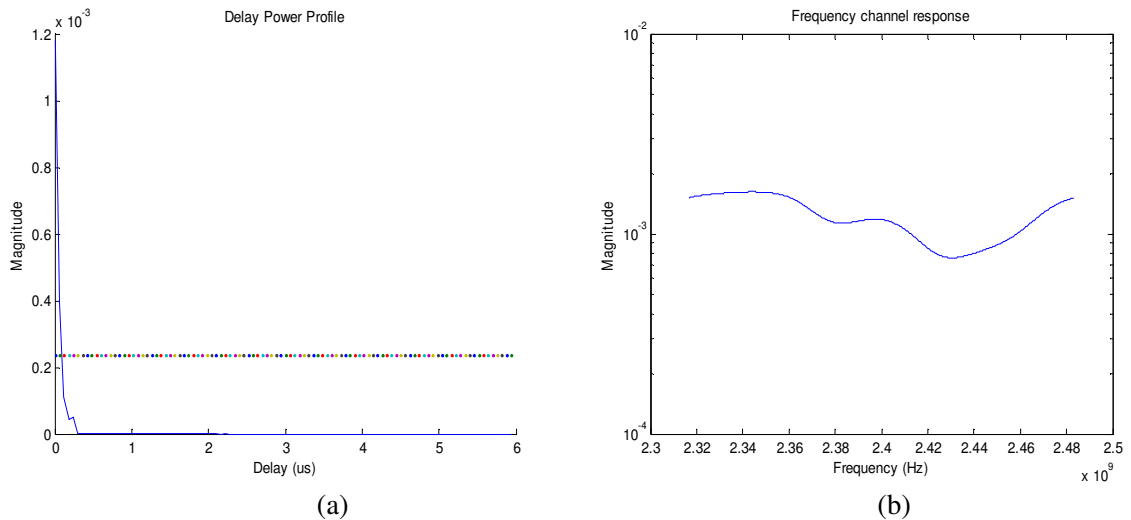
In a region with a hilly terrain the channel sounder is able to differ between the dominant peak and a few additional paths. The amplitude of these paths is few dBs below the main peak [12]. The estimate PDP for a certain estimation window has the shape in Fig2.11.



**Figure 2.11** Hilly Area where users are driving (80km/h) (a) estimate PDP, (b) frequency channel response.

### 2.4.3.3 Flat Terrain

The radio channel in flat terrain is characterized by the absence of detected multipath propagation, so the channel impulse response looks like one path and the frequency channel response is almost flat, see Fig2.12.



**Figure 2.12** Almost one path (a) estimate PDP, (b) frequency channel response.



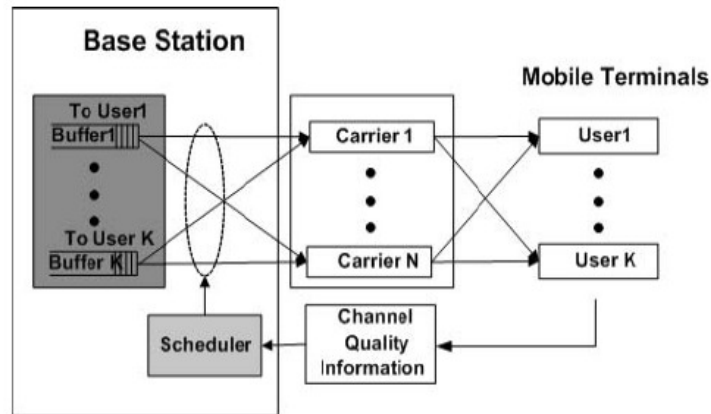
## Chapter 3

# Scheduling Algorithms

In this chapter, different scheduling algorithms have been proposed which are working in different scenarios. The goal is to improve the throughput of multiple packet-data users sharing a wireless downlink channel while preserving fairness. RR and Max Rate algorithms will take into account to improve fairness and throughput respectively and two proportional fair scheduling algorithms, PFS and RCG which maximize system throughput maintained fairness. Scheduling algorithms for wireless packet data networks exploit the fact that the propagation channels between the BS and the MSs it serves fade independently by scheduling transmission to MSs when their channels are good, giving rise to multiuser diversity [13].

The system under consideration is an OFDM system with frequency-division multiple access (FDMA) and time-division multiple access (TDMA). Perfect channel state information is assumed at both the MS and the BS, i.e. the channel gain on each sub-carrier due to path loss, shadowing, and multipath fading is assumed to be known.

Each sub-carrier can only be used by one user at any given time. Sub-carrier allocation is performed at the BS and the users are notified of the sub-carriers chosen for them. Consider a system with  $K$  users,  $N$  sub-carriers and the time divided in time slots. At each time slot, each scheduling user  $k$  will transmit on the allocated sub-carrier. Equal power allocation algorithms are taken in consideration, which simply distributes the transmission power equally among the sub-carriers. The objective is to find a sub-carrier allocation, which allows each user to satisfy its rate requirements, maximizing throughput without comprising fairness. The system is showed in Fig3.1.



**Figure 3.1** Multi-carrier transmission system.

### 3.1 Multiuser Diversity

Traditionally, diversity can be achieved over fading channels either over space (multiple antenna in reception and/or transmission), over time (interleaving) or in frequency (use of RAKE receiver in spread spectrum systems). In a multiple access wireless network, another source of diversity, called multiuser diversity [15], can be obtained exploiting the time-varying characteristic of the users' channels. To maximize the total information-theoretic capacity, they showed that the optimal strategy is to schedule at any one time only the user with the best channel to transmit to the BS. Diversity gain arises from the fact that in a system with many users, whose channels vary independently, there is likely to be a user whose channel is near its peak at any one time. Overall system throughput is maximized by allocating at any time the common channel resource to the user that can best exploit it. It can also be thought of as a form of selection diversity.

The fading rate and the dynamic range of the channel fluctuations are essential parameters for the exploitation of multiuser diversity. In fact, the larger the fluctuation, the larger the diversity gain. For example, if the channel varies too slowly compared to the delay constraint of the application, the scheduler might not be able to wait long enough for the channel to reach its peak [14]. However, as motion increases, the diversity gain will be reduced by the errors in the channel estimations provided to the scheduler through the feedback channel, thus leading to a multiuser diversity gain-mobility trade-off.

To exploit multiuser diversity in a real communication system, one must take into account two main issues: fairness and delay. In general, the statistics of the users' channels are different, and a scheduling policy of always giving the channel to the best user can lead to unfairness. In fact, usually the users in a cell are placed at different distances from the BS, and the shadowing characteristics change from location to location so that it is more likely that the BS will be transmitting always to the nearest user in the cell. Delay is not a tight bound as it is for voice applications but nevertheless we have to insure a QoS to the users, for example, in the form of a maximum average delay of the packets.



### 3.1.1 Multiuser Diversity and Mobility

It is clear that for pedestrian speeds diversity gain is the main phenomenon involved. However, as velocity of the users increases, the loss in performance is dramatic. As the Doppler spread,  $f_d = v/\lambda$  increases, the coherence time of the channel becomes shorter, and the rate request is less reliable. Fast fading may affect not only multiple-slot packets, but also single-slot ones. In fact, it can happen that the rate requested based on the pilot channel measurement made in the previous slot is not supported by the channel in the actual slot, and thus the packet can be lost.

As the speed of the users increases, this measured SNR is a less reliable prediction of the current state of the channel. Suppose we split the slot into two semi-slots, each with a pilot channel (of half length). This would double the feedback channel rate, but not result in a dramatic increase in the amount of data (with 4 bits we can address the 11 rates that the mobiles can request on the feedback channel). Naturally, the number of slots per packet is doubled, but the packets have the same duration in time. Hence we can now better exploit the multiuser diversity effect and, at the same time, reduce the impact of the coherency of the channel for high speeds. In fact, the pilot measurement is a more accurate prediction of the channel state because it is carried out twice as often. Also, if the packet duration is more than one slot, one can better adapt the transmission to the fading characteristics of the channel. Suppose we are transmitting the  $i$ th slot of an  $n$ -slots packet at a rate  $R_k$ , and that for the current slot the rate request of the user is  $R_i < R_k$ . One can avoid transmitting the  $n-i$  slots remaining, because they would probably be lost. Instead, we stop the transmission to the user (completing only the current slot), and restart the proportional fair algorithm for the assignment of the downlink. To ensure fairness, the fairness index is not computed on the whole packet, but only on the basis of the actual data transmitted to the MS ( $i$  slots in this case). The improvement is large, and the cost is only a small amount of additional information to feedback to the BS.

## 3.2 Round Robin algorithm

*Round Robin* (RR) is one of the simplest scheduling algorithms, which assigns time slices to each MS in equal portions and in order, handling all MSs as having the same priority. RR scheduling is both simple and easy to implement, and starvation-free.

In wireless networks, where many stations share one channel, this algorithm provides every MS to transmit or receive on the shared channel at a regular interval. This may make RR to appear a fair algorithm. However, because it is much less efficient than other algorithms such as PFS or RCG (as we will see in sections 3.4 and 3.4 respectively), hardly will provide very good service to the MS. The BS will also suffer reduced capacity of the network. The main reason for this is that this algorithm does not take into account the changing reception conditions at the different receivers or as we said before, RR algorithm does not take into account multiuser diversity and thus it will schedule transmissions to/from subscribers half of the time when their reception conditions are worse than average [17].

### 3.3 Max Rate algorithm

*Max Rate* scheduler transmits always to the user having the largest SNR, so users that are at a fading peak are likely to be scheduled all the time, while others that experience deep fades are not scheduled at all. Max Rate scheduler yields higher throughput than any other possible scheduling policy, but it totally ignores fairness. In wireless environments strength users' channel can be very different, due to perhaps different distances from the BS, although both channel fluctuate due to multipath fading. So, always picking the strongest user would be highly unfair.

### 3.4 Proportional Fair Scheduling algorithm

To implement the idea of multiuser diversity in a real system, one is immediately confronted with two issues: fairness and delay. In the ideal situation when users' fading statistics are the same, the strategy above maximizes not only the total capacity of the system but also the throughput of individual users. In reality, the statistics are not symmetrical; there are users who are closer to the BS with a better average SNR; there are users who are stationary and some that are moving; there are users which are in a rich scattering environment and some with no scatters around them. Moreover, the strategy is only concerned with maximizing long-term average throughputs; in practice, there are latency requirements, in which case the average throughputs over the delay time scale is the performance metric of interest. The challenge is to address these issues while at the same time exploiting the multiuser diversity gain inherent in a system with users having independent fluctuating channel conditions.

#### 3.4.1 Definition

A simple scheduling algorithm has been designed to meet this challenge, *Proportional Fair Scheduling* (PFS) algorithm in an OFDM system. In this system, the feedback of the channel quality of user  $k$  in time slot  $t$  to the BS is in terms of a requested data rate  $R_{k,n}(t)$ , this is the data rate that the  $k$ th user's  $n$  sub-carrier can currently support. The scheduling algorithm works as follows. It keeps track of the average throughput  $T_{k,n}(t)$  of each user on every sub-carrier in a past window of length  $t_c$ . In time slot  $t$ , the scheduling algorithm simply transmits at each sub-carrier to the user  $k^*$  with the largest

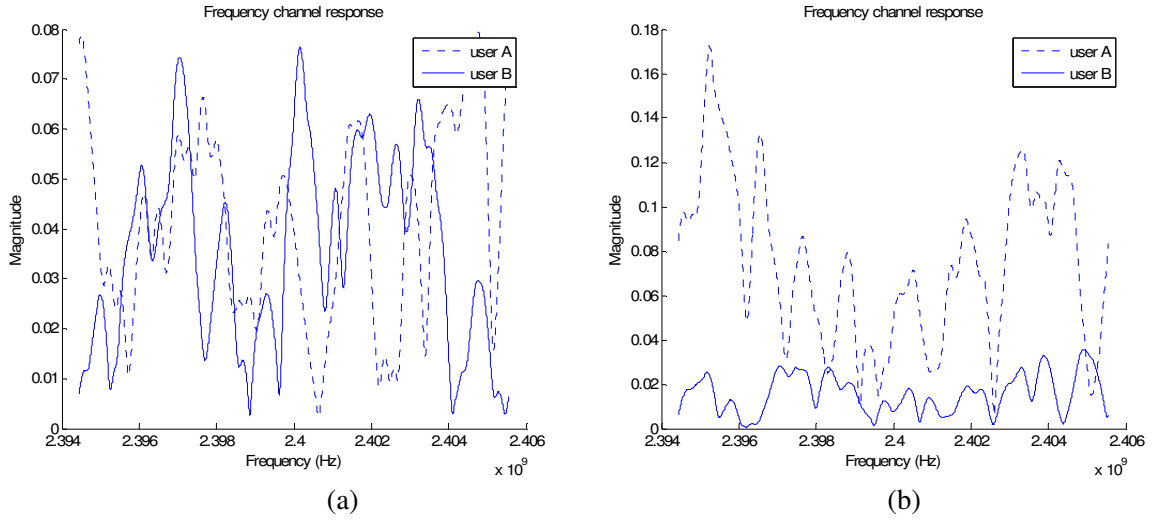
$$\frac{R_{k,n}(t)}{T_{k,n}(t)} \quad (1)$$

among all active users in the system. The average throughputs  $T_{k,n}(t)$  can be updated using an exponentially weighted low-pass filter

$$T_{k,n}(t+1) = \begin{cases} \left(1 - \frac{1}{t_c}\right) T_{k,n}(t) + \frac{1}{t_c} R_{k,n}(t) & k = k^*(t) \\ \left(1 - \frac{1}{t_c}\right) T_{k,n}(t) & k \neq k^*(t) \end{cases} \quad (2)$$

In this report, the PFS algorithm in an OFDM system is used taken sub-carriers independently each other. So, we have to compute what user has the largest value given by equation (1) at each sub-carrier and time slot in order to allocate this sub-carrier to that user. Also, we have to update the users' average throughput by equation (2) at each sub-carrier and time slot. So, the PFS algorithm treats sub-carriers independently each other and we have to update the system every time slot. This algorithm does not take into account users' rate requirements and conventional power allocation is not considered because it has some weak points that degrade the system performance when employed in heterogeneous user channel environment [31]. Equal power allocation algorithm is taken in consideration, which simply distributes the transmission power equally among the sub-carriers. To exploit the flexibility of transmission power allocation which is inherent in multiuser OFDM systems we encourage reader to see [31].

One can get an intuitive feel of how this algorithm works by inspecting Fig3.2. We plot the frequency channel response of two users as a function in two different scenarios, equal and unequal fading statistics. In Fig3.2 (a), the two users have identical fading statistics. If the scheduling time scale  $t_c$  is much larger than the correlation time scale of the fading dynamics, then by symmetry the throughput of each user converges to the same quantity. The scheduling algorithm reduces to always picking the user with the highest requested rate. Thus, each user is scheduled when its channel is good and at the same time the scheduling algorithm is perfectly fair on the long term. In Fig3.2 (b), due to perhaps different distances from the BS, one user's channel is much stronger than the other user's on the average, although both channels fluctuate due to multipath fading. Always picking the user with the highest requested rate means giving all the system resources to the statistically stronger user and would be highly unfair. In contrast, under the proposed scheduling algorithm, users compete for resources not directly based on their requested rates but only after normalization by their respective average throughputs. The user with the statistically stronger channel will have a higher average throughput. Thus, the algorithm schedules a user when its instantaneous channel quality is high relative to its own average channel condition over the time scale  $t_c$ . In short, data is transmitted to a user when its channel is near its own peaks. Multiuser diversity benefit can still be extracted because channels of different users fluctuate independently so that if there is a sufficient number of users in the system, there will likely be a user near its peak at any one time.



**Figure 3.2** Frequency channel response (a) equal users, (b) unequal users.

### 3.4.2 The Parameter $t_c$

The parameter  $t_c$  is tied to the latency time scale of the application. Peaks are defined with respect to this time scale. If the latency time scale is large, then the throughput is averaged over a longer time scale and the scheduler can afford to wait longer before scheduling a user when its channel hits a really high peak [18].

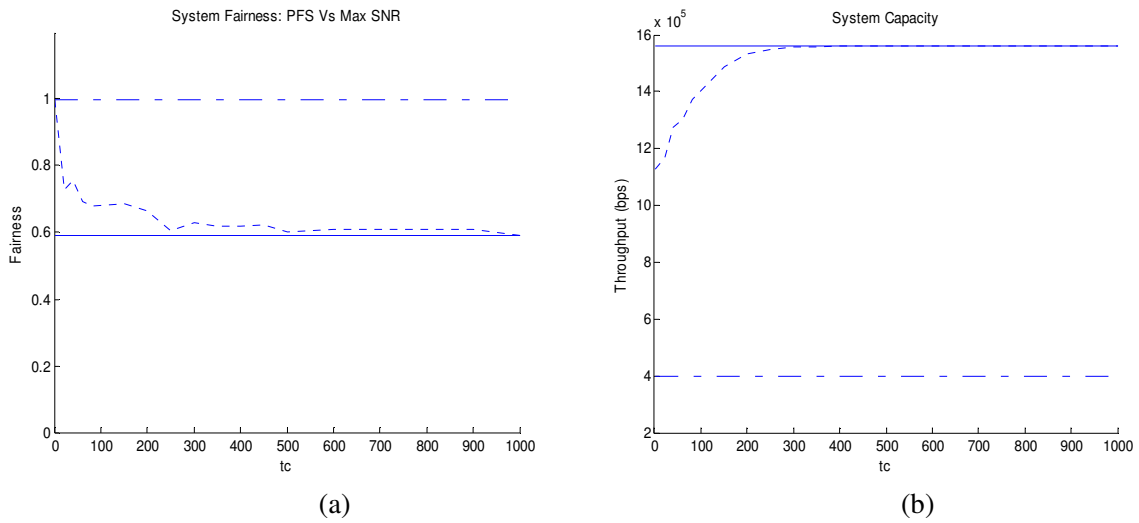
The larger value of  $t_c$  parameter is  $t_c = \infty$ , in this situation the allocation resources according to PFS algorithm is decided solely by instantaneous SNR, leading to maximum system throughput and poor fairness characteristics. On the other hand, the lower value of  $t_c$  parameter is  $t_c = 1$  in this situation scheduling becomes fair. Therefore,  $t_c$  means the trade-off between fairness and throughput. Table 3.I shows simulation parameters. The performance of this algorithm is analyzed and compared through MATLAB computer simulations with *throughput\_fairness\_Vs\_PFS.m* file, see Appendix C.3.

Time slots	100
Sub-carriers	16
$t_c$	From 1 to 1000
Users	4
Mobile Speed (km/h)	3

**Table 3.I** Simulation parameters

In Fig3.3 (a), the system fairness is showed,  $F$ , versus  $t_c$  parameter according to three different algorithms, PFS, RR and Max Rate. As we will see in chapter 4, if  $F=1$  all users share exactly the same amount of resources, and  $F=0$  occurs when only one user consumes all amount of resources. Max Rate and RR algorithms are independently of  $t_c$  parameter, so fairness according to these algorithms remains constant and represents fairness' boundaries. Max Rate algorithm represents the smallest fairness and RR algorithm represents the largest fairness. As we can see in Fig3.3 (a) if  $t_c$  tends to 1, there is an increase in fairness and PFS behaviour tends to RR behavior, but if  $t_c$  tends to infinity, there is a reduction in fairness and PFS behavior tends to Max Rate behaviour.

On the other hand, in Fig3.3 (b) throughput versus  $t_c$  parameter is showed. As we have said before, Max Rate and RR algorithms are independently of  $t_c$  parameter and represent behaviour's boundaries. Max Rate algorithm represents the largest throughput and RR algorithm represents the smallest throughput. As we can see in Fig3.3 (b) if  $t_c$  tends to infinity, there is an increase in throughput and PFS behaviour tends to Max Rate behavior, but if  $t_c$  tends to 1, there is a reduction in throughput and PFS behavior tends to RR behaviour.



**Figure 3.3** PFS (dotted line), RR (dotted continuous line) and Max Rate (continuous line) behaviour versus  $t_c$  parameter (a) fairness, (b) throughput.

### 3.5 Rate-Craving Greedy algorithm

This algorithm divides the problem of joint allocation into two steps. In the first step, the number of sub-carriers that each user will get is determined based on the users' rate requirements and average SNR. The algorithm is shown to find the distribution of sub-carriers that minimizes the total power required when every user experiences a flat-fading channel. In the second stage of the algorithm, it finds the best assignment of sub-carriers to users based on the estimated rate of transmission.

Consider a system with  $K$  users and  $N$  sub-carriers. Each user  $k$  must transmit at least  $R_{\min}^k$  bits per unit time to satisfy rate requirements. A user can transmit at most  $R_{\max}$  bits per unit time and sub-carrier, so  $R_{\max}$  is the maximum value that the user's subchannel response reaches. Let  $H_{k,n}(t)$  be the channel gain and  $r_{k,n}(t)$  the transmission rate for user  $k$  on sub-carrier  $n$  at  $t$  time slot.

The objective is to find a sub-carrier allocation, which allows each user to satisfy its rate requirements.

As we said before, the problem is solved by two algorithms, which use information about users' channel and rate requirements to find a close approximation to the solution.

1. *Resource Allocation*: Decide the number of sub-carriers each user gets - its bandwidth - based on rate requirements and the users' average channel gain, *Band Assignment Based on SNR* (BABS) algorithm.
2. *Sub-carrier Allocation*: Use the result of the resource allocation stage and channel information to allocate the sub-carriers to the users, *Rate Craving Greedy* (RCG) algorithm.

By solving these sub-problems separately, a good, but not necessarily optimal, solution is found which guarantees a certain level of service for each user.

#### 3.5.1 BABS

This section describes the bandwidth assignment based on SNR, BABS algorithm which uses the rate requirements and the average SNR for each user to decide the number of sub-carriers that user will be assigned.

Assume that each user  $k$  experiences a channel gain of  $H_{k,n}(t) = \left( \sum_{n=0}^{N-1} |H_{k,n}(t)|^2 / N \right)$  on every sub-carrier. Let user  $k$  be allocated  $m_k$  sub-carriers. When the gain on every sub-carrier is the same, the optimal rate-power allocation is to transmit  $R_{\min}^k / m_k$  bits on each sub-carrier.

To find the optimal distribution of sub-carriers among users given the flat channel assumption, a greedy descent algorithm is proposed.

### 3.5.1.1 BABS ALGORITHM

The implemented BABS algorithm works as follow: At first each user gets  $m_k$  sub-carriers according to their rate requirements, if the sum of the whole sub-carriers allocated does not fit with the available bandwidth, we have to

1. remove sub-carriers from users who are demanding less sub-carriers.
2. add sub-carriers to users according to the SNR.

In [16], this algorithm is shown to converge to the distribution of sub-carriers among users and the function  $f(r)$  is explained more accurately. In order to make relevant comparisons with the other algorithms, fix power at the BS is used. So, the BS simply distributes the transmission power equally among the sub-carriers without taking advantage on power control.

$$\begin{aligned}
 m_k &\leftarrow \left\lceil \frac{R_{\min}^k}{R_{\max}} \right\rceil, \quad k = 0, \dots, K-1 \\
 \text{while } \sum_{k=0}^{K-1} m_k &> N \text{ do} \\
 & \quad k^* \leftarrow \arg \min_{0 \leq k \leq K-1} m_k \\
 & \quad m_{k^*} \leftarrow 0 \\
 \text{end while} \\
 \text{while } \sum_{k=0}^{K-1} m_k &< N, \text{ do} \\
 & \quad G_k \leftarrow \frac{m_k + 1}{H_k} f\left(\frac{R_{\min}^k}{m_k + 1}\right) - \frac{m_k}{H_k} f\left(\frac{R_{\min}^k}{m_k}\right), \quad k = 0, \dots, K-1 \\
 & \quad l \leftarrow \arg \min_{0 \leq k \leq K-1} G_k \\
 & \quad m_l \leftarrow m_l + 1 \\
 \text{end while}
 \end{aligned}$$

### 3.5.2 RCG

Once the number of sub-carriers is determined, the next step is to assign specific sub-carriers to users. The problem is still difficult to solve since different users see different channels. The RCG algorithm begins with an estimate of the users' transmission rate on each sub-carrier and aims to maximize the total transmission rate. For the model used in simulations  $r_{k,n}(t) = B/N \log_2(1 + SNR)$  is used, according to Shannon capacity theorem.

#### 3.5.2.1 RCG ALGORITHM

The algorithm works as follow:

1. Allocate each sub-carrier  $n$  at each time slot to the user with maximum transmission rate  $r_{k,n}(t)$ .
2. While there exists some user  $k$  such that the number of sub-carriers allocated according to maximum transmission rate are more than  $m_k$ , the sub-carriers allocated according to BABS algorithm, remove a sub-carrier from user  $k$  and add a sub-carrier to user  $l$  such that the number of sub-carriers allocated according to maximum transmission rate are less than  $m_k$ , using a sequence of reallocations.

```

for each subcarrier  $n = 0 : N - 1$ , do
     $k^* \leftarrow \arg \max_{0 \leq k \leq K-1} r_k(n)$ 
     $C_{k^*} \leftarrow C_{k^*} \cup \{n\}$ 
end for
for all users  $k$  such that  $\text{num}C_k > m_k$ , do
    while  $\text{num}C_k > m_k$ , do
         $l^* \leftarrow \arg \min_{\{l: \text{num}C_l < m_l\}} \min_{0 \leq n \leq N-1} -r_k(n) + r_l(n)$ 
         $n^* \leftarrow \arg \min_{0 \leq n \leq N-1} -r_k(n) + r_{l^*}(n)$ 
         $C_k \leftarrow C_k \setminus \{n^*\}, C_{l^*} \leftarrow C_{l^*} \cup \{n^*\}$ 
    end while
end for

```



### 3.5.3 Example Run of the Algorithm

In this section an example run of the algorithm is demonstrated for  $N=8$  sub-carriers and  $K=4$  users which require 64, 45, 35 and 16 Kb per time slot, see Table 3.II (a).

At first the algorithm decides the number of sub-carriers each user gets base on rate requirements and the users' average channel gain. As we can see in Table 3.II (b), the number of sub-carriers demanded does not fit with the available bandwidth, so the sub-carriers are removed from the users who are demanded less transmission rate until the number of sub-carriers allocated to users would be 8, Table 3.II (c).

	Demanded transmission rate	Demanded sub-carriers	Allocated sub-carriers
#1	64	3	3
#2	45	3	2
#3	16	2	1
#4	35	3	2

(a)

(b)

(c)

**Table 3.II** (a) Transmission rate demanded by users, (b) sub-carriers demanded by users, (c) sub-carriers allocated by the algorithm.

Once the number of sub-carriers is determined, this information and channel state are used to allocate the sub-carriers to the users.

	#1	#2	#3	#4
#1	<b>4</b>	1	2	3
#2	<b>7</b>	6	3	5
#3	<b>6</b>	2	4	1
#4	5	<b>8</b>	1	8
#5	3	4	<b>6</b>	2
#6	<b>8</b>	7	8	6
#7	2	3	<b>5</b>	4
#8	1	5	<b>7</b>	7

(a)

	#1	#2	#3	#4
#1	<del>4</del>	<del>1</del>	<del>2</del>	<b>3</b>
#2	<b>7</b>	6	3	5
#3	<b>6</b>	2	4	1
#4	5	<b>8</b>	1	8
#5	3	4	<b>6</b>	2
#6	<b>8</b>	7	8	6
#7	2	<b>3</b>	<del>5</del>	4
#8	1	5	<del>7</del>	<b>7</b>

(b)

**Table 3.III** Sub-carrier allocation by RCG algorithm at the end of stage 1 (a) and (b) the algorithm. The number in column  $k$  row  $n$  is the estimated rate of transmission of user  $k$  on sub-carrier  $n$ , if it is allocated that sub-carrier. The blond rate has been chosen.

In Table 3.III, columns represent different users, rows are different sub-carriers. The number in row  $n$  column  $k$  represents an estimate of the rate at which user  $k$  would transmit on  $n$  if it were allocated that sub-carrier. The bold numbers show to whom each sub-carrier is allocated.

1. Allocate each sub-carrier  $n$  to the user with the maximum transmission rate.

User #1 → 4 sub-carriers

User #2 → 1 sub-carriers

User #3 → 3 sub-carriers

User #4 → 0 sub-carriers

2. Because of there exists some users such that the number of sub-carriers allocated according to Max Rate is more than the number of sub-carriers allocated according to BABS algorithm, sub-carriers are removed from these users and add to others according to Table 3.II (c).

Remove one sub-carrier from user #1 to user #4

Remove two sub-carriers from user #3, one to user #2 and one to user #4

In Appendix G, we can see an example run of this algorithm in 10 different time slots with equal and unequal users.

### 3.6 MATLAB Simulations

In this section two different scheduling policies have been compared for system resources allocating, PFS and RCG. Also, RR and Max Rate algorithms have been used for reinforcing the conclusions about system fairness and throughput.

The algorithms will be compared based on three criteria, fairness, throughput and delay but these results will be show in chapters 4, 5 and 6 respectively. In this chapter the policies used for allocating resources will be showed. The performance of these algorithms is analyzed and compared through MATLAB computer simulations with *scheduler.m* and *statistic.m* files, see Appendix C.1 and C.2 respectively.

#### 3.6.1 Model Description

In this report, the downlink transmission of a multiuser OFDM system is considered. We assume that the overall bandwidth  $B$  is divided into  $N$  orthogonal narrow-band sub-carriers. Each MS measures the channel gain of each sub-carrier and feeds back the channel state information to the BS via a separate feedback channel. We assume that channel estimation can be done perfectly and feedback channel is errorless. The data stream of each

selected user is then modulated by a proper modulation scheme that is supportable in the given channel state and the allocated transmission power of the sub-carrier.

In the frequency selective fading channel, in general, different sub-carriers experience different channel gains even for the same user. We assume that each sub-carrier is narrow enough to undergo flat fading. We also assume that the channel gain is constant in each time slot but varies from slot to slot. Under the assumptions above, the received SNR of the  $n$ th sub-carrier signal of the  $k$ th user at the  $t$ th slot can be expressed by

$$SNR_{k,n}(t) = \frac{S_{k,n}(t)H_{k,n}(t)}{N_0 B / N} \quad (3)$$

where  $S_{k,n}(t), H_{k,n}(t)$  are the allocated transmission power and channel gain on  $n$ th sub-carrier at  $t$ th time slot respectively,  $N_0$  is the power spectral density of AWGN,  $B$  is the bandwidth and  $N$  is the number of sub-carriers.

In multiuser OFDM systems, a resource allocation algorithm decides which user to serve with what power for each sub-carrier. Here, it is assumed that the total transmission power is constrained to  $\bar{S}$ . Once the resources are allocated to users, the instant data rate of each user is determined and the BS serves each user at this rate. According to information theory and as we can see in Chapter 5, the instant service rate on the  $n$ th sub-carrier at  $t$ th time slot is got by

$$R_{k,n}(t) = B / N \log_2(1 + SNR) \quad (4)$$

where  $R_{k,n}(t)$  is the  $k$ th user transmission rate at  $t$ th time slot,  $B$  is the total bandwidth and  $N$  is the number of sub-carriers.

Three classes of services are considered: data, voice, and video. It is assumed that 10% of the users will be transmitting video, 40% will be transmitting voice, and the remaining 50% of the users will be transmitting data. Video and voice traffic are given a constant transmission rate of 64 and 16 Kb per time slot, respectively, whereas data traffic is assumed to be exponentially distributed, with a mean of 30 Kb per time slot.

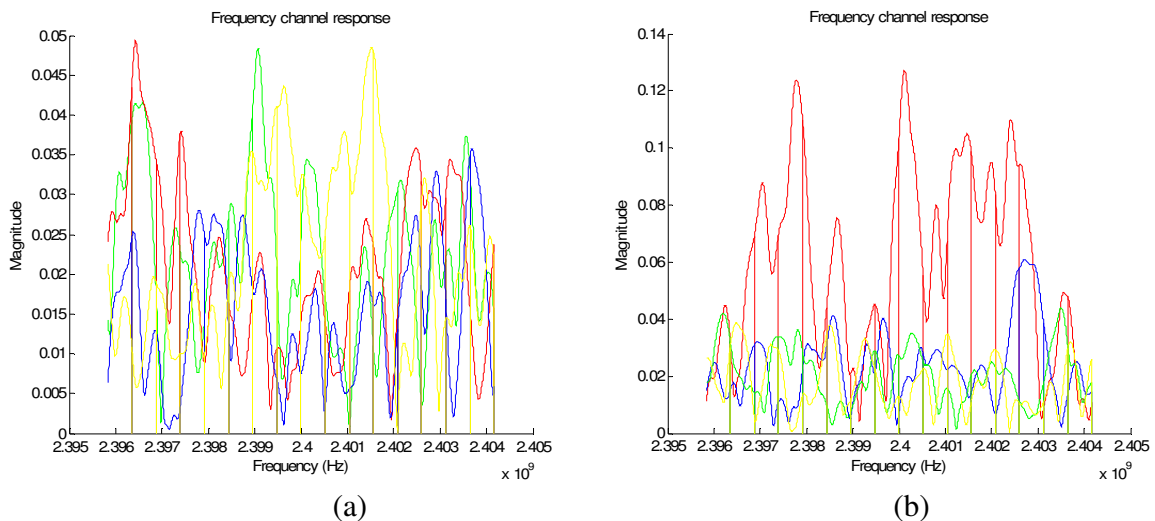
### 3.6.2 Simulation Results

In simulations there are 16 sub-carriers for the OFDM system and the carrier frequency is 2.4 GHz. In order to show how work the different scheduling algorithms we only simulate four different users, A (red), B (blue), C (green) and D (yellow), with their different channel responses. We assume that user A is transmitting video (64 Kb per time slot), user B is transmitting voice (16 Kb per time slot), users C and D are transmitting data, 30 and 40 Kb per time slot respectively. Each sub-carrier is allocated to each user in two

different scenarios according to four different algorithms, RR, Max Rate, PFS and RCG. Table 3.IV shows simulation parameters.

Time slots	16
Sub-carriers	16
fc (GHz)	2.4
$t_c$	20 and 100'000
Users	4
Rates (Kb per time slot)	16, 30, 40 and 64
Mobile Speed (km/h)	3

**Table 3.IV** Simulation parameters.



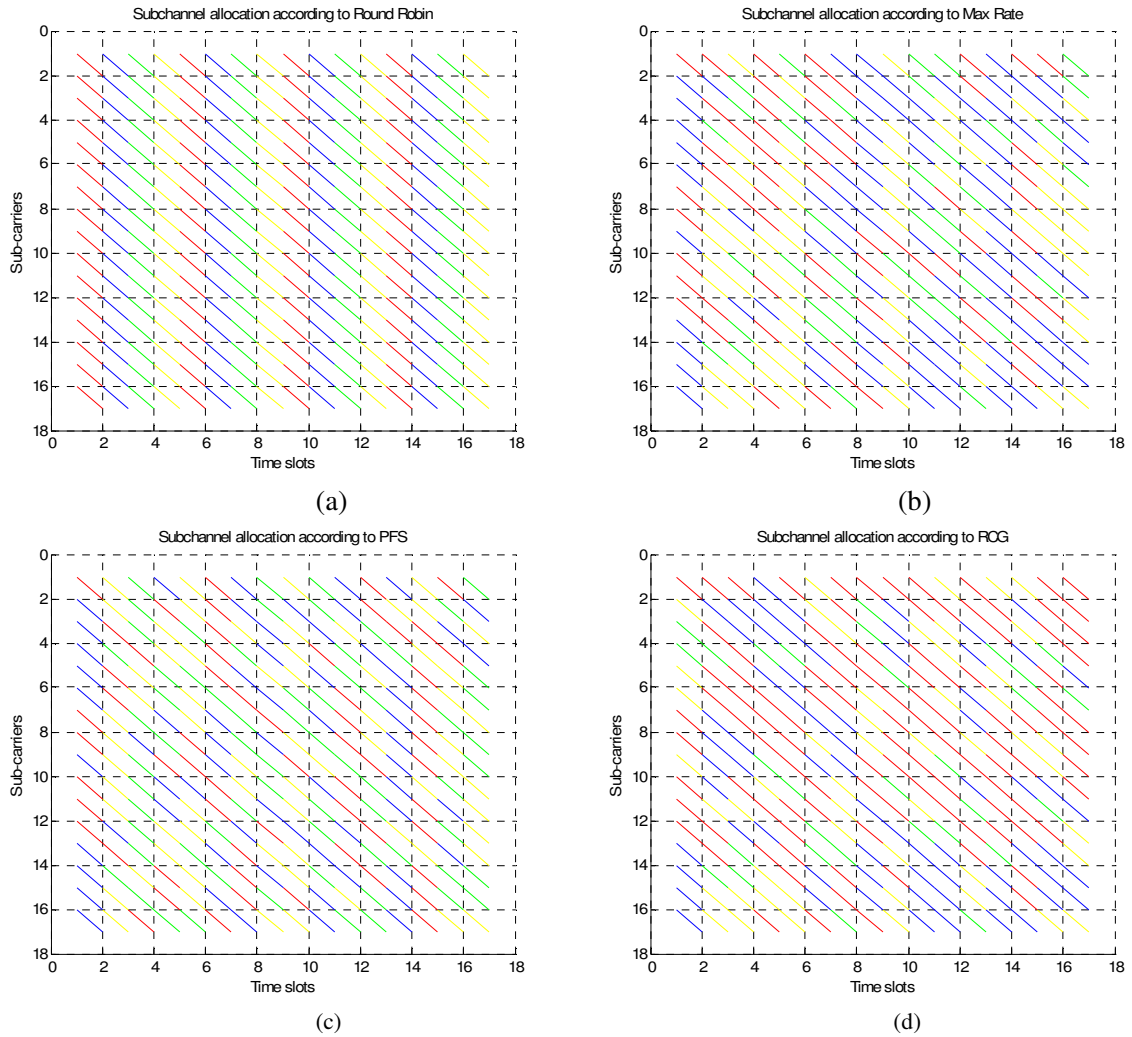
**Figure 3.4** Frequency channel response at any instant time  $t_i$ , (a) equal users, (b) unequal users.

### 3.6.2.1 Scenario A: Equal Users

As we can see in Fig3.4 (a), all the users simulated have the same channel response average. In Fig3.5, the scheduler scheme is showed according to four different scheduling policies.

As we can see in Fig3.5 (a), RR algorithm allocates all sub-carriers to one user at each time slot, so this policy is totally fair. In this scenario we cannot notice the differences between Max Rate (see Fig3.5 (b)) and PFS (see Fig3.5 (c)) policies on allocating resources

because all users have the same channel response average. But we can notice that RCG allocates more sub-carriers to user A than other scheduling algorithms, the reason is because only this algorithm takes into account rate requirements by users and user A is transmitting video and this services needs more amount of data than other services, so user A needs more sub-carriers than other users.



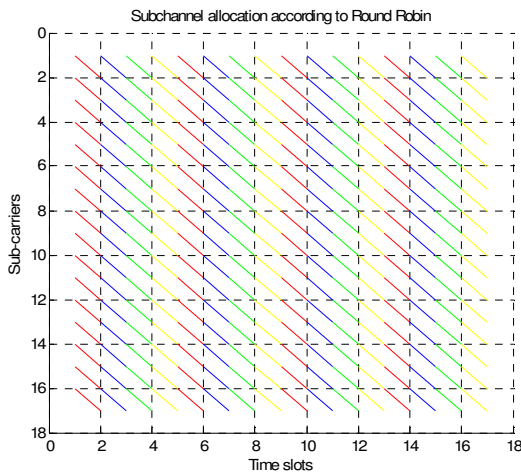
**Figure 3.5** Scheme where users have equal channel responses average and have been simulated according to filtered Gaussian noise method, (a) RR algorithm, (b) Max Rate algorithm, (c) PFS algorithm, with  $t_c = 20$ , (d) RCG algorithm.

### 3.6.2.2 Scenario B: Unequal Users

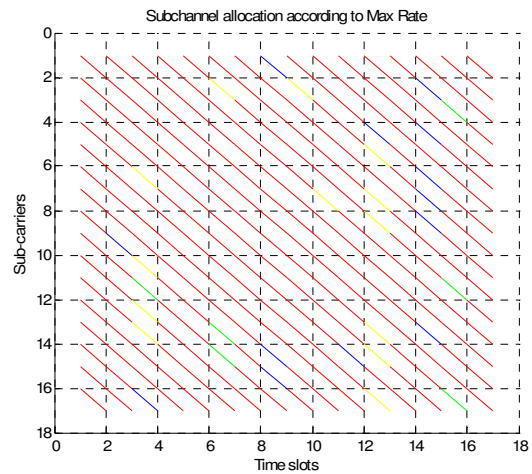
As we can see in Fig3.4 (b), the user A (red) is nearer to the BS than the other users because the channel A is stronger than others. In this scenario RR algorithm (see Fig3.6 (a)) takes the same scheduler scheme that the previous scenario, it allocates system resources to one user at each time slot.

When users have not the same channel response average, we can notice the differences between Max Rate and PFS on scheduling resources and the different PFS scheduler scheme when  $t_c$  parameter changes. As we can see in Fig3.6 (b), almost all sub-carriers are allocated to user A because of channel strength and the same results are achieved with PFS policy when  $t_c$  parameter is high (see Fig3.6 (c)) but, as we saw in section 3.4.2 the results change when we take a low  $t_c$  parameter (see Fig3.6 (d)). The larger  $t_c$  parameter, the unfairer PFS is.

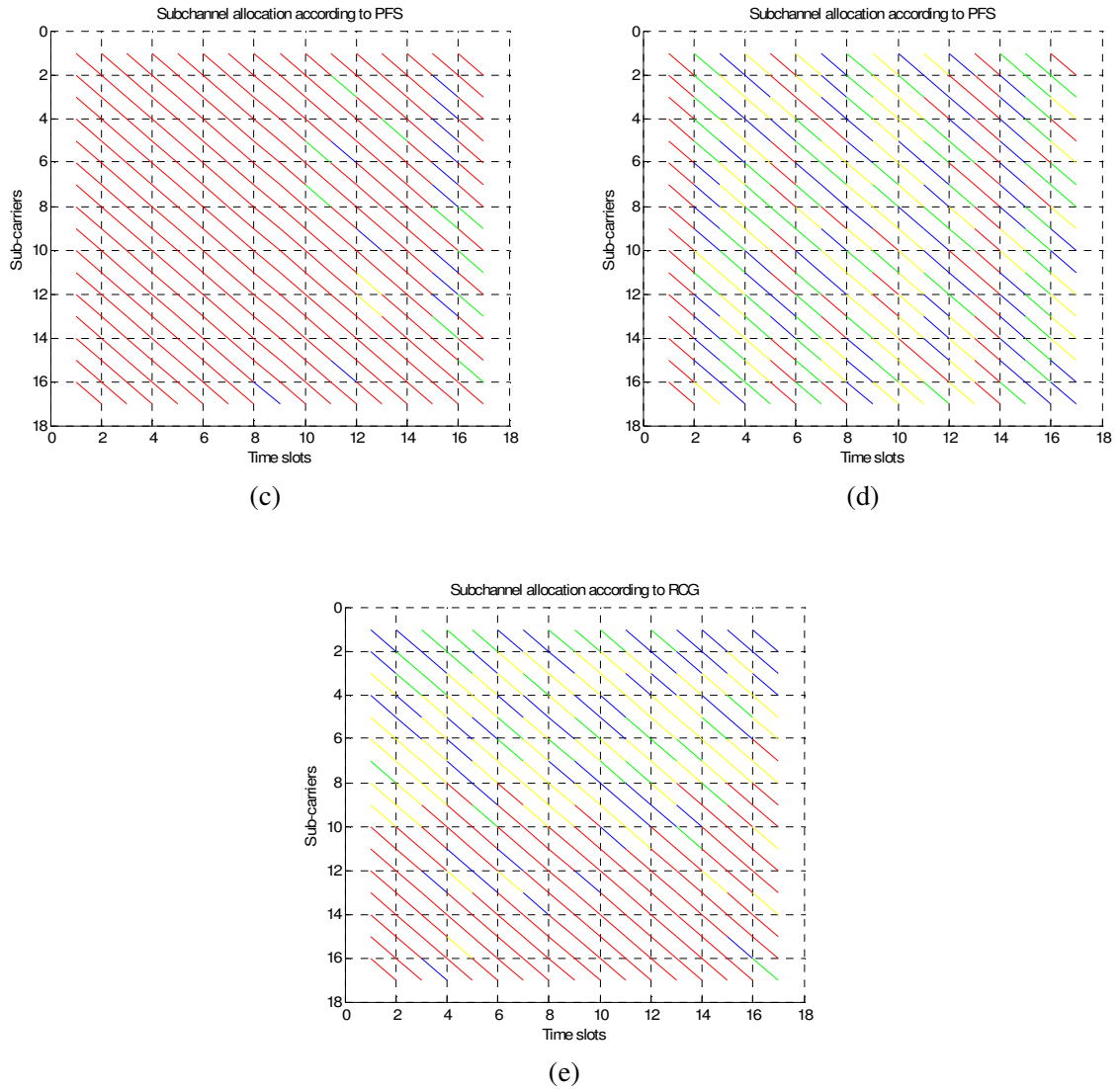
RCG policy maintains the index fairness because RCG allocates sub-carriers based on rate requirements and the users' average SNR.



(a)



(b)



**Figure 3.6** Scheme scheduled where users have unequal channel responses average and have been simulated according to filtered Gaussian noise method, (a) RR algorithm, (b) Max Rate algorithm, (c) PFS algorithm, with  $t_c = 20$ , (d) PFS algorithm, with  $t_c = 100'000$ , (e) RCG algorithm





## Chapter 4

# Fairness

In this chapter, one of the most important properties in a wireless environment where system resources are being shared by users is discussed. In order to evaluate and find the scheduling disciplines with the best performance, appropriate performance measures are required. For best effort users the major concern for the scheduling discipline should be to maximize the total throughput for all users at the same time as each user is served a fair amount of throughput.

### 4.1 Definition

Fairness is a desirable property of the wireless network as it offers protection between users. This means that the traffic flow of an ill-behaving user cannot affect the traffic flow of another user.

The fairness of a system, the users in a system, or of a scheduling algorithm is not usually expressed in terms of a quantitative value. Rather, fairness is usually expressed in broader terms. Generally, a system is deemed to be fair or unfair based on whether or not the system meets certain criteria. Such criteria are usually in terms of throughput or delay. For example, a scheduling algorithm may be deemed to be unfair if any user receives a throughput of less than  $X$  bits/sec. Another example might be that a system is deemed fair if the probability of any user experiencing a delay greater than  $\tau$  is less than  $p$ ; if the probability is greater than  $p$  the system is unfair. The rationale behind the proposed definitions is to see if a value for the “fairness” of a system or scheduling algorithm can be defined in a similar manner to how Shannon defined a value for the “information” of a source. The fairness value should also make sense from both a mathematical and a semantic sense. Shannon’s definition was based on probabilities of events. In comparison, the proposed fairness measures are based on proportions of allocated resources [20]. In order to evaluate fairness, we propose two set of users, equal and unequal weighted users.

## 4.2 Equal weighted users

The “resources” in question depend on the circumstances under which the definition is applied. Possible examples include the number of bits or packets sent to users, or the amount of bandwidth, time, or slots each user is allocated. The proportion for each user would then be the number or amount of packets, bandwidth, slots, etc. allocated to that user, relative to the total allocated to all the users in the system.

For a random variable or process  $X$ , the self-information of the event  $X = x_i$  is:

$$I(x_i) = \log \frac{1}{P(x_i)} = -\log P(x_i) \quad (1)$$

where  $P(x_i)$  is the probability of the event  $X = x_i$ .

A measure for fairness can be defined in a similar manner, but using proportions instead of probabilities. In the case when all  $N$  users of a system are considered equal, the definition for the “*self-fairness*” of a given user is:

$$F_i = \frac{-\log p_i}{-\log(1/N)} = -\frac{\log p_i}{\log N} \quad (2)$$

where  $p_i$ , is the proportion of resources allocated to user  $i$ , and the  $\log N$  term is a normalization factor. The “*self-unfairness*” of a user can also be defined simply as the reciprocal of the self-fairness:

$$U_i = \frac{1}{F_i} = \frac{-\log N}{\log p_i} \quad (3)$$

### 4.2.1 Average fairness and average unfairness

Given the self-information of the  $N$  possible events of the process  $X$  as given by, the average self-information in the process  $X$  can be found by:

$$H(X) = \sum_{k=1}^N \frac{P(x_k)}{I(x_k)} = -\sum_{k=1}^N P(x_k) \log P(x_k) \quad (4)$$

Similarly, the proposed definition for the “*average fairness*” of a system of  $N$  users is:

$$\bar{F} = \sum_{k=1}^N p_k F_k = - \sum_{k=1}^N p_k \frac{\log p_k}{\log N} \quad (5)$$

And the “average unfairness” is given by:

$$\bar{U} = \sum_{k=1}^N p_k U_k = - \sum_{k=1}^N p_k \frac{\log N}{\log p_k} \quad (6)$$

#### 4.2.2 Remarks

The above definitions yield some interesting properties:

1. When a user consumes exactly its fair share of resources (i.e. a proportion of  $1/N$ ), the value of self-fairness and self-unfairness for that user will be unity.
2. As the amount of resources a user consumes increases, it takes an increasing amount away from the other users in the system. Thus, that user becomes less fair (or more unfair), and consequently, the values of self-fairness and self-unfairness for that user decrease and increase respectively.
3. Conversely, as a user consumes fewer resources, it is giving up resources in favor of the other users in the system. Hence, that user becomes much fair (or less unfair), and correspondingly, the values of self-fairness and self-unfairness for that user increase and decrease respectively.
4. In the limit as one user consumes all the available resources, all the other users in the system will be the greedy user to be less fair (or more unfair). Thus, logically, the value for the self-unfairness for that user should be the highest possible value, infinity, which indeed proves to be the case in the proposed definition. The self-fairness for the user in question becomes zero, the lowest value the self-fairness can become.
5. Similarly, in the limit when a user consumes no resources, that user gives up all its deserved resources in favor of other users. Thus, there is no possible way for that user to be much fair (or less unfair). This is reflected in the resulting values of infinity and zero for the self-fairness and self-unfairness, respectively.
6. The value for the average fairness of the system of  $N$  users ranges from 0 to 1 inclusive. Furthermore, the maximum value of unity is only achieved when all users in the system consume exactly their fair share of the resources (i.e.  $p_1 = p_2 = \dots = p_N = 1/N$ ). The minimum of zero occurs in the limit when one user consumes all the allocated resources while the other  $N-1$  users are starved.

7. Similarly, the average unfairness ranges between 1 and infinity, with the lower and upper extreme values occurring under the same conditions as the maximum and minimum values respectively for the average fairness. That is, a value of unity will result only if all users consume their fair share of resources, and infinity will result if a single user consumes all the allocated resources.

### 4.3 Unequal weighted users

The above definitions are now extended to the more general case when individual users can have different weightings (for example, to achieve different levels of QoS). In [21], the authors state that for a system of  $N$  users with a set of backlogged flows  $B(t)$  during the time interval  $[t_1, t_2]$  a queuing system is fair if:

$$\forall i, j \in B(t_1, t_2), \left| \frac{W_i(t_1, t_2)}{r_i} - \frac{W_j(t_1, t_2)}{r_j} \right| = 0 \quad (7)$$

where  $W_j$  is the capacity granted to flow  $j$  (i.e. the resources allocated to flow  $j$ ), and  $r_j$  is the weighting of user  $j$ . From this, it can be shown that the proportion of the resources allocated to user  $j$  that is deemed to be fair (in other words, the proportion of the resources that user  $j$  should receive in order for the system to be fair) is given by:

$$P_{fair,j} = \frac{r_j}{r_T} \quad (8)$$

where  $r_T$  is the weighting sum of all users. From the above proportion, it can be deduced that the formula for the weighted self-fairness of a user should become:

$$F_j = \frac{\log p_j}{\log r_j / r_T} \quad (9)$$

The weighted self-unfairness of a user is the reciprocal of the above equation:

$$U_j = \frac{\log r_j / r_T}{\log p_j} \quad (10)$$

### 4.3.1 Weighted Average fairness and unfairness

It would be desirable to define the weighted average fairness as  $\sum_{k=1}^N p_k F_k$ , as in the case of equal weighted users. However, it is impossible to make this definition and still maintain the properties discussed before. Instead, the weighted average fairness of a system can be defined as:

$$\bar{F} = \frac{r_T \sum_{k=1}^N C_k p_k F_k}{\sum_{k=1}^N C_k r_k} \quad (11)$$

the values  $C_k$  are normalization constants.

$$C_k = \frac{1 + \frac{1}{\ln(r_1 / r_T)}}{1 + \frac{1}{\ln(r_k / r_T)}} \quad (12)$$

Their purpose is to ensure that the maximum value of the weighted average fairness is unity (as in the case of equally weighted users), and that the maximum occurs when user consume exactly its fair share of the resources (for more details see [20]).

### 4.3.2 Remarks

The above definitions yield some interesting properties:

1. The weighted definitions share all the same properties as the equal weighted definitions. The only difference is that the value of 1 occurs for weighted self-fairness, self-unfairness, average fairness, and average unfairness when  $p_k = r_k / r_T$ ,  $k=1, 2, \dots, N$ , instead of  $p_k = 1/N$  as before.
2. Note that there is an implicit time scale in the proportions. That is, the proportions of resource allocation would be measured over a certain period of time. Since the period used can be arbitrary, the proposed measures can be used to examine both the short-term fairness and the long-term fairness of systems [19].

## 4.4 MATLAB Simulations

In this section the algorithms will be compared based on fairness criteria. We use RR algorithm as reference because this policy achieves the best results on index fairness.

The performance of these algorithms is analyzed and compared through MATLAB computer simulations with *fairness.m* and *throughput\_fairness\_Vs\_users.m* files, see Appendix D.1 and Appendix E.1 respectively.

### 4.4.1 Model Description

The same system model described in chapter 3 is used. For evaluating the index fairness, users with different channel response averages in three different scenarios are considered:

- A. All users are transmitting voice, constant rate, 16 Kb per time slot.
- B. All users are transmitting video, constant rate, 64 Kb per time slot.
- C. Mix scenario where 10% of users are transmitting video, constant rate, 64 Kb per time slot, 40% of users are transmitting voice, constant rate, 16 Kb per time slot and 50% of users are transmitting data, exponentially distributed rate, with a mean of 30 Kb per time slot.

Table 4.I shows simulation parameters.

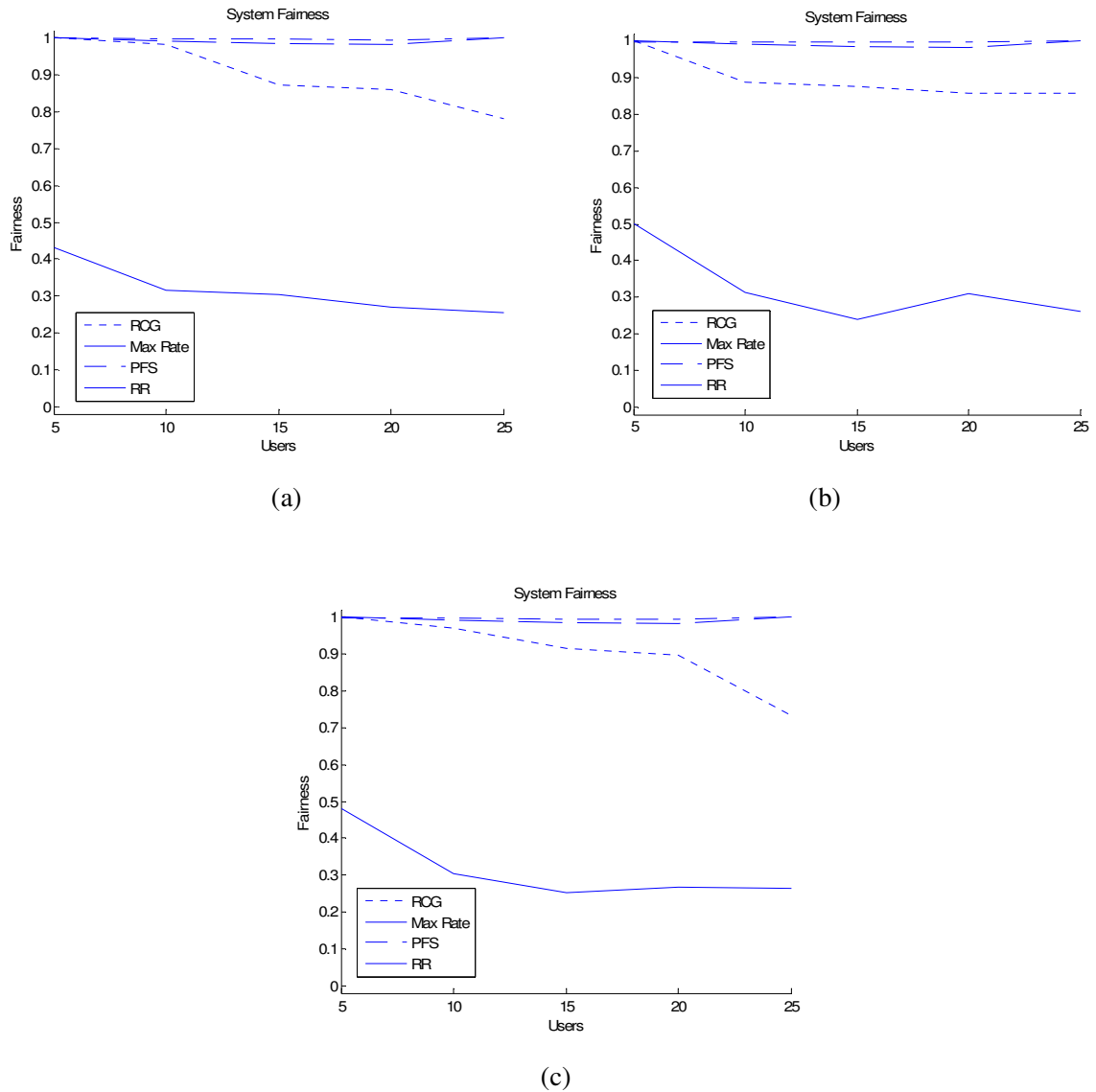
Time slots	100
Sub-carriers	16
$t_c$	20
Users	From 5 to 25
Rates (Kb per time slot)	16, 64 and $\text{exprnd}(30)$
Mobile Speed (km/h)	3

**Table 4.I** Simulation parameters

### 4.4.2 Simulation Results

MATLAB appears to be a simple and straightforward tool to demonstrate the concept of fairness. The simulation results are shown in Fig4.1. As we said in section 4.3.2 the period of time over which we can measure the index fairness can be arbitrary, but as the simulation

is so short, the whole set of 100 time slots are used to evaluate the index fairness. So, in this case the proposed measures are used to examine the short-term fairness which can be interested for applications with strength delay restrictions.



**Figure 4.1** System fairness versus users with different channel response average in three different scenarios, (a) scenario A, (b) scenario B, (c) scenario C.

Max Rate policy is the unfairness algorithm because it allocates the system resource to users who have the strongest channel, so the results achieved by this algorithm are independent from the scenario simulated. As we can see in Fig4.1, the larger the users who are demanding service in the system, the lower the index fairness, because Max Rate scheduler always serves to the same stronger users.

As Max Rate scheduler, PFS scheduler is independent from rate requirements, so the results achieved in these three different scenarios are more or less the same. As we can see in Fig4.1, PFS algorithm has a good behaviour because as we saw in chapter 3, with a low  $t_c$  parameter this algorithm maintenance index fairness without involving system throughput.

As we said in chapter 3, RCG algorithm allocates sub-carriers to users according to users' channel response and rate requirements. In Fig4.1, we can see that the larger the users in the system, the lower the index fairness, but there are differences among them. In the scenario A, Fig4.1 (a), RCG maintains total fairness until the number of users in the system is more than 10 so long as in B scenario, Fig4.1 (b), RCG maintains total fairness until 5 users. The reason of this behaviour is because when there are sub-carriers enough to satisfy rate requirements RCG allocates the share amount of resources among users, but when there are not sub-carriers enough, RCG algorithm allocates sub-carriers to stronger users until satisfying their rate requirements, after that RCG allocates the sub-carriers remained. Users who are demanding video service need more sub-carrier than users who are demanding voice service, so when there are not sub-carriers enough, users who are transmitting voice have to wait. In the scenario C, Fig4.1 (c), an intermediate situation in a mix scenario is plotted.



## Chapter 5

# Throughput

There is a considerable interest within the industry in specifying the performance of wireless data. In this chapter, we will discuss about one of the most important properties of information system, throughput. The throughput is commonly considered within a framework that optimizes the system performance [22] and is a measure of how much information can be transmitted and received per unit time with a negligible probability of error. We will evaluate and find the scheduling disciplines with the best performance.

### 5.1 Motivation

To design and deploy a WLAN, an accurate deployment procedure is required to ensure sufficient coverage and network functionality (capacity, interference, etc.). While wireless networking gear is often classified according to its standards-based signaling rate, the actual data throughput, or data being transmitted, is often just a fraction of the signaling rate's theoretical maximum. Research conducted in [23] showed that the user throughput performance changes radically when access points or clients are located near an interfering transmitter or when frequency planning is not carefully conducted. Data throughput can also be limited due to a number of important environmental and product-specific factors, including: Distance between WLAN devices, access points and network interface cards; Transmission power levels; Effect of waveguide, e.g. in hallways; Building and home materials; Radio frequency interference; Signal propagation; and Antenna type and location. So, even though the 802.11 wireless networking products available are capable of a  $C$  bps signaling rate, the practical, or "actual," data throughput rate is more likely to be much less.

### 5.2 Channel Capacity

At the input of a communication system, discrete source symbols are mapped into a sequence of channel symbols. The channel symbols are then transmitted through a wireless channel that by nature is random. In addition, random noise is added to the channel symbols.

In general, it is possible that two different input sequences may give rise to the same output sequence, causing different input sequences to be confusable at the output. To avoid this situation, a non-confusable subset of input sequences must be chosen so that with a high probability, there is only one input sequence causing a particular output. It is then possible to reconstruct all the input sequences at the output with negligible probability of error [24].

A measure of how much information that can be transmitted and received with a negligible probability of error is called the channel capacity. But by simplification, to determine this measure of channel potential, we assume that the channel can support at most  $C$  bits per time slot, where  $C$  is the channel capacity according to Shannon-Hartley Theorem.

### 5.2.1 Shannon-Hartley Theorem

In information theory, the Shannon–Hartley theorem is an application of the noisy channel coding theorem to the archetypal case of a continuous-time analog communications channel subject to Gaussian noise. The result establishes the maximum amount of error-free digital data (that is, information) that can be transmitted over such a communication link with a specified bandwidth in the presence of the noise interference [25]. The Shannon limit or Shannon capacity of a communications channel is the theoretical maximum information transfer rate of the channel.

The Shannon–Hartley theorem establishes what that channel capacity is, for a finite bandwidth continuous-time channel subject to Gaussian noise.

If we had such a thing as an infinite-bandwidth, noise-free analog channel we could transmit unlimited amounts of error-free data over it per unit of time. However, real life signals have both bandwidth and noise-interference limitations.

Surprisingly, bandwidth limitations alone do not impose a cap on maximum information transfer. This is because it is still possible (at least in a thought-experiment model) for the signal to take on an infinite number of different voltage levels on each cycle, with each slightly different level being assigned a different meaning or bit sequence [25]. If we combine both noise and bandwidth limitations, however, we do find there is a limit to the amount of information that can be transferred, even when clever multi-level encoding techniques are used. Considering all possible multi-level and multi-phase encoding techniques, the theorem gives that the theoretical maximum rate of clean (or arbitrarily low bit error rate) data  $C$  with a given average signal power that can be sent through an analog communication channel subject to AWGN interference is:

$$C = B \log_2(1 + SNR) \quad (1)$$

where  $C$  is the channel capacity in bits per second, net of error correction;  $B$  is the bandwidth of the channel in hertz and  $SNR$  is the signal-to-noise ratio of the communication signal to the Gaussian noise interference expressed as a straight power ratio (not as decibels).

The maximum information rate that can be used causing negligible probability of errors at the output is called the capacity of the channel. In simulations,  $C$  is taken as the information rate of the channel.

### 5.2.2 Outage Capacity

Another measure of channel capacity that is frequently used is *outage capacity* [24]. With outage capacity, the channel capacity is associated to an outage probability. Capacity is treated as a random variable which depends on the channel instantaneous response and remains constant during the transmission of a finite-length coded block of information. If the channel capacity falls below the outage capacity, there is no possibility that the transmitted block of information can be decoded with no errors, whichever coding scheme is employed. The probability that the capacity is less than the outage capacity denoted by  $C_{outage}$  is  $q$ . This can be expressed in mathematical terms by

$$P\{C \leq C_{outage}\} = q \quad (2)$$

in this case, (2) represents an upper bound due to fact that there is a finite probability  $q$  that the channel capacity is less than the outage capacity. It can also be written as a lower bound, representing the case where there is a finite probability  $(1-q)$  that the channel capacity is higher than  $C_{outage}$ , i.e.,

$$P\{C > C_{outage}\} = (1 - q) \quad (3)$$

## 5.3 MATLAB Simulations

In this section the algorithms will be compared based on throughput criteria. Max Rate algorithm is used as reference because this policy achieves the best results on system capacity.

The most interested concept is that in order to attain capacity, only one user should transmit at any given time over the allocated sub-carrier. This user has the strongest signal at that particular time instant, relative to the average received powers of all the users [26].

The performance of these algorithms is analyzed and compared through MATLAB computer simulations with *throughput.m* and *throughput\_fairness\_Vs\_users.m* files, see Appendix D.2 and Appendix E.1 respectively.

### 5.3.1 Model Description

The same system model described in chapter 3 is used. To evaluate the system throughput, and as we did in chapter 4, three different scenarios are considered:

- A. All users are transmitting voice, constant rate, 16Kb per time slot.
- B. All users are transmitting video, constant rate, 64Kb per time slot.
- C. Mix scenario where 10% of users are transmitting video, constant rate, 64Kb per time slot, 40% of users are transmitting voice, constant rate, 16Kb per time slot and 50% of users are transmitting data, exponentially distributed rate, with a mean of 30Kb per time slot.

Table 5.I shows simulation parameters.

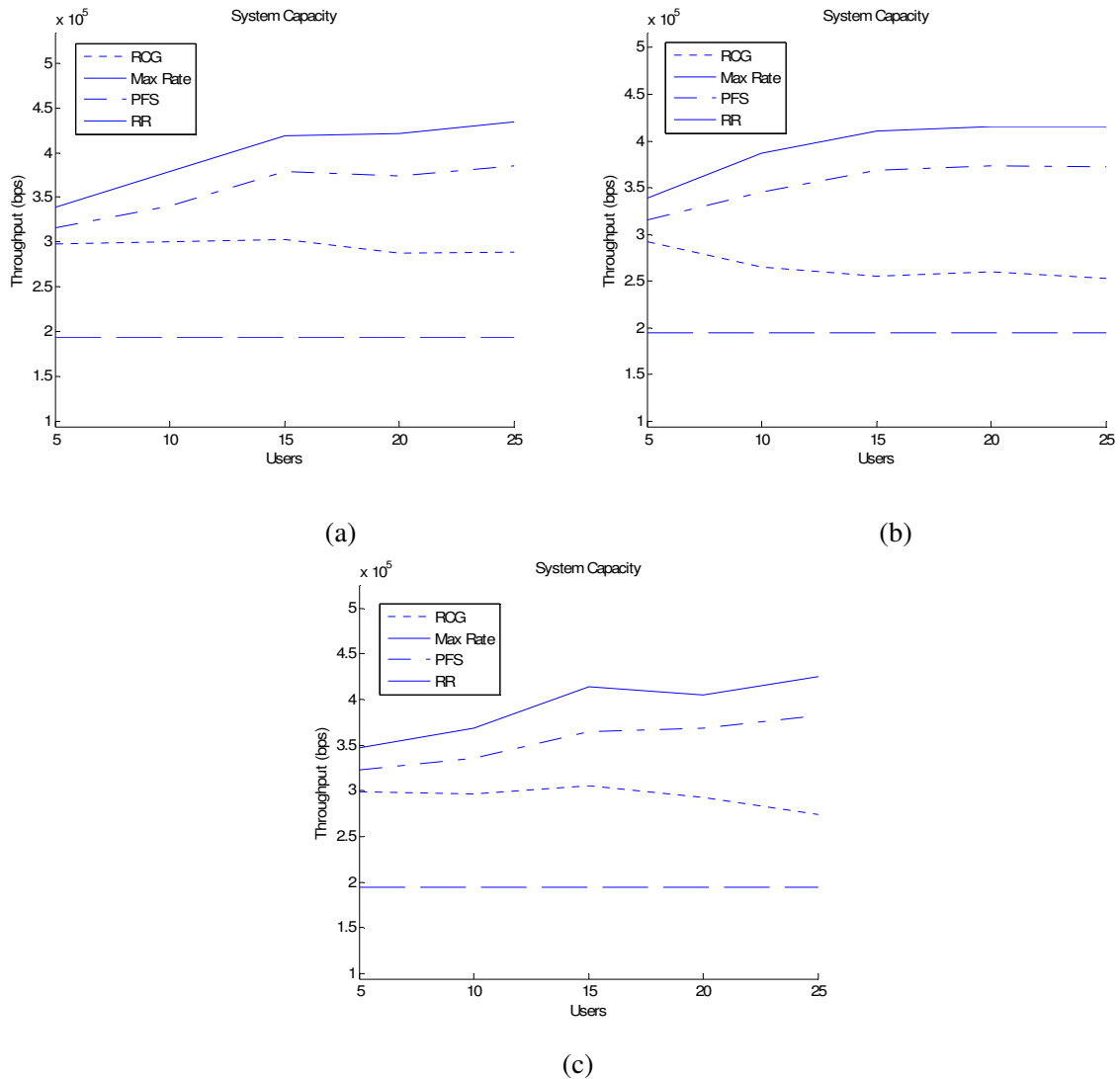
Time slots	100
Sub-carriers	16
$t_c$	20
Users	From 5 to 25
Rates (Kb per time slot)	16, 64 and $exprnd(30)$
Mobile Speed (km/h)	3

**Table 5.I** Simulation parameters.

To compute system throughput, at first the scheduler scheme is taken to know the sub-carriers over which the user is transmitting, because each channel is divided in 16 sub-carriers and there are so channels as users in the system. After that, the channel capacity is computed at each sub-carrier according to Shannon-Hartley Theorem. Finally, the mean value for all time slots simulated is computed.

### 5.3.2 Simulation Results

With the parameters showed in Table 5.I and the system model explained in chapter 3, different set of users are simulated in MATLAB, the results are showed in Fig5.1.



**Figure 5.1** System Capacity versus users in three different scenarios, (a) scenario A, (b) scenario B, (c) scenario C.

The system capacity achieved by RR policy reaches the lowest value because this algorithm does not take into account multiuser diversity and allocates all sub-carriers to one user at each time slot independently of users' channel response and rate requirements. The value reached remains constant with different set of users because the MSs are simulated with the same channel response average.

Max Rate algorithm, according to system throughput, reaches the best result because this algorithm allocates system resources to users with the strongest channel and it maximizes the system throughput. The larger the number of users in the system, the larger the system

throughput because finding a channel stronger is much more probable when there are much more users.

These two algorithms represent the system throughput's boundaries.

PFS algorithm exploits the fact that the propagation channel between the BS and MSs is independently each other, giving rise to multiuser diversity. As we can see in Fig5.1, this algorithm has a good behaviour because it reaches a good level of system throughput without comprising fairness. In this case, users compete for resources not directly based on their SNRs but only after normalization by their respective average throughputs. As we said, before, the larger the users in the system the larger the system throughput. Because getting a user stronger is much more probable. But the value reaches with this policy is lower than the value got by Max Rate policy because of maintenance fairness.

Until now, the different scheduler policies are independently of which scenario is treating because algorithms allocate resources without taking into account users' rate requirements.

As we can see in Fig5.1, when the number of users in the system increases, the system throughput of RGS algorithm is not so good as the system throughput achieves with previous algorithms.

When the scheduler scheme is computed by RCG algorithm, the system throughput remains constant or decreases when users increase. The reason of this behaviour is because of RCG policy. At first RGC algorithm computes the number of sub-carriers each user gets based on rate requirements, after that RCG algorithm allocates sub-carriers to users based on channel state information. So this policy only takes into account multiuser diversity after allocating the number of sub-carriers each user gets based on rate requirements.

## Chapter 6

# Delay

In a real-time wireless environment, every packet generated by applications has a deadline associated with it. If the system cannot allocate enough resources to serve the packet before the deadline, the packet would be dropped. Different applications have different delay requirements that should be guaranteed by the system so as to maintain some given packet dropped probabilities. Compared to the wired network, it is more challenging to provide QoS guarantees in the wireless network because of resource limitation, error characteristic and mobility.

Scheduling the available bandwidth to different services is a non-trivial task. Some applications, e.g. video, require a huge amount of bandwidth and are very delay sensitive, whereas some others, e.g. voice, require a relatively small amount of bandwidth with less strict delay requirements [18]. These applications should all be satisfied if the system has enough resources to support and maintain QoS of each individual application.

### 6.1 Previous work

In a wireless environment, any packet can be dropped mainly because of the following reasons: not enough resources for sending a critical packet; wireless channel quality; or collision of the packets from different applications. We want to evaluate the probability that a user is transmitting less than  $R_{min}$  using different scheduling algorithms, which is related to the packet dropping rate.

In order to ensure the availability of time slots, there is a BS which allocates the system resources. There is a queue in the MS for holding ready-to-send packets. If the BS informs a MS that the number of time slots available, the MS would select some packets from the queue for transmission. The BS would schedule the available bandwidth to different MS, as we could see in chapter 3, according to four different algorithms, RCG, Max Rate, PFS and RR.

## 6.2 MATLAB Simulations

In this section the probability that a user is transmitting less than  $R_{min}$  is evaluated, where  $R_{min}$  is the demanded rate for a user according to the service on demand. The performance is analyzed and compared through MATLAB computer simulations with *probability\_delay.m* file, see Appendix F.1.

### 6.2.1 Model Description

The model used is the same model explained in chapter 3, but in this case, three different scenarios to improve performance are considered.

- A. All users are transmitting voice, constant rate, 16Kb per time slot.
- B. All users are transmitting video, constant rate, 64Kb per time slot.
- C. 10% of users are transmitting video, constant rate, 64Kb per time slot, 40% of users are transmitting voice, constant rate, 16Kb per time slot and 50% of users are transmitting data, exponentially distributed rate, with a mean of 30Kb per time slot.

Table 6.I shows simulation parameters.

Time slots	100
Sub-carriers	16
$t_c$	20
Users	From 5 to 25
Rates (Kb per time slot)	16, 64 and $exprnd(30)$
Mobile Speed (km/h)	3
Voice deadline (time slots)	1
Video deadline (time slots)	2
Data deadline (time slots)	4

**Table 6.I** Simulation parameters

### 6.2.2 Simulation Results

The simulation results are showed in Figs6.1-6.7. In these examples, the channel is split in 16 sub-carriers, so each sub-carrier can reach about 25Kb per time slot according to filtered Gaussian noise model to simulate MS. Suppose that users are transmitting video,



64Kb per time slot, and the deadline of these packets is one time slot, so one user needs at least 3 sub-carriers at each time slot to satisfy QoS requirements. In this situation, the system could give service to 5 users with zero probability. These simulations give us a measure of the number of users this system can serve with zero probability or in other words, the users' happiness level.

On the other hand, we suppose that all users are transmitting data, and that to achieve QoS, one user cannot be more than 4 time slots without transmitting, so the probability that one user is transmitting less than  $R_{min}$  shows us the probability that users achieve QoS.

In Fig6.1-6.6, we can see different results according to different scheduling policies. If we allocate the system resources using RR algorithm, the packet deadline should be so time slots as users in the system in order to be able to give service to all users at least one time and to be able to avoid packet drop. RR algorithm allocates all sub-carriers to one user at each time slot and this amount of data is enough to satisfy the data requirements, so the delay in time slots, according to this algorithm, is equal to the number of users in the system. The larger the number of users in the system, the larger the delay, so if we want to reach QoS, our system will only accept so users as the delay restriction. In results showed, RR algorithm has a linear behaviour. It means that at each time slot, each user gets data enough to satisfy demanded data service.

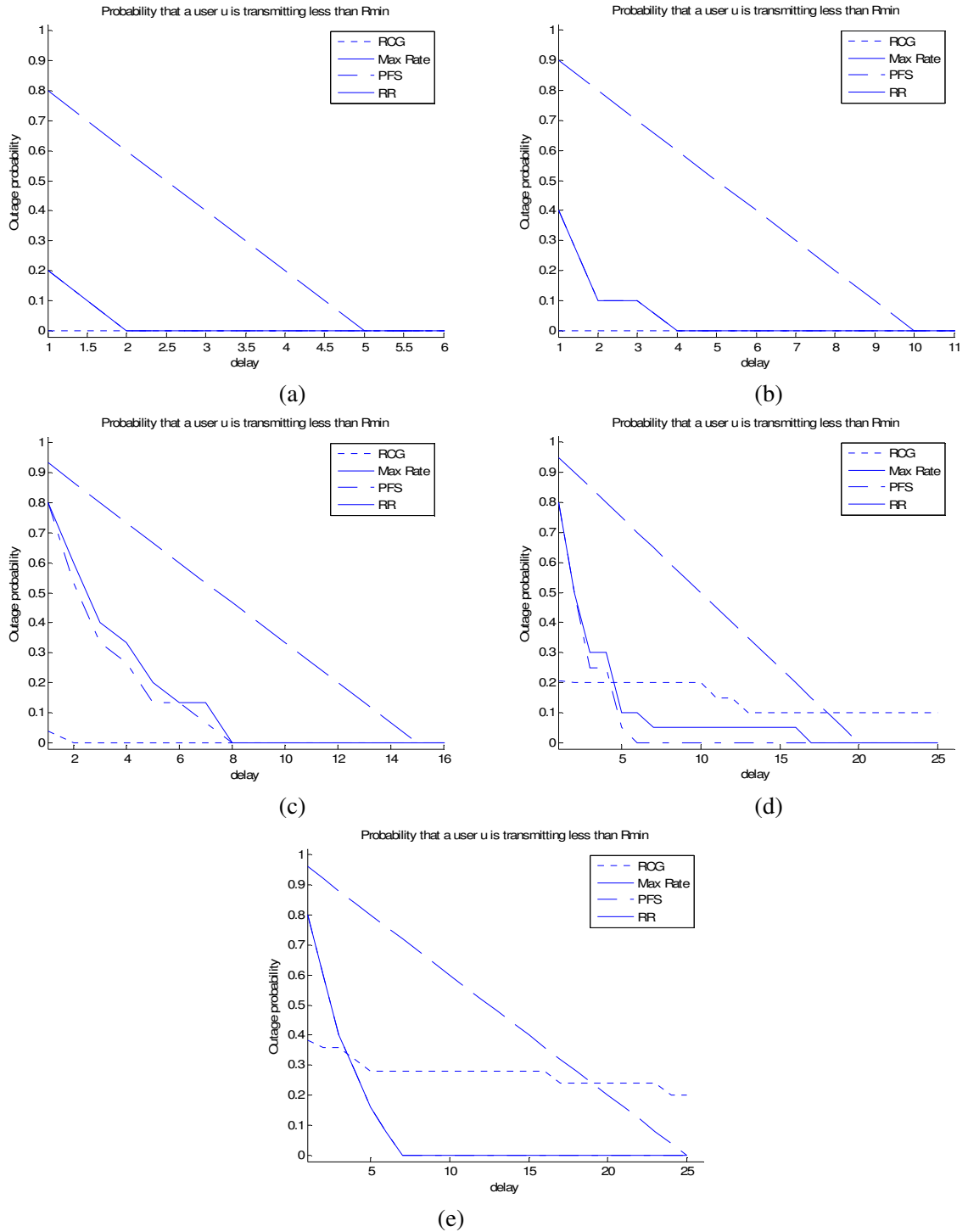
If we allocate the system resources using RCG algorithm, we can see that when there are sub-carrier enough to satisfy demanded services by users, the algorithm reaches zero delay, so we can say that RCG algorithm carry out QoS. For example when there are 5 users transmitting voice (see Fig1 (a)) there are sub-carriers enough to serve users without waiting. But the larger the number of users in the system, satisfying data requirements for all users is more difficult, because RCG algorithm allocates more sub-carriers to users who are demanding more data, so the larger the users, the larger the probability that one user does not get sub-carriers enough because the system resources are limited. As we can see in Fig6.3 and Fig6.6, RCG algorithm does not reach zero probability because at least there is one user who does not get sub-carriers enough. This scenario takes place when the services demanded by users are very different, in mix scenarios. For example, in Fig6.3, there is one user demanding 100kbps and other demanding 2kbps, according to RCG algorithm, the scheduler allocates sub-carriers enough to satisfy the user who are demanding more resources, because of limited resources in the system, the user who is demanding 2kbps never get one sub-carrier. This scenario shows us that RCG algorithm is not fairness enough.

As we can see in Fig6.1-6.6, according to delay requirements, PFS algorithm has the best behaviour. It reaches zero probability faster than the other algorithms because there are no users without getting service in all time slots simulated and the sub-carriers allocated to users are enough in order to satisfy data requirements. The larger the number of users in the system, the larger the delay, so if we want to carry out QoS, we have to limit the number of users. In Fig6.1 and Fig6.4, we can see that the users are transmitting voice and the delay restriction for this service is two time slots, so if we want to reach QoS, we have to restrict the number of users to 5 at least.

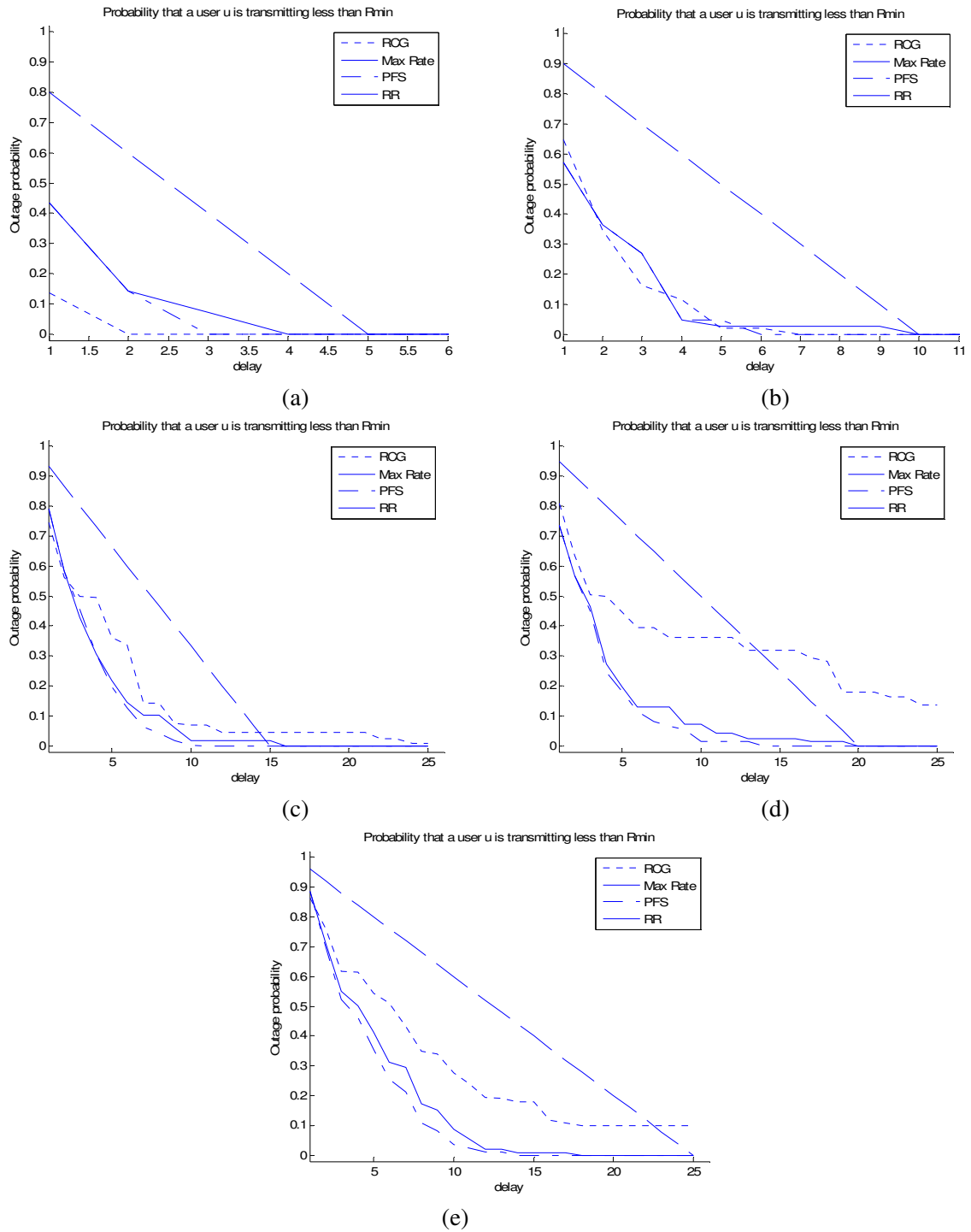
Max Rate algorithm allocates sub-carriers to users who reach the strongest SNR at each time slot, so when all users have the same channel response average, is probably that

one user gets resources enough to satisfy data requirements in few time slots. For example when users are transmitting voice, in Fig6.1, we can see that the delay is no much more than the delay according to PFS algorithm, so we can say that Max Rate algorithm achieves good results comparing with PFS algorithm in scenarios where all users have the same channel response average. But when there are users with channel responses stronger than others, Fig6.4-6.6, for example when these users are nearer to BS than others, the whole system resources are allocated to these users, so there are users who never get sub-carriers to transmit. In this case, the probability that one user is transmitting less than  $R_{min}$  increase when users increase and remain constant in time because users with weak channel will never get sub-carriers.

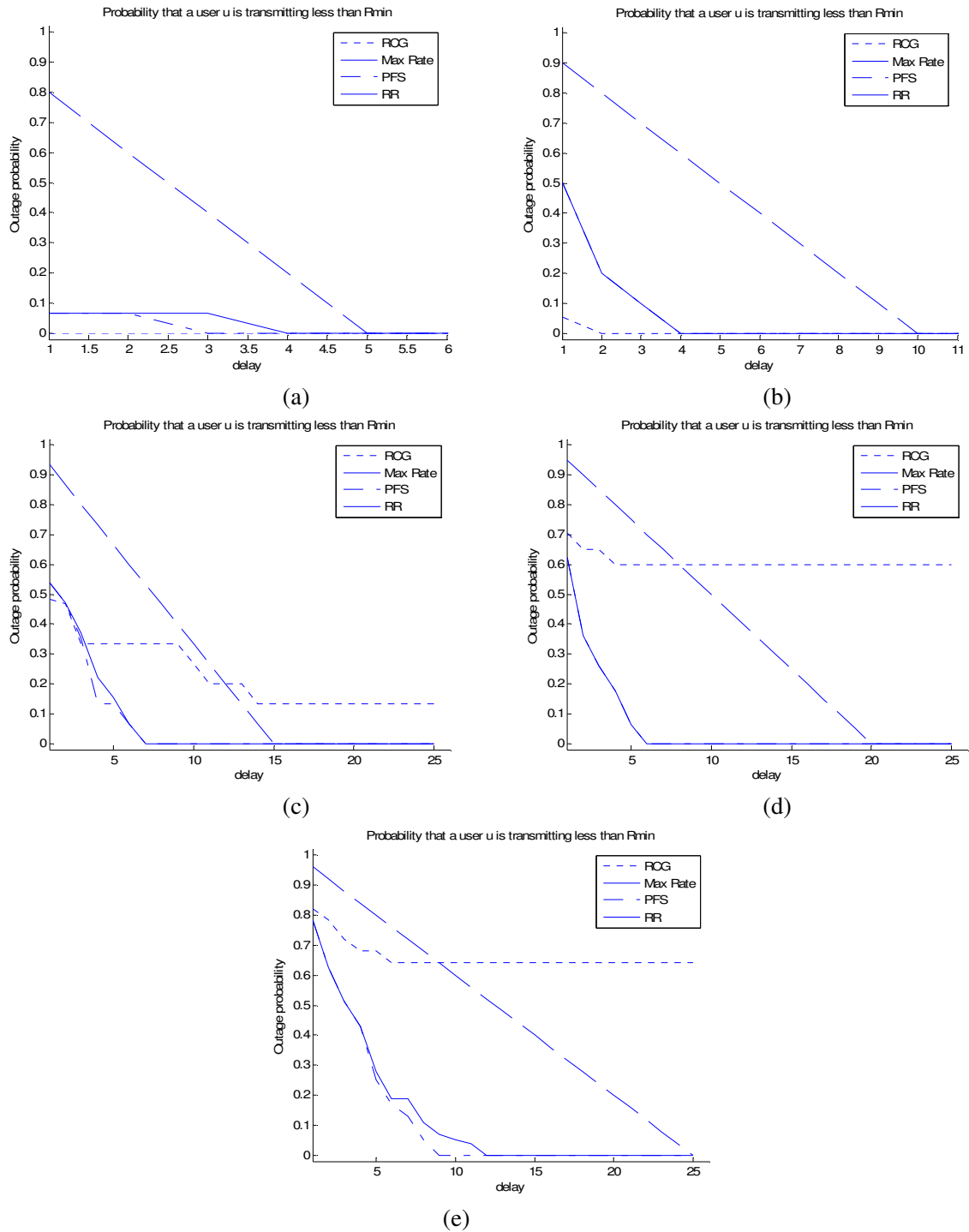
As we can see in Fig6.1 and Fig6.2, algorithms reach zero probability when users are transmitting video later than when users are transmitting voice. This occurs because of video service demand more resources than voice services, so the amount of data demanded by users who are transmitting video is larger than the amount of data demanded by users who are transmitting voice.



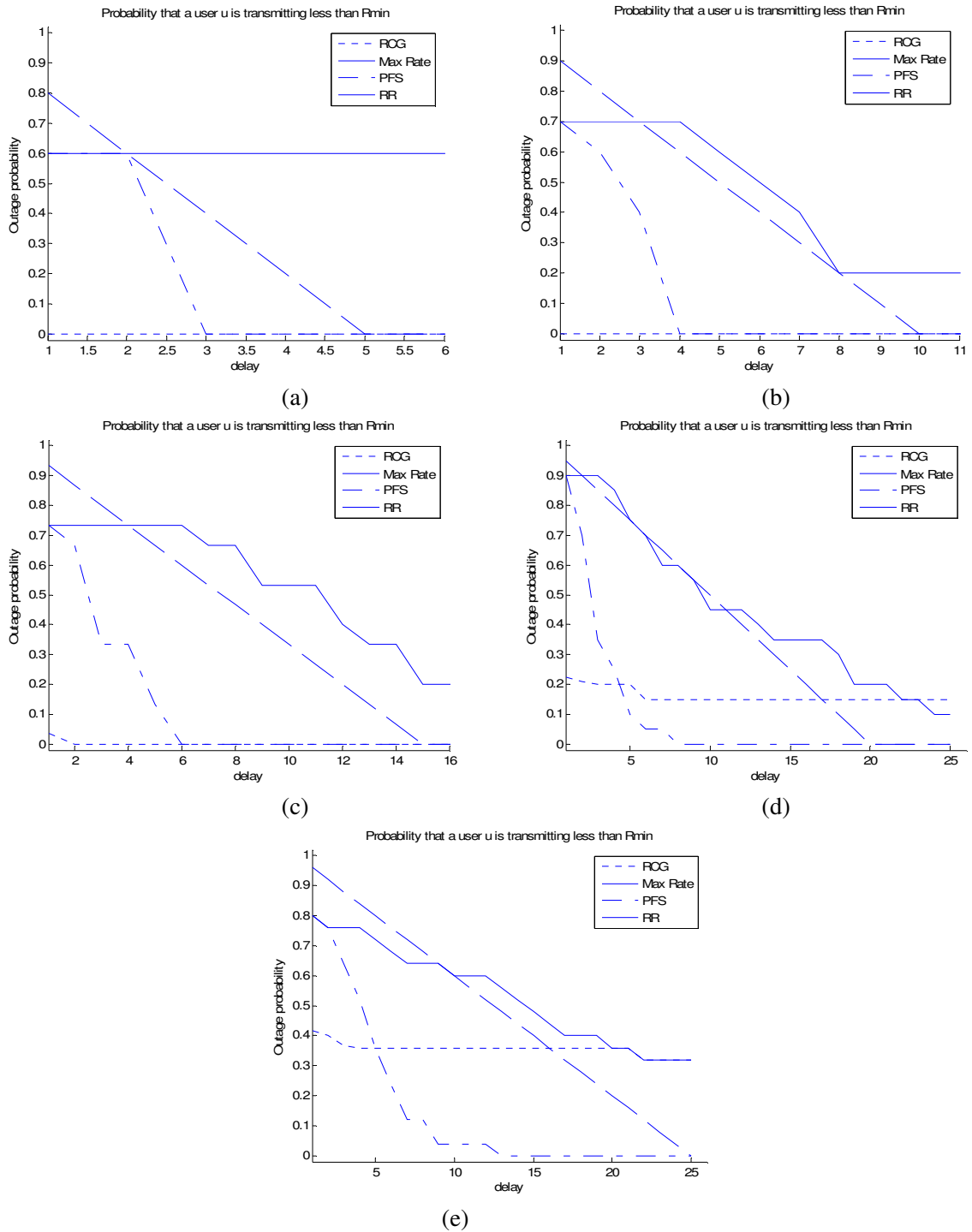
**Figure 6.1** Probability that a user is transmitting less than  $R_{min}$  in scenario A with equal channel response average. (a) 5 users, (b) 10 users, (c) 15 users, (d) 20 users, (e) 25 users.



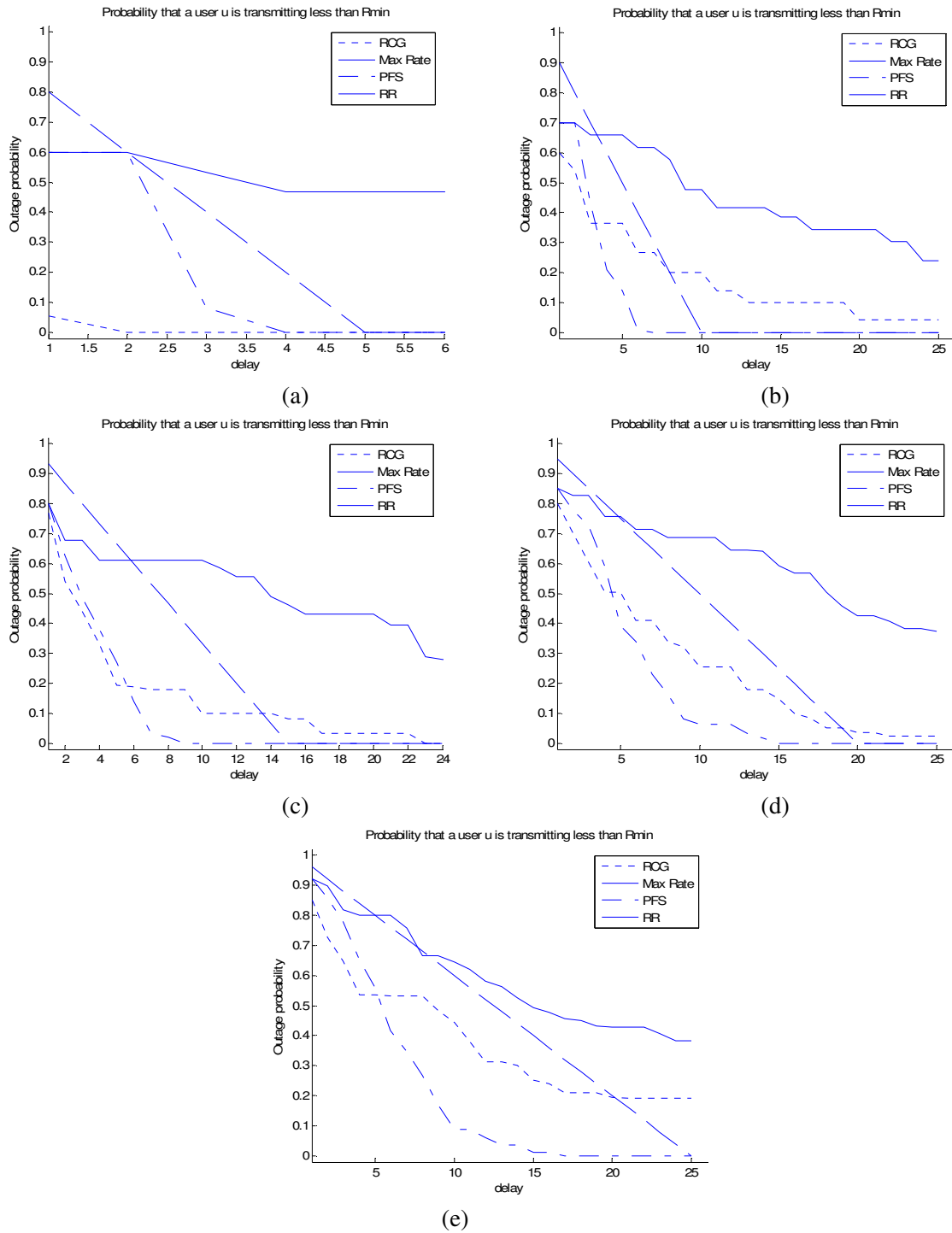
**Figure 6.2** Probability that a user is transmitting less than  $R_{min}$  in scenario B with equal channel response average. (a) 5 users, (b) 10 users, (c) 15 users, (d) 20 users, (e) 25 users.



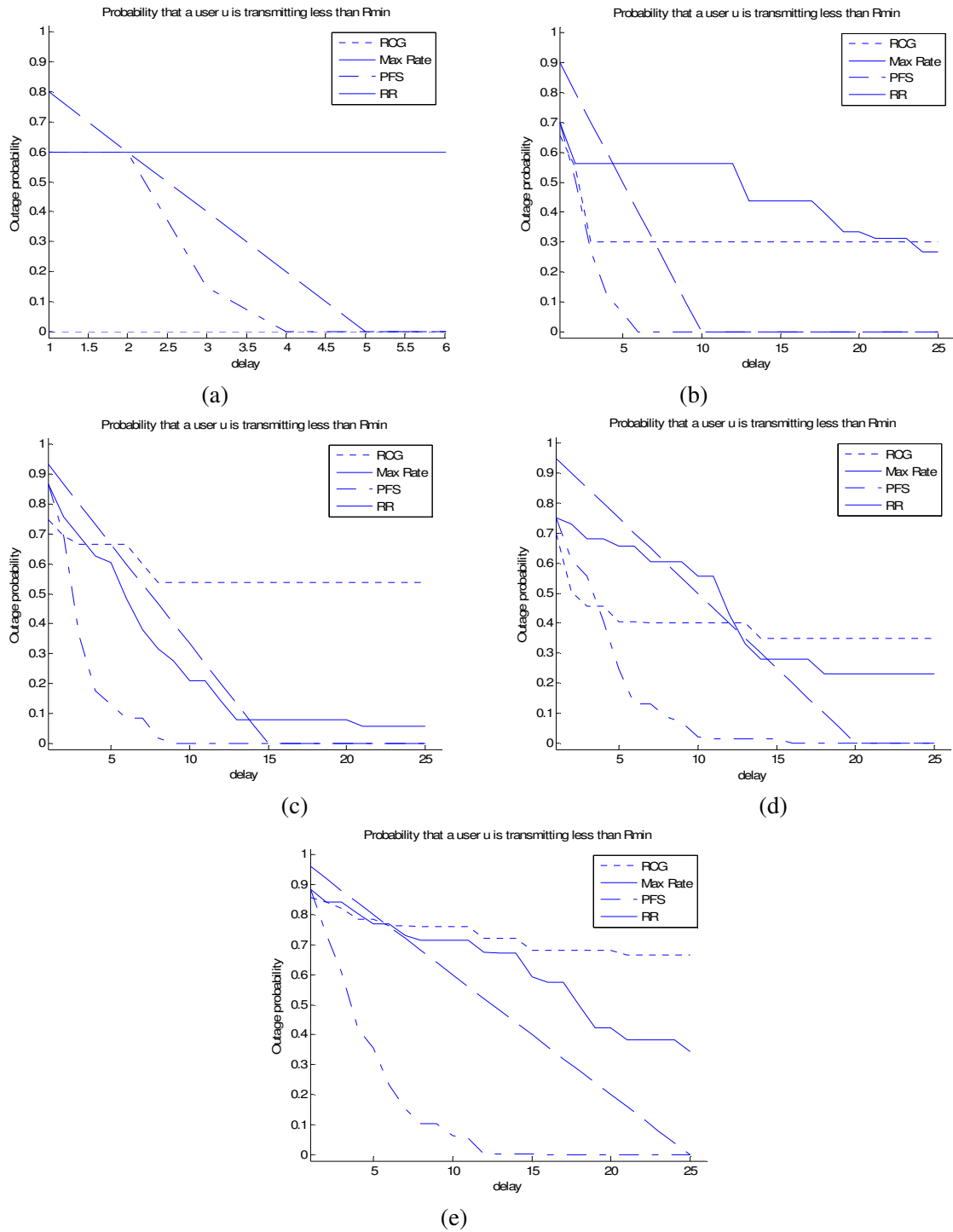
**Figure 6.3** Probability that a user is transmitting less than  $R_{min}$  in scenario C with equal channel response average. (a) 5 users, (b) 10 users, (c) 15 users, (d) 20 users, (e) 25 users.



**Figure 6.4** Probability that a user is transmitting less than  $R_{min}$  in scenario A with unequal channel response average. (a) 5 users, (b) 10 users, (c) 15 users, (d) 20 users, (e) 25 users.



**Figure 6.5** Probability that a user is transmitting less than  $R_{min}$  in scenario B with unequal channel response average. (a) 5 users, (b) 10 users, (c) 15 users, (d) 20 users, (e) 25 users.



**Figure 6.6** Probability that a user is transmitting less than  $R_{min}$  in scenario C with unequal channel response average. (a) 5 users, (b) 10 users, (c) 15 users, (d) 20 users, (e) 25 users.

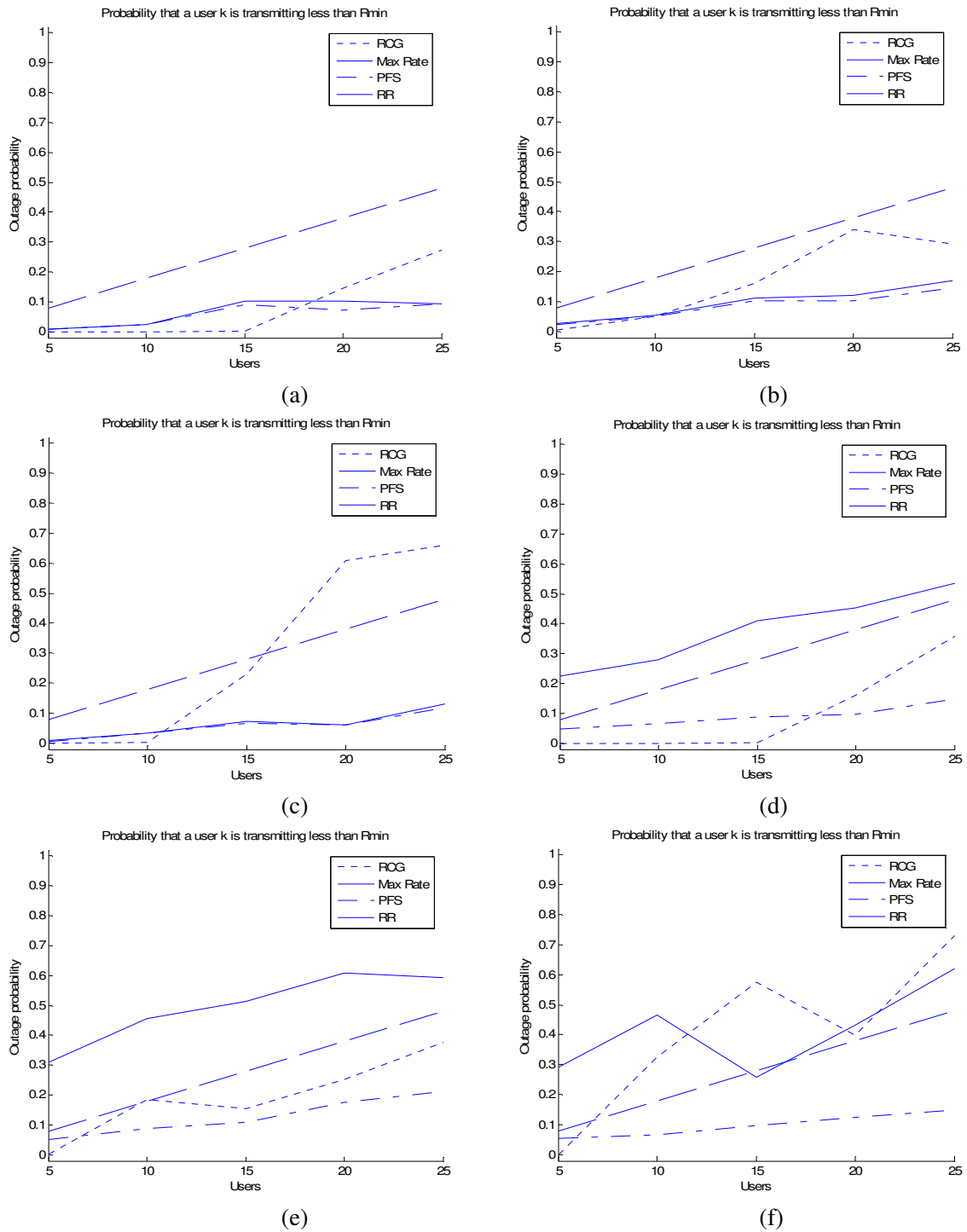


Finally in Fig6.7, we show that the larger the number of users in the system, the larger the probability that one user is transmitting less than  $R_{min}$  and the larger the packets dropped.

As we can see in Fig6.7 (a)-(c), Max Rate algorithm has the same behaviour that PFS algorithm, as we said before it takes place when users have the same channel response average. But when the channel response average is different, see Fig6.7 (d)-(f), Max Rate algorithm has the worst behaviour because of resource allocation policy.

In Fig6.7 we can notice the behaviour of RCG algorithm in different scenarios. For example, in Fig6.7 (a) and (b), we can see that RCG algorithm has the best outage probability until the number of sub-carriers demanded by users are more than the system resources, in this case the outage probability increase rapidly. The worst behaviour of this algorithm takes place in mix scenarios where there are users who are demanding very different services. In this case, at first, the system serves to users who are demanding more data and after satisfying these users, the remaining sub-carriers are shared out amongst the others.

RR algorithm outage probability increases linearly with users because of resources allocated to users at each time slot are enough for satisfying data requirements.



**Figure 6.7** Probability that a user is transmitting less than  $R_{min}$ , with equal channel response average, (a) scenario A, (b) scenario B, (c) scenario C, with unequal channel response average (d) scenario A, (e) scenario B, (f) scenario C.

## Chapter 7

# Conclusions

In this report have been discussed important properties which take place in a wireless environment where the system resources are shared among users and where channel user responses change randomly and independently each other. OFDM has been proposed as a technique to transmit signals over wireless channels because of efficient bandwidth use. The target of this Thesis has been to exploit the idea of multiuser diversity, which can be obtained exploiting the time-varying users' channel characteristics.

In order to develop this issue, a set of wireless channel simplifications have been required because of time and tool limitations. To analyze different scheduling policies, the simulated wireless environments have been urban area and pedestrian speed because finding this scenario in real life is more probable.

In this report, the most important section has been the performance analysis done over the suggested scheduling policies, according to three different QoS criteria, fairness, throughput and delay. Max Rate and RR algorithms have been suggested as a reference because the first, achieves the maximum system throughput, but is totally unfair and the second is totally fair, but it does not take into account the idea of multiuser diversity, so it does not get a good system throughput.

RCG and PFS algorithms exploit the idea of multiuser diversity, but they work differently. RCG policy computes the number of sub-carriers each user need according to rate requirements and it adapts them to achieve the same number of sub-carriers that the system can offer, removing sub-carriers from the user who is transmitting less data. After that sub-carriers are allocated according to channel response getting rise to multiuser diversity. This means that when there are users demanding services with very different data rates, users who are transmitting less data never get sub-carriers enough and these users will not be served. So, RCG algorithm works well in environments where users are demanding seemed services and when there are resources enough.

PFS algorithm does not take into account rate requirements, it works keeping track of the average throughput of each user in a past window of length  $t_c$  and allocating sub-carriers to the user with the largest  $R_{k,n}(t)/T_{k,n}(t)$ . This algorithm treats sub-carries independently each other and has been demonstrated that PFS policy with a low  $t_c$  parameter, reaches

better results about fairness, throughput and delay than RCG policy in mix scenarios and when there are not sub-carrier enough to satisfy all users.

The performance reaches from the suggested algorithms has been analyzed and compared through MATLAB computer simulations, a good theoretical tool to take a first approximation, but not necessary the best in this field.

## Chapter 8

### Future Work

A lot of work remains to be done and can be continued in a future Master Thesis. Some of them would be the solution to some “fallacies and pitfalls” that appeared throughout the Thesis and also, the design suggests several extensions which are showed in the next points:

- The simulation results showed in this paper have been done taken into account at most 100 time slots because of simulation length. So, in order to reinforce conclusions it would be more interesting to extend the number of time slots simulated, especially for the PFS algorithm which takes time to converge properly for large  $t_c$  values. Increasing simulation length, improving scripts or using other technique to reduce simulation time.
- Playing with different parameters to simulate different scenarios and different channel responses. It would be able to widen conclusions and analyze more accurately the algorithm’s behaviour.
- Reducing simplifications in order to get results closer to real life.
- Implementing an algorithm taken into account the PFS showed in this report. In this paper we have been working with a PFS algorithm which takes each sub-carrier independently each other, so an interested work could be to consider a PF scheduling that handles transmission with multiple carriers but without treating independently each other, see [27].
- To improve the algorithm’s performance could be interesting to predict time-varying channels. There are some papers [28] and [29], which propose adaptive channel predictors for OFDM communications over time-varying channels, and which also consider the use of adaptive channel predictors for delay-free equalization, thereby avoiding the need for regular transmission of pilot symbols.

- All the work developed in this Thesis has been theoretical, getting results from MATLAB simulations, so undergoing these theoretical ideas in a real environment could be an interesting future work.

## Appendix A

### A.1 FGN\_model.m

```

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Name: FGN_model
%Author: Alfonso Bahillo
%Date: February 2006
%Last modification: March 2006
%Support: MATLAB 7.0.019920 (R14)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Description: this function creates a fading signal from in-phase and
% quadrature Gaussian noise sources. Applying the appropriate filtering
% provides the Doppler spectrum. A tap line is simulated and the channel
% response is divided in different subchannels.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%input arguments:
%v, mobile speed in km/h (between 3 and 80 km/h).
%C, number of sub-carriers per channel (power by 2).
%T, number of time slots
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%output arguments:
%fo, coherence frequency.
%CH, matrix which stores the users' channel response.
%Noise, storing noise level.
%I, fft points (sample rate)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

function [fo CH Noise I]=FGN_model(v,C,T)

%Parameters
fc=2.4*10.^9;           %carrier frequency
v=v*1000/3600;         %mobile speed (m/s)
c=3*10.^8;             %light speed (m/s)
fd=fc*v/c;            %max Doppler frequency
wd=2*pi*fd;
Q=100;                %taps resolution
I=2048;               %fft resolution in tau direction
N=500;                %vector length t and f
t=0.001:10/N:10;     %vector t
f=-2*fd:4*fd/N:2*fd-4*fd/N; %vector f
tau=0.0001:4/Q:4;    %tap delay until 4us, typical urban area

%matrix initialization
real=zeros(Q,N);
imag=zeros(Q,N);
expo=zeros(Q,N);
envcomplex=zeros(Q,N);

```

```

expo=zeros(Q,N);
complex=zeros(Q,N);
auxTm=zeros(1,Q);
yt=zeros(Q,I);
Yf=zeros(Q,N);
ppf=zeros(I,I);
CH=zeros(C,(I/C)*T);    %Channel filtered to generate subchannels

%complex white Gaussian process, unit variance, zero mean
real=randn(Q,N);
imag=randn(Q,N);
envcomplex=(real+j*imag)/sqrt(2);
expo=exp(j*(2*pi*rand(Q,N)));    %phase uniformly distributed [0 2pi]
complex=envcomplex.*expo;        %wi(t)
complexf=fft(complex);           %Wi(f)

figure(1)
semilogy(t,abs(complex(1,:))/max(abs(complex(1,:))));
title('Typical Rayleigh fading envelope, normalized');
xlabel('Time (s)');
ylabel('Signal level dB, normalized');

%one tap loss model
loss=2-exp(t/15);
complexloss=complex(1,:).*loss;

figure(2)
semilogy(t,abs(complexloss(1,:)));
title('Pat loss envelope');
xlabel('Time (s)');
ylabel('Signal level dB');

%Power Spectral Density by Jake's model
for i=1:N
    if f(i)>-fd && f(i)<fd
        S(i)=1/(pi*fd*(sqrt(1-(f(i).^2)/fd.^2)));
    else
        S(i)=0;
    end
end
end

figure(3)
plot(f,abs(S));
title('Power Spectrum');
xlabel('Frequency (Hz)');
ylabel('Magnitude');

%filtering Gaussian process
for q=1:Q
    Yf(q,:)=sqrt(S).*complexf(q,:);
end

%one tap in frequency domain (t direction)
figure(4)
plot(f,abs(Yf(1,:)));
title('Tap Frequency Response');
xlabel('Frequency (Hz)');
ylabel('Magnitude');

%Matrix in time domain
for q=1:Q
    yt(q,:)=ifft(Yf(q,:),I);
end

```



```

%PDP at any given time without scaling
figure(5)
plot(tau,abs(yt(:,10)));
title('Power Delay Profile (without being scaled)');
xlabel('Delay (us)');
ylabel('Magnitude');

%Exponential decaying according to different environments
%Urban Area
for q=1:Q
    e(q)=exp(-q/10);
    yt(q,:)=e(q).*yt(q,:);
end

%almost one tap (one way)
%for q=1:Q
%    e(q)=exp(-q);
%    yt(q,:)=e(q).*yt(q,:);
%end

%Hilly Area
%for q=1:Q
%    if q<40
%        e(q)=exp(-q/10);
%    else
%        e(q)=exp(-q/10)+0.5*exp(-(q-40)/10);
%    end
%    yt(q,:)=e(q).*yt(q,:);
%end

%computing Tm
threshold=0.2; %threshold level 20%
for tslot=1:T
    auxTm(tslot)=threshold.*max(abs(yt(:,tslot)));
end
for tslot=1:T
    aux2Tm(tslot)=0; %aux to compute position
    for q=1:Q
        if abs(yt(q,tslot))>auxTm(tslot)
            aux2Tm(tslot)=aux2Tm(tslot)+1;
        end
    end
    Tm(tslot)=tau(aux2Tm(tslot)); %delay spread
    fo(tslot)=1/(Tm(tslot)*10.^(-6)); %coherence bandwidth in Hz
end

%for plotting one example
tauf=2400*1e6-2*fo(10):4*fo(10)/I:2400*1e6+2*fo(10)-(4*fo(10)/I);
auxTm(1,:)=auxTm(10); %to plote threshold level

%PDP at any given time
figure(6)
hold on
plot(tau,abs(yt(:,10)));
plot(tau,auxTm,'r');
title('Power Delay Profile');
xlabel('Delay (us)');
ylabel('Magnitude');
hold off

%Spaced-frequency correlation function
for tslot=1:T
    ppf(:,tslot)=fftshift(fft(yt(:,tslot),I));
end

```

```

figure(7)
semilogy(tauf,abs(ppf(:,10)));
title('Frequency channel response');
xlabel('Frequency (Hz)');
ylabel('Magnitude');

figure(8)
hold on
semilogy(tauf,abs(ppf(:,10)),'b');
for c=1:C
    semilogy([tauf(I/C*c)tauf(I/C*c)],[abs(ppf(I/C*c,10))0],'r');
end
title('Frequency channel response');
xlabel('Frequency (Hz)');
ylabel('Magnitude');
hold off

%channel response now is a row
ppf=ppf';
for tslot=1:T
    Noise(tslot)=mean(abs(ppf(tslot,:)))/4;
end

%CH, matrix which stores subchannel responses at each row of length I/C
aux2=floor(I/C);
aux1=aux2-1;

for tslot=1:T
    for c=1:C
        CH(c,1+(I/C)*(tslot-1):(I/C)*tslot)=ppf(tslot,1+aux1*(c-1):aux2+aux1*(c-1));
    end
end
end

```

## A.2 Jakes\_model.m

```

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Name: jakes_model
%Author: Alfonso Bahillo
%Date: February 2006
%Last modification: February 2006
%Support: MATLAB 7.0.019920 (R14)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Description: this function simulates mobile channel behaviour according
%             to Jakes' model by summing a set of complex sinusoids.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%input arguments:
%v, mobile speed in km/h (between 3 and 100 km/h).
%C, number of sub-carrier per channel.
%T, number of time slots simulated.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%output arguments:
%fo, coherence frequency (to compute Capacity, Shannon Theorem).
%CH, matrix which stores the users' channel responses.
%Noise, store noise level.
%I, fft points (sample rate).
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

function [fo CH Noise I]=jakes_model(v,C,T)

%Parameters
fc=2.4*10.^9;           %carrier frequency
v=v*1000/3600;         %mobile speed
c=3*10.^8;             %light speed
fd=fc*v/c;             %max Doppler frequency
wd=2*pi*fd;           %2*pi*f
N=5120;                %vector length t and f
No=20;                 %number of complex lower-frequency oscillators
Eo=1;
fiN=0;                 %initial phase
Q=100;                 %taps resolution
I=2048;                %fft points in tau direction
tt=2.001:8/N:10;      %t direction
ttaux=0.001:10/N:10;  %Plot from 0 to 10 sec
taut=0.0001:6/Q:6;    %tau direction
f=-2*fd:4*fd/I:2*fd-4*fd/I; %f direction

%Matrix initalitation
xc=zeros(Q,N);
xs=zeros(Q,N);
Tt=zeros(Q,N);
Ttf=zeros(Q,I);
Ttauf=zeros(I,I);
real=zeros(Q,N);
imag=zeros(Q,N);
expo=zeros(Q,N);
auxTm=zeros(1,Q);
H=zeros(C,I);          %Each row is a filter to select subchannels
CH=zeros(C,I/C);      %Channel filtered to generate the subchannels

%xc(t), in-phase component
for q=1:Q
    for t=1:N
        for n=1:No

```

```

xc(q,t)=xc(q,t)+2*cos((pi*n)/(No+1))*cos(tt(t)*wd*cos(2*pi*n/(4*No+2)));
    end
    xc(q,t)=xc(q,t)+sqrt(2)*cos(fiN)*cos(wd*tt(t));
    end
end

%xs(t), quadrature component
for q=1:Q
    for t=1:N
        for n=1:No

xc(q,t)=xs(q,t)+2*sin((pi*n)/(No+1))*cos(tt(t)*wd*cos(2*pi*n/(4*No+2)));
        end
        xs(q,t)=xs(q,t)+sqrt(2)*sin(fiN)*cos(wd*tt(t));
        end
    end

%complex envelope
Tt=(Eo/sqrt(2*No+1))*(xc+j*xs);

%plot one example, row 1
figure(1)
semilogy(ttaux,abs(Tt(1,:)));
title('Typical Rayleigh fading envelope');
xlabel('Time (s)');
ylabel('Signal level dB');

%path loss model
loss=2-exp(tt/15);
Tloss=Tt(1,:).*loss;

figure(2)
semilogy(ttaux,abs(Tloss(1,:)));
title('Path loss envelope');
xlabel('Time (s)');
ylabel('Signal level dB');

%Fourier Transform in t domain
for q=1:Q
    Ttf(q,:)=fftshift(fft(Tt(q,:),I));
end

%power spectral density
figure(3)
plot(f,abs(Ttf(1,:)).^2)
title('Tap Frequency Response');
xlabel('Frequency (Hz)');
ylabel('Magnitude');

%Tap delay line generation
real=randn(Q,N); %Unit variance, zero mean
imag=randn(Q,N);
envcomplex=(real+j*imag)/sqrt(2); %matrix with Wi(t)'s envelope
expo=exp(j*(2*pi*rand(Q,N))); %phase uniformly distributed [0 2pi]
complex=envcomplex.*expo;

%random amplitude
Tt=Tt.*complex;

%Exponential decaying
for q=1:Q
    e(q)=exp(-q/20); %decaying factor
    Tt(q,:)=e(q).*Tt(q,:);
end

```

```

end

%Tm compute
threshold=0.2; %threshold level 20%
for tslot=1:T
    auxTm(tslot)=threshold.*max(abs(Tt(:,tslot)));
end

for tslot=1:T
    aux2Tm(tslot)=0; %aux to compute vector T position
    for q=1:Q
        if abs(Tt(q,tslot))>auxTm(tslot)
            aux2Tm(tslot)=aux2Tm(tslot)+1;
        end
    end
    Tm(tslot)=taut(aux2Tm(tslot)); %delay spread
    fo(tslot)=1/(Tm(tslot)*10.^(-6)); %coherence bandwidth in Hz
end

%to plot one example
tauf=2400*1e6-5*fo(1):10*fo(1)/I:2400*1e6+5*fo(1)-(10*fo(1)/I);
auxTm(1,:)=auxTm(1); %plot threshold level

figure(4)
hold on
plot(tauf,abs(Tt(:,1)));
plot(tauf,auxTm,'r');
title('Power Delay Profile');
xlabel('Delay (us)');
ylabel('Magnitude');
hold off

%Spaced-frequency correlation function
for t=1:I
    Ttauf(:,t)=fftshift(fft(Tt(:,t),I));
end

figure(5)
hold on
semilogy(tauf,abs(Ttauf(:,1)))
for c=1:C
    semilogy([tauf(I/C*c)tauf(I/C*c)],[abs(Ttauf(I/C*c))0],'r');
end
title('Frequency channel response');
xlabel('Frequency (Hz)');
ylabel('Magnitude (dB)');
hold off

%Ttauf is now a row
Ttauf=Ttauf';
for tslot=1:T
    Noise(tslot)=mean(abs(Ttauf(tslot,:)))/4;
end

%CH, matrix which stores subchannel responses at each row of length I/C
aux2=floor(I/C);
aux1=aux2-1;

for tslot=1:T
    for c=1:C
        CH(c,1+(I/C)*(tslot-1):(I/C)*tslot)=Ttauf(tslot,1+aux1*(c-1):aux2+aux1*(c-1));
    end
end
end

```

## Appendix B

### B.1 algorithms.m

```

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Name: algorithms
%Author: Alfonso Bahillo
%Date: February 2006
%Last modification: April 2006
%Support: MATLAB 7.0.019920 (R14)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Description: this function computes TxU different channel responses, one
% per user and time slot. After that, the scheduler scheme is computed
% according to four different algorithms, PFS, Max Rate, Round Robin and
% RCG.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%input arguments:
%users, number of different users.
%C, number of sub-carriers per channel.
%T, number of time slots.
%tc, effective memory of the average throughput.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%output arguments:
%XPFSS, XMaxRate, Xroundrobin and XRCG matrix that store the user scheduled
% per sub-carrier at each time slot, according to PFS, Max Rate, Round
% Robin and RCG algorithms respectively.
%auxPFS, auxMaxSNR, auxroundrobin and auxRCG matrix that store the rate
% which each user transmits per sub-carrier at each time slot.
%Rmin, deman rate vector.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

function
[XPFSS,XMaxRate,Xroundrobin,XRCG,auxPFS,auxMaxRate,auxroundrobin,auxRCG,Rmin]=algorithms(users,C,T,tc)

%matriz initialization
%auxiliar matrix to store rates
auxPFS=zeros(C,T);
auxroundrobin=zeros(C,T);
auxMaxRate=zeros(C,T);
auxRCG=zeros(C,T);
%matrix to store user ID
XPFS=zeros(C,T);
Xroundrobin=zeros(C,T);
XMaxRate=zeros(C,T);
XRCG=zeros(C,T);

%avoiding zero division

```

```

epsilon=0.0001;

%Storing in a matrix, R (Cx(T*users)), the different values of rate, for each
%user, sub-carrier and time slot.
R=zeros(C,T*users);

%matrix in use for RCG algorithm
Haux=zeros(C,T*users);
H=zeros(users,T);
Rmin=zeros(1,users);
aux1Rmin=zeros(1,users);
aux2Rmin=zeros(1,users);
Rmax=zeros(users,T);
subcuser=zeros(C,users); %storing the estimated transmission rate of
%different users in the same time slot
m=zeros(users,T); %sub-carriers demanded according to services
q=zeros(users,T); %sort m, to compute
mm=zeros(users,T); %sub-carriers allocated according to channel
%conditions
G=zeros(users,T); %matrix to compute second stage of RCG
aux1RCG=zeros(C,T*users);

%matrix in use for PFS algorithm
aux1PFS=zeros(1,users);
aux1MaxRate=zeros(1,users);
Th=zeros(C,T*users);

%generating TxU different channel responses
for u=1:users
    [fo CHH Noise I]=FGN_model(3,C,T); %v=3km/h user walking
    for tslot=1:T
        for c=1:C
            R(c,tslot+T*(u-
1))=(fo(tslot)/C)*log2(1+mean(abs(CHH(c,1+(I/C)*(tslot-
1):(I/C)*tslot))/Noise(tslot)));
            Haux(c,tslot)=abs(mean(CHH(c,1+(I/C)*(tslot-1):(I/C)*tslot))).^2;
        end
        H(u,tslot)=1/C*sum(Haux(:,tslot));
    end
end

%if you want to take channel B (user B) stronger than other channels
R(:,T+1:2*T)=2*R(:,T+1:2*T);

%PFS ALGORITHM

%At first time slot, the same average throughput, for all users
for u=1:users
    Th(:,1+T*(u-1))=epsilon; %Initial average throughput
end

%XPFs and auxPFS matrix generation
u=1:users;
for s=1:T
    for c=1:C
        aux1PFS=R(c,s:T*T*users)./Th(c,s:T*T*users);
        aux1MaxRate=R(c,s:T*T*users);
        for t=s:T*T*users
            if (R(c,t)./Th(c,t))==max(aux1PFS)
                XPFS(c,s)=ceil(t/T);
                auxPFS(c,s)=R(c,t);
                Th(c,t+1)=(1-1/tc)*Th(c,t)+1/tc*R(c,t);
            else
                Th(c,t+1)=(1-1/tc)*Th(c,t);
            end
        end
    end
end

```

```

        end
        %XMaxRate and auxMaxRate matrix generation
        if R(c,t)==max(aux1MaxRate)
            XMaxRate(c,s)=ceil(t/T);
            auxMaxRate(c,s)=R(c,t);
        end
    end
end
%Xroundrobin and auxroundrobin matrix generation
for p=1:T
    if s<=p*users
        Xroundrobin(:,s)=u(s-(p-1)*users);
        break;
    end
end
if s+(s-1)*T<=users*T
    auxroundrobin(:,s)=R(:,s+(s-1)*T);
end
end

%BABS and RCG ALGORITHM

%it is assumed that 10% of the users will be transmitting video (constant
%rate 64kbps), 40% will be transmitting voice (constant rate 16Kbps) and
%the remaining 50% will be transmitting data (exponentially distributed
%rate with a mean of 30Kbps)
UVd=floor(users*0.1);
UVz=floor(users*0.4);
UDt=users-(UVd+UVz);

RminVd=zeros(1,UVd);
RminVz=zeros(1,UVz);
RminDt=zeros(1,UDt);

RminVd(1,:)=64e3;
RminVz(1,:)=16e3;
for dt=1:UDt
    RminDt(1,dt)=exprnd(30e3);
end
Rmin(1,:)=[RminVd,RminVz,RminDt];

%Reordering Rmin, because different user are demanded different services
aux1Rmin=Rmin;
aux2Rmin=randperm(length(Rmin));
for f=1:length(Rmin)
    Rmin(f)=aux1Rmin(aux2Rmin(f));
end

for u=1:users
    for t=1:T
        Rmax(u,t)=max(R(:,(u-1)*T+t));
    end
end

%BABS algorithm
%use the average SNR for each user to decide the number of sub-carriers
%that user will be assigned
for t=1:T
    %at first we allocate sub-carriers according to floor(Rmin/Rmax)
    for u=1:users
        m(u,t)=floor(Rmin(1,u)/Rmax(u,t))+1;
    end
    %while sum(m)>C, we allocate 0 sub-carriers to user with less
    %sub-carriers

```



```

q(:,t)=sort(m(:,t));
for s=1:length(m(:,t))
    if sum(q([s:length(m(:,t))],t))<=C
        break;
    end
end
for p=1:s-1
    for r=1:length(m(:,t))
        if q(p,t)==m(r,t)
            m(r,t)=0;
            break;
        end
    end
end
%while sum(m)<C, we add one sub-carrier to user who minimize G function
while sum(m(:,t))<C,
    for u=1:users
        if m(u,t)==0
            G(u,t)=(m(u,t)+1)/H(u,t)*(0.6*(Rmin(u)/(m(u,t)+1)).^3)-
            (epsilon/H(u,t))*(0.6*(Rmin(u)/epsilon).^3);
        else
            G(u,t)=(m(u,t)+1)/H(u,t)*(0.6*(Rmin(u)/(m(u,t)+1)).^3)-
            (m(u,t)/H(u,t))*(0.6*(Rmin(u)/m(u,t)).^3);
        end
        for u=1:users
            if G(u,t)==min(G(:,t))
                m(u,t)=m(u,t)+1;
                break;
            end
        end
    end
end
end

%XRCG and auxRCG initialization
XRCG=XMaxRate;
auxRCG=auxMaxRate;

%RCG algorithm
%Allocating each sub-carrier to the user with the maximum transmission rate
for t=1:T
    for u=1:users
        for c=1:C
            if u==XMaxRate(c,t)
                mm(u,t)=mm(u,t)+1;
            end
        end
    end
end
%While there exists some user such that mm(u,t)>m(u,t), remove a sub-carrier
%from this user and add a sub-carrier to a user such that mm(u,t)<m(u,t)
for t=1:T
    for u=1:users
        subcuser(:,u)=R(:,t+(u-1)*T);
    end
    for u=1:users
        aux2RCG=zeros(C,users);
        if mm(u,t)>m(u,t)
            for p=1:users
                if mm(p,t)<m(p,t)
                    for c=1:C
                        if subcuser(c,u)==max(subcuser(c,:))
                            aux2RCG(c,p)=subcuser(c,u)-subcuser(c,p);
                        end
                    end
                end
            end
        end
    end
end

```

```
end
end
end
while mm(u,t)>m(u,t),
for c=1:C
if sum(aux2RCG(c,:))~=0
for k=1:users
if sum(aux2RCG(:,k))==0
%avoiding zero conflict for finding minimum
aux2RCG(c,k)=NaN;
end
end
change=min(aux2RCG(c,:));
for r=1:users
if aux2RCG(c,r)==change
%XRCG, auxRCG and mm update
XRCG(c,t)=r;
auxRCG(c,t)=auxRCG(c,t)-change;
mm(u,t)=mm(u,t)-1;
mm(r,t)=mm(r,t)+1;
aux2RCG(c,:)=0;
if mm(r,t)>=m(r,t)
aux2RCG(:,r)=0;
end
break;
end
end
break;
end
end
end
end
end
end
end
```

## Appendix C

### C.1 scheduler.m

```

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Name: scheduler
%Author: Alfonso Bahillo
%Date: March 2006
%Last modification: April 2006
%Support: MATLAB 7.0.019920 (R14)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Description: this function plots the scheduler scheme according to four
%  different algorithms, PFS, Max Rate, Round Robin and RCG. The scheme
%  is represented as a matrix with C rows (subchannels) and T columns (time
%  slots). The user scheduled is showed with a representative color.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%input arguments:
%users, number of different users.
%C, number of sub-carriers per channel.
%T, number of time slots to compute.
%tc, effective memory of the average throughput.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%no output arguments
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

function scheduler(users,C,T,tc)

[XPFS,XMaxRate,Xroundrobin,XRCG,auxPFS,auxMaxRate,auxroundrobin,auxRCG,Rmin]=algo
rithms(users,C,T,tc);

%Each user is represented by one color, 7 different colors
color=['r' 'b' 'g' 'y' 'k' 'c' 'm'];

%Plotting scheduler scheme, according to Round Robin algorithm
figure(1)
axis ij
hold on
grid on
for c=1:C
    for t=1:T
        plot([t t+1],[c c+1],color(Xroundrobin(c,t)));
    end
end
title('Subchannel allocation according to Round Robin');
xlabel('Time slots');
ylabel('Sub-carriers');
hold off

```

```

%Plotting scheduler scheme, according to PFS algorithm
figure(2)
axis ij
hold on
grid on
for c=1:C
    for t=1:T
        for u=1:users
            if XPFS(c,t)==u
                plot([t t+1],[c c+1],color(u));
                break;
            end
        end
    end
end
title('Subchannel allocation according to PFS');
xlabel('Time slots');
ylabel('Sub-carriers');
hold off

%Plotting scheduler scheme, according to Max Rate algorithm
figure(3)
axis ij
hold on
grid on
for c=1:C
    for t=1:T
        for u=1:users
            if XMaxRate(c,t)==u
                plot([t t+1],[c c+1],color(u));
                break;
            end
        end
    end
end
title('Subchannel allocation according to Max Rate');
xlabel('Time slots');
ylabel('Sub-carriers');
hold off

%Plotting scheduler scheme, according to RCG algorithm
figure(4)
axis ij
hold on
grid on
for c=1:C
    for t=1:T
        plot([t t+1],[c c+1],color(XRCG(c,t)));
    end
end
title('Subchannel allocation according to RCG');
xlabel('Time slots');
ylabel('Sub-carriers');
hold off

%finally statistics are generated in a .xls file
statistic(users,C,T,XPFS,XMaxRate,Xroundrobin,XRCG,Rmin);

```

## C.2 statistic.m

```

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Name: statistic
%Author: Alfonso Bahillo
%Date: March 2006
%Last modification: April 2006
%Support: MATLAB 7.0.019920 (R14)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Description: this function generates a file 'statistic.xls' which stores
% the scheduler scheme statistics.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%input arguments:
%users, number of different users.
%C, number of subchannels per channel.
%T, number of time slots to compute.
%XPFS, XMaxRate and XRCG matrix which the scheduler scheme is stored
% according to PFS, Max Rate and RCG algorithms respectively.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%output arguments:
%file .xls where the statistics are stored.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

function statistic(users,C,T,XPFS,XMaxRate,Xroundrobin,XRCG,Rmin)

%matrix initialitiation
user_T=zeros(3,T*users);
user_C=zeros(3,C*users);

%how many sub-carriers are allocated to user u at time slot t according to
%PFS algorithm, user_T(1,t), according to MaxRate algorithm, user_T(2,t)
%and according to RCG algorithm user_T(3,t)
for t=1:T
    for c=1:C
        for u=1:users
            if XPFS(c,t)==u
                user_T(1,t+T*(u-1))=user_T(1,t+T*(u-1))+1;
                break;
            end
        end
        for u=1:users
            if XMaxRate(c,t)==u
                user_T(2,t+T*(u-1))=user_T(2,t+T*(u-1))+1;
                break;
            end
        end
        for u=1:users
            if XRCG(c,t)==u
                user_T(3,t+T*(u-1))=user_T(3,t+T*(u-1))+1;
                break;
            end
        end
    end
end

%how many time slots are allocated to user u per sub-carrier c according to
%PFS algorithm, user_C(1,c), according to MaxRate algorithm, user_C(2,c)
%and according to RCG algorithm, user_C(3,c)
for c=1:C
    for t=1:T

```

```

        for u=1:users
            if XPFS(c,t)==u
                user_C(1,c+C*(u-1))=user_C(1,c+C*(u-1))+1;
                break;
            end
        end
        for u=1:users
            if XMaxRate(c,t)==u
                user_C(2,c+C*(u-1))=user_C(2,c+C*(u-1))+1;
                break;
            end
        end
        for u=1:users
            if XRCG(c,t)==u
                user_C(3,c+C*(u-1))=user_C(3,c+C*(u-1))+1;
                break;
            end
        end
    end
end

[FPFS,FRR,FMaxRate,FRCG]=fairness(users,C,T,XPFS,Xroundrobin,XMaxRate,XRCG,Rmin);

%creating .xls file, fi is the identifier, text is error control
[fi,text]=fopen('statistic.xls','a');

fprintf(fi,'\tACCORDING TO PFS ALGORITM\n\n');
fprintf(fi,'\tFAIRNESS: %f\n\n',FPFS);
fprintf(fi,'USER\t TIME SLOT\t N.SUBCARRIERS\t USE PERCENTAGE\n');

for u=1:users
    for t=1:T
        s=user_T(1,t+(u-1)*T);
        p=(s/C)*100;
        fprintf(fi,'%d\t %d\t %d\t %d\n',u,t,s,p);
    end
end

fprintf(fi,'\nUSER\t SUBCARRIER\t N.TIME SLOTS\t USE PERCENTAGE\n');

for u=1:users
    for c=1:C
        s=user_C(1,c+(u-1)*C);
        p=(s/T)*100;
        fprintf(fi,'%d\t %d\t %d\t %d\n',u,c,s,p);
    end
end

fprintf(fi,'\n\tACCORDING TO MaxSNR ALGORITHM\n\n');
fprintf(fi,'\tFAIRNESS: %f\n\n',FMaxRate);
fprintf(fi,'USER\t TIME SLOT\t N.SUBCARRIERS\t USE PERCENTAGE\n');

for u=1:users
    for t=1:T
        s=user_T(2,t+(u-1)*T);
        p=(s/C)*100;
        fprintf(fi,'%d\t %d\t %d\t %d\n',u,t,s,p);
    end
end

fprintf(fi,'\nUSER\t SUBCARRIER\t N.TIME SLOTS\t USE PERCENTAGE\n');

for u=1:users
    for c=1:C

```

```

        s=user_C(2,c+(u-1)*C);
        p=(s/T)*100;
        fprintf(fi,'%d\t %d\t %d\t %d\n',u,c,s,p);
    end
end

fprintf(fi,'\tACCORDING TO RCG ALGORITM\n\n');
fprintf(fi,'\tFAIRNESS: %f\n\n',FRCG);
fprintf(fi,'USER\t TIME SLOT\t N.SUBCARRIERS\t USE PERCENTAGE\n');

for u=1:users
    for t=1:T
        s=user_T(3,t+(u-1)*T);
        p=(s/C)*100;
        fprintf(fi,'%d\t %d\t %d\t %d\n',u,t,s,p);
    end
end

fprintf(fi,'\nUSER\t SUBCARRIER\t N.TIME SLOTS\t USE PERCENTAGE\n');

for u=1:users
    for c=1:C
        s=user_C(3,c+(u-1)*C);
        p=(s/T)*100;
        fprintf(fi,'%d\t %d\t %d\t %d\n',u,c,s,p);
    end
end

%close .xls file
st=fclose(fi);

```

### C.3 throughput\_fairness\_Vs\_PFS.m

```

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Name: throughput_fairness_Vs_tc_PFS
%Author: Alfonso Bahillo
%Date: March 2006
%Last modification: March 2006
%Support: MATLAB 7.0.019920 (R14)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Description: this function computes the system fairness and system
% capacity versus tc parameter according to three different algorithms,
% PFS, MaxRate and Round Robin.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%input arguments:
%users, number of different users.
%C, number of subchannels per channel.
%T, number of time slots to compute.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%no output arguments
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

function throughput_fairness_Vs_tc_PFS(users,C,T)

%possible set of tc parameter values
tc=[1.2:20:100 150:50:500 600:100:1000];

%matrix initialitation
%auxiliar matrix to store rates
auxPFS=zeros(C,T);
auxMaxRate=zeros(C,T);
auxroundrobin=zeros(C,T);
auxRCG=ones(C,T);
%matrix to store user ID
XPFS=zeros(C,T);
XMaxRate=zeros(C,T);
Xroundrobin=zeros(C,T);
XRCG=ones(C,T); %avoiding zero division

%Store in a matrix, R (Cx(T*users)), the different values of rate, for each
%user, subcarrier and time slot.
R=zeros(C,T*users);
Th=zeros(C,T*users);
Rmin=ones(1,users); %avoiding zero division
aux3=zeros(1,users);
epsilon=0.001; %average throughput initialization

%Store in a matrix R (Cx(T*users)), the different values of rate, for each
%user, subcarrier and time slot
for u=1:users
    [fo CHH Noise I]=FGN_model(3,C,T); %v=3km/h, user walking
    for tslot=1:T
        for c=1:C
            R(c,tslot+T*(u-
1))=(fo(tslot)/C)*log2(1+mean(abs(CHH(c,1+(I/C)*(tslot-
1):(I/C)*tslot)))/Noise(tslot));
        end
    end
end

%At first time slot, average throughput = epsilon for all users

```



```

for u=1:users
    Th(:,1+T*(u-1))=epsilon;           %Initial average throughput
end

%if you want to take channel B (user B) stronger than other channels
R(:,T+1:2*T)=1.4*R(:,T+1:2*T);

u=1:users;
%MaxRate algorithm:
%XMaxRate, matrix which stores who is scheduled according to MaxRate
for s=1:T
    for c=1:C
        aux3=R(c,s:T:T*users);
        for t=s:T:T*users
            if R(c,t)==max(aux3)
                XMaxRate(c,s)=ceil(t/T);
                auxMaxRate(c,s)=R(c,t);
            end
        end
    end
end
%Round Robin algorithm:
%Xroundrobin, matrix which stores who is scheduled according to RR
for p=1:T
    if s<=p*users
        Xroundrobin(:,s)=u(s-(p-1)*users);
        break;
    end
end
if s+(s-1)*T<=users*T
    auxroundrobin(:,s)=R(:,s+(s-1)*T);
end
end

%PFS algorithm:
%XPFS, matrix which stores who is scheduled according to PFS
for p=1:length(tc)
    for s=1:T
        for c=1:C
            aux3=R(c,s:T:T*users)./Th(c,s:T:T*users);
            for t=s:T:T*users
                if (R(c,t)./Th(c,t))==max(aux3)
                    XPFS(c,s)=ceil(t/T);
                    auxPFS(c,s)=R(c,t);
                    Th(c,t+1)=(1-1/tc(p))*Th(c,t)+1/tc(p)*R(c,t);
                else
                    Th(c,t+1)=(1-1/tc(p))*Th(c,t);
                end
            end
        end
    end
end

%fairness and throughput functions compute index system fairness and
%system throughput respectively

[FPFS(p),FRR(p),FMaxRate(p),FRCG]=fairness(users,C,T,XPFS,Xroundrobin,XMaxRate,XR
CG,Rmin);

[throughput_PFS(p),throughput_MaxRate(p),throughput_RR(p),throughput_RCG]=through
put(auxPFS,auxMaxRate,auxroundrobin,auxRCG,users,T);
end

%system fairness Vs tc
figure(1)
hold on

```

```
axis([0 1000 0 1.1]);
plot(tc,FPFS,':');
plot(tc,FMaxRate,'-');
plot(tc,FRR,'-.');
title('System Fairness');
xlabel('tc');
ylabel('Fairness');
hold off

%system throughput VS tc
figure(2)
hold on
plot(tc,throughput_PFS,':');
plot(tc,throughput_MaxRate,'-');
plot(tc,throughput_RR,'-.');
title('System Capacity');
xlabel('tc');
ylabel('Throughput (bps)');
hold off
```

## Appendix D

### D.1 fairness.m

```

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Name: fairness
%Author: Alfonso Bahillo
%Date: March 2006
%Last modification: April 2006
%Support: MATLAB 7.0.019920 (R14)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
% Description: this function computes fairness of different scheduling
% algorithms, with equal and unequal weighted users. 0<F<1,
% if F=1, then all users in the system consume exactly their fair share
% of resources.
% if F=0, then one user consumes all resources allocated.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%input arguments:
%users, number of different channel responses.
%C, number of subchannels per channel.
%T, number of time slots.
%XPFSS, Xroundrobin, XRCG and XMaxRate, matrix that stores the user scheduled
% according to PFS, Round Robin, RCG and MaxRate algorithms respectively.
%Rmin, demand vector rate.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%output arguments:
%FPFS, FRR, FRCG and FMaxRate fairness index according to PFS, Round Robin
% RCG and Max Rate algorithms respectively.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

function
[FPFS,FRR,FMaxRate,FRCG]=fairness(users,C,T,XPFSS,Xroundrobin,XMaxRate,XRCG,Rmin)

%matrix initialization
%vector that store the number of time that one user is scheduled
PRCG=zeros(1,users);
PMaxRate=zeros(1,users);
PPFS=zeros(1,users);
PRR=zeros(1,users);

%PFS algorithm
for c=1:C
    for t=1:T
        for u=1:users
            if XPFSS(c,t)==u
                PPFS(u)=PPFS(u)+1;
                break;
            end
        end
    end
end
end
end

```

```

%computing index fairness according to equal weighted users
for u=1:users
    if PPFs(u)==0
        aux1PPFS(u)=0;
    else
        aux1PPFS(u)=- (PPFS(u) / (C*T)) * (log(PPFS(u) / (C*T))) / (log(users));
    end
end

FPFS=sum(aux1PPFS);          %0<F<1

%Round Robin algorithm
for c=1:C
    for t=1:T
        for u=1:users
            if Xroundrobin(c,t)==u
                PRR(u)=PRR(u)+1;
                break;
            end
        end
    end
end

%computing index fairness according to equal weighted users
for u=1:users
    if PRR(u)==0
        aux1rr(u)=0;
    else
        aux1rr(u)=- (PRR(u) / (C*T)) * (log(PRR(u) / (C*T))) / (log(users));
    end
end

FRR=sum(aux1rr);           %0<F<1

%RCG algorithm
%PRCG matrix generation that stores how many times a user is scheduled
%according to RCG algorithm
for c=1:C
    for t=1:T
        for u=1:users
            if XRCG(c,t)==u
                PRCG(u)=PRCG(u)+1;
                break;
            end
        end
    end
end

%computing index fairness according to unequal weighted users
for u=1:users

auxRCGden(u)=((1+1/(log(Rmin(1)/sum(Rmin))))/(1+1/(log(Rmin(u)/sum(Rmin))))) * Rmin(u);
    if PRCG(u)==0
        auxRCGnum(u)=0;
    else

auxRCGnum(u)=((1+1/(log(Rmin(1)/sum(Rmin))))/(1+1/(log(Rmin(u)/sum(Rmin))))) * (PRCG(u) /
(C*T)) * ((log(PRCG(u) / (C*T))) / (log(Rmin(u) / sum(Rmin))));
    end
end

FRCG=sum(Rmin) * (sum(auxRCGnum) / sum(auxRCGden));          %0<F<1

%Max Rate algorithm
for c=1:C
    for t=1:T
        for u=1:users
            if XMaxRate(c,t)==u
                PMaxRate(u)=PMaxRate(u)+1;
            end
        end
    end
end

```

```
                break;
            end
        end
    end
end
%computing index fairness according to equal weighted users
for u=1:users
    if PMaxRate(u)==0
        aux1MaxRate(u)=0;
    else
        aux1MaxRate(u)=- (PMaxRate(u)/(C*T)) * (log(PMaxRate(u)/(C*T)))/(log(users));
    end
end
FMaxRate=sum(aux1MaxRate);           %0<F<1
```

## D.2 throughput.m

```

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Name: throughput
%Author: Alfonso Bahillo
%Date: March 2006
%Last modification: April 2006
%Support: MATLAB 7.0.019920 (R14)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Description: this function computes the system throughput according to
% four different algorithms (for allocating resources) PFS, Max Rate,
% Round Robin and RCG.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%input arguments:
%auxPFS, auxMaxSNR, auxroundrobin and auxRCG matrix that store the rate
% which each user transmits per sub-carrier at each time slot.
%users, number of different channel responses.
%T, number of time slots.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%output arguments:
%throughput_PFS, throughput_MaxRate, throughput_RR matrix and
% throughput_RCG which store mean value of throughput according to PFS,
% MaxRate, Round Robin and RCG algorithms respectively.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

function
[throughput_PFS,throughput_MaxRate,throughput_RR,throughput_RCG]=throughput (auxPFS,aux
MaxRate,auxroundrobin,auxRCG,users,T)

%matrix initialization
%auxiliary matrix
auxthroughput_PFS=zeros(1,T);
auxthroughput_MaxRate=zeros(1,T);
auxthroughput_RR=zeros(1,T);
auxthroughput_RCG=zeros(1,T);

for t=1:T
    auxthroughput_PFS(t)=sum(auxPFS(:,t));           %PFS algorithm
    auxthroughput_MaxRate(t)=sum(auxMaxRate(:,t));  %Max Rate algorithm
    auxthroughput_RR(t)=sum(auxroundrobin(:,t));    %RR algorithm
    auxthroughput_RCG(t)=sum(auxRCG(:,t));          %RCG algorithm
end

throughput_PFS=mean(auxthroughput_PFS);           %PFS algorithm
throughput_MaxRate=mean(auxthroughput_MaxRate);  %Max Rate algorithm
throughput_RR=sum(auxthroughput_RR([1:users]))/users; %RR algorithm
throughput_RCG=mean(auxthroughput_RCG);          %RCG algorithm

```

## Appendix E

### E.1 throughput\_fairness\_Vs\_users.m

```

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Name: throughput_fairness_Vs_users
%Author: Alfonso Bahillo
%Date: March 2006
%Last modification: April 2006
%Support: MATLAB 7.0.019920 (R14)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Description: this function computes the system throughput and fairness
% with different set of users and plot the results comparing four
% different algorithms PFS, Round Robin, RCG and MaxRate.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%input arguments:
%us, system throughput is computed us by us users.
%maxusers, the system throughput is computed over 'maxuser' users.
%C, number of subchannels per channel.
%T, number of time slots.
%tc, effective memory of the average throughput.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%no output arguments
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

function throughput_fairness_Vs_users(us,maxusers,C,T,tc)

%matrix initialization
u=[us:us:maxusers];
%throughput values
throughput_RCG=zeros(1,length(u));
throughput_PFS=zeros(1,length(u));
throughput_MaxRate=zeros(1,length(u));
throughput_RR=zeros(1,length(u));
%fairness values
FRCG=zeros(1,length(u));
FPFS=zeros(1,length(u));
FMaxRate=zeros(1,length(u));
FRR=zeros(1,length(u));

%computing throughput and fairness for us by us users
for uu=us:us:maxusers

    [XPFS,XMaxRate,Xroundrobin,XRCG,auxPFS,auxMaxRate,auxroundrobin,auxRCG,Rmin]=al
    gorithms(uu,C,T,tc);

    [FPFS(uu/us),FRR(uu/us),FMaxRate(uu/us),FRCG(uu/us)]=fairness(uu,C,T,XPFS,Xroun
    drobin,XMaxRate,XRCG,Rmin);

    [throughput_PFS(uu/us),throughput_MaxRate(uu/us),throughput_RR(uu/us),throughpu
    t_RCG(uu/us)]=throughput(auxPFS,auxMaxRate,auxroundrobin,auxRCG,uu,T);

```

```
end

%System capacity Vs users
figure(1)
hold on
plot(u,throughput_RCG,':');
plot(u,throughput_MaxRate,'-');
plot(u,throughput_PFS,'-.');
plot(u,throughput_RR,'--');
legend('RCG','Max Rate','PFS','RR');
title('System Capacity');
xlabel('Users');
ylabel('Throughput (bps)');
hold off

%System Fairness Vs users
figure(2)
hold on
plot(u,FRCG,':');
plot(u,FMaxRate,'-');
plot(u,FPFS,'-.');
plot(u,FRR,'--');
legend('RCG','Max Rate','PFS','RR');
title('System Fairness');
xlabel('Users');
ylabel('Fairness');
hold off
```



## Appendix F

### F.1 Probability\_delay.m

```

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Name: probability_delay
%Author: Alfonso Bahillo
%Date: April 2006
%Support: MATLAB 7.0.019920 (R14)
%Last modification: May 2006
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%Description: this function computes the probability that a user u is
% transmitting less than Rmin(u) bits per unit time. Plotting this
% probability versus time slots and different set of users.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%input arguments:
%us, the probability is computed us by us users.
%maxusers, the probability is computed over 'maxuser' users.
%C, number of subchannels per channel.
%T, number of time slots.
%tc, effective memory of the average throughput.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%no output arguments
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

function probability_delay(T,C,tc,us,maxusers)

%Matrix initialization
aux_users=zeros(4,maxusers/us);

for uu=us:us:maxusers

    %storing total rate per user and time slot
    auxRCGprobl=zeros(uu,T);
    auxMaxRateprobl=zeros(uu,T);
    auxPFSprobl=zeros(uu,T);
    auxRRprobl=zeros(uu,T);
    %storing total data transmitted at each time slot
    aux_time_RCG=zeros(uu,T);
    aux_time_MaxRate=zeros(uu,T);
    aux_time_PFS=zeros(uu,T);
    aux_time_RR=zeros(uu,T);
    %mean values at each time slot
    aux_time=zeros(4,T);

[XPFS,XMaxRate,Xroundrobin,XRCG,auxPFS,auxMaxRate,auxroundrobin,auxRCG,Rmin]=algorithm
s(uu,C,T,tc);

    for t=1:T
        for c=1:C

```

```

%computing auxiliary variable for RCG probability
for u=1:uu
    if XRCG(c,t)==u
        auxRCGprobl(u,t)=auxRCGprobl(u,t)+auxRCG(c,t);
        break;
    end
end
%computing auxiliary variable for MaxRate probability
for u=1:uu
    if XMaxRate(c,t)==u
        auxMaxRateprobl(u,t)=auxMaxRateprobl(u,t)+auxMaxRate(c,t);
        break;
    end
end
%computing auxiliary variable for PFS probability
for u=1:uu
    if XPFS(c,t)==u
        auxPFSprobl(u,t)=auxPFSprobl(u,t)+auxPFS(c,t);
        break;
    end
end
%computing auxiliary variable for RR probability
for u=1:uu
    if Xroundrobin(c,t)==u
        auxRRprobl(u,t)=sum(auxroundrobin(:,u));
        break;
    end
end
end
end
for d=1:T
    for u=1:uu
        aux_time_RCG(u,d)=1-sum(auxRCGprobl(u,[1:d]))/Rmin(u);
        aux_time_MaxRate(u,d)=1-sum(auxMaxRateprobl(u,[1:d]))/Rmin(u);
        aux_time_PFS(u,d)=1-sum(auxPFSprobl(u,[1:d]))/Rmin(u);
        aux_time_RR(u,d)=1-sum(auxRRprobl(u,[1:d]))/Rmin(u);
        if aux_time_RCG(u,d)<=0
            aux_time_RCG(u,d)=0;
        end
        if aux_time_MaxRate(u,d)<=0
            aux_time_MaxRate(u,d)=0;
        end
        if aux_time_PFS(u,d)<=0
            aux_time_PFS(u,d)=0;
        end
        if aux_time_RR(u,d)<=0
            aux_time_RR(u,d)=0;
        end
    end
end
%computing mean value for all users
aux_time(1,:)=mean(aux_time_RCG);
aux_time(2,:)=mean(aux_time_MaxRate);
aux_time(3,:)=mean(aux_time_PFS);
aux_time(4,:)=mean(aux_time_RR);

%plotting results
for d=uu:T
    if aux_time(1,d)==0
        break;
    end
end
end

%Plotting outage probability versus time slots
t=1:T;
figure(uu)

```

```

hold on
axis([1 d+1 -0.1 1.1]);
plot(t,aux_time(1,:), ':');
plot(t,aux_time(2,:), '-');
plot(t,aux_time(3,:), '-. ');
plot(t,aux_time(4,:), '--');
legend('RCG', 'Max Rate', 'PFS', 'RR');
title('Probability that a user u is transmitting less than Rmin');
xlabel('delay');
ylabel('Outage probability');
hold off
%computing mean value for all time slots
aux_user(1,uu/us)=mean(aux_time(1,:));
aux_user(2,uu/us)=mean(aux_time(2,:));
aux_user(3,uu/us)=mean(aux_time(3,:));
aux_user(4,uu/us)=mean(aux_time(4,:));
end

%Plotting outage probability versus users
u=[us:us:maxusers];
figure(40)
hold on
axis([us maxusers -0.1 1.1]);
plot(u,aux_users(1,:), ':');
plot(u,aux_users(2,:), '-');
plot(u,aux_users(3,:), '-. ');
plot(u,aux_users(4,:), '--');
legend('RCG', 'Max Rate', 'PFS', 'RR');
title('Probability that a user k is transmitting less than Rmin');
xlabel('Users');
ylabel('Outage probability');
hold off

```

## Appendix G

In this appendix an example run of the RCG algorithm in 10 time slots is showed within two different scenarios, equal and unequal users. We have 4 users who are demanded different services, see Table G.I.

	Demanded transmission rate
#1	64
#2	45
#3	16
#4	35

**Table G.I** Transmission rate demanded by users.

### G.1 Equal Users

In this scenario, four users with the same channel response average are simulated in 10 time slots. The system has 8 sub-carriers and the sub-carriers allocated by this algorithm at first 10 time slots are showed in Table G.II.

	#1	#2	#3	#4	#5	#6	#7	#8	#9	#10
#1	3	3	3	3	3	4	3	3	3	3
#2	2	2	2	2	2	2	2	2	2	2
#3	1	1	1	1	1	1	1	1	1	1
#4	2	2	2	2	2	1	2	2	2	2

**Table G.II** Sub-carriers allocated to users at 10 first time slots. Rows represent different users, columns are different sub-carriers.

On the other hand, the scheduler scheme according to RCG algorithm is plotted, see Table G.III.

	#1	#2	#3	#4	#5	#6	#7	#8	#9	#10
#1	1	1	1	1	2	1	4	1	4	1
#2	1	1	1	4	2	1	4	4	1	1
#3	3	4	1	1	1	2	2	1	3	1
#4	1	4	2	1	4	1	3	3	4	4
#5	4	1	4	4	4	3	2	2	2	4
#6	2	2	2	2	1	1	1	1	2	2
#7	2	2	4	2	1	4	1	2	1	2
#8	4	3	3	3	3	2	1	4	1	3

**Table G.III** Scheme scheduled according to RCG algorithm with equal users. Rows represent different sub-carriers, columns are different time slots.

## G.2 Unequal Users

In this scenario, four users with different channel response average are simulated in 10 time slots. The user number #2 is stronger than others. The system has 8 sub-carriers and the sub-carriers allocated by this algorithm at first 10 time slots are showed in Table G.IV.

	#1	#2	#3	#4	#5	#6	#7	#8	#9	#10
#1	3	4	3	2	4	3	3	3	3	3
#2	2	2	2	2	2	2	2	2	2	2
#3	1	1	1	1	1	1	1	1	1	1
#4	2	1	2	3	1	2	2	2	2	2

**Table G.IV** Sub-carriers allocated to users at 10 first time slots. Rows represent different users, columns are different sub-carriers.

On the other hand, the scheduler scheme according to RCG algorithm is plotted, see Table G.V, and compare this with the scheduler scheme according to Max Rate algorithm, see Table G.VI.

	#1	#2	#3	#4	#5	#6	#7	#8	#9	#10
#1	4	4	3	1	4	3	1	1	4	3
#2	3	3	4	1	1	1	1	1	4	4
#3	4	1	4	4	1	4	4	4	1	4
#4	1	1	1	4	1	4	1	4	2	1
#5	1	1	1	4	2	1	4	3	1	1
#6	1	1	1	2	2	1	3	2	1	1
#7	2	2	2	3	1	2	2	2	3	2
#8	2	2	2	2	3	2	2	1	2	2

**Table G.V** Scheme scheduled according to RCG algorithm with unequal users. Rows represent different sub-carriers, columns are different time slots.

	#1	#2	#3	#4	#5	#6	#7	#8	#9	#10
#1	2	2	2	2	2	2	3	1	2	2
#2	2	2	2	2	3	2	3	2	3	2
#3	2	2	2	2	2	2	2	2	2	2
#4	2	2	2	3	2	2	2	2	2	2
#5	2	2	2	2	2	2	4	3	3	2
#6	2	2	2	2	2	2	3	2	3	2
#7	2	2	2	3	3	2	2	2	3	2
#8	2	2	2	2	3	2	2	1	2	2

**Table G.VI** Scheme scheduled according to Max Rate algorithm with unequal users. Rows represent different sub-carriers, columns are different time slots.

## REFERENCES

- [1] Theodore S. Rappaport, “*Wireless Communications, Principles and Practice*” Prentice Hall Communications Engineering and Emerging Technologies Series, Second Edition, 2002.
- [2] Matthew S. Gast, “*802.11 Wireless Networks, the definitive guide*” O’reilly, 2002.
- [3] <http://www.itu.int>, “*Web official, International Telecommunications Union*”.
- [4] Randall K. Nichols and Panos C. Lekkas, “*Wireless Security, Models, Threats and Solutions*” Mc Graw-Hill Telecom, 2002.
- [5] Merritt Maxim and David Pollino, “*Wireless Security*” Mc Graw-Hill/Osborne, 2002.
- [6] K. Pahlavan and Allen H. Levesque, “*Wireless Information Networks*” Wiley series in telecommunications and signal processing, Second Edition, John G. Proakis.
- [7] N. Blaunstein and Jorgen B. Andersen, “*Multipath Phenomena in Cellular Networks*” Artech House mobile communications series, 2002.
- [8] William Web, “*The Complete Wireless Communications Professional*”, Artech House Boston, 1999.
- [9] Michel C., Philip B. and K. Sam Shanmugan, “*Simulation of Communication Systems. Modeling, Methodology, and Techniques*” Kluwer Academic/Plenum Publishers, Second Edition, 2000.
- [10] R. H. Clarke, “*A statistical Theory of Mobile-Radio Reception*” Bell Syst. Tech. J., 957-1000, 1968.
- [11] W. C. Jakes (Ed.), “*Microwave Mobile Communications*” Wiley, New York, 1974.
- [12] H. Safer, G. Berger and F. Seifert, “*Wideband Propagation Measurements of the Mobile Radio Channel in Different Areas of Austria*” University of Technology Vienna, applied electronics laboratory.
- [13] A. Jalai, R. Padovani and R. Pankaj, “*Data Throughput of CDMA-HDR a High Efficiency-High Data Rate Personal Communications Wireless System*” in Proc. IEEE Veh. Technology Conf. 2000-Spring, pp 1854-1858, 2000.

- [14] D. Piazza and L. B. Milstein, "Multiuser Diversity-Mobility Trade-Off: Modeling and Performance Analysis of a Proportional Fair Scheduling" Global Telecommunication Conference, 2002, IEEE press, Vol. 1, pp. 906-910, November 2002.
- [15] David N. Tse, "Multiuser Diversity in Wireless Networks" Wireless communications seminar, Stanford University, April 2001.
- [16] D. Kivanc, G. Li, H. Lui, "Computationally Efficient Bandwidth Allocation and Power Control for OFDMA" IEEE transactions on wireless communications, vol. 2, no. 6, November 2003.
- [17] E. L. Hahne, "Round-Robin Scheduling and Window Flow Control for Max-Min Fairness in Data Networks" IEEE, technical report, November 1987.
- [18] P. Viswanath, D. Tse, and R. Laroia, "Opportunistic Beamforming Using Dumb Antennas" IEEE transactions on information theory, vol. 48, no. 6, June 2002.
- [19] K. Norlund, T. Ottosson, A. Brunstrom, "Fairness Measures for Best Effort Traffic in Wireless Networks" Personal, Indoor and Mobile Radio Communications, 2004, IEEE, Vol. 4, pp.2953-2957, September 2004.
- [20] R. Elliott, "A Measure of Fairness of Service for Scheduling Algorithms in Multiuser Systems" Electrical and Computer Engineering, 2002, IEEE vol 3, pp 1583-1588, August 2002.
- [21] V. Bharghavan, S. Lu, and T. Nandagopal, "Fair Queuing in Wireless Networks: Issues and Approaches" IEEE Pers. Commun., vol. 6, no. 1, pp. 44-53, February 1999.
- [22] Qi Bi, C. Huang, P. Li and Mark E. Newbury, "Measures of Wireless Data Performance" Lucent Technologies Inc. 2003.
- [23] T. L. Marzetta and B. M. Hochwald, "Capacity of a Mobile Multiple-Antenna Communication Link in Rayleigh Flat Fading" IEEE Trans. Inform. Theory, vol. 45, pp. 139-157, January 1999.
- [24] B. Holter, "On the Capacity of the MIMO Channel - a Tutorial Introduction -" Department of Telecommunications, NTNU, Trondheim, Norway.
- [25] C. E. Shannon, "The Mathematical Theory of Communication" Urbana, IL: University of Illinois Press, 1949 (reprinted 1998).
- [26] R. Knopp and P. A. Humblet, "Information Capacity and Power Control in Single-Cell Multiuser Communications", IEEE International Conference, vol. 1, pp. 331-335, June 1995.
- [27] H. Kim and Y. Han, "A Proportional Fair Scheduling for Multicarrier Transmission System" IEEE Communications letters, vol. 9, no. 3, March 2005.
- [28] A. Stamoulis, S. N. Diggavi and N. Al-Dhair "Estimation of Fast Fading Channels in OFDM" IEEE Wireless Communications and Networking Conference, vol. 1, pp. 465-470, July 2002.
- [29] D. Schafhuber, G. Matz and F. Hlawatsch "Adaptive Prediction of Time-Varying Channels for Coded OFDM Systems" IEEE International Conference, vol. 3, pp. 2549-2552, July 2002.



[30] Joint Technical Committee for PCS T1 R1P1.4, “*Technical Report on RF Channel Characterization and System Deployment*”, September 1994.

[31] H. Seo and B. Gi Lee, “*A Proportional Fair Power Allocation for Fair and Efficient Multiuser OFDM Systems*”, School of Electrical Engineering, Seoul National University, April 2004.