“A LOW COMPLEXITY ALGORITHM FOR DYNAMIC FAIR RESOURCE ALLOCATION IN OFDMA SYSTEMS”

Por

ANDRÉ LUIS CAVALCANTI MOREIRA

Dissertação de Mestrado

Universidade Federal de Pernambuco
posgraduacao@cin.ufpe.br
www.cin.ufpe.br/~posgraduacao

RECIFE, JULHO/2008
André Luis Cavalcanti Moreira

“A Low complexity algorithm for dynamic fair resource allocation in OFDMA systems"

Este trabalho foi apresentado à Pós-Graduação em Ciência da Computação do Centro de Informática da Universidade Federal de Pernambuco como requisito parcial para obtenção do grau de Mestre em Ciência da Computação.

ORIENTADOR: Prof. Dr. Djamel Fawzi Hadj Sadok

RECIFE, JULHO/2008
Moreira, André Luis Cavalcanti


Inclui bibliografia e glossário.


025.04 CDD (22.ed.) MEI2008-112
To my wife and my parents.
First and foremost, I would like to thank to God for his guidance, faithfulness and wisdom, granting me the chance and ability to complete this work.

I also would like to thank my supervisor Prof Djamel Sadok who gave me this opportunity to work in this fascinating area of computer networks.

I also would like to thank Ericsson Research, the sponsor of the RRM project that motivated this work.

I would like to express sincere gratitude to Professor Judith Kelner for the support, without which, I would not be finished.

I also would like to thank all the staff at the Networking and Telecommunications Research Group for the discussions, help and companion, making this research much more enjoyable.

Finally, I would like to thank my parents for their continual encouragement to my studies. And to my wife, Érica, for her understanding and support.
Abstract

The popularization of the Internet and the demand for high-speed access has led the development of Broadband Wireless Access. Despite its great potential, the wireless communications impose some challenges. A main limitation is the transmission medium itself because the inherently effects of radio propagation such as path loss, frequency selective fading, Doppler spread and multipath delay-spread.

In this context, OFDM is a promising technology because of its tolerance to loss and multipath problems. Due to the combination of independent channels, it is possible to use different modulation in each sub-carrier, according to the channel conditions, a technique referred to as adaptive modulation and coding. Also, in a point-to-multipoint architecture, multiple users can share the spectrum by dynamically assigning different sets of sub-carriers to different users, taking advantage from an effect referred to as multi-user diversity. In comparison to other multiple-access techniques, OFDMA allows an increased exploration of multiuser diversity and scheduling with the possibility of fine grained allocation. A great deal of research has looked into adaptive techniques capable of improving network spectral efficiency in multi-user systems. They are usually formulated as a constraint optimization problem and know to be NP-hard complex.

In this work, we adopt a heuristic approach to deal with this kind of problem. The main objective is to design an allocation strategy that makes efficient use of available resources and maximizes overall spectral efficiency. However, an allocation strategy that targets only the higher spectral efficiency can generate a problem regarding the justice of the resource sharing strategy being used. Also, as the wireless systems networks are spreading, it is expected that they can support a wide variety of different services with different requirements of QoS and bandwidth. Hence, we propose an algorithm that let the network operator balance these requirements. By taking into account the QoS requirement of the users or applications, the algorithm must give the maximum throughput possible under these restrictions, established by the network operator or the system policies.

Keywords: OFDMA systems, fair resource allocation, QoS awareness.
A popularização da Internet e a demanda por acesso de alta velocidade levou ao desenvolvimento da Broadband Wireless Access. Apesar do seu grande potencial, a comunicação via rádio impõe alguns desafios. Uma grande limitação é o próprio meio de transmissão devido a efeitos inerentes à propagação de radio como o path loss, frequency selective fading, espalhamento Doppler e multipath delay-spread.

Nesse contexto, o OFDM é uma tecnologia promissora por causa de sua tolerância a problemas de perdas e multi-caminho. Devido à combinação de canais independentes, é possível usar diferentes modulações em cada sub-carrier, de acordo com as condições do canal. Esta técnica é conhecida como adaptive modulation and coding. Além disso, em uma arquitetura ponto a multi-ponto, múltiplos usuários podem compartilhar o espectro ao se atribuir diferentes conjuntos de sub-carriers, tirando vantagem do um efeito conhecido como diversidade multi-usuário. Em comparação com outras técnicas de múltiplo acesso, o OFDMA permite um melhor aproveitamento da diversidade multi-usuário com a possibilidade de uma alocação com alta granularidade. Muitas pesquisas têm investigado técnicas adaptativas capazes de melhorar a eficiência espectral em sistemas multi-usuário. Essas técnicas são normalmente formuladas como constraint optimization problems, conhecidos por serem NP-hard.

Neste trabalho, adotamos uma abordagem heurística para lidar com esse tipo de problema. O objetivo principal é desenvolver uma estratégia de alocação fazendo uso eficiente dos recursos disponíveis e maximizando a eficiência espectral total. Entretanto, um estratégia que apenas procura maximizar a eficiência espectral pode gerar um problema relacionado à justiça no compartilhamento de recursos. Outrossim, com a popularização das redes sem fio, é esperado que elas sejam capazes de prover uma maior variedade de serviços com diferentes requisites de QoS e largura de banda. Portanto, procuramos desenvolver um algoritmo que permita ao operador da rede definir esses requisitos. De acordo com eles, o algoritmo deve fornecer o maior throughput possível dentro dos limites estabelecidos por essas restrições.

Keywords: sistemas OFDMA, alocação justa de recursos, QoS awareness.
# Table of contents

ACKNOWLEDGEMENTS ............................................................................................................... III
ABSTRACT ................................................................................................................................. IV
RESUMO .................................................................................................................................. V
TABLE OF CONTENTS .............................................................................................................. VI
LIST OF TABLES ...................................................................................................................... IX
LIST OF FIGURES .................................................................................................................. X
GLOSSARY ............................................................................................................................... XII

CHAPTER 1 .............................................................................................................................. 14
INTRODUCTION ......................................................................................................................... 14
  1.1 MOTIVATION ..................................................................................................................... 14
  1.2 STATE OF THE ART ......................................................................................................... 16
  1.3 OBJECTIVE ...................................................................................................................... 17
  1.4 DISSERTATION OUTLINE ............................................................................................... 17

CHAPTER 2 .............................................................................................................................. 19
WIRELESS COMMUNICATION ................................................................................................. 19
  2.1 WIRELESS CHALLENGES .............................................................................................. 20
    2.1.1 Path loss .................................................................................................................... 21
    2.1.2 Shadowing ................................................................................................................. 21
    2.1.3 Multipath fading ........................................................................................................ 22
    2.1.4 Doppler spread .......................................................................................................... 23
    2.1.5 Delay spread ............................................................................................................ 24
2.1.6 Noise ................................................................................................................. 24
2.2 Cellular Systems ........................................................................................................ 24

CHAPTER 3 .................................................................................................................... 25
ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING ........................................ 25
3.1 Basic Principles ........................................................................................................ 26
3.2 Generation of Subcarriers ...................................................................................... 28
  3.2.1 Subcarrier modulation ....................................................................................... 28
  3.2.2 Error correction ................................................................................................. 30
  3.2.3 Guard time ......................................................................................................... 31
  3.2.4 Cyclic prefix ....................................................................................................... 32
3.3 The PAPR Problem ................................................................................................. 34

CHAPTER 4 .................................................................................................................... 36
MULTIUSER-OFDM ....................................................................................................... 36
4.1 Multiuser Diversity ................................................................................................. 38
4.2 Adaptive Modulation and Coding ........................................................................... 39
4.3 Fairness ................................................................................................................... 41

CHAPTER 5 .................................................................................................................... 45
RESOURCE ALLOCATION TECHNIQUES ................................................................. 45
5.1 Single-Cell ............................................................................................................... 46
  5.1.1 Maximum Sum Rate Algorithm ....................................................................... 47
  5.1.2 Maximum Fairness Algorithm ......................................................................... 48
  5.1.3 Proportional Rate Constraints algorithm ....................................................... 49
  5.1.4 Proportional Fair Scheduling .......................................................................... 50
  5.1.5 Performance Comparison ............................................................................... 52
5.2 Multi-cell Allocation ............................................................................................... 52
CHAPTER 6 .......................................................... 54
PROPOSED ALGORITHM .................................................. 54
  6.1 SYSTEM MODEL .................................................. 55
  6.2 ALGORITHM .................................................. 56
  6.3 A GENERAL FUNCTION TO EVALUATE GAIN .......... 58
  6.4 FAIRNESS .................................................. 59
  6.5 QUALITY OF SERVICE ........................................ 60

CHAPTER 7 .......................................................... 63
PERFORMANCE EVALUATION ........................................ 63
  7.1 THE SIMULATOR .................................................. 63
    7.1.1 Simulator working description ........................... 64
    7.1.2 SNR calculation and modulation of subcarriers ...... 64
    7.1.3 Metrics .................................................. 65
  7.2 SIMULATING PARAMETERS ...................................... 65
  7.3 SCENARIO 1 .................................................. 69
  7.4 SCENARIO 2 .................................................. 72
  7.5 SCENARIO 3 .................................................. 74

CHAPTER 8 .......................................................... 77
CONCLUSION .................................................. 77

REFERENCES .................................................. 79
List of tables

Table 4.1 - Relation between modulation, bit number and SNR ........................................ 41
Table 4.2 – Fairness Index for allocation schemes ............................................................... 43
Table 5.1 – Symbols used ...................................................................................................... 46
Table 5.2 – Comparison between OFDMA allocation techniques ........................................ 52
Table 6.1 – Symbols used ...................................................................................................... 56
Table 7.1 – OFDMA Parameters used in the simulation ...................................................... 66
Table 7.2 – System parameters ............................................................................................ 66
Table 7.3 – Parameters of Scenario 1 ................................................................................... 69
Table 7.4 – Parameters of Scenario 2 ................................................................................... 72
Table 7.5 – Parameters of Scenario 3 ................................................................................... 74
List of figures

Figure 2.1 – Point-to-multipoint applications [15] ................................................................. 20
Figure 2.2 – Shadowing effect ................................................................................................. 22
Figure 2.3 – Multipath propagation ......................................................................................... 23
Figure 3.1 – Concept of the OFDM signal: (a) Conventional multicarrier technique, (b) orthogonal multicarrier technique [18] .................................................................................. 25
Figure 3.2 – Spectra of (a) OFDM subchannel and an (b) OFDM channel ......................... 27
Figure 3.3 – OFDM block diagram .......................................................................................... 28
Figure 3.4 – QPSK, 16-QAM and 64-QAM constellations [18] .............................................. 29
Figure 3.5 – 16-QAM constellation: a) original vector, b) vector after a 6dB noise, c) vector points crossing the decision boundary ......................................................................................... 30
Figure 3.6 – Reed-Solomon codeword .................................................................................... 31
Figure 3.7 – a) Guard interval between OFDM symbols, b) delay spread ......................... 32
Figure 3.8 – Effect of the multipath: the delayed subcarrier #2 causes ICI on delayed subcarrier #1 .......................................................................................................................... 33
Figure 3.9 – Addition of the cyclic prefix .............................................................................. 33
Figure 3.10 – OFDM symbol with cyclic extension ............................................................... 34
Figure 3.11 – A typical amplifier response .......................................................................... 35
Figure 4.1 - In OFDMA, the base station allocates to each user a fraction of the subcarriers where they have a strong channel .................................................................................. 38
Figure 4.2 - PDF of the maximum K user’s channel gain ....................................................... 39
Figure 4.3 – A simple rate allocation problem ...................................................................... 43
Figure 6.1 – Proposed algorithm ............................................................................................ 57
Figure 6.2 – A example of cost function: \( w(x) = x^2 \) .............................................................. 58
Figure 7.1 – Positioning of terminals: a) Configuration that benefits the MSR algorithm, b) Configuration that may starve some users in MSR ................................. 67
Figure 7.2 – Throughput and Mean Throughput after replications ..................................... 68
Figure 7.3 – System throughput in relation to the number of users .................................... 70
Figure 7.4 – Jain’s fairness index in relation to the number of users ......................... 71
Figure 7.5 – System throughput in relation to the traffic rate.................................. 73
Figure 7.6 – Jain’s fairness index in relation to the traffic rate ................................ 73
Figure 7.7 – Dianati’s fairness index in relation to the bandwidth requirement......... 75
Figure 7.8 – Jain’s fairness index in relation to the bandwidth requirement.......... 76
3G – 3rd Generation
3GPP – 3rd Generation Partnership Project
4G – 4th Generation
AM – Amplitude Modulation
ASK – Amplitude Shift Keying
AWGN – Additive White Gaussian Noise
BER – Bit Error Rate
BES – Best Effort Service
BS – Base Station
BWA – Broadband wireless access
CBR – Constant Bit-Rate
CDMA – Code Division Multiple-Access
CSI – Channel State Information
dB – Decibel
DFT – Discrete Fourier Transform
DSL – Digital Subscriber Line
FDM – Frequency Division Multiplexing
FDMA – Frequency Division Multiple-Access
FEC – Forward Error Correction
FFT – Fast Fourier Transform
FIFO – First In First Out
FM – Frequency Modulation
ICI – Inter-Carrier Interference
IDFT – Inverse Discrete Fourier Transform
IFFT – Inverse Fast Fourier Transform
IQ – In-phase Quadrature-phase
ISI – Inter-Symbol Interference
LAN – Local Area Network
MC-CDMA – Multi-Carrier Code Division Multiple-Access
MF – Maximum Fairness
MSR – Maximum Sum Rate
OFDM – Orthogonal Frequency Division Multiplexing
OFDMA – Orthogonal Frequency Division Multiple-Access
PAM – Pulse Amplitude Modulation
PAPR – Peak-to-Average-Power Ratio
PAR – Peak-to-Average Ratio
PDF – Probability Distribution Function
PF – Proportional Fair
PM – Phase Modulation
PRC – Proportional Rate Constraints
PSK – Phase Shift Keying
QAM – Quadrature Amplitude Modulation
QoS – Quality of Service
QPSK – Quadrature Phase-Shift Keying
SINR – Signal-to-Interference-plus-Noise Ratio
SIR – Signal-to-Interference Ratio
SNR – Signal-to-Noise Ratio
TDM – Time Division Multiplexing
TDMA – Time Division Multiple-Access
UCS – Urgent and Crucial Service
WLAN – Wireless Local Area Network
WMAN – Wireless Metropolitan Area Network
Chapter 1

Introduction

1.1 Motivation

Wireless communication has been played a major role in the telecommunications industry in recent years. The market demand for Internet access, multimedia services and home networking has been driven the development of broadband wireless systems. Such systems must be able to provide high-data-rate communications at anytime and in anyplace. The bandwidth of wireless communication systems is often limited by the cost of the radio spectrum required. Hence, to increase the bandwidth is necessary to improve the spectral efficiency of those systems.

The improvement of the spectral efficiency is achieved by increasing the bit rate. In digital communications, the bit rate is related to the symbol-rate, and the number of bits in each symbol. Increasing any of these values leads to the improvement of spectral efficiency. However, conveying many bits per symbol may cause bit errors as the receiver has to distinguish many signal levels. On the other hand, a higher symbol-rate leads to smaller symbol duration, which is more sensitive to the problems of radio transmissions such as delay spread and multipath fading.

This increasing demand for bandwidth in modern wireless equipment and the inherent problems of the systems that rely on conventional single-carrier techniques, has been increasingly driven wireless systems to adopt the multi-carrier technique orthogonal frequency division multiplexing (OFDM) [18], which has become widely used in radio transmission systems. In Europe and Australia, it is used for the broadcast of digital audio and video [1]. Also, it is the modulation adopted for the emerging IEEE standards WLAN 802.11g and WMAN 802.16 (WiMax). Its success is due to the high spectral efficiency and tolerance to loss and multipath problems.

The basic principle of OFDM is parallelization. Instead of transmitting symbols sequentially over the communication channel, the channel is split into many sub-channels
(sub-carriers) and the data symbols are transmitted in parallel over these sub-channels. The smaller the sub-channel bandwidth, the longer the transmission period of the data symbol on that channel. Therefore, the impact on intersymbol interference (ISI) decreases [2]. These sub-channels are computationally combined through an Inverse Fast Fourier Transform (IFFT) [3].

Due to the combination of independent channels, it is possible to use different modulations in each sub-carrier, according to the channel conditions [1], a technique referred to as adaptive modulation and coding. The modulation used in the sub-carrier must balance between spectral efficiency and bit error rate (BER). Thus, the overall efficiency can be increased by choosing the best modulation possible for a sub-carrier under an acceptable BER, given the sub-channel conditions. Further, in a point-to-multipoint communication architecture, multiple users can share the spectrum by dynamically assigning different sets of sub-carriers to different users. This is known as Multiuser-OFDM or OFDMA. The OFDMA is the medium access control mechanism for the standard IEEE 802.16a [5] and the natural choice for the 4th generation (4G) of wireless network systems [4].

In multi-user systems, resource allocation among users can be divided into two classes. In the fixed allocation scheme, like TDMA and FDMA, each terminal receives a time slot or an exclusive spectrum share. Consequently, resources can be wasted as some channels can present deep fade or users may not have enough power to transmit useful information [6]. In the adaptive allocation scheme, an opportunity arises from an effect referred to as multi-user diversity. As several terminals are located in the cell, sub-channels are likely to be in different quality states for different terminals. In other words, the multi-user communication scenario is characterized by a spatial selectivity of the sub-channels [7]. The OFDMA spectral efficiency depends on the many physical factors and adaptive schemes such as power control, interference mitigation using advanced channel allocation techniques and making use of time and space diversity.

In the adaptive allocations schemes, an allocation strategy that targets the higher spectral efficiency can generate a problem regarding to the justice of the resource sharing strategy being used [8]. Users with better channel conditions will always receive most part of resources while the other users may starve. So, fairness is an issue in such
systems. Also, as the wireless systems networks are spreading, it is expected that they can support a wide variety of different services with different requirements of Quality of Service (QoS) and bandwidth. Therefore, cross-layer optimizations techniques must be considered in order to balance fairness, QoS and spectral efficiency.

1.2 State of the art

Orthogonal Frequency Division Multiple Access (OFDMA) is an attractive multiple access technique, because it provides a high degree of channel orthogonality within a cell and is capable of allocating radio resources with a fine granularity to competing users. A great deal of recent research has looked into adaptive techniques capable of improving network spectral efficiency using (1) dynamic control of sub-carrier allocation, (2) transmission power and (3) modulation and coding scheme for every sub-carrier [1][3]. In addition, a number of works have shown that spectral efficiency can be increased by the successful manipulation of channel variation, multi-user diversity, and inter-cell diversity. Along another line, a series of contributions have shown that slowing down elastic traffic increases cellular capacity [9]. There is a trade off between allocated bit rates and session durations when considering slowing down elastic applications as a factor for capacity increase. Recently, it has been shown that under mild conditions, opportunistic scheduling combined with opportunistic power allocation maximizes the throughput of a single OFDMA cell [7]. Unfortunately, the basic schemes do not inherently take into account user-specific or QoS class specific fairness constraints.

Existing applications may be classified in terms of QoS mainly into two distinct service classes: delay sensitive real time services such as voice and video applications and best effort elastic data services such as web access. Intuitively, we look for sub-carrier strategies that allocate guaranteed numbers of radio resources to demanding multimedia real-time applications whereas elastic sessions may receive variable resources above a pre-established minimum threshold.
Chapter 1 – Introduction

Existing models in the literature are NP-hard joint optimization problems and hence computationally complex. Consequently, many heuristics have been developed to turn the problem more tractable [10]. Some of these have been based on the over simplified assumption that modulation is fixed. An observation made in [11] shows that adaptive power control has little additional benefits when adaptive coding is employed within a single cell. Furthermore, since there are constraints on the power level used to transmit at the base station, users at the cell border may experience outage and low QoS in their services. Using similar assumptions, the problem is often reduced from a non-linear problem into a linear one [12]. Most existing approximations rely on the use of iterative algorithms to then build a sub-optimal solution. There has also been a great deal of work on techniques for sub-carrier-bit dynamic allocation and power control [6], [10] and [13]. A recent study of a simple two cells model may be found in [14]. The results however are based on statistical average values.

1.3 Objective

The main objective of our work is to design an allocation strategy that makes efficient use of available radio resources and maximizes overall spectral efficiency while user’s QoS requirements, such as bit-error rate (BER) and throughput, remain guaranteed. This is a known NP-hard joint optimization complex problem and we therefore adopt an incremental approach to dealing with it. In order to achieve this, we start with the study of OFDMA sub-carrier allocation within a single cell. Next, we propose a strategy for dynamic resource allocation in such scenario and evaluate its performance and fairness.

1.4 Dissertation outline

Chapter 2 focuses on the wireless systems and their inherent problem. It presents a general architecture of a wireless system and the challenges related to the radio propagation. It ends with a brief overview of a cellular system.

---

1 A problem is said NP-Hard if an algorithm for solving it can also solve any nondeterministic polynomial time problem, i.e. is can solved in a nondeterministic Turing machine in polynomial time.
Chapter 1 – Introduction

Chapter 3 introduces the orthogonal frequency division multiplexing technique. It shows how OFDM deals with the problems discussed in chapter 2 and its advantages such as adaptive modulation and coding.

Chapter 4 shows how OFDM takes advantage of a multiuser environment, introducing the orthogonal frequency division multiple-access, comparing it to other multiple access techniques and introducing the concept of multiuser diversity.

Chapter 5 presents the problems related to resource allocation and the approaches to deal with it and discuss some resource allocation techniques for OFDMA.

Chapter 6 proposes a heuristic approach for a fair adaptive resource allocation in a single-cell OFDMA. It also discusses some desired properties of an allocation strategy concerning fairness, throughput maximization and QoS awareness.

In chapter 7, a performance evaluation is performed by comparing the proposed algorithm with two of presented algorithms of chapter 5. Finally, chapter 8 concludes this work.
Chapter 2

Wireless Communication

Wireless communication has experienced a remarkable growth in the telecommunication industry in recent years. The growth of the Internet has been driving the demand for high-speed access, and wireless technology has an important appeal to it as a Broadband Wireless Access (BWA), alternative to fixed access technologies. There are two types of broadband wireless services: fixed wireless broadband, which can sometimes be seen as an alternative to DSL service; and the wireless mobile broadband technology, that brings new opportunities like device portability and the mobility itself.

The fixed wireless solution has been used as a substitute in developing countries like Brazil for a wired infrastructure in voice telephony systems: the wireless local-loop. Also, it can be used to provide interconnectivity between buildings within a campus or to offer a broadband access to small offices or homes. It has two types of applications: the point-to-point (interconnection between buildings) and the point-to-multipoint configuration (wireless broadband access). Figure 2.1 presents a scenario for a point-to-multipoint application over a fixed wireless broadband access.
Chapter 2 – Wireless Communication

Figure 2.1 – Point-to-multipoint applications [15]

Such wireless access is likely to be used in rural or underserved areas that lack a good wired infrastructure. The full potential of wireless communications increases when used in nomadic and mobile broadband applications. Nomadicity implies the ability to connect to the network from different locations via different radio base stations. Mobility implies the ability to keep an ongoing connection active while moving [15].

Many wireless broadband technologies have emerged recently such as WiMax (IEEE 802.16). The advantage of WiMax over other technologies like Wi-Fi [16] and 3rd generation (3G) is the combination of the throughput capabilities with the multicellular system. The throughput advantages of WiMax rely on the capability of OFDMA to exploit the multiuser and frequency diversity to improve capacity. As a matter of fact, WiMax is presented as the next 4G system.

2.1 Wireless challenges

A main limitation for wireless communication is the transmission medium itself. The wireless channel is not always a friendly environment. Radio propagation effects such as path loss, frequency selective fading, Doppler spread and multipath delay-spread limit the effectiveness of wireless communications [18]. Understanding these effects is
necessary to understand how OFDMA can take advantage of them. Thus, some basic concepts concerning these effects will be outlined in this chapter.

2.1.1 Path loss

During the propagation of the signal, the power is weakened with the distance. The rate of the decay is much faster than in wired conditions. This is due to fact that power is distributed on a propagating sphere, thus the density power decreases as the area of the sphere increases. Since the area of such sphere is proportional to the radius squared, the power strength reduces proportionally to the distance squared.

In free space, the propagation is very predictable allowing the creation of an accurate model to describe the wireless communication in a link with no obstructions. However, in terrestrial communications, the path loss depends on a number of variables, such as terrain, foliage, obstructions, weather and antenna height. Hence, the environment is much more complex, making propagation modeling much more difficult.

2.1.2 Shadowing

Large obstructions, such as buildings, hills, trees and walls may cause some localized blocking of the signal. These objects cause attenuation as the signal passes through them, resulting in areas with severe loss of power, causing problems in the communication. This loss or missing radio coverage is referred to as to shadowing.
The amount of the shadow depends on the type of the obstructing material and the frequency range of the signal. Some materials may reflect the signal, making them opaque, resulting in large shadows behind them. Others may be transparent to the signal, but absorb part of the energy while the signal passes through them. Under these conditions, most of the energy received at the destination comes from reflection or diffraction rather than a direct path.

2.1.3 Multipath fading

Because of the reflection in multiple objects, the propagation of the signal may take different paths (see Figure 2.2). Thus, the resulting received signal is the sum of the signals received through these different paths, creating a multipath environment. The paths are also different in distance, causing the signal to achieve different propagation times. The propagation time results in a phase rotation in the wavelength, and the amount of the rotation depends on the wavelength and the path length travelled. As signals with the same wavelength propagate through different distances, the phase rotation is different. This causes constructive or destructive interference to the signal at the receiver. A destructive interference occurs when the vector sum is zero. In this case, the resulting signal would experience a deep fade.
Another problem is due to the variation of the path loss with the movement of the receiver or the transmitter. It occurs on distances about 10 to 100 wavelengths and depends on the size of the object causing the path loss rather than the frequency of the signal. It is known as slow fading, due to the slow changing nature of the variation.

The multipath problem also causes fading that changes with the frequency. Hence, some frequencies may experience more destructive interference than others, leading to frequency selective fading.

### 2.1.4 Doppler spread

The Doppler spread is the frequency dispersion due to the relative motion between the transmitter and the receiver. The Doppler spread is proportional to the relative velocity between the transmitter and the receiver and to the frequency of the signal. Because of the multipath components of the signal, the relative velocity of each component leads to a random variation on the frequency caused by the Doppler Effect.
This leads to a degradation of the signal. In OFDM systems, as it will be shown, it causes the loss of orthogonality between subcarriers.

2.1.5 Delay spread

The delay spread is the measure of the time dispersion of the signal. It causes time blurring, where the energy from previous data symbols becomes mixed in with current symbols, resulting in an interference known as Inter-Symbol-Interference (ISI). The interference occurs because the symbols are uncorrelated, adding noise to the signal. At higher data rates, the symbol time is shorter, thus, a small delay can cause ISI. Hence, the ISI is a concern in broadband wireless systems.

2.1.6 Noise

Additive White Gaussian Noise (AWGN) is the most basic problem in any communication channel. It is caused by thermal variations in the environment. Since the amount of noise is proportional to the bandwidth, its impact is much higher in broadband than in narrowband systems.

2.2 Cellular systems

From the path loss, an opportunity arises. Its combination with short-range transmission increases the overall system capacity by allowing simultaneous transmission to occur. It is done by using the so-called cellular systems. The idea is to subdivide the serviced area into smaller geographic areas called cells [15]. Then, a base station is placed in each cell, and the transmit power is adjusted to be just enough to provide the required signal strength to reach the cell boundaries. Hence, the path loss allows an isolation of the cells from each other allowing them to operate at the same frequency. Therefore, it is possible to reuse the frequency band.

When cellular systems support user mobility, it is necessary to transfer an ongoing connection from one cell to another. This is referred to as the handoff process. Achieving a smooth and seamless transfer is a challenging aspect of cellular system design.
Chapter 3

Orthogonal Frequency Division Multiplexing

Orthogonal Frequency Division Multiplexing (OFDM) [18] is a multiplexing technique very similar to Frequency Division Multiplexing (FDM). The idea behind FDM, is the use of different frequencies for each channel to be multiplexed. Information on these channels are transmitted at the same time but do not interfere with each other because they use different carrier frequencies. Also, the channels are bandwidth limited and spaced sufficiently far apart so that the signals do not overlap in the frequency domain.

OFDM uses the main principles of FDM, but the main difference is that it chooses the frequencies in a specific manner in order to increase the spectral efficiency. There is a coordination or synchronization in the combination of the signals to be multiplexed. This leads to a dense packing of subcarriers, which are synchronized with each other allowing them to overlap while controlling the interference (Figure 3.1). They do not cause Inter-Carrier Interference (ICI) due to the orthogonal nature of the modulation [17]. OFDM achieves higher spectral efficiency, in comparison to FDM, because of the reduction of the large guard band used to prevent the interference.

![Figure 3.1 – Concept of the OFDM signal: (a) Conventional multicarrier technique, (b) orthogonal multicarrier technique [18]](image)
Chapter 3 – Orthogonal Frequency Division Multiplexing

In wireless communication, a modulation scheme is used to map the information signal to a waveform that can be transmitted over the channel [18]. Many modulation schemes have been developed: for analog waveform information, Frequency Modulation (FM), Amplitude Modulation (AM), Phase Modulation (PM) and others may be used. For digital sources, schemes include Amplitude Shift Keying (ASK), Phase Shift Keying (PSK), and Quadrature Amplitude Modulation (QAM) [19]. In FDM, the carriers can mix different modulation schemes or even the type of the source signal (digital or analog), since there is no synchronization between the signals. In OFDM, only digital modulation schemes can be used.

The basic principle of OFDM is to split a signal of high data rate into various low rate data signals each one transmitted simultaneously on a subcarrier [18]. Since multiple carriers form a single OFDM transmission, they are commonly referred to as “subcarriers”, while the term carrier is reserved for describing the mixing of the signal. When data is transmitted in parallel, the rate is decreased in a manner that is inversely proportional to the number of subcarriers. Thus, the higher number of subcarrier is, the lower the data rate. Because of this lower rate, the dispersion in time caused by the delay spread is decreased. Also, the intersymbol interference (ISI) is eliminated almost completely by introducing a guard time in every OFDM symbol.

3.1 Basic principles

Orthogonality is the property that allows multiple information signals to be transmitted over a common channel, without interference. It means that the signals are independent of each other. The loss of orthogonality implies the blurring of the signals and the consequent degradation of the communication. Some multiplexing schemes are inherently orthogonal, like Time Division Multiplexing (TDM). The division of the signal in time slots prevents the interference between them, thus they are orthogonal in nature. In FDM, the orthogonality is achieved by separating the signal in space frequencies to avoid interference. In OFDM, the orthogonality is achieved mathematically [18]. A set of functions are orthogonal if they match the conditions of Equation 3.1.

\[
\int_0^\tau s_i(t)s_j(t)dt = \begin{cases} 
C & \text{if } i = j \\
0 & \text{if } i \neq j
\end{cases}
\]

Equation 3.1
Chapter 3 – Orthogonal Frequency Division Multiplexing

If two different signals are multiplied and integrated over a symbol, the result must be zero, for orthogonal functions. Hence, in OFDM, the subcarriers are orthogonal because when waveforms of any subcarriers are multiplied and integrated in a symbol, the result is zero.

This property can be seen by observing the spectrum. Figure 3.2 (a) shows a subchannel of a QAM modulated signal, with symbol duration \( T \), that crosses zero every \( 1/T \). In Figure 3.2 (b), the spacing between subchannels is carefully selected in order to place each subchannel where the others cross zero. Although there is some overlap between the modulated subcarriers, they are isolated from each other through a correlation filter. The orthogonal nature of the transmission is a result of the peak of each subcarrier corresponding to the zeros of the all others. When the resulting signal is detected with a Discrete Fourier Transform (DFT), the spectrum is not continuous as in (b) but has discrete samples. These samples are the peaks in Figure 3.2. If DFT is time synchronized, the frequency samples of the DFT correspond to only the peaks of the subcarriers, hence the overlapping region between subcarriers does not affect the receiver. The peaks occur where the zero occurs for all other subcarriers, leading to the orthogonality.

![Figure 3.2 – Spectra of (a) OFDM subchannel and an (b) OFDM channel.](image-url)
3.2 Generation of subcarriers

The analog generation of OFDM signals requires a set of coherent oscillators that is hard to implement, mainly for a high number of subcarriers. Thus, OFDM signals are typically digitally generated. Although a complex task, it is feasible because of the increasing computational power available nowadays [17]. The transmission starts with the conversion of the serial data into a mapping of subcarrier amplitude and phase, see Figure 3.3. Then, the spectral representation is transformed into the time domain using a Inverse Discrete Fourier Transform (IDFT) [20]. In practice, the systems use an IFFT since it is much more computationally efficient [21]. To transmit, the calculated time domain signal is mixed to the required frequency.

![Figure 3.3 – OFDM block diagram](image)

The receiver (lower block of Figure 3.3) performs the reverse operation. It receives the signal from the channel, which is mixed to the based band for extraction, and a Fast Fourrier Transform (FFT) is applied to convert the signal to the frequency domain. Then, the amplitude and phase of the subcarrier is extracted to create back the digital data. The IFFT and the FFT are complementary functions.

3.2.1 Subcarrier modulation

The data allocated in each OFDM symbol depends on the modulation scheme used and the number of subcarriers. This is known as bit loading. For instance, for a subcarrier modulation of 64-QAM, each subcarrier would carry 6 bits of data, thus if transmission
uses 1000 subcarriers, the number of bits per symbol would be 6000. Once the data is allocated to a subcarrier for transmission, it is then mapped using a modulation scheme to a subcarrier amplitude and phase, which are represented by a complex In-phase and Quadrature-phase (IQ) vector [18]. Quadrature Amplitude Modulation (QAM) is the most popular type of modulation in OFDM. Especially rectangular constellations are easy to implement as they can be split into independent Pulse Amplitude Modulation (PAM) for both In-phase and quadrature parts. Hence, subcarrier modulation can be implemented using a lookup table. Figure 3.4 shows the constellations for QPSK, 16-QAM and 64-QAM.

At the receiver, the demodulation is performed by mapping the received IQ vector back to data. However, during the transmission, noise and distortion are added to the signal. Figure 3.5 (a) shows a 16-QAM constellation generated in the transmitter, Figure 3.5 (b) shows the same constellation received after a 6dB noise. Each of the received IQ points is blurred due to the channel noise. Thus, the receiver has to estimate the original transmitted vector. If the noise is too high, like in Figure 3.5 (c), errors may occur since the points cross over the decision boundary.
3.2.2 Error correction

When the transmission occurs in a multipath radio environment, frequency selective fading can result in groups of subcarriers being heavily attenuated. In fact, some may be completely lost due to deep fade [18]. Hence, despite most subcarriers being able to reach the receiver without errors, the overall Bit Error Rate (BER) will be dominated among a few subcarriers, leading to a high error probability on those. To avoid this, Forward Error Correction (FEC) codes are essential. By using codes across the subcarriers, errors of weak subcarriers can be corrected. Thus, the performance of the OFDM link depends on the average of the received power, rather than by the power of the weakest subcarrier.

3.2.2.1 Block codes

A block code is an error correction code that adds redundancy to a block. It encodes a block of \( k \) input symbols into a block of \( n \) symbols, with \( n \) larger than \( k \). The purpose is the increase of the minimum Hamming distance, which is the minimum number of different symbols between any pair of code words. The most popular block codes are the Reed-Solomon codes. The Reed-Solomon encoder takes \( k \) data symbols of \( m \) bits and adds parity symbols to form an \( n \) symbol codeword (see Figure 3.6).
There are \( n-k \) parity symbols of \( m \) bits each. A Reed-Solomon decoder can correct up to \( t \) symbols that contain errors in a codeword, where \( 2t = n-k \), given \( n = 2^m - 1 \). However, it is only true if the errors occur with the maximal amount of correctable errors, according to the Hamming distance definition. Thus, if a Reed-Solomon code is designed to correct up to two symbol errors containing 8 bits per symbol, it cannot correct an arbitrary combination of three bit errors [18]. This characteristic makes Reed-Solomon codes useful for correcting bursty channels. For instance, in OFDM, the errors would be concentrated on a few subcarriers that perceive the deep fade.

### 3.2.2.2 Convolutional codes

A convolutional code maps the input stream to the output stream by convolving the input bits with a binary impulse response. It can be implemented with shift registers and modulo-2 adders. For instance, in a convolutional coder of rate \( \frac{1}{2} \) code, an input stream of single data could generate two outputs \( A_i \) and \( B_i \), interleaved to form the output code sequence \( \{A_1B_1A_2B_2\ldots\} \). The interleaving is specified by generator vectors or generator polynomials [18]. For instance, a generator could specify for a code rate of \( \frac{3}{4} \), the following sequence: \( \{A_1B_1A_2B_3A_4B_4A_5B_5A_6B_6\ldots\} \).

### 3.2.3 Guard time

One of the main advantages of OFDM is its tolerance to the multipath delay spread. The symbol rate for an OFDM signal is much lower than a single carrier transmission scheme, because of the division of the symbol in many subcarriers. This low symbol rate makes OFDM naturally resistant to effects of Inter-Symbol Interference (ISI), caused by multipath propagation. To completely eliminate the ISI, a guard interval is added between
transmissions of the OFDM symbols, as shown in Figure 3.7 (a). If the duration of the guard time is larger than the delay spread, see Figure 3.7 (b), it is possible to guarantee that there is no ISI between subsequent OFDM symbols.

![Figure 3.7 – a) Guard interval between OFDM symbols, b) delay spread](image)

### 3.2.4 Cyclic prefix

Although the guard time can mitigate the ISI, the problem of Intercarrier Interference (ICI) may arise. The ICI is the loss of orthogonality due to the crosstalk between them. This effect is shown in Figure 3.8, when the FFT on the receiver tries to demodulate the OFDM symbol, it will encounter some interference from the other subcarriers, because there is no integer number of cycles between the subcarriers, hence, the loss of orthogonality.
To eliminate the ICI, the channel must appear to provide a circular convolution. Each subcarrier in the data section of the symbol (the symbol without the guard period), has an integer number of cycles. Because of this, adding a cyclic prefix results in a continuous signal (see Figure 3.10). The cyclic prefix is a copy of the end of the symbol placed in the guard period (see Figure 3.9). This ensures that delayed replicas of the OFDM symbol always have an integer number of cycles [18].
3.3 The PAPR problem

To transmit an OFDM signal, the wireless device has to amplify it in order for the signal to have enough power to be picked up by the receivers. This is done by the power amplifier, which tends to be the biggest consumer of energy in wireless devices. Hence, the efficiency of the power amplifier is crucial, and depends basically on two factors: a) the amplifier must be able to amplify the highest peak of the wave, hence, the peak is the decider of the power consumption of the amplifier; and b) the peak does not transport more information, therefore, the speed does not depend on the peak but on the average power level.

The problem is that typical amplifiers are not a linear device (see Figure 3.11). If the signal is amplified in the nonlinear region, high peak signals will experience a distortion (constellation tilting and scattering). This affects severely the performance of the system. Hence, the amplitude range of the signal must be kept in the boundary of the linear region of the amplifier.
In OFDM, the signals have a high Peak-to-Average Ratio (PAR) or Peak-to-Average-Power Ratio (PAPR). Thus, most of the time, the signal in OFDM remains in the average area of the Figure 3.11. This leads to a sub utilization of the amplifier, reducing the efficiency and hence increasing its cost. However, if the average signal fits the linear region of the amplifier, the peaks would cross it and reach the nonlinear region, leading to distortion. The PAR problem is an important challenge because the amplifier is one of the most expensive components in the radio.
Chapter 4

Multiuser-OFDM

OFDM is not a multiple-access technique but rather a modulation that creates many independent streams of data and can be used by different users. Thus, a different approach can be implemented: the orthogonal frequency division multiple-access (OFDMA), where users share subcarriers besides timeslots. It allows an increased exploration of multiuser diversity, an increased freedom in scheduling with the possibility of fine grained allocation of users on channels (subcarriers). This comes at a small cost, i.e. the user must know what channels were assigned to him and the transmitter needs the channel state information for each user in order to take a decision on how to allocate channels.

The strategies for multiple-access systems focus on providing non-interfering communication channels for each user. There are several ways to achieve that. The most common are the multiplexing of frequency, time or code division. In Frequency Division Multiple Access (FDMA), the frequency spectrum is divided in frequency bands, with each user having access to some of these bands. In Time Division Multiple Access (TDMA), the users periodically take turns, either on demand or on a round-robin fashion, of the entire available bandwidth to burst during a period of time. Code division multiple access (CDMA) is similar to TDMA in the sense that does not divide the frequency range in narrow channels. Instead, it employs a spread spectrum technology and a special coding scheme (where each transmitter is assigned a code) to allow multiple users to be orthogonally multiplexed over the same physical channel [19]. However, all these techniques give the same normalized radio capacity, in case of AWGN [22].

One advantage of OFDMA is that it can be seen as a combination of the previously mentioned techniques. In some way, it can take advantage of the best features of each one [15]. For instance, FDMA can be implemented with OFDMA by using static allocation of channels (frequencies) among users. Also, as the number of users in the system becomes higher than can be carried on a single symbol, OFDMA also makes then use of the time
division technique. Thus, OFDMA can accommodate multiple users in the same way TDMA does. In this context, static TDMA can be viewed as round-robin scheduling while OFDMA is an adaptive scheduling.

Another similarity can rise from CDMA, since it is not fundamentally incompatible with OFDMA [18]. They can be combined to create a multicarrier CDMA (MC-CDMA). In one of its schemes, it spreads the original data stream over different subcarriers using a given spread code in the frequency domain, i.e. each chip of spread code is transmitted through a different subcarrier [23]. Nowadays, CDMA is the most common multiple-access technique for cellular systems, but its flaw is the need for considerably larger bandwidth than the data rate in order to suppress interference.

This problem does not occur in OFDM, and, as OFDMA is an extension of OFDM, it inherits its advantages like the robust multipath suppression and frequency diversity. Besides, it can explore the multiuser diversity and adaptively allocate the power and subcarrier, thus maximizing the throughput. This adaptive allocation is based on the channel conditions and can also be performed by an adaptive modulation. Each subcarrier can use a different modulation based on the conditions (given by the CSI – Channel State Information) at the user it will be assigned to. Thus, higher data rates can be achieved by choosing the appropriate modulation scheme, and higher gains can be achieved by giving channels to users with “good” conditions. In OFDMA, the base station allocates to each user subcarriers preferably in a range where they have a strong channel (see Figure 4.1).

In comparison to OFDM, there is a potential in OFDMA to reduce the power and to relax the PAPR problem. This problem is more acute in the uplink, where the power efficiency and cost of power amplifiers of the user terminals are highly sensitive. As the PAPR increases with the number of subcarriers, the small number of subcarriers allocated to the user tends to decrease the PAPR.
4.1 Multiuser Diversity

The multiuser diversity is an effect experienced in wireless systems when many users are present. Due to mobility and other interferences a time-variation perception of the user channel quality appears, leading to moments of severe decay and moments of stronger channel gains. It can be exploited by scheduling transmissions when a user has favorable channels condition. Hence, it is a cross-layer technique in which the medium access control layer uses CSI from the physical layer in order to perform the allocation. The more users in the system, the better the probability of finding a user with good channels condition, thus, the total throughput tends to improve with the number of users [24].

In a system where the subcarriers experiences Rayleigh fading, the Probability Density Function (PDF) of the $k$ user’s channel gain is given by:
Chapter 4 – Multiuser OFDM

\[
p(h_k) = \begin{cases} 
2h_k e^{-h_k^2} & \text{if } h_k \geq 0 \\
0 & \text{if } h_k < 0 
\end{cases}
\]

Equation 4.1

Thus, if the base station transmits only to users with maximum channel gain, the PDF of \( h_{\text{max}} \) is:

\[
p(h_{\text{max}}) = 2K h_{\text{max}} \left(1 - e^{-h_{\text{max}}^2}\right)^{K-1} e^{-h_{\text{max}}^2}
\]

Equation 4.2

Figure 4.2, shows the PDF for the users with the highest gain \( h_{\text{max}} \) [15] for various values of \( K \). In practice, these gains may be reduced by averaging effects, such as spatial diversity and the need to assign users continuous block of subcarriers to run their applications adequately.

4.2 Adaptive Modulation and Coding

Another concept of OFDMA is the use of adaptive modulation and coding to take advantage of the fluctuations in the channel conditions. The goal is to avoid dropping of
packets by adjusting the modulation, and consequently, the data rate of the channel according to the channel state. For poor quality channels, a lower rate is used to transmit. This can be achieved by using a small constellation, such as QPSK, and low-rate error-correcting codes, such as rate $\frac{1}{2}$ convolutional or turbo codes [15].

In order to decide which modulation scheme will be used, the transmitter needs to know the channel Signal-to-Noise Ratio (SNR) $\gamma$ to choose the modulation scheme given a target BER. The desired BER is a function of the SNR which depends on the type of digital modulation. For instance, for BPSK, the BER is given by $BER_{BPSK} = Q(\sqrt{2SNR})$, for QPSK, it is given by $BER_{BPSK} = Q(\sqrt{SNR})$, where $Q$ denotes the Gaussian-Q function [25]:

$$Q(x) = \frac{1}{\sqrt{2\pi}} \int_{x}^{\infty} e^{-t^2/2} dt$$  \hspace{1cm} \text{Equation 4.3}$$

Some works have been made to describe the BER expression for an arbitrary modulation scheme, in particular for 16-QAM and 64-QAM [26] [27]. The work described in [28] gives a BER model for M-ary square QAM for an AWGN channel and Gray code mapping:

$$BER = P_b = \begin{cases} \frac{1}{2} \text{erfc}(\sqrt{\gamma}), & m = 1 \\ \frac{2}{m} \left(1 - \frac{1}{\sqrt{2^m}}\right) \text{erfc} \left(\frac{3\gamma}{\sqrt{2^m} - 1}\right), & m > 1 \end{cases}$$  \hspace{1cm} \text{Equation 4.4}$$

Where, $m = \{1,2,\ldots,\log_2 M\}$, and $\text{erfc}$ represents the complementary error function:

$$\text{erfc}(x) = \frac{2}{\sqrt{\pi}} \int_{x}^{\infty} e^{-u^2} du$$  \hspace{1cm} \text{Equation 4.5}$$

Which gives the same results as those from [29]:

$$m = \begin{cases} 3.4\log(\gamma) - 2.05, & m \geq 1 \\ 0.14\gamma, & 0 < m < 1 \end{cases}$$  \hspace{1cm} \text{Equation 4.6}$$
Hence, the following table can be obtained[28]:

<table>
<thead>
<tr>
<th>Mod.</th>
<th>Bits/sym ( (m) )</th>
<th>( \gamma_n (\text{db}) )</th>
</tr>
</thead>
<tbody>
<tr>
<td>BPSK</td>
<td>1</td>
<td>2.799419</td>
</tr>
<tr>
<td>QPSK</td>
<td>2</td>
<td>3.756657</td>
</tr>
<tr>
<td>8-QAM</td>
<td>3</td>
<td>5.041215</td>
</tr>
<tr>
<td>16-QAM</td>
<td>4</td>
<td>6.765017</td>
</tr>
<tr>
<td>32-QAM</td>
<td>5</td>
<td>9.078258</td>
</tr>
<tr>
<td>64-QAM</td>
<td>6</td>
<td>12.18249</td>
</tr>
<tr>
<td>128-QAM</td>
<td>7</td>
<td>16.3482</td>
</tr>
<tr>
<td>256-QAM</td>
<td>8</td>
<td>21.93833</td>
</tr>
</tbody>
</table>

The consequence of this approach is that it requires the transmitter to have the knowledge of the current channel state information. However, in high mobility environments, where the Doppler frequency is high and the channel changes rapidly, the CSI might be outdated due to processing and feedbacks delays, which causes significant performance degradation. A mean to overcoming this is the use of channel prediction [30][31].

### 4.3 Fairness

From the operator’s perspective, system capacity is the foremost performance measure. The system capacity is often measured as the throughput of its subsystems while attending some QoS levels, hence, maximizing the overall system throughput. The problem with the maximization of the total throughput is that the system looses fairness among the users as some of them may always have “bad” conditions and never have a subcarrier allocated. There must be a balance between the maximization of the system capacity and a fair distribution of resources among users. One approach is the use of utility functions to maximize capacity where negative return values may be used to ensure fairness. However, finding the right comparable functions between the different service requirements is not straightforward. Also, negative utility function may not be meaningful; hence, its design is not trivial at all.

The concept of fairness, or equality, arises in various contexts such as economy, operating systems and telecommunication networks [32]. It is an inherent problem when
a limited amount of resources must be allocated among various entities. Thus, it is highly desirable to find a method to measure and compare the degree of fairness for a specific allocation policy.

The problem of fairness measurement has been extensively studied and some traditional methods have been proposed as a general purpose fairness index to compare different allocation schemes:

- The Gini Index [33]:
  \[ I_{\text{Gini}} = \frac{1}{2n^2 \bar{x}} \sum_i \sum_j (x_i - x_j) \]  
  Equation 4.7

- The Min-max index[32]:
  \[ I_{\text{min-max}} = \frac{\min\{x_i\}}{\max\{x_i\}} \]  
  Equation 4.8

- The Jain fairness index[32]:
  \[ I_{\text{Jain}} = \frac{\sum_{i=1}^{n} x_i^2}{n \sum_{i=1}^{n} x_i^2} \]  
  Equation 4.9

Where \( x_i \), \( n \) and \( \bar{x} \) are the amount of allocated resource to user \( i \), the total number of users, and the average allocated resource, respectively.

The key issue in adopting a metric of fairness is the ambiguities of some indexes. Suppose a network model as shown in Figure 4.3. There are two links represented by the rectangles with limited capacity \( C_1 = 1 \) and \( C_2 = 2 \). The users denoted by \( X_0 \), \( X_1 \), and \( X_2 \) share the resources of each link according to the model. Let \( S_1 = \left( \frac{1}{2}, \frac{1}{2}, \frac{1}{2} \right) \), and \( S_2 = \left( \frac{1}{2}, \frac{1}{2}, \frac{3}{2} \right) \) be the allocation vector corresponding to two different allocation schemes. Table 4.2 shows the fairness index of the above mentioned metrics of each allocation scheme.
According to the common sense of fairness, the first allocation vector appears to be fairer. But, considering the network configuration, the resource $C_2$ can only be used by $X_0$ and $X_2$. Thus, the allocation in $S_2$ may not be less fair than $S_1$, because the resource in $C_2$ could not be given to $X_1$. Thus, $S_1$ wastes resources. However, according to Table 4.2, the allocation $S_1$ is always fairer than $S_2$ for the given indexes.

![Figure 4.3– A simple rate allocation problem](image)

Another problem concerning fairness indexes is the meaning of the value. How to compare different values for fairness among allocation options? For instance, how big is a value of 1.0 when compared to 0.5? In this context, in [32], some well-accepted desirable properties for fairness metric were identified:

- **Population size independence**: the index must support any number of users, finite or infinite. Also, it must be applicable to only two users sharing a resource;
- **Scale and Metric independence**: the index must be independent from the metric or scale used. The unit of measurement must not interfere in the result of the metric;
• **Boundedness:** the index must be bounded between 0 and 1, i.e. a totally fair system must be 1 and a totally unfair system must be 0. Hence, it can be expressed in percentage. For instance, an index of 0.1 means that the allocation scheme is fair to 10% of users and unfair to 90% of the users.

• **Continuity:** the index must be continuous. Any change in allocation must be perceived in the index.

Dianati et al [34] proposed a utility-basis framework to evaluate the degree of fairness for resource allocation schemes in wireless access networks. It offers a methodology that takes into account both effort and service unfairness, and can be customized for different application types and QoS requirements. The Dianati fairness index is given by:

\[
F = \frac{\left( \sum_{i=1}^{n} U_i(s_i) \right)^2}{\sum_{i=1}^{n} U_i(1)}. \tag{Equation 4.10}
\]

Where \( s \) is the amount of resources allocated, and \( U_i \) is a utility function given by:

\[
U_i = \begin{cases} 
0 & s \leq s^{(\text{min})} \\
\frac{s}{s^{(\text{min})}} & s^{(\text{min})} \leq s
\end{cases} \tag{Equation 4.11}
\]

and

\[
U_2 = \begin{cases} 
0 & s \leq s^{(\text{min})} \\
\sqrt{s - s^{(\text{min})}} & s^{(\text{min})} \leq s \leq s^{(u)} \\
\sqrt{s^{(u)} - s^{(\text{min})}} & s^{(u)} \leq s
\end{cases} \tag{Equation 4.12}
\]

Where \( s^{(\text{min})} \) is the minimum resource required and \( s^{(u)} \) is the resource effectively used (not the resource allocated). Note that the utility function returns zero when the minimum resource requirement is not satisfied. The denominator on Equation 4.10 is the utility when all the requested resource is allocated.
Chapter 5

Resource allocation techniques

There is a great deal of research focusing on taking advantage of multiuser diversity and adaptive modulation in OFDMA. The goal of a scheduling algorithm for dynamic allocation is to determine which users to schedule, how to allocate subcarriers to them, and how to determine the appropriate power levels for each user in each subcarrier [15]. All users estimate and feedback the CSI to a base station, which executes the allocation procedure, where subcarrier and power allocation are determined according to the users’ CSI. Once the allocation is determined by the base station, it must be informed to the user which subcarriers have been assigned to him. This task is performed by a messaging system. This procedure repeats itself periodically changing the subcarrier allocation overtime.

The resource allocation problem is usually formulated as a constraint optimization problem. Two classes of adaptive optimization have been proposed: margin-adaptive [6] and rate-adaptive [3][35][36]. In margin-adaptive optimization, the idea is to minimize the total transmit power with a constraint in the user data rate [6][37]. In rate-adaptive optimization, the goal is to maximize the total data rate while the constraint is the total power [12][38][39][40]. The first class of optimization is useful for fixed-rate applications such as voice, whereas the second class accommodates better bursty applications such as data and IP services.

Both classes belong to the group of integer programming problem [12]. Finding an optimal solution is a difficult task despite the finite set of possible solutions, in fact, these problems are NP-hard [42] [43]. A lot of research has been conducted to design an optimal solution or even near optimal with low computational costs. These proposals belong to one of three different methods:

- **Relaxation**: Consists of relaxing the integer constraints on the allocation problem of bits or subcarrier. With the relaxation, both margin and rate adaptive problems become linear programming problems that can be
resolved more easily. This is done by accepting real numbers on allocation, but once a solution is reached, the resulting values must be reevaluated to find the integer solution (feasible solution). It was first proposed in [6];

- **Problem splitting:** It applies the technique of divide and conquer; this big problem is divided in two complex problems. The first one tries to allocate all users’ requirements of resources. Then, the second allocation performs the rest of the operation, which, for instance, would try to maximize the throughput. This technique is used in [9];

- **Heuristics:** Try to solve both classes of problems by using heuristics mostly based on sort procedures and presented in [3]. In comparison with the relaxation, this technique can decrease the use of CPU power by a factor of up to 100.

### 5.1 Single-Cell allocation

In this section, some techniques will be described according to the classification made in [15]. The table below summarizes the notation used.

<table>
<thead>
<tr>
<th>Notation</th>
<th>Meaning</th>
</tr>
</thead>
<tbody>
<tr>
<td>( K )</td>
<td>Number of users</td>
</tr>
<tr>
<td>( N )</td>
<td>Number of subcarriers</td>
</tr>
<tr>
<td>( h_{k,n} )</td>
<td>Channel gain of user ( k ) on subcarrier ( n )</td>
</tr>
<tr>
<td>( P_{k,n} )</td>
<td>Transmit power of user ( k ) on subcarrier ( n )</td>
</tr>
<tr>
<td>( \sigma^2 )</td>
<td>AWGN power spectrum density</td>
</tr>
<tr>
<td>( P_{tot} )</td>
<td>Total transmit power of the base station</td>
</tr>
<tr>
<td>( B )</td>
<td>Total transmission bandwidth</td>
</tr>
</tbody>
</table>
5.1.1 Maximum Sum Rate Algorithm

The goal of Maximum Sum Algorithm (MSR) is to maximize the total rate (sum of rates of each user), given a total power constraint [12]. This algorithm gives an optimal solution if the final goal is to maximize the overall throughput. The drawback is that users near to the base station will have excellent channel conditions and will therefore have the preference in allocation of the system resources. The consequence is the starvation of users in bad propagation conditions, i.e. users far from the base station. The algorithm works as follow:

Let $P_{k,n}$ denote the user $k$’s transmit power on subcarrier $n$. The Signal-to-Interference-plus-Noise Ratio (SINR) for user $k$ on subcarrier $n$ ($SINR_{k,n}$) is can be expressed as:

$$SINR_{k,n} = \frac{P_{k,n} h_{k,n}^2}{\sum_{j=1, j\neq k}^{K} P_{j,n} h_{j,n}^2 + \sigma^2 \frac{B}{N}}$$  \hspace{1cm} \text{Equation 5.1}

Using the Shannon formula as the throughput measure, the maximum sum rate algorithm maximizes the following quantity:

$$\max_{P_{k,n}} \left\{ \sum_{k=1}^{K} \sum_{n=1}^{N} \frac{B}{N} \log\left(1 + SINR_{k,n}\right) \right\},$$  \hspace{1cm} \text{Equation 5.2}

With the total power constraint: $\sum_{k=1}^{K} \sum_{n=1}^{N} P_{k,n} \leq P_{tot}$. It is easy to see that the sum capacity is maximized if the total throughput in each subcarrier is maximized. Thus, this problem can be divided into $N$ simpler problems, each one for each subcarrier. Further, the sum capacity of each subcarrier $n$ is given by:

$$C_n = \sum_{k=1}^{K} \log \left(1 + \frac{P_{k,n}}{P_{tot,n} - P_{k,n} + \sigma^2 \frac{B}{h_{k,n}^2 N}} \right),$$  \hspace{1cm} \text{Equation 5.3}
Chapter 5 – Resource allocation techniques

Where \( P_{tot,n} - P_{k,n} \) denotes the other users’ interference to user \( k \) on subcarrier \( n \). Consequently the sum capacity is maximized when all available power is allocated to the user with the maximum gain. It is very intuitive: give the channel to the user with the best gain in it. That is why this technique is often referred to as a “greedy” optimization.

5.1.2 **Maximum Fairness Algorithm**

Although the Maximum Sum Rate algorithm achieves the maximum throughput of an OFDMA system, it creates a problem concerning the fairness of the distribution of resources. As explained in section 3.3, the path loss attenuation varies by several orders of magnitude between users, and some might be, most of the time, with bad channel conditions leading to no allocation of subcarriers to them. Thus, as an alternative to MSR, the Maximum Fairness Algorithm (MF) [45] does the opposite by allocating subcarriers and power so that the minimum user’s data rate is maximized.

The maximum fairness algorithm can be referred to as a max-min class problem, since the goal is to maximize the minimum data rate. This maximization is a complex problem due to difficulty to simultaneously find the optimal subcarrier and power allocation. To solve it, the problem is formulated as a convex optimization problem. Despite the inherent complexity, it can be solved with linear programming.

Thus, the problem of maximum fairness algorithm is formulated as:

\[
\begin{align*}
\max_{\mathcal{P}_{k,n}, \omega_{k,n}} t, \text{ subject to } \\
t & \leq \sum_{n=1}^{N} \frac{\omega_{k,n} B}{N} \log_{2} \left( 1 + \frac{P_{k,n} h_{k,n}^2}{N_0 \frac{\omega_{k,n} B}{N}} \right) \quad \text{for all } k,
\end{align*}
\]

Equation 5.4

With the constraints:

\[
\begin{align*}
\sum_{k=1}^{K} \sum_{n=1}^{N} P_{k,n} & \leq P_{\text{max}}, \\
P_{k,n} & \geq 0, \text{ for all } k, n,
\end{align*}
\]
\[
\sum_{k=1}^{K} \omega_{k,n} \leq 1, \text{ for all } n,
\]
\[
\omega_{k,n} \geq 0, \text{ for all } k, n.
\]

Where, \( \omega_{k,n} \) represents the portion of subchannel \( n \) assigned to \( k \).

Solving this convex optimization problem gives the optimal solution to max-min problem. However, it requires an intense computation due to the recursive nature of solutions for these types of problems [45]. A common approach to avoid it is to assume initially that equal power is allocated to each subcarrier, and then iteratively assign the available subcarriers to the low-rate users with good channel gain. This sub-optimal approximation achieves very close performance in comparison with the exhaustive search for the best joint subcarrier-power allocation, in terms of fairness and total throughput.

### 5.1.3 Proportional Rate Constraints algorithm

Another weakness of the MSR algorithm is its lack of flexibility. In wireless systems, different users have different QoS requirements, thus, the application-specific data rates vary substantially. Hence, the maximum fairness algorithm can be extended to add some flexibility. In the Proportional Rate Constraints Algorithm (PRC), there is a constraint to the user data rate, which must be proportional to a predefined parameter. This parameter depends on the QoS of the users’ application and the policies of the system.

Mathematically, the proportional data rate constraint can be expressed as [46]:

\[
\frac{R_1}{\beta_1} = \frac{R_2}{\beta_2} = \frac{R_3}{\beta_3} = \ldots = \frac{R_k}{\beta_k}, \tag{Equation 5.5}
\]

Where, \( \{\beta_k\}_{k=1}^{K} \) is the set of predetermined parameters and the user’s achieved data rate is:
\[ R_k = \sum_{n=1}^{N} \rho_{n,k} \frac{B}{N} \log_2 \left( 1 + \frac{P_{k,n} h_{k,n}^2}{\sigma^2 B} \right), \]

Equation 5.6

And \( \rho_{n,k} \) is either 0 or 1, indicating that the channel \( n \) is allocated to the user \( k \). If \( \beta_k = 1 \forall k \), it behaves exactly as the Maximum Fairness Algorithm. The advantage is that any arbitrary data rate can be achieved by adjusting the value of \( \beta_k \).

The PRC optimization problem is also generally very difficult to solve directly, since it involves both continuous and binary variables, and the feasible set is not convex [15]. Some near-optimal approaches were outlined in [3][35] and a low complexity algorithm was developed in [46].

5.1.4 Proportional Fair Scheduling

The last discussed algorithm attempts to achieve an objective instantaneously, such as the total sum (MSR), the maximum fairness or proportional rates for each user. It is a memory-less approach, i.e. in each allocation, the new decision does not take into account the last decisions (allocations). Alternatively, this algorithm attempts to achieve its objective over the time, giving some flexibility to the scheduling procedure. Hence, in addition to the compromise of throughput and fairness, there is another QoS component: the latency.

In a system with latency tolerance, the base station could wait for the user to get close to start the transmission. It can be viewed as an opportunistic scheduling. Using this approach, the Proportional Fair (PF) algorithm [47] can achieve maximum throughput and fairness on the long term, since there is no constraint on the latency. This long term could be seconds, minutes or even hours.

The scheduler is designed to take advantage of the multi-user diversity while maintaining a reasonable long-term throughput to all users [15]. Let \( R_k(t) \) be the instantaneous data rate that the user \( k \) can achieve at time \( t \). Let \( T_k(t) \) be the average
throughput of the user $k$ up to the tie slot $t$. The goal is to select the user $k^*$ with the highest $R_k(t)/T_k(t)$ for transmission. Then the average throughput $T_k(t)$ is updated to all users as follows:

$$ T_k(t + 1) = \begin{cases} 
(1 - \frac{1}{t_c})T_k(t) + \frac{1}{t_c}R_k(t) & k = k^* \\
(1 - \frac{1}{t_c})T_k(t) & k \neq k^* 
\end{cases} $$

Equation 5.7

As the user selection is based on the instantaneous data rate relative to the average throughput, “bad” channels are unlikely to be selected to each user. However, on the long term, underserved users will receive priority, hence, promoting fairness. The parameter $t_c$ is the time scale of the system. With large $t_c$, the latency increases, and the system will benefit the instantaneous throughput, since the average throughput will have a slight change. On the other hand, with small $t_c$, the average throughput changes quickly, forcing the schedule to prioritize fairness, at the expense of instantaneous throughput.

Under proportional fair algorithm with averaging time scale $t_c \rightarrow \infty$, the long term average throughput of each user exists almost surely, and the algorithm maximizes [47]:

$$ \sum_{k=1}^{K} \log T_k $$

Equation 5.8

almost surely.

The Proportional Fair algorithm was designed for single-channel, time-slotted systems, but it can be adapted in OFDMA systems, by treating each subcarrier independently [48]. Let $R_k(t, n)$ be the instantaneous data rate that the user $k$ can achieve at time $t$ in subcarrier $n$. Then, for each subcarrier, the user with the highest $R_k(t, n)/T_k(t)$ is chosen. Let $\Omega_k(t)$ denote the set of subcarriers in which the user $k$ is scheduled for transmission at time slot $t$, then the average user throughput is updated as
\[ T_k(t+1) = \left(1 - \frac{1}{t_c}\right) T_k(t) + \frac{1}{t_c} \sum_{n \in \Omega_k(t)} R_k(t,n). \]  

Equation 5.9

### 5.1.5 Performance Comparison

Table 5.2 shows a brief comparison of the four allocation techniques discussed in this section. Of all allocation techniques, MSR is the best in terms of maximizing throughput. Of course, this occurs at a complete expense of any kind of fairness, including the possibility of some users to starve. Hence, it is not a feasible solution because it requires that all users have nearly identical channel conditions, which is not a reasonable requirement.

The MF does the contrary, at the expense of the total throughput; it tries to bring an “equal” distribution of resources among all users. Although the PRC algorithm allows a flexible trade-off between these extremes, it may be not reasonable to apply data rate constraints in real time. The PF algorithm is simple to implement and also achieves a practical balance between throughput and fairness, but it requires the acceptance of a latency to achieve the fairness.

<table>
<thead>
<tr>
<th>Algorithm</th>
<th>Total Throughput</th>
<th>Fairness</th>
<th>Complexity</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum Sum Rate (MSR)</td>
<td>Best</td>
<td>Poor and inflexible</td>
<td>Low</td>
</tr>
<tr>
<td>Maximum Fairness (MF)</td>
<td>Poor</td>
<td>Best but inflexible</td>
<td>Medium</td>
</tr>
<tr>
<td>Proportional Rate Constraint (PRC)</td>
<td>Good</td>
<td>Most flexible</td>
<td>High</td>
</tr>
<tr>
<td>Proportional Fair (PF)</td>
<td>Good</td>
<td>Flexible</td>
<td>Low</td>
</tr>
</tbody>
</table>

### 5.2 Multi-cell Allocation

In cellular systems, there is the possibility of channel reuse, when the distance between two base stations is long enough so there is no interference between these channels. In such systems, the allocation technique can be performed in two stages. The first stage is the inter-cell, where the channels are distributed among the base stations...
located in the cells. If two base stations do not interfere with each other, the same channel
can be assigned to both. The next stage is the intra-cell allocation. In this stage, the same
techniques for the single-cell allocation can be used, but restricted to the channels
allocated to the base station.

In [49], an allocation scheme for IEEE 802.11 Wireless LANs is proposed, where
neighboring base stations must use different channels in order to avoid interference. The
channel assignment is modeled as a game problem. Then, the Nash equilibrium of the
game in relation to the total coverage is found for an optimal solution.

In [50], the same problem was treated as a weighted variant of the graph coloring
problem [51]. Since it is a NP-Hard problem, some scalable distributed algorithms were
proposed.
Chapter 6

Proposed Algorithm

Most dynamic allocation techniques aim to achieve a maximum throughput based on the users’ channel quality. Others try to bring fairness to the system at the expense of improving the throughput, or try to establish data rates to their users. The main problem faced by allocation technique rises from the fact that throughput maximization and fairness are conflicting goals. Thus, we argue that a good allocation technique must let the network operator define the balance between the fairness and throughput maximization.

The goal is to achieve a low complexity algorithm to dynamic subcarrier allocation that can be parameterized by the network operator in order to improve throughout, fairness, or give privileges to some user or certain classes of applications. The algorithm must take into account the QoS requirement of users or applications, and try to give the maximum throughput possible under these restrictions, established by the network operator or the system policies.

In this context, we propose a strategy for dynamic resource allocation based on the use of an utility function, where users would compete for the available resources. The idea is to maximize the overall system gain during each allocation cycle by carefully allocating users on channels (subcarriers) based on an index calculated for each user. The index is calculated by a utility function that can be more easily understood as the cost for a user to maintain data on his buffer. Thus, the overall gain is higher when the resulting cost after the allocation cycle is lower. The resulting cost is the sum of all costs associated to the users after an allocation cycle. For each channel, the allocation strategy must choose the user that leads to the highest cost reduction if that channel is allocated to him. Therefore, the algorithm runs iteratively, choosing the best user (i.e. the one with the higher cost reduction), until all the channels are allocated.
6.1 System Model

This strategy is designed for adaptive allocation on a single OFDMA cell. However, it can be extended to a multi-cell scenario by assuming the base stations as a special type of competitors and performing a two stage allocation. Let $N$ be the number of OFDM sub-carriers available on a BS (base station). We assume that $k$ users compete for radio resources. Let $\delta_{k,n}$ be the allocation matrix of user $k$ on sub-carrier (channel) $n$. Hence $\delta_{k,n} = 1$ when user $k$ receives channel $n$ and is null otherwise. Note that channel allocation is exclusive, which means that at any moment a given channel may only be allocated to a single user. Thus, if $\delta_{k,n} = 1$ then $\delta_{k',n} = 0$ for any $k' \neq k$. It is also assumed that the base station has access to the channel state information (CSI) which provides the current signal to noise ratio $SNR_{k,n}$ of the user $k$ on subcarrier $n$. The SNR, together with other information (e.g. the target BER) is used by the base station to dynamically choose the best modulation for transmitting to user $k$ using subcarrier $n$.

Using the selected modulation, it is assumed that in a given timeslot, the user $k$ is capable to transmit $b_{k,n}$ bits while using subcarrier $n$. We define $s_k$ as the amount of bits the user $k$ has awaiting on his queue for transmission and $\sigma_{k,n} = \min(b_{k,n}, s_k)$ as the maximum amount of data the user $k$ is able to transfer using subcarrier $n$, considering both the data waiting on his queue and the maximum allowed to transfer at the timeslot given the channel conditions. The Table 6.1 summarizes the used symbols.
Chapter 6 – Proposed algorithm

Table 6.1 – Symbols used

<table>
<thead>
<tr>
<th>Symbol</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ℵ</td>
<td>The set of available subcarriers on a single OFDM cell</td>
</tr>
<tr>
<td>N</td>
<td>The number of available subcarriers, ( N =</td>
</tr>
<tr>
<td>K</td>
<td>The set of users competing for resources on a single OFDM cell</td>
</tr>
<tr>
<td>Κ</td>
<td>The number of users competing for subcarriers, ( Κ =</td>
</tr>
<tr>
<td>( δ_{k,n} )</td>
<td>The subcarrier allocation matrix. ( δ_{k,n} = 1 ) means user ( k ) can use subcarrier ( n )</td>
</tr>
<tr>
<td>( s_k )</td>
<td>Amount of bits queued for user ( k )</td>
</tr>
<tr>
<td>B</td>
<td>The maximum amount of bytes that can be queued on the BS</td>
</tr>
<tr>
<td>( b_{k,n} )</td>
<td>The maximum amount of data that user ( k ) is able to transfer using subcarrier ( n ), given the current subcarrier conditions.</td>
</tr>
<tr>
<td>( σ_{k,n} )</td>
<td>The maximum amount of data that user ( k ) is able to transfer using subcarrier ( n ), considering the data waiting on his queue and the maximum allowed to transfer, given the current subcarrier conditions.</td>
</tr>
<tr>
<td>( G_k(\alpha, \sigma) )</td>
<td>The gain a user ( k ) obtain when flushes out ( b ) bits from a total of ( z ) existing on his buffer</td>
</tr>
</tbody>
</table>

6.2 Algorithm

We now describe in details the proposed algorithm, which is executed in each timeslot to perform resource allocation. During the execution of the algorithm, it is assumed that both CSI information and the amount of bytes on the user buffers are static, i.e. any changes during the algorithm execution must be considered only for the next timeslot. The algorithm is depicted in Figure 6.1.
Chapter 6 – Proposed algorithm

1. Proposed Algorithm:
2. Begin
3. for each $k \in K$ do
4. $z_k \leftarrow s_k$
5. for each $n \in \mathcal{N}$ and $k \in K$ do
6. $\delta_{k,n} \leftarrow 0$
7. while $\mathcal{N} \neq \emptyset$ do
8. Begin
9. $k^* = \arg \max_k \left(G_k(z_k, \sigma_{k,n})\right)$
10. $\delta_{k^*,n} = 1$
11. $z_{k^*} = z_{k^*} - \sigma_{k^*,n}$
12. $\mathcal{N} = \mathcal{N} \setminus \{n\}$
13. End
14. End

Figure 6.1 – Proposed algorithm

The time complexity of the algorithm is $O(KN \log N)$ assuming the complexity of gain processing $G_k(z, \sigma)$ as $O(1)$. Thus, the complexity of the algorithm is highly sensitive to the complexity of the gain function. Therefore, we only propose gain functions with complexity $O(1)$.

The rationale behind the algorithm is that the user who obtains the greater gain by using a particular subcarrier will receive that subcarrier, where the gain is evaluated by the function $G_k(z, \sigma)$. This is reflected on line 9 of Fig 6.1, which selects the user $k^*$ for using subcarrier $n$. Line 10 assigns subcarrier $n$ to the selected user $k^*$ and line 11 updates the amount of data remaining on the user’s virtual buffer $z_{k^*}$, decrementing the amount of data user will flush. It is important to observe that no data is transmitted yet, and future queries to the amount of data queued for the users are done considering virtual amount of data $z_{k^*}$ instead of real amount $s_{k^*}$. This is done to keep track of the resources already received by the selected user so that on the next iteration the gain for that user will be lesser than the current one, since he has already received a channel. This is performed until all subcarriers have been assigned.
6.3 A general function to evaluate gain

We define function $G_k(z, \sigma)$ as the gain a user $k$ obtains when being able to transmit $\sigma$ bits from a total of $z$ bits waiting on its buffer. An easier approach is to invert and think about the cost, so $G$ is calculated as:

$$G_k(z, \sigma) = W(z) - W(z - \sigma)$$

Equation 6.1

where $W$ is a steadily increasing function that represent the cost of keeping $x$ amount of data in user buffer. Therefore, it is easy to see that $G_k(z, \sigma)$ accounts for the difference between the cost of keeping $z$ units of data and the cost of keeping $z - \sigma$ units of data. Also, is important to observe that $W(z) - W(z - \sigma)$ is different from $W(\sigma)$ when $W(\cdot)$ is not linear, see Figure 6.2 for $W(x) = x^2$.

Figure 6.2 – A example of cost function: $W(x) = x^2$

$z = 5, \quad \sigma = 1$

$W(z) - W(z - \sigma) \Rightarrow W(5) - W(5 - 1) \Rightarrow 25 - 16 \Rightarrow 9$

$W(\sigma) = 1$
The idea of function $G_k(z,\sigma)$ is to allow the algorithm to support different behaviours according to the method of cost calculation. The cost function must be constructed to match a certain desired behaviour. To achieve that, some ideas will be developed in next sections.

6.4 Fairness

Consider a $W(.)$ linear to $z$. By doing so, the best user will always be the one with the best SNR because he will be able to use the best modulation, giving the highest cost reduction. In fact, if $W(.)$ increases linearly, then $G_k(z,\sigma) = W(z) - W(z - \sigma)$ reduces to $G_k(z,\sigma) = W(\sigma)$. Hence, and the expression $k^* = \arg\max_k G_k(z_k,\sigma_{k,n})$ (see line 9 of the algorithm in Figure 6.1) will always find the user $k^*$ with greatest $\sigma_{k,n}$, that is necessarily the user that is able to transmit higher amount of data using subcarrier $n$. Using such linear function, the algorithm behaves like the Maximum Sum Rate. This function gives the maximum throughput to the system to the detriment of the common sense of fairness (i.e. equality) including the possibility of starvation of some users.

Now, consider the following equation:

$$G_k(z,\sigma) = B - z \quad \text{Equation 6.2}$$

Equation 6.2 gives highest gain to the user with the less resources allocated to him. Hence, the system will give channels to the users by considering only the amount of resources already received forcing the system to equally distribute resource among users. Obviously, this is done in complete detriment of the overall system throughput. At first, all the users have the same gain and receive the best channel to them. Then, the remaining channels are distributed in a way that will almost give the same amount of resources for all users. Therefore, the algorithm behaves like the MF.

Another approach is:

$$W(z) = 0 \quad \text{Equation 6.3}$$
Chapter 6 – Proposed algorithm

The equation 6.3 gives a cost function so that \( W(z) - W(z - \sigma) = 0 \), \( \forall \sigma, z \). Thus, the resources will be distributed in a round robin fashion. Again, this is done in complete detriment of the overall system throughput. A better approach is to define the cost function as:

\[
W = f(z, \sigma)
\]

Equation 6.4

Thus, \( f \) can trade-off throughput and equal resource distribution. For the sake of simplicity, suppose \( f \) as a steadily increasing non-linear function of \( z \). In this case, \( W \) benefits throughput but, at same time, does not allow the buffer of a user with bad SNR to increase too much. Note that buffer occupation is tightly related to delay and this proposal tries somehow to control delay by controlling buffer size. For instance, consider \( f(z, \sigma) = z^2 \). At higher values of \( z \), \( W(z) - W(z - \sigma) \) is bigger, i.e. the rate of the cost reduction increases faster than \( z \) (Figure 6.2).

All the equations present memory-less functions for \( W \). This is important for the reduction of algorithm complexity. It is also possible to define functions that keep the state among allocation cycles as well as the amount of resources allocated on the last cycles. This approach would allow the algorithm to calculate the cost based on long time fairness, like the PF.

6.5 Quality of Service

An important observation about fairness is that it does not necessarily mean equality. Users or services may have different requirements in terms of bandwidth or delay. One of the advantages of OFDM systems is the possibility of adapting the resource allocation for the user according to their QoS requirements. So far, the proposed strategy does not use a cross-layer optimization technique. The strategy can be improved to gather information from the data-link layer to meet the user QoS requirements, as in [52].

Firstly, we assume initially two types of QoS classes:

- Urgent and Crucial Service (UCS): supports real time data (voice, video-conferencing);
• Best Effort Service (BES): supports data streams without delay requirements (file transfer, WEB browsing)

A good way to represent this model is to think about these requirements in terms of probabilities. For instance, if a service packet with a delay constraint does not reach its destination in a certain time, there is a high probability for this packet to be useless and therefore it will be dropped by the service. This gives us an approach to design a cost function based on the probability of a packet to be discarded if it waits too long on the queue (user buffer). Even BES packets must not wait too long. Of course, the definition of “long” depends on the type of the QoS requirement.

The user buffer is divided into two queues, each one associated to a class (type of QoS requirement). Each queue has a queue cost function based on the class. Hence, the overall cost is given by:

\[ W = f(W_{UCS}(z_k, b_k), W_{BES}(z_k, b_k)) \]  

Equation 6.5

where \( W_{UCS} \) and \( W_{BES} \) are the queue function for the classes and \( z_k \) and \( b_k \) are the amount of bytes for the queues. The simplest possible cost \( f \) is the straight sum \( f = W_{UCS} + W_{BES} \). When a user receives a resource, he serves first the queues with higher priorities. The relationship between the queue functions must be adjusted to match the fairness definitions of the system. For example, suppose that \( W_{UCS} = \alpha_{UCS} z_k \), if \( \alpha \) is too high, it may cause starvation of the other queues or other users. However, this might be a desired situation under some setups. Whether it is fair or not is a subjective matter, and must be adjusted by the network operator. In another situation, it could be desired that certain users have more proportional resources allocated to them. For instance, a user that pays more for the service.

Also, suppose a class of QoS with bandwidth requirement. This is a case for a cost function that must keep the state of allocated resources for the last cycles. For instance:

\[ W = \begin{cases} \alpha z^2, & \text{if the requirement was not met} \\ z^2, & \text{if the requirement was met} \end{cases} \]  

Equation 6.6
with $\alpha \geq 1$. The cost of a queue that has not met the bandwidth requirement is higher. This enforces the algorithm to privilege the users that have not yet met the requirement. The higher $\alpha$, the bigger the privilege.
Chapter 7

Performance Evaluation

The evaluation of the proposed algorithm is performed by simulation. A discrete event simulator written in Java has been implemented. It reproduces a radio environment with support for algorithms that perform the subcarrier allocation. The radio environment is composed by a base station and a set of terminals attached to it. The main goal of the base station is to allocate subcarriers among the terminals according to an allocation technique. In practical systems, besides the subcarrier allocation, the base station is also responsible for power control, admission control and handover control. However, these services will not be evaluated and implemented in the simulation environment, since the main objective is to evaluate the spectral efficiency and the fairness achieved with the proposed algorithm. During the simulations, all algorithms will be given the same number of terminals and will use the same power transmission.

Since the main metrics are the spectral efficiency and fairness, the proposed algorithm is compared to the allocation techniques described in section 4: the maximum sum rate and the maximum fairness; which give, respectively, the maximum spectral efficiency and the maximum fairness possible. The maximum sum rate is able to achieve the maximum spectral efficiency, thus, it is used to compare how close the proposed algorithm can get to this limit. On the other hand, the maximum fairness is used to establish the fairness boundaries.

For the sake of simplification, the base stations only perform the downlink allocation in the simulations.

7.1 The Simulator

The Simulator uses the Object-Oriented paradigm to represent the components and interactions of a simple OFDM system. It uses the Desmo-J [53] library that provides a framework to discrete-event modeling and simulation, licensed under the Apache License. The Desmo-J makes easy the simulation programming by abstracting the tasks
inherent to discrete-event modeling. In models of this type, all system state changes are
supposed to happen at discrete points in time. Between such events the system state is
assumed to remain constant. Discrete-event simulation is therefore particularly suitable
for systems like network queuing, in which relevant changes of state occur suddenly and
irregularly.

7.1.1 Simulator description

The main components of the simulator are the base station, the terminals and the
application traffic generator. The base station is the most important component since it
performs subcarrier allocation to the terminals attached to it, according to a given
allocation technique. In the base station, there is a FIFO queue that receives packets
generated by the application generators.

Attached to each terminal, there is an application generator that generates a
certain type of traffic, for instance, Constant-Bit-Rate (CBR) traffic, to be delivered to the
terminals. According to the traffic generation parameters, the application generator
creates a packet and queues it up in the base station queue. After each allocation cycle,
the base station dequeues the packets and sends them to the terminals, according to the
bit-rate determined by the modulation type (see section 7.1.2) of the subcarriers assigned
to the terminals.

A coverage area is defined for the base station. It is a circle centered at the base
station position. All terminals are randomly placed in the coverage area, defining the
terminals’ positions, which are used to calculate the distances to the base station. Because
the SNR depends on the distance between the terminal and the base station, the bigger the
coverage area, the smaller the mean SNR for the terminals.

7.1.2 SNR calculation and modulation of subcarriers

It is assumed that the base station has the complete knowledge of the channel
(subcarrier) state for each terminal. SNR is calculated according to the 3GPP propagation
model of a macro-cell defined in 3GPP TR 25.942 V7.0.0 [54]. The local-mean Signal-
to-Interference Ratio (SIR) is calculated by dividing the received signal by the
interference and multiplying by the processing gain. The signals from the other users are
summed together and seen as interference. Signal-to-interference-ratio in downlink can be expressed as [54]:

$$SIR_{DL} = \frac{G_p \cdot S}{\alpha \cdot I_{OWN} + I_{OTHER} + N_0}$$

Equation 7.1

Where $S$ is the received signal, $G_p$ is processing gain, $I_{OWN}$ is interference generated by those users that are connected to the same base station that the observed user, $I_{OTHER}$ is interference from other cells, $\alpha$ is the orthogonality factor and $N_0$ is thermal noise [54]. The orthogonality factor takes into account the fact that the downlink is not perfectly orthogonal due to multipath propagation. The assumed value for the orthogonality factor alpha for macro-cell (for instance, a city) is 0.4, and the thermal noise is -99dB, according to [54].

The received signal depends on the path loss due to distance and carrier frequency. Based on the SNR of the subcarrier, the modulation chosen is the one which gives the maximum data rate for a BER of $10^{-5}$, according to table 4.1.

7.1.3 Metrics

The main metrics collected by the simulator are: throughput, Jain fairness index and Dianati fairness index. These values are collected in intervals that depend on the duration of the simulation and the number of samples to collect. For instance, if the number of samples is 60 and the simulation time is 60s, the metrics will be collected every 1s. At the end of the simulation, a mean of the values measured in the samples is provided. For throughput, the per-terminal value is also provided.

7.2 Simulating parameters

This work does not intend to deeply simulate all aspects of OFDMA. Hence, some parameters of an OFDMA system (e.g. the time guard interval) will be ignored and only the parameters that have a direct impact on the performance of algorithm will be considered. The parameters were chosen based on the WiMax OFDMA specification.
[55], in order to provide a simulation environment similar to real conditions. Table 7.1 shows the fixed parameters used in all simulations.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Propagation Model</td>
<td>3GPP TR 25.942</td>
</tr>
<tr>
<td>Power</td>
<td>1 W</td>
</tr>
<tr>
<td>System Frequency</td>
<td>6GHz</td>
</tr>
<tr>
<td>System Bandwidth</td>
<td>1.25 MHz</td>
</tr>
<tr>
<td>Number of subcarriers</td>
<td>128</td>
</tr>
<tr>
<td>Frame duration</td>
<td>10ms</td>
</tr>
<tr>
<td>Target Bit Error Rate</td>
<td>0.000010</td>
</tr>
</tbody>
</table>

Table 7.1 – OFDMA Parameters used in the simulation

The duration of each simulation experiment was 60 seconds. During the simulation, a hundred samples of the desired data were collected. This means that a sample is collected every 0.6 sec. The queue buffer of the base station was fixed to 500 Kbytes. If the queue reaches this limit, the packets are dropped. Table 7.2 summarizes the system parameters.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Simulation time</td>
<td>60 sec</td>
</tr>
<tr>
<td>Cell Radius</td>
<td>1.00 Km</td>
</tr>
<tr>
<td>Queue Buffer</td>
<td>500.00 Kbytes</td>
</tr>
<tr>
<td>Number of samples</td>
<td>100</td>
</tr>
<tr>
<td>Number replications of the Simulation</td>
<td>30</td>
</tr>
</tbody>
</table>

Table 7.2 – System parameters

The cell radius is one kilometer large, which means that the coverage area of the base station is a circle with a radius of one kilometer. All terminals are randomly placed in this circle. Because of the large area, a small number of terminals can create a number of different configurations in terms of positions that have a relevant impact in the system throughput. For instance, in Figure 7.1 (a), the position of the terminals gives a very good throughput to the system. It benefits the MSR algorithm as it can achieve a high spectral efficiency and also a high fairness index. It happens because all the terminals are near to the base station, giving a good SNR to all terminals and a small variation between their SNR, leading to a near equal distribution.
On the other hand, in Figure 7.1 (b), the variation of SNR of the terminals tends to bring an unfair distribution of resources if the MSR algorithm is used. Terminals that are far from the base station may experience very low data rates or even may not receive resources at all. Also, the overall system throughput tends to decrease as the distance of the terminals to the base station increases.

Hence, to perform a fair evaluation, it is necessary to eliminate these variations. This is achieved by performing various replications of the same experiment. By replicating the same conditions to all algorithms (same number of terminals), after a number of replications, the variations caused by different positioning of terminals tend to be the same (or equally distributed) to all algorithms. Thus, the result considered is the mean of the results of each replication. Figure 7.2 presents the throughput (a) and the mean throughput (b) of the same simulation with eight (8) terminals and CBR traffic of 1024Kbits/s. It can be seen that when the number of replications increases, the mean result of the replications stabilizes.
Figure 7.2 – Throughput and Mean Throughput after replications
Chapter 7 – Performance evaluation

7.3 Scenario 1

This scenario focuses on the evaluation of overall system throughput with regard to the number of users on the system. The system throughput is the sum of the throughput achieved by each terminal during the simulation. In addition to the throughput, the fairness of resource distribution is also evaluated. The metric used for fairness is the Jain index described in section 4.3. This metric is appropriate to measure the equality of resource distribution. The main goal of this scenario is to determine the degree of fairness achieved in comparison with the optimal possible fairness of the MF algorithm. The proposed strategy must achieve a balance between the system throughput and fairness. It was configured with the cost function depicted in Figure 6.2.

The application generators associated to the users were configured to generate constant bit rate (CBR) traffic of 128 Kbits/s. The simulation parameters were described in Tables 7.1 and 7.2. The number of users in the system varies from 2 to 128. Table 7.3 summarizes the specific parameters used in this scenario.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of users</td>
<td>2 ~ 128</td>
</tr>
<tr>
<td>Type of traffic</td>
<td>CBR</td>
</tr>
<tr>
<td>Data rate</td>
<td>128 Kbits/s</td>
</tr>
<tr>
<td>Cost function</td>
<td>Figure 6.2</td>
</tr>
</tbody>
</table>

All configurations were simulated separately and the only parameter that differs is the number of users according to Table 7.3. As explained before, each simulation is replicated many times in order to eliminate the variations caused by the placement of users in the cell. Then, there is as many data as the number of samples collected during the simulation. For instance, after running the simulation for two users, there will be sixty (60) values for the system throughput, each one corresponding to the time it was collected. Since these values remain almost constant, because there is no user mobility, the registered value is taken from the mean of the throughput of all samples. The same procedure is adopted to calculate the fairness index. The result is presented on Figure 7.3.
These results show that, with constant bit rate traffic of 128 Kbits/s, the system is not stressed when the number of users is within the range of 2 to about 30. This can be seen in the linear portion of the graphic above. As the number of user increases, the overall system throughput also increases. In this range, the system has sufficient resources available to meet all users’ demand. This allows the three algorithms to, in addition to achieve the maximum throughput, also achieve the maximum fairness.

When the system reaches its limits, the overall system throughput stabilizes and, as expected, the Maximum Sum Rate algorithm achieves the best results. The users with the best SNR acquire the available subcarriers guaranteeing the maximum throughput. In the implementation of the MSR, the users only receive subcarriers when there is a demand to send traffic. Hence, there is no waste of resources because users do not receive subcarriers if they have nothing to send.

On the other hand, the Maximum Fairness algorithm has the poorest throughput performance. By guaranteeing the same rate to all users, the subcarriers can be allocated to users that do not necessarily have the best SNR.
Chapter 7 – Performance evaluation

The proposed algorithm achieves a performance comparable to that of the MSR algorithm, being able to achieve a system throughput very close to it. With the cost function used, the proposed algorithm tries to maximize the system capacity. However, it also tries to maintain the fairness in resource distribution. This is shown in Figure 7.4.

![Jain’s Fairness Index](image)

As explained before, the three algorithms achieved the maximum fairness within the range of 2 to about 30 users. However, only the Maximum Fairness algorithm maintained the fairness index very close to 1. Observe that when the number of users increases, the fairness index has a subtle decrease. It occurs because when the number of users is high, there is also a high heterogeneity between the signal quality of the users and it is a little more difficult to maintain the same rate among all of them.

In the Maximum Sum Rate algorithm, an abrupt fall in the fairness index is observed when the system reaches its limit. In fact, it has the worst distribution of resources in terms of fairness since the main goal is to maximize the system capacity. On the other hand, the proposed algorithm, despite maintaining a throughput comparable to the MSR, was able to provide a fairer distribution. Observe that a little decrease in system
throughput allows the proposed algorithm to offer a much better fairness index. The lowest value was 0.85, 30% better than the MSR. Also, it has a subtle fall in the fairness index while the heterogeneity increases (caused by the increasing number of users).

### 7.4 Scenario 2

This second scenario focuses again on the evaluation of overall system throughput but with regard to the traffic generated by the users on the system. There is a fixed number of thirty-two (32) users in the system. The metrics of the simulations are the same as the ones used in the scenario 1, using the same procedures to calculate them. The main goal of this scenario is also to determine the degree of fairness achieved. The cost function is the same used in scenario 1 as well as the other parameters concerning the simulation. Table 7.4 summarizes the parameters used for this scenario.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of users</td>
<td>32</td>
</tr>
<tr>
<td>Type of traffic</td>
<td>CBR</td>
</tr>
<tr>
<td>Data rate</td>
<td>64 ~ 1024 Kbits/s</td>
</tr>
<tr>
<td>Cost function</td>
<td>Figure 6.2</td>
</tr>
</tbody>
</table>

The results are shown by Figures 7.5 and 7.6. They are very similar to the results of the scenario 1. In fact, the effect of increasing the number of users is similar to the effect of increasing the traffic rate: they both increase the users’ demands for resources and the system is not able to meet them. However, the heterogeneity of the users’ channels quality in this scenario is smaller than in the first one. This justifies why the decrease in the fairness index is lesser. Again, the proposed algorithm offers a good trade-off between throughput and fairness. In comparison to the MSR algorithm, the proposed algorithm achieves a fairness index 40% better.
Chapter 7 – Performance evaluation

Figure 7.5 – System throughput in relation to the traffic rate

Figure 7.6 – Jain’s fairness index in relation to the traffic rate
### Scenario 3

This scenario focuses on the evaluation of fairness in the distribution of resources in a constrained environment where the users have QoS requirements. The QoS requirement defined is a minimum bit rate that all users must achieve. Fairness will be measured by using the Jain’s fairness index and the Dianati’s fairness. The Jain’s fairness index measures the equality in resource distribution but it is not appropriate to a system with QoS requirements because it does not show if the requirement was met. It will be used only to show the behavior of the resource distribution when the requirements change. The metric that actually measures the fairness in a QoS restricted environment is the Dianati’s fairness index.

The main goal of this scenario is to determine how well the algorithms meet the bandwidth requirements. The proposed strategy must achieve a good system throughput while also meeting the QoS requirements of the users. It was configured with the cost function depicted in Equation 6.6.

The application generators associated to the users were configured to generate constant bit rate traffic of 512 Kbits/s. The simulation parameters are the same as those for scenario 1 and were described in Tables 7.1 and 7.2. The number of users in the system is fixed in thirty-two (32) as in scenario 2. The bandwidth requirements vary from 32 to 128 Kbits/s. Table 7.5 summarizes the specific parameters used in this scenario.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of users</td>
<td>32</td>
</tr>
<tr>
<td>Type of traffic</td>
<td>CBR</td>
</tr>
<tr>
<td>Data rate</td>
<td>512 Kbits/s</td>
</tr>
<tr>
<td>Bandwidth requirement</td>
<td>32 ~ 128 Kbits/s</td>
</tr>
<tr>
<td>Cost function</td>
<td>Equation 6.6</td>
</tr>
</tbody>
</table>

The Dianati’s fairness index was calculated with $\alpha = 0.2$, as recommended in [34]. The result is presented on Figure 7.7.
The proposed algorithm achieved a higher fairness in comparison to Maximum Sum Rate while maintaining a close value in the system throughput. The Maximum Fairness achieves the best fairness but has an abrupt fall when the system is not able to provide the bandwidth requirements to all users. This is seen in Figure 7.7 when the bandwidth requirement is above 112 Kbit/s. Since the MF delivers the same bit rate to all users, none of the users achieved the requirement hence, the fairness index was 0. Eventually, all algorithms will suffer this effect if the requirement continues to increase. At that point, this did not happen to the proposed and the MSR algorithms because only part of the users achieved a bit rate below the requirements. But in the MSR algorithm more user data rates were below the requirements as well as the distribution was unfairer than in the proposed algorithm, hence, the fairness index was higher than the last one.

The issue discussed above can be easily understood by answering the following question: when the system is not able to provide the bandwidth requirements of all users, how to proceed? For the MF algorithm, the answer is to give all users a data rate below
the requirements but the same to all of them. However, it could be more useful to meet the requirements of some users at least, since it is not possible to meet everyone’s requirements. And, to improve the system throughput, the users chosen to meet the requirements are those with the best SNR. This is what the MSR and the proposed algorithms do, but the last is fairer. It adjusts the distribution to meet the requirements, thus, it is the only one that changes the Jain’s index (see Figure 7.8). The index increases because the distribution tends to be more equal.

![Figure 7.8 – Jain’s fairness index in relation to the bandwidth requirement](image)
Motivated by the increasing popularization of mobile communication, in addition to new applications and advanced services, there is a growing demand for efficient usage of the available bandwidth in wireless systems. The broadband wireless communication has a great potential to meet these necessities and enabling technologies play a fundamental role. OFDM offers a good flexibility to maintain a maximum spectral efficiency, by matching the system parameters, such as subcarrier modulation and frequency, based on the conditions of the radio channel. The adaptive modulation technique in OFDM reduces the BER allowing the system to improve the data rate.

Also, in a multiuser environment, there is an opportunity to improve the adaptability in subcarrier allocation by exploiting the variations in channel response between users, caused by the frequency selective fading. In this context, various adaptive techniques aimed to maximize the system capacity were proposed. Such maximization has a side effect in systems that have a great heterogeneity in the channel’s quality of users. It regards to the fairness in resource distribution. Because throughput maximization and fairness are conflicting goals, an algorithm that tries to establish a fixed balance between both lacks of flexibility. Also, in practical systems, an allocation technique must take into account the QoS requirements of users or applications.

Hence, an adaptive allocation technique should not decide how to balance these requirements but must provide the means to the network operator do that. For instance, the algorithm could provide support for traffic prioritization for a user that pays more or for a specific application.

This work presented an algorithm the meet the above mentioned requirements. The idea was to offer a low complexity algorithm to dynamic subcarrier allocation that can be parameterized by the network operator in order to improve throughout, fairness, or to give privileges to some user or certain classes of applications. It is achieved by defining a utility function that controls the behavior of the algorithm. The simplest way to
understand how this function works is to think about it as the calculation of the cost a user would give to the system if a subcarrier is not assigned to him.

It was shown, how the proposed algorithm could assume different behaviors comparing it to some of the resource allocation techniques presented previously: the Maximum Sum Rate algorithm, the Maximum Fairness, the Proportional Rate Constraints, and the Proportional Fair Scheduling.

The evaluation of the proposed algorithm was performed by means of simulation. A simulator of OFDMA system was implemented for compare the performance of the proposed algorithm to the MSR and MF algorithms. It was written in Java using a framework for discrete-event modeling. It is able to simulate some aspects of an OFDMA environment including the SNR of each user on each subcarrier. Also, it is possible to define the OFDMA parameters and the wireless infrastructure of the simulating system.

In simulations, it was shown that the proposed strategy achieves a performance comparable to the MSR but maintaining a good fairness between the users in different scenarios.

The main contribution of this work was the proposed strategy aimed to provide a flexible and low complexity approach for adaptive resource allocation. By taking the decision of allocation off the algorithm and putting it in a utility function, the network operator can fully parameterize the behavior of allocation, defining the goals in terms of system capacity, fairness, and attendance of QoS requirements.

Another contribution is the Simulator implemented. It was developed in the project RRM and provides the means to implement different strategies of resource allocations in order to compare them, using metrics such as throughput and fairness indexes in different simulating scenarios.

For future work, it could be considered and implemented other utility functions for the proposed algorithm to compare with other algorithms using different scenarios. Also, it could be done a generalization of the model to a multi-cell one with support for coordinated radio resource among the different cells or technologies.


[8] Wu Yu, Wei Ji-bo and Xi Yong, ”Opportunistic Scheduling with Statistical Fairness Guarantee in Multi-Rate Wireless Networks”.


References


[35] Ian Wong, Zukan...


Dissertação de Mestrado apresentada por André Luis Cavalcanti Moreira à Pós-Graduação em Ciência da Computação do Centro de Informática da Universidade Federal de Pernambuco, sob o título “A Low Complexity Algorithm for Dynamic fair Resource Allocation in OFDMA Systems”, orientada pelo Prof. Djamel Fawzi Hadj Sadok e aprovada pela Banca Examinadora formada pelos professores:

Prof. Aluízio Fausto Ribeiro Araújo  
Centro de Informática / UFPE

Prof. Carlos Alberto Kamienski  
Pós-Graduação em Engenharia da Informação / UFABC

Prof. Djamel Fawzi Hadj Sadok  
Centro de Informática / UFPE

Visto e permitida a impressão.  
Recife, 29 de agosto de 2008.

Prof. FRANCISCO DE ASSIS TENÓRIO DE CARVALHO  
Coordenador da Pós-Graduação em Ciência da Computação do  
Centro de Informática da Universidade Federal de Pernambuco.