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Scheduling Algorithms for OFDM Broadband Wireless Systems

By

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A thesis submitted to

The Faculty of Graduate Studies and Research

In partial fulfillment of

The requirements for the degree of

Doctor of Philosophy

Ottawa-Carleton Institute for Electrical and Computer Engineering

Department of Systems and Computer Engineering

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Ottawa, Ontario

Canada

December 200

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**Dedicated to Dr. Yan for his enlightening,
to my parents and family for their love,
and to my friends for their supports and helps.**

Abstract

Scheduling algorithms are playing a key role in radio resource management (RRM) of 4G broadband wireless systems due to spectrum scarcity. The Proportional Fair (PF) scheduling algorithm was proposed in seeking the optimal compromise between system throughput and user fairness. It has been well studied and successfully applied to CDMA/HDR systems, but not to OFDM based networks, which is a competitive candidate for 4G wireless systems. The performance of PF algorithm depends on the observation window and can only provide short term fairness. In this thesis, we modify it to eliminate the impact of the observation window. In addition, we apply it to OFDM broadband wireless system and name it OFDM Proportional Fairness (OPF). OPF is tailored to comply with IEEE 802.16a standard for Medium Access Control (MAC) scheduling scheme which is an area open for research. In combination with adaptive modulation and coding (AMC) scheme, system performance is evaluated both from efficiency and fairness. Efficiency is in terms of throughput, mean packet delay and delay variance. Fairness is in terms of user satisfaction rate and average user rate. For comparison purpose, we also implement 4 algorithms in the 802.16a OFDM system. Two of them are conventional algorithms: Maximal SNR and Round Robin (RR). The others are two versions of PF: PF with observation window and weighted PF. Joint MAC and Physical (PHY) layer simulation results show that, compared with PF and conventional methods, OPF gives a better compromise between system throughput and fairness by providing the highest user satisfaction rate under wireless channel conditions of both time varying and frequency selective fading.

In addition, a theoretical framework is developed by presenting the mathematical model of the system performance for OFDM Broadband Wireless System (BWS). We give the detailed derivation of the system throughput expression and the system upper bound is calculated. A mathematical model of user satisfaction rate and average user rate are developed as well. The relationship of efficiency and fairness, i.e., the user satisfaction rate and throughput, is described with mathematical expressions.

In the effort of seeking new algorithms for multimedia services, we proposed 3 algorithms based on our previous work. They are Adaptive OPF (AOPF), Multimedia AOPF (MAOPF), and Normalized MAOPF (NMAOPF). Under the same traffic type with different rate, which is the simplest form of multimedia traffic, simulation results show that AOPF and MAOPF improve system throughput and delay statistics, but the drawback is poor fairness. While NMAOPF provides the highest throughput and better delay statistics, it also offers good user satisfaction rate and the highest average user rate.

To further evaluate the performance of OPF and the proposed algorithms, simulation is implemented in real multimedia traffic. The impact of user location is also evaluated. Packet drop ratio is added as another term of performance. We discuss each performance term under different traffic range and the overall performance is defined by the performance score.

Abstract

In conclusion, we have: 1) under multimedia traffic, for near group users, our proposed NMAOPF and MAOPF outperform conventional MaxSNR and RR. 2) The modified version of PF, namely OPF, performs the best for far group users.

The three proposed algorithms and the modified PF improve the efficiency without sacrificing fairness under certain conditions. They provide better tradeoff between throughput and user fairness. This conclusion is not limited to the IEEE802.16a standard. The proposed techniques can be applied to any 3G/4G OFDM based broadband wireless system.

Acknowledgements

At the outset I would like to express my most sincere gratitude to my thesis supervisor Dr. Roshdy H. M. Hafez for his constructive suggestions and guidance, as well as his kindness, patience and encouragement during the entire course of six years. His passion for research, knowledge and rich experience has greatly enhanced my enjoyment of this process.

My sincere gratitude to Dr. Yan who enlightens my life. Special thanks to my classmates in YXLST™ learning room, we are so close as sisters and brothers.

My great thanks to my parents, aunties and uncles, family members, they are wonderful and always standing besides me, helping with love and encouragements.

At last, many thanks to my colleagues and all my friends, everyone and everything, for their helps and supports in some way.

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List of Acronyms

3G	The Third Generation
3GPP	the 3rd Generation Partnership Project
4G	The Fourth Generation
ADSL	Asymmetric Digital Subscriber Lines
AMC	Adaptive Modulation and Coding
APF	Adaptive Proportional Fair
AOPF	Adaptive OFDM Proportional Fair
ARQ	Automatic Repeat reQuest
AWGN	Addictive White Gaussian Noise
BE	Best Effort
BER	Bit Error Rate
bps	Bits per second
BPSK	Binary Phase Shift Keying
BS	Base Station
BWS	Broadband Wireless System
cdf	cumulative distribution function
C/I	Carrier over Interference
CBQ	Class-Based Queuing
CBR	Constant Bit Rate

List of acronyms

CC	Convolutional Codes
CDMA	Code Division Multiple Access
CP	Cyclic Prefix
CSDPS	Channel State Dependent Packet Scheduling
CIPF-Q	Channel-condition Independent Packet Fair Queuing
CN	Core Network
DAB	Digital Audio Broadcasting
DCA	Dynamic Channel Allocation
DFT	Discrete Fourier Transform
DGPS	Dynamic Generalized Processor Sharing
DL	Down Link
DRC	Data Rate Control
DS	Direct-Sequence
DVB-T	Terrestrial Digital Video Broadcasting
EDF	Earliest Deadline First
ETSI	European Telecommunications Standards Institute
FACH	Forward Access Channel
FCS	Fast Cell Selection
FDD	Frequency Division Duplex
FER	Frame Error Rate
FT	Fourier Transform

List of acronyms

FFT	Fast Fourier Transform
FIFO	First In First Out
FTP	File Transfer Protocol
GI	Guard Interval
GPRS	General Packet Radio Service
GPS	Generalized Processor Sharing
H-ARQ	Hybrid-Automatic Repeat reQuest
HDR	High Data Rate
HDSL	High-bit-rate Digital Subscriber Lines
HDTV	High Definition Television
HS-DSCH	High Speed-Downlink Shared Channel
HSDPA	High Speed Downlink Packet Access
ICI	Inter Carrier Interference
I-CSDPS	Improved Channel State Dependent Packet Scheduling
IDFT	Inverse Discrete Fourier Transform
IEEE	Institute of Electrical and Electronics Engineers
IFFT	Inverse Fast Fourier Transform
ISI	Inter Symbol Interference
IWFQ	Idealized Wireless Fair Queuing
LOS	Line Of Sight
MAC	Medium Access Control

List of acronyms

MC	Multi_Carrier
MCS	Modulation and Coding Schemes
MIMO	Multiple-Input Multiple-Output
MS	Mobile Station
MAC	Medium Access Control
MAOPF	Multimedia Adaptive OFDM Proportional Fair
MaxC/I	Maximum Carrier over Interference
MaxSNR	Maximum Signal Noise Ratio
Mbps	Mega bits per second
MPEG	Moving Picture Experts Group
NLOS	Non Line Of Sight
NMAOPF	Normalized Multimedia Adaptive OFDM Proportional Fair
NRT	Non-Real-Time
nrtPS	non-real-time Polling Service
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
OPF	OFDM Proportional Fair
PF	Proportional Fair
PHY	Physical
PDF	probability density function
PDU	Protocol Data Units

List of acronyms

PS	Packet Scheduler
PSD	Power Spectrum Density
PSK	Phase Shift Keying
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
QoS	Quality of Service
RACH	Random Access CHannel
RF	Radio Frequency
RNC	Radio Network Controller
RR	Round Robin
RS	Reed-Solomon codes
rtPS	real-time Polling Service
SC	Single Carrier
SF	Spreading Factor
SI	Self Index
SNR	Signal to Noise Ratio
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TGPS	Truncated Generalized Processor Sharing
UE	User Equipment
UGS	Unsolicited Grant Service

List of acronyms

UL	Up Link
UMTS	Universal Mobile Telecommunications System
UTRA	Universal Terrestrial Radio Access
UTRAN	UMTS Terrestrial Radio Access Network
VBR	Variable Bit Rate
VC	Virtual Clock
VDSL	Very-high-speed Digital Subscriber Lines
VFEC	Variable Forward Error Correction
WCDMA	Wideband Code Division Multiple Access
WFQ	Weighted Fair Queuing
WWW	World Wide Web

1 Introduction

1.1 Research Motivation

The technology for the third generation (3G) and the fourth generation (4G) wireless network systems have been advancing rapidly in recent years. Orthogonal Frequency Division Multiplexing (OFDM) plays a significant role in proposed 3G/4G systems. The focus of the technology development has shifted from improving systems that provide traditional voice and other circuit-switched services to enabling high-speed wireless packet data systems. As the concept of wireless Internet and multimedia services gradually turns into reality, and with an increased demand for wireless data services, there is a need for an efficient high-speed data system. Specifically, the research effort has been mainly focused on improving the downlink capacity. Several enhancement techniques, such as Adaptive Modulation and Coding (AMC), Hybrid ARQ (H-ARQ), Fast Cell Selection (FCS), Multiple Input Multiple Output (MIMO) Antenna Processing, and Frame-by-Frame Scheduling, are proposed in order to provide very high speed downlink packet access (HSDPA) on shared channels [1]. Frame-by-frame scheduling is one of the active research areas. Thereafter, applying these technologies to OFDM based broadband wireless systems and evaluating its performance is an important research area. Selecting a simple and efficient scheduling algorithm is very important in the sense that it has considerable impact on the system performance, such as throughput and service fairness. Based on this background, our

research is focused on the studies of efficient scheduling strategies for OFDM broadband wireless systems.

1.2 Thesis Overview

Round Robin (RR) and Maximal SNR (MaxSNR) are two well-known scheduling strategies in packet data services. RR runs the scheduling in a fixed cycle. All active users are identified by their ID numbers. Each user is allocated an equal, fixed number of time slot(s) in a ring fashion. Transmission service will not be re-granted to the same user before all other users had been served. In MaxSNR scheduling method, all active users are periodically ranked by their reported SNR values. At every scheduling event, the user with the highest SNR value is first selected, allocated a number of time slots depending on the service requirement and the availability of slot resources. The same user can be re-scheduled for the next available resource providing that it still possesses the highest SNR value at the next scheduling event. It is widely accepted that RR has the advantage of being “fair” in the sense of giving equal slots to every user. It is also known that MaxSNR provides high system throughput but not fairness.

In order to quantify the performance of RR and MaxSNR within OFDM broadband wireless system under time varying and frequency selective fading channel conditions, simulation is carried out based on the system parameters of IEEE 802.16a standard.

The performance is evaluated from two aspects: efficiency and fairness. Efficiency is presented in terms of system throughput as function of the system load or number of users, mean packet delay and delay variance which are function of the throughput. Fairness is measured in terms of user satisfaction rate and average user rate which are also functions of the throughput.

The theoretical framework is developed by presenting the mathematical model of the system performance for OFDM Broadband Wireless System (BWS). We give the detailed derivation of the system throughput expression and the system upper bound is calculated. The mathematical model of user satisfaction rate and average user rate are developed as well. The relationship of efficiency and fairness, i.e., the user satisfaction rate and throughput, is described with mathematical expressions.

As expected, simulation results show that RR is very poor in system throughput and MaxSNR has a considerably higher system throughput. However, this high throughput is obtained at the expense of fairness. In wireless systems, users near the base station have statistically higher SNR values than those far from the base station. In other words, near users have higher possibilities to obtain services. The inequality between near and far users is the main disadvantage of the MaxSNR.

In order to improve the fairness in service to all users without sacrificing much in the obtainable throughput, one should incorporate both aspects in scheduling algorithm

figure of merit. This is one of the main objectives and contribution of this thesis. We try to develop a scheduling scheme that provides a trade-off between throughput and fairness. Based on previous work in this area, we focus on the Proportional Fair (PF) scheduling algorithm which was originally proposed in [2] and [3]. In this scheme, users are selected for service based on two factors: the supportable data rate at the instant of scheduling, and the average data rate that had been experienced by that user during an observation time window preceding the scheduling instant. This time span for the calculation of the average experienced rate is defined as the observation window. PF defines its figure of merit criteria as the ratio of these two factors. This algorithm and its several versions are well studied in the context of Code Division Multiple Access (CDMA) systems. As shown in the reference paper, the performance of PF is a function of the observation window.

Many efforts are made to evaluate PF in CDMA, Time Division Multiple Access (TDMA), and TDMA/CDMA hybrid systems. However, there is little work done in applying PF to OFDM systems. Since OFDM is a competitive candidate for 4G broadband wireless systems, we apply a modified version of PF to OFDM system and name it OPF (OFDM-PF). We introduce the concept of full degree of fairness from the instant of user activation to termination. Assume the user is keeping the “on” state from the moment of its activation, and also this user’s mean traffic rate does not change till its termination, so, the time span to calculate its average experienced rate is from the moment of its activation till the current time. Hence, our modification

eliminates the effect of observation window. We update the average rate of each user symbol by symbol. This is to capture the fast variation of wireless channel and hence to achieve higher accuracy. Quality of Service (QoS) requirement is also considered in the calculation of supportable data rate for each carrier of all users. All of these considerations are added up together to give better compromise between overall system throughput and fairness. Joint layers of Media Access Control (MAC) and Physical (PHY) are completely simulated. System performance is evaluated in terms of system throughput, mean packet delay, delay variance, user satisfaction rate, and average user rate.

To go a step further, we tailor OPF to fully comply with the IEEE 802.16a standard. Many works had already been done in the physical layer of 802.16a, and there already exist a lot of simulation models and results. However, in the standard, the scheduling algorithm in MAC layer is not defined and this area is open for research. We apply the OPF scheduling algorithm and evaluate its performance in IEEE 802.16a standard. Complete two-layer, MAC and PHY, simulation is implemented under the realistic urban and suburban wireless environments. Both impact of time varying and frequency selective fading is embedded in the generation of wireless channel in our simulation.

For comparison purpose, we also implement 4 algorithms in 802.16a OFDM system. Two of them are conventional algorithms: Maximal SNR and Round Robin (RR). The

others are two versions of PF: PF with observation window and weighted PF. Simulations are carried out to show the performance of OPF and the other 4 algorithms as well. To compare OPF with other algorithms, three noticeable results are obtained:

- 1). OPF is superior to RR in terms of system throughput. It increases the system throughput by 10Mbps on the average compared with RR.
- 2). OPF is superior to both RR and MaxSNR in terms of user satisfaction rate (to be defined later) as well as average user rate before the system is saturated.
- 3). OPF outperforms PF with observation window and weighted PF in terms of system throughput, average delay and average user rate. However, the performance of OPF is inferior under light traffic conditions.

Up till now, we may say that OPF gives the better compromise between throughput and fairness compared with known algorithms under heavy traffic conditions.

As multimedia service is one of the features of 3G/4G systems, we make efforts in developing new algorithms for multimedia services. We proposed 3 algorithms based on all the previous work. They are Adaptive OPF (AOPF), Multimedia AOPF (MAOPF), and Normalized MAOPF (NMAOPF). We change the simulation environment with two different rate requirements to evaluate the performance of the 3 proposed algorithms. Results show that AOPF and MAOPF improve system

throughput, average delay and delay variance, but they have worse user satisfaction rate and average user rate. The NMAOPF, however, breaks through the existing system throughput upper bound, improves delay statistics, as well as offers good user satisfaction rate and highest average user rate.

To further evaluate the performance of OPF and proposed algorithms, simulation is implemented in real multimedia traffic. The impact of user location is also evaluated. Packet drop ratio is added as another term of performance. We discuss each term under different traffic range and the overall performance is defined by performance score.

In summary, we have: 1) under multimedia traffic, for near group users, our proposed NMAOPF and MAOPF outperform conventional MaxSNR and RR. 2) The modified version of PF, namely OPF, performs the best for far group users.

All the three proposed algorithms and the modified PF improve the efficiency without sacrificing fairness under certain conditions. They provide better tradeoff between throughput and user fairness. This conclusion can be extended to any OFDM systems without the constraints of IEEE 802.16a. They can be competitive candidate for 3G/4G OFDM based broadband wireless system.

1.3 Thesis Organization

The thesis is organized as follows. In Chapter 2, background knowledge of OFDM is briefly described. In Chapter 3, we summarize the previous works in the area of scheduling algorithms and the contribution of this thesis. In Chapter 4, the overall system description is introduced through system level and link level assumptions. At the same time, we list the key parameters in IEEE 802.16a standard. We also give the definition of system performance which is used to evaluate the system. In Chapter 5, theoretical framework is developed by presenting the mathematic model of performance for OFDM system. In Chapter 6, detailed description of three scheduling algorithms and their OFDM implementation are presented. Two of them are conventional, that is, Round Robin (RR) and maximal SNR (MaxSNR). The third one is our modified version of Proportional Fair, namely OFDM Proportional Fair (OPF). In addition, two previous versions of PF are briefly described for comparison purpose. They are PF with observation window and weighted PF. In Chapter 7, we show how to implement the whole joint MAC and PHY layer simulation with OPNET as a tool. In Chapter 8, preliminary simulation results and discussions are presented. In Chapter 9, we propose three new algorithms for multimedia service and present the simulation results under simplest form of multimedia traffic. In Chapter 10, we further evaluate and compare the performance of the algorithms under real multimedia traffic, with the impact of user location. Another term of performance, that is, packet drop ratio is also introduced in this chapter and we use score method to evaluate the overall

performance considering both efficiency and fairness. Finally, in Chapter 11, we give summary and conclusions. At the same time, we list some future study areas as potential direction for our next phase of research.

2 Background of OFDM

2.1 History and basics OFDM

OFDM is a special case of multicarrier transmission, where a single datastream is transmitted over a number of lower rate subcarriers. It is worth mentioning here that OFDM can be seen either as a modulation technique or as a multiplexing technique. One of the main reasons to use OFDM is to increase the robustness against frequency selective fading or narrowband interference. In a single carrier system, a single fade of interferer can cause the entire link to fail, but in a multicarrier system, only a small percentage of the subcarriers will be affected. Error correction coding can then be used to correct for the new erroneous subcarriers. The concept of using parallel data transmission and frequency division multiplexing was published in the mid-1960s [4], [5]. Some early development is traced back to the 1950s [6]. A U.S. patent was filed and issued in January, 1970 [7].

In a classical parallel data system the total signal frequency band is divided into N nonoverlapping frequency subchannels, each subchannel is modulated with a separate symbol and then the N subchannels are frequency-multiplexed. It seems good to avoid spectral overlap of channels to eliminate interchannel interference. However, this leads to inefficient use of the available spectrum. To cope with the inefficiency, the ideas proposed from the mid-1960s were to use parallel data and FDM with

overlapping subchannels, in which each carrying a signaling rate b symbols/sec is spaced b Hz apart in frequency to avoid the use of high-speed equalization and to combat impulsive noise and multipath distortion, as well as to fully use the available bandwidth efficiently.

Figure 2-1 Concept of OFDM signal: (a) Conventional multicarrier (b) Orthogonal multicarrier

Figure 2-1 illustrates the difference between the conventional nonoverlapping multicarrier technique and the overlapping multicarrier modulation technique. As shown in the figure, by using the overlapping multicarrier modulation scheme, we save almost 50% of bandwidth. To realize the overlapping multicarrier technique,

however we need to reduce the crosstalk between subcarriers, which means that we want orthogonality between the different modulated carriers.

The word orthogonal indicates that there is a precise mathematical relationship between the frequencies of the carriers in the system. In a normal frequency-division multiplex system, many carriers are spaced apart in such a way that the signals can be received using conventional filters and demodulators. In such receivers, guard band are introduced between the different carriers and in the frequency domain which results in low spectral utilization efficiency.

Much of the research focuses on the high efficient multicarrier transmission scheme based on “orthogonal frequency” carriers. In 1971 Weinstein and Ebert [8] applied the discrete Fourier transform (DFT) to parallel data transmission systems as part of the modulation and demodulation process. Figure 2-2 (a) shows the spectrum of the individual data of the subchannel. The OFDM signal, multiplexed in the individual spectra with a frequency spacing b equal to the transmission speed of each subcarrier, is shown in Figure 2-2 (b). This figure shows that at the center frequency of each subcarrier, there is no crosstalk from the channels. Therefore, if we use DFT at the receiver and calculate the correlation values with the center of frequency of each subcarrier, we recover the transmitted data without crosstalk. In addition, using the DFT-based multicarrier technique, frequency-division multiplex is achieved not by bandpass filtering but by baseband processing.

(a)

(b)

Figure 2-2 Spectra of (a) OFDM subchannel; (b) OFDM signal

In the 1960s, the OFDM technique was used in several high-frequency military systems such as KINEPLEX [6] and ANDEFT [9]. In the 1980s, OFDM was studied for high-speed modems, digital mobile communications, and high-density recording as in [10]. In the 1990s OFDM was exploited for wideband data communications over mobile radio FM channels, high-bit-rate digital subscriber lines (HDSL; 1.6Mbps),

asymmetric digital subscriber lines, (ADSL; up to 6 Mbps), very-high-speed digital subscriber lines, (VDSL; 100 Mbps), digital audio broadcasting (DAB), and high-definition television (HDTV) terrestrial broadcasting [11], [12], [13], [14], [15].

The OFDM transmission scheme has the following key advantages:

- OFDM is an efficient way to deal with multipath; for a given delay spread, the implementation complexity is significantly lower than that of a single carrier system with an equalizer.
- In relatively slow time-varying channels, it is possible to significantly enhance the capacity by adopting the data rate per subcarrier according to the signal-to-noise ratio of these particular subcarriers.
- OFDM is robust against narrowband interference, because such interference affects only a small percentage of the subcarriers.
- OFDM makes single-frequency networks possible, which is especially attractive for broadcasting applications.

On the other hand, OFDM also has some drawbacks compared with single-carrier modulation:

- OFDM is more sensitive to frequency offset and phase noise.
- OFDM has a relatively large peak-to-average power ratio, which tends to reduce the power efficiency of the RF amplifier.

2.2 Generation of Subcarriers using the IFFT

An OFDM signal consists of a sum of subcarriers that are modulated by using phase shift keying (PSK) or quadrature amplitude modulation (QAM). If d_i are the complex QAM symbols, N_s is the number of subcarriers, T the symbol duration, and f_c the carrier frequency, then one OFDM symbol starting at $t=t_s$ can be written as:

$$s(t) = \begin{cases} \operatorname{Re} \left\{ \sum_{i=-\frac{N_s}{2}}^{\frac{N_s}{2}-1} d_{i+N_s/2} \exp \left(j2\pi \left(f_c - \frac{i+0.5}{T} \right) (t-t_s) \right) \right\}, & t_s \leq t \leq t_s + T \\ s(t) = 0 & t < t_s \wedge t > t_s + T \end{cases} \quad \text{Equation 2-1}$$

In the literature, often the equivalent complex baseband notation is used, which is given by:

$$s(t) = \begin{cases} \operatorname{Re} \left\{ \sum_{i=-\frac{N_s}{2}}^{\frac{N_s}{2}-1} d_{i+N_s/2} \exp \left(j2\pi \frac{i}{T} (t-t_s) \right) \right\}, & t_s \leq t \leq t_s + T \\ s(t) = 0 & t < t_s \wedge t > t_s + T \end{cases} \quad \text{Equation 2-2}$$

In this representation, the real and imaginary parts correspond to the in-phase and quadrature parts of the OFDM signal, which have to be multiplied by a cosine and sine of the desired carrier frequency to produce the final OFDM signal. Figure 2-3 shows the operation of the modulator in a block diagram.

Figure 2-3 OFDM modulator

Figure 2-4 Four subcarriers of one OFDM signal

As an example, Figure 2-4 shows four subcarriers from one OFDM signal. In the example, all subcarriers have the same phase and amplitude, but in practice the amplitudes and phases may be modulated differently for each subcarrier. Note that each subcarrier has exactly an integer number of cycles in the interval T , and the number of cycles between adjacent subcarriers differs by exactly one. This property accounts for the orthogonality between the subcarriers. For instance, if the j th subcarrier from equation 2-2 is demodulated by down converting the signal with a frequency of j/T and then integrating the signal over T seconds, the result is as written in:

$$\begin{aligned}
 & \int_{t_s}^{t_s+T} \exp\left[-j2\pi \frac{j}{T}(t-t_s)\right] \sum_{i=-\frac{N_s}{2}}^{\frac{N_s}{2}-1} d_{i+N_s/2} \exp\left[j2\pi \frac{i}{T}(t-t_s)\right] dt \\
 &= \sum_{i=-\frac{N_s}{2}}^{\frac{N_s}{2}-1} d_{i+N_s/2} \int_{t_s}^{t_s+T} \exp\left[j2\pi \frac{i-j}{T}(t-t_s)\right] dt \\
 &= d_{j+N_s/2} T
 \end{aligned}
 \tag{Equation 2-3}$$

By looking at the intermediate result, it can be seen that a complex carrier is integrated over T seconds. For the demodulated subcarrier j , this integration gives the desired output $d_{j+N/2}$ (multiplied by a constant factor T), which is the QAM value for that particular subcarrier. For all other subcarriers, the integration is zero, because the frequency difference $(i-j)/T$ produces an integer number of cycles within the integration interval T , such that the integration results is always zero.

The orthogonality of the different OFDM subcarriers can also be demonstrated in another way. According to equation 2-1, each OFDM symbol contains subcarriers that are nonzero over a T -second interval. Hence, the spectrum of a single symbol is a convolution of a group of Dirac pulses located at the subcarrier frequencies with the spectrum of a square pulse is equal to $\text{sinc}(\pi fT)$, which has zeros for all frequencies f that are an integer multiple of $1/T$. This effect is shown in Figure 2-5 which shows the overlapping *sinc* spectra of individual subcarriers. At the maximum of each subcarrier spectrum, all other subcarrier spectra are zero. Because an OFDM receiver essentially calculates the spectrum values at those points that correspond to the maxima of individual subcarriers, it can demodulate each subcarrier free from any interference from the other subcarriers. Basically Figure 2-5 shows that the OFDM spectrum fulfills Nyquist's criterion for an inter-symbol interference free pulse shape. Notice that the pulse shape is present in the frequency domain and not in the time domain, for which the Nyquist criterion usually is applied. Therefore, instead of inter-symbol interference (ISI), it is inter-carrier interference (ICI) that is avoided by having the maximum of one subcarrier spectrum correspond to zero crossings of all the others.

Figure 2-5 Spectra of individual subcarriers

The complex baseband OFDM signal as defined by equation 2-2 is in fact nothing more than the inverse Fourier transform of N_s QAM input symbols. The time discrete equivalent is the inverse discrete Fourier transform (IDFT), which is given by equation 2-4, where the time t is replaced by a sample number n .

$$s(n) = \sum_{i=0}^{N_s-1} d_i \exp\left(j2\pi \frac{in}{N_s}\right) \quad \text{Equation 2-4}$$

In practice, this transform can be implemented very efficiently by the inverse fast Fourier transform (IFFT). An N point IDFT requires a total of N^2 complex multiplications which are actually only phase rotations. Of course, there are also additions necessary to do an IDFT, but since the hardware complexity of an adder is significantly lower than that of a multiplier or phase rotator, only the multiplications are used here for comparison. The IFFT drastically reduces the amount of calculations by exploiting the regularity of the operations in the IDFT. Using the radix-2 algorithm, an N -point IFFT requires only $(N/2) \cdot \log_2(N)$ complex multiplications [16]. For a 16-point transform, for instance, the difference is 256 multiplications for the IDFT versus 32 for the IFFT, a reduction by a factor of 8! This difference grows for larger numbers of subcarriers, as the IDFT complexity grows quadratically with N , while the IFFT complexity only grows slightly faster than linear.

2.3 Guard time and cyclic extension

One of the most important reasons to do OFDM is the efficient way it deals with multipath delay spread. By dividing the input data stream in N_s subcarriers, the symbol duration is made N_s times smaller, which also reduces the relative multipath delay spread, relative to the symbol time, by the same factor. To eliminate intersymbol interference almost completely, a guard time is introduced for each OFDM symbol. The guard time is chosen larger than the expected delay spread, such that multipath components from one symbol cannot interfere with the next symbol.

The guard time could consist of no signal at all. In that case, however, the problem of intercarrier interference (ICI) would arise. ICI is crosstalk between different subcarriers, which means they are no longer orthogonal. This effect is illustrated in Figure 2-6.

Figure 2-6 Effect of multipath with zero signal in the guard time

In this example, a subcarrier 1 and a delayed subcarrier 2 are shown. When an OFDM receiver tries to demodulate the first subcarrier, it will encounter some interference from the second subcarrier, because within the FFT interval, there is no integer number of cycles difference between subcarrier 1 and 2. At the same time there will be crosstalk from the first to the second subcarrier for the same reason.

To eliminate ICI, the OFDM symbol is cyclically extended in the guard time, as shown in Figure 2-7. This ensures that delayed replicas of the OFDM symbol always have an integer number of cycles within the FFT interval, as long as the delay is smaller than the guard time. As a result, multipath signals with delays smaller than the guard time cannot cause ICI.

Figure 2-7 OFDM symbol with cyclic extension

As an example of how multipath affects OFDM, Figure 2-8 shows received signals for a two-ray channel, where the dotted curve is a delayed replica of the solid curve. Three separate subcarriers are shown during three symbol intervals. In reality, an

OFDM receiver only sees the sum of all these signals, but showing the separate components makes it more clear what the effect of multipath is. From the figure, we can see that the OFDM subcarriers are BPSK modulated, which means that there can be 180 degree phase jumps at the symbol boundaries. For the dotted curve, these phase jumps occur at a certain delay after the first path. In this particular example this multipath delay is smaller than the guard time, which means there are no phase transitions during the FFT interval. Hence, an OFDM receiver “sees” the sum of pure sine waves with some phase offsets. This summation does not destroy the orthogonality between the subcarriers, it only introduces a different phase shift for each subcarrier. The orthogonality does become lost if the multipath delay becomes larger than the guard time. In that case, the phase transitions of the delayed path fall within the FFT interval of the receiver. The summation of the sine waves of the first path with the phase modulated waves of the delayed path no longer gives a set of orthogonal pure sine waves, resulting in a certain level of interference.

Figure 2-8 An OFDM signal with three subcarriers in a two-ray multipath channel

2.4 Choice of OFDM parameters

The choice of various OFDM parameters is a tradeoff between various, often conflicting requirements. Usually, there are three main requirements to start with: bandwidth, bit rate, and delay spread. The delay spread directly dictates the guard time. As a rule, the guard time should be about two to four times the root-mean-squared delay spread. This value depends on the type of coding and QAM modulation. Higher order QAM is more sensitive to ICI and ISI than QPSK, while heavier coding obviously reduces the sensitivity to such interference.

Now that the guard time has been set, the symbol duration can be fixed. To minimize the signal-to-noise ratio (SNR) loss caused by the guard time, it is desirable to have the symbol duration much larger than the guard time. It cannot be arbitrarily large, however, because larger symbol duration means more subcarriers with a smaller subcarrier spacing, a larger implementation complexity, and more sensitivity to phase noise and frequency offset [17], as well as an increased peak-to-average power ratio [18]. Hence, a practical design choice is to make the symbol duration at least five times the guard time, which implies a 1-dB SNR loss because of the guard time.

After symbol duration and guard time are fixed, the number of subcarriers follows directly as the required -3dB bandwidth divided by the subcarrier spacing, which is the inverse of the symbol duration less the guard time. Alternatively, the number of

subcarriers may be determined by the required bit rate divided by the bit rate per subcarrier. The bit rate per subcarrier is defined by the modulation type, coding rate and symbol rate.

An additional requirement that can affect the chosen parameters is the demand for an integer number of samples both within the FFT/IFFT interval and in the symbol interval.

2.5 OFDM signal processing

The previous sections described how the basic OFDM signals are formed using the IFFT and adding a cyclic extension. However, there is more to it to build a complete OFDM modem. Figure 2-9 shows the block diagram of an OFDM modem, where the upper path is the transmitter chain and the lower path corresponds to the receiver chain. In the center we see the IFFT, which modulates a block of input QAM values onto a number of subcarriers. In the receiver, the subcarriers are demodulated by an FFT, which performs the reverse operation of an IFFT. An interesting feature of the FFT/IFFT is that the FFT is almost identical to an IFFT. In fact, an IFFT can be made using an FFT by conjugating input and output of the FFT and dividing the output by the FFT size. This makes it possible to use the same hardware for both transmitter and receiver. Of course, this saving in complexity is only possible when the modem does not have to transmit and receive simultaneously.

Figure 2-9 OFDM signal processing

The functions before the IFFT have not been discussed till now. Binary input data is first encoded by a forward error correction code. The encoded data is then interleaved and mapped onto QAM values. In the receiver path, after passing the RF part and the analog-to-digital conversion, the digital signal processing starts with a training phase to determine symbol timing and frequency offset. And FFT is used to demodulate all subcarriers. The output of the FFT contains N_s QAM values, which are mapped onto binary values and decoded to produce binary output data. To successfully map the QAM values onto binary values, first the reference phases and amplitudes of all subcarriers have to be acquired. Alternatively, differential techniques can be applied.

3 Literature research in scheduling algorithms

3.1 Importance of the Role of Scheduling Algorithm

Making efficient use of the radio resources is becoming a very challenging task for 4G wireless communication systems. The scarcity of radio resources, diverse Quality of Service (QoS) requirements and time-varying channel conditions complicate the scheduling and radio resource management. Scheduling algorithms play an important role in resource management. The main function of the scheduler is to intelligently allocate radio resources to individual users in order to achieve high system performance. The scheduling algorithm is a key component in optimizing the system across the physical and MAC layers. In terms of scheduling, OFDM-based systems have two degrees of freedom: time and frequency. In [19] and [20] the authors do not describe specific scheduling algorithms in detail, instead they describe the architecture of the system and available information that enables diverse scheduling algorithms.

The scheduler uses information from the physical layer and other sources to the network layer to make decisions. The information set is rich which includes user profile, wireless channel conditions as well as radio resource. In user profile, there are traffic model for different user groups, QoS requirements, data rate requirements of different users, user locations in the cell and user mobility. At the same time, the scheduler has to take into account of the wireless channel condition, such as available

bandwidth; large scale propagation including path loss and shadowing model; small scale channel fading of time varying and frequency selective fading; antenna model; as well as coding and modulation schemes. In radio resource, there are also other requirements to consider, such as power constraints, frame structure and so on. As shown in Figure 3-1, the scheduler is a central place in which the performance objectives of both the MAC layer and physical layer are jointly optimized.

Role of Scheduling in BWS

Figure 3-1 Role of scheduler

3.2 Overview of Scheduling Algorithm in Broadband Wireless Systems

The coming 4G wireless broadband network will be an integral part of the global communications infrastructure. With the fast growth of wireless data service and the increasing demand for multimedia applications, future wireless networks will provide services for heterogeneous classes of traffic with different QoS requirements. Scheduling algorithms provide mechanisms for bandwidth allocation and multiplexing at the packet level. Compared with wire line networks, wireless communication poses special problems such as high error rate and bursty errors, location-dependent and time-varying link capacity, scarce bandwidth, user mobility and power constraint. All of these characteristics make developing efficient and effective scheduling algorithms for wireless networks very challenging. As stated in [21], major issues in wireless scheduling are: 1) wireless link variability; 2) fairness; 3) QoS; 4) data throughput and channel utilization; 5) power constraint and simplicity. To fulfill these special requirements of wireless link, some initial efforts in scheduling algorithms are made. For example, [22] proposed Channel State Dependent Packet Scheduling (CSDPS). It was the first paper that addressed the problem of location-dependency and bursty errors, while it did not propose any mechanism to guarantee bandwidth for each mobile station. In [23] the CSDPS algorithm was modified through Class-Based Queuing (CBQ) in order to address the fairness issue in sharing the wireless channel. Other algorithms that are focused on different level of fairness are: Weighted Fair Queuing (WFQ) in [24], Idealized Wireless Fair Queuing (IWFQ) in [25], Channel-

condition Independent Packet Fair Queuing (CIPF-Q) in [26], and Improved Channel State Dependent Packet Scheduling (I-CSDPS) in [27]. These early contributions all tried to achieve fairness while maintaining high throughput. However, these algorithms are suitable only for simple wireless communication systems such as the typical TDMA systems. As time passes, more sophisticated networks, namely CDMA, OFDMA, or hybrid networks are employed with an ever increasing speed in modern communication systems. They aim at providing multimedia services, faster data rate with wideband access. They also put strict requirements on scheduling algorithms which can achieve the optimal compromise between throughput and fairness.

3.3 Scheduling Algorithms Based on Trading-off Throughput and Fairness

Round Robin (RR) and Maximal SNR (MaxSNR) are two well-known scheduling strategies in packet data services. RR runs the scheduling in a fixed cyclic order. All active users are identified by their ID numbers. Each user is allocated time-frequency resources in a ring fashion. Transmission service will not be re-granted to the same user before all other users have been served. MaxSNR scheduler ranks all active users periodically by their reported SNR values. At every scheduling event, the user with the highest SNR value is selected and allocated proper time-frequency resources, depending on the service requirement and the availability of resources. It is possible

for the same user to be re-scheduled for the next available resource provided that it still possesses the highest SNR value at the next scheduling event. It's been widely accepted that RR has the advantage of being "fair" in the sense of giving equal chances to every individual user. It is also known that MaxSNR provides high channel throughput. However, achieving fairness and high throughput are contradictory requirements. Trade-off between them has to be reached according to system requirements. Many efforts are made addressing this issue, such as Proportional Fair (PF). Here we list the previous works in scheduling methods from 2000 to 2004 addressing the compromise between throughput and fairness.

In [2] and [3], a proportional fair (PF) scheduling algorithm was proposed. The scheduler selects the user with a maximal priority metric defined as:

$$j = \arg \max_{1 \leq i \leq N} D_i(t) / R_i(t-1) \quad \text{Equation 3-1}$$

where i is the user index, and j is the selected user, $D_i(t)$ is the current supportable data rate of user i by the channel, and $R_i(t)$ is the average data rate of user i at time t :

$$R_i(t) = \begin{cases} 0 & i \neq j \\ (1 - \frac{1}{T_c})R_j(t-1) + \frac{1}{T_c} D_j(t) & i = j \end{cases} \quad \text{where } T_c > 1 \quad \text{Equation 3-2}$$

where $R_i(t-1)$ is the average data rate of user i at time " $t-1$ ". T_c is an observation window. With this algorithm, the scheduler selects the user with the highest priority matrix to receive the next resource. Notice that good channel means higher

numerator, namely $D_i(t)$, and bad channel means smaller $D_i(t)$. A user is selected when it has a good channel so that high system throughput is reached. In the meantime, users in bad channel are also considered since their low average rate makes the denominator $R_i(t)$ small, thus the ratio of $D_i(t)/R_i(t)$ will increase. Hence their chance of being selected for the next scheduling event improves as their average data rate drops. The performance of this algorithm is a function of the observation window T_c . When this observation window is small, only short term fairness can be reached. This scheduler is implemented in High Data Rate (HDR) networks of CDMA2000-1X. It is recently studied by many authors as those of [28], [29], and [30]. Their research showed unevenness in the data rates achieved by different users when they experience different channel conditions in terms of the corresponding fading properties.

Another version of PF is the modified PF algorithm with Data Rate Control (DRC) exponent rule in [31]. In this paper, the selection criterion is modified to:

$$j = \arg \max_{1 \leq i \leq N} \frac{D_i^n}{R_i} \quad , \quad \text{Equation 3-3}$$

where n is the weighted parameter introduced to manage the relationship between the data rates of the users with different channel conditions. However, the control parameter n is a constant for all users, and two problems exist in this approach: 1) the control parameter is fixed in time, it does not adapt to the current radio conditions of

each user; 2) n is set to the same value for all users. It is not possible to choose a value for n that ensures fairness among all users at the same time.

In 2004, an Adaptive Proportional Fairness (APF) scheduling algorithm is proposed in [32]. It takes a further step in the user selection criterion as:

$$j = \arg \max_{1 \leq i \leq N} \frac{D_i^{c_i}}{R_i}, \quad \text{Equation 3-4}$$

where c_i is the control parameter corresponding to user i . To be able to track the fast variations of the channels and the difference between the channels for different users, the APF algorithm goes a step further to update the value of c_i according to this formula:

$$\begin{cases} c_i = c_i + \Delta c & \text{if } \left(\frac{R_i}{r_i} - \frac{1}{N} \sum_{j=1}^N \frac{R_j}{r_j} \right) < -\varepsilon \\ c_i = c_i - \Delta c & \text{if } \left(\frac{R_i}{r_i} - \frac{1}{N} \sum_{j=1}^N \frac{R_j}{r_j} \right) > \varepsilon \end{cases}, \quad \text{Equation 3-5}$$

where r_i is the instantaneous data rate of the user equipment i can support under its current channel conditions, and $\bar{r}_i$ is the average channel supportable data rate of user i . c_i measures the difference between the proportional data rate allocated to user i which is $\frac{R_i}{r_i}$, and the average value over all users which is $\frac{1}{N} \sum_{j=1}^N \frac{R_j}{r_j}$. If this

difference is not within acceptable values defined by the interval $[-\varepsilon, \varepsilon]$, c_i is updated

by adding or subtracting of Δc . The choice of Δc depends on the speed of convergence sought and the desired strength of variations around the value at convergence.

In the current year of 2005, the authors of [33] proposed a bin packing algorithm to assure fairness and a layered prefetching algorithm to efficiently utilize the available link capacity for video traffic. This paper tries to provide tradeoff through software implementation combining fair scheme and efficient scheme instead of considering both fairness and efficiency in the same algorithm as [2], [3], [31] and [32].

All of the above efforts are made in seeking the optimal compromise between throughput and fairness. Compared with the original ideas in fair scheduling in wireless broadband algorithms before 2000 which are applied in simple TDMA systems, those algorithms are fitted for and implemented in more sophisticated modern wireless communication systems such as CDMA, WCDMA and TDMA/CDMA hybrid networks. PF is well discussed in the forum of High Speed Downlink Packet Access (HSDPA).

3.4 Fast-growing Deployment of OFDM in BWS

Another feature of the wireless channel is that the received signal consists of a number of time-varying multipath components. Time varying introduces slow or fast fading.

Multipath with different path delays causes frequency selective fading. Orthogonal Frequency Division Multiplexing (OFDM) is well known for its flexibility and good performance in combating the frequency selective fading. It eliminates the Inter Symbol Interference (ISI) by inserting a Cyclic Prefix (CP) longer than the delay spread of the channel. At the same time, due to its feature of multi-carrier, it can fulfill the ever-increasing demand of high data rate communication as well as the multi-media requirements. OFDM is a competitive candidate for the 3rd generation (3G) and 4G broadband wireless communication systems [34], [35], [36]. This technique has been successfully applied in the ADSL applications, Digital Audio Broadcasting (DAB), Terrestrial Digital Video Broadcasting (DVB-T) system, HIPERLAN/2, and satellite communications. Recently, there has been a growing interest in applying this technique to the broadband multiple access system [37], [38]. It is also adopted by IEEE standards such as 802.11a, 802.11g and 802.16a.

3.5 Compromise between Throughput and Fairness in OFDM systems

There was few literatures considering fair scheduling in the OFDM systems before 2001. In early works such as [39], a modified fair queuing algorithm is presented to give priority to the terminals that have acquired lowest data rate since they became active. This centralized scheduling scheme is implemented in the DVB_T system. In this system, groups of transmitters are sending the same information simultaneously over the same frequency channel. Only a handful of papers have been found in

seeking the optimal compromise between throughput and fairness in OFDM systems.

We list these papers as follows:

In 2003, the author of [40] proposed the Truncated Generalized Processor Sharing (TGPS) scheduling for OFDM system. It took a step further based on GPS. The basic principle of ideal GPS is to assign each user a fixed weight, instead of a fixed bandwidth. It dynamically allocates bandwidth to all the users according to their weights and traffic load. The newly proposed TGPS allocated the available subcarriers to each user by:

$$M_i^k = \left\lfloor \frac{\phi_i}{\sum_{j \in B} \phi_j} M_{effective}^k \right\rfloor, \quad \text{Equation 3-6}$$

where $M_{effective}^k$ is the number of effective subcarriers at any instant k in the busy period, B is the set of backlogged users, ϕ_j is the predefined weight for user j by GPS.

In order to minimize the difference of this TGPS and ideal GPS, a compensation algorithm is introduced to allocate the unallocated subcarriers to the users with maximum accumulated quantization errors. Compared with GPS, the proposed TGPS scheduler can guarantee capacity allocation to all backlogged users at any instant according to the predefined weight. However, how the weights are defined for each user is not discussed.

Based on the TGPS, the author proposed Dynamic GPS (DGPS) in 2004 in [41]. The scheduling scheme adapts the scheduling weights as:

$$\eta_{i,k}^d = \eta_i \left(1 + \frac{(\eta_{i,k}^w - \eta_i) \alpha}{\eta_i} \right) = (1 - \alpha) \eta_i + \alpha \eta_{i,k}^w, \quad \text{Equation 3-7}$$

where η_i is the GPS weight for user i , $\eta_{i,k}^w$ are the weights to achieve minimum power consumptions, $\eta_{i,k}^d$ are the dynamic weights at the k th symbol interval, α is a constant. Actually, this α represents the tradeoff between fairness and traffic load. Like the previous paper, how α is defined is not given in this paper. Its simulation results show that the performance is the function of the preset α , which is a constant between 0 and 1.

In [42], the virtual clock (VC) discipline is adopted as the scheduling scheme in OFDM system. This algorithm aims to imitate the TDM system while maintaining statistical multiplexing flexibility. The basic idea of VC is to assign each data flow a virtual clock, which ticks at every packet arrival from that flow. The tick step is equal to the average inter-packet interval. The packets are stamped by the flow's virtual clock and are transmitted according to the increasing order of the virtual timestamp values. It is quite similar to the previously mentioned IWFQ in [25].

Paper [43] is the one discussing the PF scheduling algorithm in multi-user OFDM system which we are interested in. This paper focuses on the proposed of a scheduling

interval that balances achievable gains with the amount of feedback. It also emphasizes adaptive modulation and coding method as well as MIMO configurations. Only two short paragraphs point out the PF as an additional gain to the system, with MaxSNR as upper bound and RR as lower bound. No detailed impact of PF to the system is given.

Paper [44], 3 modified PF schedulers are elaborated and fit into the OFDM system. The criterion is exactly the same as the formula in [2] and [3]. All of the three schemes could be regarded as frequency domain scheduler, which divided the subcarriers into several sub-bands. Each sub-band is scheduled according to the PF criteria. The difference among the 3 schemes lies in how the average rate is updated. The first two schemes update the average rate subband by subband while the 3rd doing the update when all subbands are finished. All the proposed 3 scheduling algorithms could be regarded as frequency domain scheduler. There exist some problems. From the frequency point of view, depends on the total bandwidth and FFT size of the OFDM system plus multi-path effect, the correlation function of subcarriers is hard to predict. When several subcarriers are grouped together and assigned to the same scheduler, it is under the assumption that those subcarriers are highly correlated. It may not be true for frequency selective fading channel. The scheme is not optimal because some information in the frequency domain is lost. From the time domain, during the allocation of subbands in the same slot, the average rate is supposed to stay

constant. This is not true for fast fading wireless channels, and the scheme is not optimal because some information is lost in rate updating.

3.6 Scheduling Algorithms for IEEE 802.16 Standard

We further narrow down the literature research to scheduling algorithms for IEEE 802 standards. There are many papers for 802.11 families. But there is little record for 802.16 since the 802.16a was approved recently in June of 2004. In the standard, MAC scheduling algorithms is not defined and open for research. The only papers we found on MAC scheduling scheme for 802.16a are [45] and [46] by the same author. In the paper, the packet scheduling for QoS in uplink is described. First, service types such as Unsolicited Grant Service (UGS), real-time Polling Service (rtPS), non-real-time Polling Service (nrtPS), Best Effort service (BE) is ranked by their priority and strict priority discipline is employed for bandwidth allocation among these services. Second, within each service flow, different algorithm is applied: fixed bandwidth allocation for UGS, Earliest Deadline First (EDF) for rtPS, Weight Fair Queue (WFQ) for nrtPS and equal bandwidth allocation for BE. In summary, it is a combination of strict priority service, EDF and WFQ.

3.7 Thesis Contributions

Based on the literature research, we list the main contributions of this thesis:

1. We implement the existing algorithms, namely RR, MaxSNR, PF, Weighted PF in OFDM broadband wireless system.
2. For the PF algorithm, its performance depends on the observation window. We modify PF algorithm and name it OPF, the impact of observation window is eliminated. Hence the calculation complexity of the algorithm is greatly reduced. In other words, the efficiency of the scheduler itself is largely improved.
3. We apply OPF in broadband OFDM systems. Compared with [44], instead of a frequency domain scheduler, we apply OPF frame by frame in the OFDM system as time domain scheduler. We update the experienced average rate of each user symbol by symbol, to capture the fast variation of channel conditions and provide higher accuracy. At the same time, Adaptive Modulation and Coding (AMC) scheme is applied to each sub-carrier, while QoS requirement is considered as well in the calculation of supportable rate for each carrier. All of these considerations add up to give better compromise between overall system throughput and fairness. Both the MAC and the PHY layers are completely simulated and system performance is shown in terms of system throughput, delay mean and variance, user satisfaction rate, and average user rate.
4. We tailor OPF to fit into IEEE 802.16a standard under realistic urban and suburban environment with both time varying and frequency selective fading channel. Scheduling algorithm in 802.16a MAC layer is open for research and there is little literature in this area. As far as we know, PF is applied to

CDMA/WCDMA systems. Now we implement and evaluate the modified PF over 802.16a OFDM system, hence the name OPF.

5. In the effort of seeking new algorithms for multimedia services, we proposed 3 algorithms. They are Adaptive OPF (AOPF), Multimedia AOPF (MAOPF), and Normalized MAOPF (NMAOPF). As all existing algorithms, they are implemented in the OFDM system of IEEE 802.16a. Simulation is done under different rate requirements, and performance of the proposed algorithms is compared with known and modified algorithms. We conclude that NMAOPF can be competitive candidate for 3G/4G OFDM based broadband wireless system with multimedia services.

4 System Description

In this section, we will give an overall description of the OFDM broadband wireless system. We first briefly introduce the IEEE 802.16a standard. This is the environment we are going to implement all the scheduling algorithms. Then, both detailed PHY layer and OFDM system structure will be presented. Based on the system structure, system scenario will be described. Finally, we give the definition of system performance which will be used in the following sections to evaluate the system.

4.1 IEEE 802.16a requirements

We are going to implement the scheduling algorithms as MAC scheduling scheme in the broadband IEEE 802.16a standard. First of all, we will extract all the related requirements from the standard [47], [48] and [49]. Table 4-1 shows the carrier frequency range, bandwidth and OFDM parameters of the IEEE 802.16a. We will tailor the scheduling algorithms and implement simulations based on these requirements. Every reference is also listed in the table for reader's convenience and for verification purpose.

Table 4-1 Assumption of IEEE 802.16a standard (OFDM) mode

	Assumption	Reference and remarks
Frequency	2.500-2.520	Table 189, IEEE P802.16a/D3-2002, p.108 of ref. [48].

Band(GHz)		
Bandwidth (MHz)	20	The IEEE 802.16 WirelessMAN TM Standard for Broadband Wireless Metropolitan Area Networks [50].
N of carriers	256	Table 228, IEEE P802.16a/D3-2002, p. 186 of ref. [48].
N_{used}	200	Table 228, IEEE P802.16a/D3-2002, p. 186 of ref. [48].
Ratio of CP	25%	P802.16a/D3-2002, p.165 of ref. [48].
Symbol duration (μ s)	16 μ s $T_{FFT}=12.8\mu$ s, $T_{GI}=3.2\mu$ s,	In [51], in mobile cellular environment, (macro cells, radius of 1-20km), the value of τ_{RMS} falls between 0.1 and 10 μ s.
Frame length	2ms	We choose the same frame length as stated in ref. [52].
Physical time slot (μ s)	64 μ s (4 symbol)	P802.16a/D3-2002, p.221 of ref. [48].
Overhead symbol number	25% of total number of symbols	Overhead includes: Preamble, FCH (LD Fr Prefix, DCD, UCD, MAPs, padding) and other headers. Figure 196, P802.16a/D3-2002, p.179 of ref. [48].
Coding	RS, CC code rate: $\frac{1}{2}, \frac{3}{4}, \frac{2}{3}$	Table 229, P802.16a/D3-2002, p.187 of ref. [48].
modulation	QPSK,	Table 229, P802.16a/D3-2002, p.187 of ref. [48].

	16 QAM, 64QAM	
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Detailed modulation and coding combination adopted by 802.16a are listed in Table 4-2.

Table 4-2 Channel coding per Modulation

Modulation	Uncoded block size (Bytes)	Coded block size (bytes)	Overall coding rate
QPSK	24	48	$\frac{1}{2}$
QPSK	36	48	$\frac{3}{4}$
16 QAM	48	96	$\frac{1}{2}$
16 QAM	72	96	$\frac{3}{4}$
64 QAM	96	144	$\frac{2}{3}$
64 QAM	108	144	$\frac{3}{4}$

4.2 Link level description

Besides the 802.16a requirements, more assumption should be made in link level. We summarized these assumptions in Table 4-3. Detailed description for each issue will follow.

Table 4-3 Link level assumption for urban scenario in wireless environment

	Assumptions	Reference and remarks
Number of cells/link type	Single cell, downlink	
Cell radius	1.5 km	
Path loss model	<p>1. $L=40\log_{10}R+30\log_{10}f+49(\text{dB})$</p> <p>2.</p> <p>$L = 40(1 - 4 \times 10^{-3} \Delta h_b) - 18\log_{10} \Delta h_b + 21\log_{10} f + 80[\text{dB}]$</p> <p>(both already include shadowing, NLOS assumed)</p>	<p>1. Outdoor-to-indoor and pedestrian, R in kilometers, f is the carrier frequency around 2GHz (in GHz)</p> <p>2. for urban and suburban areas outside the high-rise core, BS antenna height of 15m above average rooftop, R in kilometers.</p> <p>Both from [53], p.618-619.</p>
Shadowing model	<p>Lognormal distribution, with Standard deviation = 10 dB (This is included in the 49dB in Path Loss Model (1))</p>	<p>Proakis' book [54], p. 813</p> <p>Michel C. Jeruchim, Philip Balaban, and K. Sam Shanmugan, "Simulation of Communication Systems-Modeling, Methodology, and Techniques", 2nd edition, Kluwer Academic/Plenum</p>

	and in the 80dB in the Path Loss Model (2) above.)	Publishers, p578. p.546 shows the relationship of shadowing and multipath fading.
Channel model	Multi-path; each multipath component is Rayleigh distribution (802.16a assumes NLOS)	IEEE standard 802.16: A technical overview of the wirelessMAN™ air interface for broadband wireless access <u>Eklund, Carl</u> (Nokia Research Center); <u>Marks, Roger B.</u> ; <u>Stanwood, Kenneth L.</u> Source: IEEE Communications Magazine, v 40, n 6, June, 2002, p 97-107
Time varying model	Jake's model	[55], p. 67.
MS distribution	Uniform distribution within the cell radius.	[75]
Noise Density	-174dBm/Hz	T1E1.4 T1E1.4/2001-002, "Draft proposed American National Standard, Spectrum Management for Loop Transmission Systems, Issue 2", Project: T1E1-38, Spectral

		Compatibility Aspects for Facilities between a Central Office and the Network-to-Customer Interface, p. 53.
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4.2.1 Cell topology

We are going to set up a single cell wireless communication system. As in Figure 4-1, one Base Station (BS) is located at the center. N number of users or Mobile Stations (MS) are uniformly distributed within the range of the cell radius. Only downlink (from BS to MS) is simulated. Cell radius is configurable according to requirement.

Figure 4-1 Down link of multi-user single cell

4.2.2 Antenna model

Antenna model is assumed to be omni-directional antenna. Antenna height will be described in the next section together with the propagation model. More antenna patterns may be considered in the future.

4.2.3 Large scale fading—propagation model

For testing purpose, we implement three different large scale path loss models. They are:

- Ideal channel: no path loss, with frequency response equals to 1.
- free space model in [56]:

$$L_s = 20 \log_{10} \left(\frac{4\pi d}{\lambda} \right) \quad \text{Equation 4-1}$$

where L_s is path loss in dB, λ is the wavelength of the carrier in meters and d is the distance between BS and MS in meters.

- Vehicular test environment: this model is applicable for scenarios in urban and suburban areas outside the high-rise core, where structures are of nearly uniform height. This is a typical environment for high speed data packet switch broadband wireless system such as 802.16a:

$$L_s = 40(1 - 4 \times 10^{-3} \Delta h_b) \log_{10} R - 18 \log_{10} \Delta h_b + 21 \log_{10} f + 80 [dB] \quad \text{Equation 4-2}$$

where L_s is path loss in dB, R is the distance between BS and MS in kilometers, f is the carrier frequency in MHz and Δh_b is the base station antenna height in meters, measured from the average rooftop level. This formula is obtained from [53], p. 618-619. Please note that:

- The path loss model is valid for a range of Δh_b from 0 to 50m, and we set it to 15m in our simulation;
- L_s shall in no circumstances be less than the free-space loss. This model is valid for the non-line-of-sight case only and describes worst-case propagation. Log-normal shadow fading with a standard deviation of 10 dB is assumed in both urban and suburban areas.

4.2.4 Large scale fading – shadowing model

Log-Normal shadowing is chosen as the large scale fading model as used in [53]. At any instant, the shadowing process is modeled statistically as lognormally distributed [54]. Its probability density function is:

$$p(g) = \begin{cases} \frac{1}{\sqrt{2\pi\sigma_{pl}^2} g} e^{-(\ln g - \mu)^2 / 2\sigma_{pl}^2} & (g \geq 0) \\ 0 & (g < 0) \end{cases} \quad \text{Equation 4-3}$$

If we define a new random variable X as $X = \ln g$, then,

$$p(x) = \frac{1}{\sqrt{2\pi\sigma_{pl}^2}} e^{-(x-\mu)^2 / 2\sigma_{pl}^2}, (-\infty < x < \infty) \quad \text{Equation 4-4}$$

The random variable X represents the path loss measured in dB, μ is the mean path loss in dB, and σ_{pl} is the standard deviation of the shadow fading in dB. For typical cellular and microcellular environments, σ_{pl} is in the range of 5-12 dB [54], p. 814.

4.2.5 Channel model

We are going to simulate a typical wireless channel to evaluate the scheduling algorithms. Currently, the three types of channel model are set up in the simulator:

- Addictive White Gaussian Noice (AWGN) channel;
- Two-way fixed channel with the path setting as:

$$\begin{aligned} s_1(t) &= a \\ s_2(t) &= b \end{aligned} \quad \text{Equation 4-5}$$

where a and b are constants. They are the time domain impulse response. The setting of a and b should satisfy $a^2 + b^2 = 1$. For example: $a=0.6$, $b=0.8$. This parameter setting is to guarantee that $s_1^2(t) + s_2^2(t) = 1$.

- Time varying multipath Rayleigh fading channel. In IEEE 802.16a standard, NLOS is assumed. Based on this assumption, there are N_p number of paths and N_p is configurable in our simulation setting. Each path is modeled as Rayleigh fading with PDF as:

$$f_x(x) = \frac{x}{\sigma^2} \exp\left(\frac{-x^2}{2\sigma^2}\right), \quad x > 0 \quad \text{Equation 4-6}$$

Its mean is $\mu_x = \sigma(\pi/2)^{1/2}$, and variance is: $\sigma_x^2 = \sigma^2(2 - \pi/2)$.

To simulate the time varying channel, from the time domain point of view, channel time impulse response is changing from time to time due to the motion of the MS. There are several ways to do the simulation. The simplest way is to implement it in a random fashion, either AWGN or Rayleigh distribution. At each frame time, channel time domain impulse response will be updated according to one of the random number distributions. And this time varying span is also configurable.

Another popular and practical approach is to use Jake's model in [55]. The speed and angle of the moving object determine the channel coefficient.

Another important parameter is the product of $f_d T_s$, where f_d is the maximal Doppler frequency and T_s is the sample duration. We will discuss the impact of fast and slow fading on the system performance in the future work.

In this thesis, we use Jake's method to simulate the time varying multipath channel. Suppose there are N_p number of paths, each multipath component is Rayleigh distributed. Here is how we simulate the Rayleigh distributed channel:

$$y_{n_modulated}(t) = x_{cJ}(t) \cos \omega_c t + x_{sJ}(t) \sin \omega_c t \quad \text{Equation 4-7}$$

where ω_c is the carrier frequency; n is the n th path of N_p paths, we use symbol “ J ” to distinguish Jake's model, thus nJ represents the number of path in Jake's model, it is different from n which represents one of the N_p path in multipath model. In simulation, we use the base band equivalent channel, that is:

$$y_n(t) = x_{cJ}(t) + jx_{sJ}(t) \quad \text{Equation 4-8}$$

where $x_{cJ}(t)$ is the in-phase component and $x_{sJ}(t)$ is the quadrature component in Jake's model, when they are both Gaussian random process, then $|y_n|$ is Rayleigh distributed. We let

$$x_{cJ}(t) = \sqrt{\frac{2}{N_{Jake}}} \sum_{nJ=1}^{N_{Jake}} \cos \beta_{nJ} \quad \text{Equation 4-9}$$

$$x_{sJ}(t) = \sqrt{\frac{2}{N_{\text{Jake}}}} \sum_{nJ=1}^{N_{\text{Jake}}} \sin \beta_{nJ} \quad \text{Equation 4-10}$$

where N_{Jake} is the total number of path with different arrival angles of the mobile object. Only when N_{Jake} is large enough, we may invoke the Central Limit Theorem to conclude that $x_{cJ}(t)$ and $x_{sJ}(t)$ are approximately Gaussian process, so that $|y_n|$ is Rayleigh as desired. We set this N_{Jake} as 34 in our simulation according to [55]. β_{nJ} is the phase of the nJ th path in Jake's model, it is:

$$\beta_{nJ} = 2\pi f_d t \cos \alpha_{nJ} + \phi_{nJ} \quad \text{Equation 4-11}$$

and ϕ_{nJ} is a random variable with uniform distribution between 0 and 2π . We assume that the arrival angles are uniformly distributed within the range of $[0, 2\pi]$ and,

$$\alpha_{nJ} = \frac{2\pi nJ}{N_{\text{Jake}}}, \quad nJ = 1, 2, \dots, N_{\text{Jake}} \quad \text{Equation 4-12}$$

Through this process, we get the time domain impulse response of y_n . It is a function of time, and also it is determined by the mobile speed in term of f_d .

From a frequency domain viewpoint, when the channel impulse response is $\{y_n, 1 \leq n \leq N_p\}$, its frequency response at the subcarrier frequencies $f_k = k/N_{\text{tone}}$ is obtained by FT:

$$Y_k = \sum_{n=1}^{N_p} y_n e^{-j2\pi nk / N_{\text{tone}}} \quad \text{Equation 4-13}$$

where k stands for the subcarrier index and N_{tone} is the total number of carriers, it equals to FFT size.

When the frequency response for each subcarrier is obtained, then we can calculate the average of the square of frequency response on the whole frequency band as follows:

$$Y_{\text{ave}} = \frac{\sum_{k=1}^{N_{\text{tone}}} |Y_k|^2}{N_{\text{tone}}} \quad \text{Equation 4-14}$$

This will be used to calculate the estimated received SNR for each user in the next section.

Here are several points we would like to mention:

- When multi-path is assumed, all path are time varying according to the movement of MS. In other words, MS' speed and angle determine whether the channel is fast or slow fading. The frequency response of the channel is calculated based on the time domain impulse response.
- The varying feature is the property of channel due to motion. This has nothing to do with the OFDM symbol duration or frame duration. We may update

channel impulse response at each symbol time, or several symbol time, or frame time, this is the sampling time.

- To measure channel time varying, we often use the parameter of $f_D \cdot T_{\text{samp}}$, f_D is the maximal Doppler frequency and T_{samp} is the sampling interval. When this product is far less than one, we say it is a slow fading channel. Or in other words, $f_D \ll f_{\text{samp}}$, we regard the channel as slow fading channel.

4.2.6 Power issue

In the power consideration, we have the following assumptions:

- Overhead in forward link power usage is 25% of the total power. This is used for pilot, paging and synchronization, control, etc. This is implemented through the frame structure which will be shown in detail in the following sections.
- No power control is adopted because OFDM systems are not as sensitive to power level as CDMA systems. Every subcarrier uses the constant maximal power to transmit data to the selected MS.

4.2.7 Other physical layer parameters

- Transmit power spectrum density (PSD) (in dBm/Hz): this will be an input parameter. We will evaluate the system performance based on various setting of this number. But in our single cell case, this parameter is not very important. In the future if we build

more complicated network with several BS, it may have much impact on the whole system performance.

- Estimated received SNR: based on previous frame, each user estimates its received SNR and sends this information in the uplink to the BS. Here is the formula we use to calculate the estimated received SNR:

$$P_r(dB) = P_t(dBm/Hz) - F_{channel}(dB) - L_S(dB) - P_{shadow}(dB) - N_{floor}(dBm/Hz) \quad \text{Equation 4-15}$$

where, $P_r(dB)$ is the estimated received symbol SNR, P_t is the transmit PSD in dBm/Hz, $F_{channel}$ stands for the loss introduced by the channel, for each individual frequency, it is the square of Y_k obtained from equation 4-13, L_S is the larger scale pass loss which is defined in equation 4-2, P_{shadow} is the loss due to shadowing effect, and N_{floor} is the noise power spectrum density floor. We choose this $N_{floor} = -174$ dBm/Hz [57]. We also define an upper and lower limit of the estimated received SNR to be 200dB and -200dB respectively. This is to ensure that this number is in a reasonable range.

- Tone capacity: tone capacity is the number of bits that each tone can carry. The dynamic channel allocation (DCA) algorithm, which will be explained in detail in the following section, will calculate the tone capacity according to the estimated received SNR value. This value is obtained from equation 4-15. We use Shannon capacity theorem on page 525 of [56] to calculate the tone capacity:

$$C(bits/s/Hz) = \log_2[1 + (SNR_{rx} - SNR_{margin} - SNR_{required})] \quad \text{Equation 4-16}$$

where C is the tone capacity; SNR_{rx} is the estimated received SNR on this tone, i.e., the P_r (dB) in equation 4-15; SNR_{margin} is set to 3dB under the consideration to combat any unexpected interference [57]; and $SNR_{required}$ is the required SNR to guarantee symbol error rate below a certain margin, such as 10^{-4} . This is needed to meet the QoS requirements for different types of multimedia communication services. We set $SNR_{required}$ to 6.75dB according to [57] in this thesis.

We also set a maximal number of bits for each tone, to ensure each tone has a reasonable maximum number of bits to carry. Now this is set at 6 because the highest modulation level in IEEE 802.16a is 64 QAM as stated in Table 4-2.

- Upper limit of the system data rate: based on the above calculation, and the given BW is 20MHz, consider 25% of the overhead defined in the frame structure in terms of number of symbols, and another 25% for the cyclic prefix in OFDM symbol in terms of symbol duration defined in Table 4-1, we have the upper limit of system data rate to be:

$$R (/ Mbps / Hz) = C * 20 * (1 - 25\%) * (1 - 25\%) \quad \text{Equation 4-17}$$

- Received bits in error:

First, we have to calculate the received signal to noise ratio per bit E_b/N_0 for each tone.

$$\frac{E_b}{N_0} = SNR_{rx} / N_{bitspertone} \quad \text{Equation 4-18}$$

SNR_{rx} is the received SNR per symbol obtained from equation 4-15 and $N_{bitspertone}$ is the tone capacity calculated from equation 4-16.

In the current simulation, we consider the modulation to be Quadrature Amplitude Modulation (QAM) and we use the theoretical formula to compute the BER. In the future, we may use Simulink or other software to simulate the bit in error in the physical layer. Here, we can calculate the probability of error through [54], p. 255:

$$P_e = 0.5 \operatorname{erfc} \sqrt{\frac{E_b}{N_0}} \quad \text{Equation 4-19}$$

Consider there are a total of n bits per tone, and the probability of k out of n bits in error is [54], p. 36-37:

$$P_{(k,n)} = \binom{n}{k} * (1 - P_e)^{n-k} P_e^k \quad \text{Equation 4-20}$$

Now given n bits and knowing all $P_{(k,n)}$, we need to generate a random variable (k) which represents the number of bits in error and following the distribution of $P_{(k,n)}$. To do this, we line up all $P_{(k,n)}$ in the (0,1) interval as shown below:

Figure 4-2 Uniform distribution of the random variable $P_{(k,n)}$

Where all $P_{(k,n)}$ line up and their sum is equal to 1. From the above, we know that we can generate a random variable uniformly distributed within (0, 1). If this random variable falls into the interval $P_{(1,n)}$, we say 1 bit is in error; if it falls into the interval $P_{(2,n)}$, we say 2 bits are in error; and so on so forth. This is the approach to simulate the received bits in error for one tone. The total number of bits in error is the sum of bits in error for each tone.

4.3 OFDM Description

In IEEE 802.16a, there are three modes: Single Carrier (SC), OFDM and OFDMA. In this thesis, we adopt the OFDM mode. In this mode, OFDM is employed as the modulation scheme. In our implementation, the setting of number of carriers, frame duration, symbol time, physical slot, cyclic prefix (CP) are all defined by the standard, as listed in Table 4-1. However, our implementation is very flexible. Almost all parameters are configurable. It can be easily switched to other OFDM systems without the constraint of the standard.

4.3.1 Frame structure

Figure 4-3 is the frame structure we use in this thesis. It is derived from IEEE 802.16a standard [48]. Each frame is composed of a number of OFDM symbols. Suppose 25% of total number of symbols is used for overhead for control and management purpose. Because the main purpose of this thesis is to evaluate the performance of the scheduling algorithm in terms of efficiency and fairness, we do not need to implement the overhead in detail according to the standard and leave this percentage of resource empty. We only consider the data stream which occupies 75% of total number of symbols for the system performance evaluation. Also, in our simulation setting, almost all parameters are configurable. This design provides flexibility for any future research.

Figure 4-3 Frame structure

4.3.2 OFDM symbol structure

Figure 4-4 is extracted from [47]. It shows the OFDM symbol structure. Here in 802.16a, we employ the single reception mode as shown in (a). The Guard Interval (GI) or Cyclic Prefix (CP) is set at 25% of the symbol time, which is $3.2\mu\text{s}$. According to the OFDM theory, this GI should be larger than the channel coherence time so that it can eliminate inter symbol interference (ISI) effectively in the multipath

fading channel. We adopt the propagation model which was described in the previous section, and this is a typical mobile cellular environment. Based on the investigating result in [51], in macro cells with radius of 1-20km, the value of the average root mean square of the delay spread τ_{RMS} falls between 0.1 and 10 μs . When our cell radius is set at 1.5km, the GI of 3.2 μs is quite reasonable to ensure that the average delay spread falls in the GI. Thus ISI is eliminated. In our simulation, here is how symbol time is determined:

$$\text{Symbol_duration} = \frac{\text{FFTSize}}{\text{Bandwidth}} \times (1 + \text{GI_percentage}) \quad \text{Equation 4-21}$$

Figure 4-4 802.11a OFDM frame with cyclic extension and windowing for (a) single reception or (b) two receptions of the FFT period

4.4 System Scenario

We will present the time-frequency resource diagram before introducing the scheduling algorithm. Then, system scenario is described in details.

4.4.1 Two-dimensional resource allocation

Figure 4-5 is the time-frequency diagram. From the time domain, each frame is composed of OFDM symbols. From the frequency domain, each OFDM symbol is composed of number of tones, or carriers. We regard each time-frequency block as the smallest unit of our two dimensional resource. The overhead percentage of the frame is stated in Table 4-1. In the rest of the thesis, we leave this overhead percentage empty and only traffic stream symbols are considered as available resource. The scheduler is called at the beginning of each frame and it does the scheduling for the whole frame.

The result of the resource allocation of the whole frame will be carried by the symbols reserved for overhead as shown in Figure 4-3 Frame structure. From this point of view, the scheduling algorithm costs some symbols in the overhead. This cause a little loses in throughput.

Figure 4-5 Time-frequency diagram

4.4.2 System scenario

Based on the frame structure in Table 4-1, the number of OFDM symbols in each frame, N_{symbol} , can be calculated. For each OFDM symbol, there are M_{tone} number of subcarriers, or tones. It equals the FFT size. N number of users, or MSs are uniformly distributed within the cell radius. The distance between user i and the BS is d_i .

Based on the previous frame, each user estimates its received SNR and sends this information in the uplink (we do not simulate the uplink in this thesis) to the BS. In the simulation, we use the channel condition of the previous frame in order to calculate the received SNR value for the current frame. Large scale fading such as path loss and shadowing, small scale fading such as time varying and multipath effects are included in the estimated SNR computation from equation 4-15.

4.4.3 Common scheduling procedure

At the beginning of each frame, BS obtains the estimated received SNR from each user. The scheduler does the scheduling for the whole frame. It skips the number of symbols reserved for overhead and starts from the first available symbol. For each symbol, the scheduling procedure consists of two steps:

- First step is user selection. The scheduler picks up one MS from N number of users based on the algorithm criterion which will be introduced in the next section.
- Second step is bits allocation and AMC.
 - Bits allocation: The scheduler dynamically allocates different number of bits of data on each tone based on its tone capacity to the selected MS. The total number of tone is equal to the FFT size of the OFDM parameters listed in Table 4-1. The bits allocation starts from tone index 1 to the last tone in this OFDM symbol.

- AMC: for each tone, tone capacity is calculated from equation 4-16 and this number of bits is allocated on this individual tone. According to the tone capacity, different modulation and coding combination will be applied to this tone as stated in Table 4-2. This process is called Adaptive Modulation and Coding scheme. No power control is adopted. Every subcarrier uses the constant maximal power to transmit data to the selected MS.

4.5 System performance definitions

The performance of the system will be evaluated in two aspects: efficiency and fairness. Efficiency is evaluated in terms of system throughput, mean packet delay, delay variance. Fairness is evaluated in terms of user satisfaction rate and mean user rate. Here we give all these definitions as follows.

- Throughput: there are many definitions of this parameter and we list some of them here for reference;
 1. Throughput is defined as the number of transmittable bits per subcarrier (bit/subcarrier)[58].
 2. Throughput is defined as the successfully transmitted number of bits per second (BPS) [59].
 3.
$$S = R \cdot \frac{1}{T} \cdot \frac{K}{M}$$

where in each data block, K symbols are allocated for data and M-K for redundancy. R is introduced for a fair comparison of the throughput

when different operation modes are used. R is defined

$$\text{as } R = \begin{cases} 1, & \text{for the } L\text{-mode} \\ \frac{n_H}{n_L}, & \text{for the } H\text{-mode} \end{cases}, \text{ where } n_H, n_L \text{ represents the number}$$

of bits/symbol corresponding to the constellation used for the L (H) mode (it indicates the modulation level in this paper) [60].

$$4. \text{ Throughput} = R * (1 - \text{overhead}) * [(1 - \text{FER}) / N_{tr}]$$

Where: R is the transmitted information bit rate. FER stands for frame error rate. N_{tr} is the average number of transmission attempts. If we exclude the effects of overheads and retransmission times, the equation becomes:

$$\text{Throughput} = R * [1 - \text{FER}] \text{ [61].}$$

5. Throughput is defined as the number of information bits transmitted in each second (bps). It's equal to the number of information bits that are received and accepted by the User Equipment (UE) receiver within a time period T divided by that time period [62].

6. Throughput is regarded as the spectral efficiency, which is the number of bits (rate) assigned to the j -th subcarrier, in bits/s/Hz [63].

In this thesis, we apply the 5th definition (which is in bold letter) to be our system throughput. To be more accurate, it is the received number of information bits per second (bps). It is equal to the total number of information bits that are received and accepted by the MSs within a time period T divided by that time period. Other system performance such as the system drop rate, block rate could also be added if it is needed. The throughput definition can also be changed according to any requirements.

- Delay: the mean value of packet completion time, measured from the instant when the packet arrives at the BS's buffer queue until the MS receives this packet.
- User satisfaction rate: ratio of the satisfied number of MS and total number of MS:

$$R_{satis} = \frac{N_{satis}}{N} \quad \text{Equation 4-22}$$

where R_{satis} is the user satisfaction rate, N_{satis} is the number of satisfied user and N is the total number of users.

A MS is defined as satisfied [64] when the following criteria are fulfilled:

- the user does not get blocked when arriving to the system;
- the ratio of received bit and total generated traffic of user i is greater than a threshold, that is:

$$\frac{N_{i_rec}}{N_{i_req}} \geq \sigma \quad \text{Equation 4-23}$$

where N_{i_rec} and N_{i_req} are the total number of received bits and total number of required bits by user i respectively, σ is the predefined threshold between 0 and 1.

- the user's packet does not get dropped.

It should be mentioned that in the literature, there are other way to define fairness and the user satisfaction. In [65], [66] and [67], the fairness is profit oriented. In [68], the user's satisfaction is based on its bit error rate requirements which is suitable for data services only. In this thesis, under multimedia traffic environment with different data rate requirements, we apply the satisfaction rate definition as in [64] which emphasizes the ratio of the received and required data rate.

- Average user rate: user rate R_i is defined as the ratio of received bits and total generated traffic for this user i :

$$R_i = \frac{N_{i_rec}}{N_{i_req}} \quad \text{Equation 4-24}$$

Average user rate is the average user rate among total number of N users:

$$\bar{R} = \sum_{i=1}^N \frac{R_i}{N} \quad \text{Equation 4-25}$$

5 Theoretical framework

In this chapter, we will deduct the calculation of system performance in terms of throughput and average user rate. In addition, we develop the mathematical model for user satisfaction rate with respect to throughput. The relationship between throughput and user satisfaction rate shows the tradeoff between system efficiency and fairness.

5.1 System throughput

5.1.1 System throughput expression

We take user No. 1 as the user of interest to calculate its throughput, and then we generalize the formula to obtain the overall system throughput.

Based on Figure 4-5 Time-frequency diagram and equation 4-16, we start at the 1st available symbol and proceed to calculate the number of bits that each OFDM carrier can carry:

$$C_{ik} (\text{bits / tone}) = \log_2 [1 + (SNR_{ik_rx} - SNR_{ik_margin} - SNR_{ik_required})] \quad \text{Equation 5-1}$$

where C_{ik} is the number of bits the k th tone can carry when user i is selected, k is the tone index, SNR_{ik_rx} is the estimated received SNR value for user i on k th tone, SNR_{ik_margin} is the SNR margin to combat any unexpected fading or interference, $SNR_{ik_required}$ is the required SNR to guarantee symbol error rate below a certain

margin for this user. We notice that the tone capacity depends on the wireless channel condition, or we can say it depends on the value of SNR_{ik_rx} . To be more specific, the estimated received SNR of user i on tone k is:

$$SNR_{ik}(dB) = P_t(dBm/Hz) - F_{ik_channel}(dB) - L_{ik_S}(dB) - P_{ik_shadows}(dB) - N_{floor}(dBm/Hz) \quad \text{Equation 5-2}$$

where P_t is the transmit PSD in dBm/Hz, $F_{ik_channel}$ stands for the loss introduced by the channel, for each individual frequency, it is the square of Y_k obtained from equation 4-13, L_{ik_S} is the larger scale pass loss which is defined in equation 4-2, P_{ik_shadow} is the loss due to shadowing effect, and N_{floor} is the noise power spectrum density floor.

Then, we have the total bits of the symbol capacity for user i . It is the sum of each tone capacity. When we choose the number of OFDM carriers to be K , the symbol capacity is:

$$C_{i_sym} = \sum_{k=1}^K C_{ik} \quad \text{Equation 5-3}$$

where C_{i_sym} is the symbol capacity for user i .

In one frame, if the symbol capacity for user i is C_{i_sym} , we have the frame capacity as:

$$C_{frame} = \sum_{i=1}^N (C_{i_sym} \cdot N_{i_sel}) \quad \text{Equation 5-4}$$

where C_{frame} is the frame capacity, N_{i_sel} is the number of symbols which user i is selected, it is between 0 and the total available number of symbols in this frame. Please note that this N_{i_sel} is different from N . If the total number of available symbols in one frame is M , which is defined as the floor of frame duration divided by symbol duration and times 75% based on Figure 4-3 Frame structure. Then, under the condition of each symbol capacity is large enough to transmit at least one packet from one user:

$$M = \sum_{i=1}^N N_{i_sel} \quad \text{Equation 5-5}$$

Equation 5-5 is correct only when traffic is large enough that in every symbol there is a user been chosen. This is not always the case. When traffic is light,

$M = \sum_{i=1}^N N_{i_sel} + N_{vacant}$, where N_{vacant} is the number of vacant symbol in the frame. No

matter under light or heavy traffic, when frame duration is T_{frame} , we obtain the system throughput S as:

$$S = \frac{C_{frame}}{T_{frame}} = \frac{\sum_{i=1}^N (C_{i_sym} \cdot N_{i_sel})}{T_{frame}} = \frac{\sum_{i=1}^N \left(\sum_{k=1}^K C_{ik} \cdot N_{i_sel} \right)}{T_{frame}} \quad \text{Equation 5-6}$$

$$= \frac{\sum_{i=1}^N \left(\sum_{k=1}^K \log_2 [1 + (SNR_{ik_rx} - SNR_{ik_margin} - SNR_{ik_required})] \cdot N_{i_sel} \right)}{T_{frame}}$$

In the above equation, even under the assumption that the SNR_{ik_margin} and the $SNR_{ik_required}$ are constant, the overall system throughput is depending on the SNR_{ik_rx}

and N_{i_sel} . In other words, the system throughput is determined by the scheduling algorithm which gives the number of symbols assigned to a specific user, and the wireless channel condition of the user.

Even though we obtain the theoretical throughput value, but it is hard to give the closed form, because SNR_{ik_rx} and N_{i_sel} are both hard to predict. Since in the SNR_{ik_rx} , there are three random variables: $F_{ik_channel}$, L_{ik_S} and P_{ik_shadow} , in the wireless environment, they are changing all the time, even if we know the distribution of them, it is difficult to obtain the joint distribution. Above that, N_{i_sel} is determined by the algorithm, the algorithms consider user traffic model, threshold and adaptive parameters. These are all random variables. It is beyond the scope of this thesis to obtain the joint distribution of random variables with dozens of dimensions.

5.1.2 System throughput upper bound

Even though it is hard to obtain the closed form of the system throughput, but we can calculate the upper bound of it under certain assumptions.

With the assumption that the channel is perfect, in other words, each tone can carry its maximal bits which are determined by the IEEE 802.16a standard. And in this case, the scheduling algorithm does not have any impact on throughput because no matter which user is selected, the symbol is under full capacity. Thus, when M is the total

number of available symbol in each frame, K is the total number of OFDM carrier, C_{max} is the maximal tone capacity, and equation 5-6 turns out to be:

$$S_{max} = \frac{\sum_{i=1}^N \left(\sum_{k=1}^K C_{ik_max} \cdot N_{i_sel} \right)}{T_{frame}} = \frac{M \cdot \sum_{k=1}^K C_{ik_max}}{T_{frame}} = \frac{M \cdot K \cdot C_{max}}{T_{frame}} \quad \text{Equation 5-7}$$

From Table 4-2 Channel coding per Modulation, we know that the highest modulation level is 64 QAM, that means each tone can carry 6 bits, or $C_{max}=6$. Also, from Table 4-1 Assumption of IEEE 802.16a standard (OFDM) mode, we have $K=256$, and

$$M = \lfloor (\text{frame length/symbol duration}) \cdot (1 - \text{Overhead DL percentage}) \rfloor = 93 ,$$

So, we have:

$$S_{max} = \frac{93 \times 256 \times 6}{2 \times 10^{-3}} = 71424 = 71.424(Mbps) \quad \text{Equation 5-8}$$

Thus, we obtain the upper bound for the system throughput. We will verify this value in our simulation result to see if the entire obtained throughput is smaller than this value.

5.2 Mathematical model of user satisfaction rate

In this section, we are going to develop the mathematical model of user satisfaction rate.

Suppose that the throughput of user i is S_i , the required rate of user i is R_{i_req} , the ratio of the two is r_i . From equation 5-6, we have:

$$S_i = \frac{\sum_{k=1}^K C_{ik} \cdot N_{i_sel}}{T_{frame}} \quad \text{Equation 5-9}$$

$$r_i = \frac{S_i}{R_{i_req}} \quad \text{Equation 5-10}$$

With the definition of user satisfaction rate R_{satis} in section 4.5, we compare this r_i to the pre-set threshold value σ , when $r_i \geq \sigma$, we say this user is satisfied. And R_{satis} is the number of satisfied users over total number of users.

Now we try to develop a mathematical model of random variables. When we regard the r as a random variable between 0 and 1, $r_i \geq \sigma$ is regarded as an “success” event, $r_i < \sigma$ is regarded as “fail”, the total number of trials is the total number of r_i , or, total number of user N , so, the R_{satis} could be regarded as the probability of success. That is:

$$\begin{aligned} R_{satis} &= P(r > \sigma) \\ &= 1 - P(r < \sigma) \\ &= 1 - \int_0^{\sigma} f(r) dr \end{aligned} \quad \text{Equation 5-11}$$

where $f(r)$ is the pdf of random variable r . As we can see from equation 5-10, r_i is defined by the throughput and the required rate. For a single user, because S_i and R_{i_req} are also random variables, r_i is defined by the joint distribution of S_i and R_{i_req} . And if there is total number of 10 users, then the R_{satis} is defined by the 10 dimensional joint distribution of r_i . In addition, the S_i itself is also a joint distribution of several

dimensions according the deduction of throughput in section 5.1.1. The complexity of R_{satis} is beyond the scope of this thesis.

Even though it is hard evaluate the R_{satis} with respect to the distribution of $f(r)$, but equation 5-11 reveals an important relationship between satisfaction rate and the throughput. Assume that the R_{i_req} is a constant C , in other words, we assume the traffic is simplest form of constant rate, then, equation 5-11 could be simplified as:

$$R_{satis} = 1 - \frac{1}{C^N} \int_0^C f(s) ds \quad \text{Equation 5-12}$$

From equation 5-12, when we further simplify it with $f(s)$ is uniform distribution, which is reasonable since the throughput could be any number from 0 to the system capacity, then we may notice that, the larger the throughput, the smaller the R_{satis} . This trend matches with the trend of our simulation results.

For further study on $f(s)$, it may not be true for equal probability between 0 and the system capacity. The distribution of $f(s)$ depends on the mean rate of the traffic and how many percentages of the users are having this type of traffic. In a more complicated case of variable bit rate service, a user's mean traffic rate is also changing from time to time. These scenarios will add complexity to the distribution of $f(s)$. This may be one of the directions of our future study.

5.3 Average user rate

From equation 5-9 and 5-10, we have:

$$r_i = \frac{S_i}{R_{i_req}} = \frac{\frac{\sum_{k=1}^K C_{ik} \cdot N_{i_sel}}{T_{frame}}}{R_{i_req}} \quad \text{Equation 5-13}$$

By the definition of average rate $\bar{R}$ in section 4.5 (equation 4-25) , we have:

$$\bar{R} = \frac{\sum_{i=1}^N r_i}{N} = \frac{\sum_{i=1}^N \frac{\sum_{k=1}^K C_{ik} \cdot N_{i_sel}}{R_{i_req} \cdot T_{frame}}}{N} \quad \text{Equation 5-14}$$

Again, it is hard to have the closed form due to the complexity of the distribution of throughput.

6 Scheduling Algorithm Description

In the first phase of our research, we implement 5 MAC scheduling algorithms in OFDM broadband wireless systems. They are Round Robin (RR), maximal SNR—time multiple accesses (MaxSNR), OFDM Proportional Fair (OPF), PF with observation window, weighted PF. Among the five algorithms, OPF is the one we modified based on PF. The other four are known algorithms. We implement the known method for comparison purpose. Among these four, two are conventional algorithms: RR and MaxSNR; and two are versions of PF: PF with observation window and weighted PF. In some papers, the word Dynamic Channel Allocation (DCA) is mentioned instead of scheduling algorithm. To avoid confusion, we state that in this thesis, DCA and scheduling algorithm have the same meaning. We tailor the scheduling algorithms to fully comply with IEEE 802.16a standard as described in the previous sections. We evaluate the efficiency and fairness as indicators of system performance. Efficiency is presented in terms of throughput, mean delay, delay variance. Fairness is in terms of user satisfaction rate and average user rate. These definitions could be found in section 4.5. We will give the traffic model before introducing the scheduling algorithms. Following the traffic model, RR, MaxSNR and OPF will be described in detail. PF with observation window and weighted PF will be introduced briefly for comparison purpose.

6.1 Traffic model

Multimedia service is one of the features of 4G wireless networks. System should give good performance for all types of traffic. We choose two typical traffic models in this thesis, constant rate and exponential distribution. At the present stage, for all the mobile stations in the cell, they either have constant rate traffic model, or have the exponential distribution traffic model. In the future, each user can have their traffic model as required. Here are descriptions of the two types of traffic:

- Constant rate: this is the simplest form of traffic. The inter-arrival time between consecutive packets is a constant. The data rate is equal to the packet size (bits) divided by the packet inter-arrival time (sec). that is:

$$Data_rate = Packet_size(bit) / interarrival(sec) \quad \text{Equation 6-1}$$

We can vary the time span of interval, or the packet size to get different rate for each MS, and the overall system traffic rate is the sum of all MSs' traffic rates.

- Exponential distribution: the inter-arrival time between two packets is a random variable with PDF (probability density function) as:

$$f_x(x) = \lambda e^{-\lambda x}, x > 0 \quad \text{Equation 6-2}$$

with $1/\lambda$ as its mean. We can vary the mean of the inter-arrival time to simulate different traffic load.

In the future, we may develop more complicated types of traffic such as internet traffic, real time video and audio. We evaluate the system performance under different traffic load of the same traffic type or under mixed type of traffics. When mixed type of traffic is the input to the system, we can implement priority among traffic which is another important feature of QoS. At the current stage, we begin with one traffic type with different rate, to see the impact of the scheduling algorithms on system performance.

6.2 Common assumptions of all algorithms

Before presenting each specific algorithm, we list the common assumptions for all the algorithms as follows:

- We regard the resource as two dimensional, namely time and frequency resource unit as shown in Figure 4-5. Each OFDM symbol duration is regarded as the smallest time unit, and each tone in frequency domain is regarded as the smallest frequency unit.
- From the point of view of traffic load, we regard one packet as the smallest unit of requested traffic. When the channel condition is too bad to support even one packet for any MS, we skip this frame and go to the next frame.
- Only 75% of the total number of OFDM symbols in the frame is available for data stream, the other 25% is for control and header and we skip these overhead symbols at each beginning of the frame.

- Scheduling is done at each beginning of the frame. The total traffic waiting to be scheduled includes two parts: the new traffic which is generated during one frame time, and the accumulated traffic which is unscheduled or left-over by all the previous frames.
- We are assuming the packet size is a constant and it does not change;
- Channel is supposed to remain stable during one frame. For the next frame, channel will change based on the time varying and frequency selective fading model. We choose Jake's model [55] to simulate one path of the time varying, Rayleigh distribution channel. There are multiple paths which cause frequency selective fading.
- The estimated received SNR for each user is based on the previous frame.
- Different MSs are not allowed to access the same symbol. In other words, for the current symbol, when one MS is selected, this MS takes over all tones no matter how many tones it needs. In the next symbol if another user is picked up, that user will start from the first tone of that symbol.
- AMC is applied to each tone when MS is selected. No matter which algorithm is operating, the allocation of bits to each tone is according to the specific frequency response of that tone for the selected MS. In this way, different tone carries different number of bits based on its own channel characteristics. Corresponding modulation and coding combination will be determined by the 802.16a standard in Table 4-2.

- All MS have the same traffic model, either constant rate or exponential distribution.

Under the common assumptions, each of the algorithms has its own properties. In the following section, all 5 algorithms will be described in details.

6.3 RR DCA

Common scheduling procedure is already described in section 4.4.3, here we focus on each scheduling criterion and their OFDM implementation.

6.3.1 RR algorithm criterion and description

- **Criterion:** In RR, each MS is allocated an ID, which is used by the BS to poll the MS sequentially. In summary, RR criterion is the user sequence number.
- **Description:** Starting from the first available symbol in the frame, the scheduler checks if a MS has data in its buffer. If there is nothing in the MS' buffer, the scheduler skips the current MS and checks the next one. Otherwise, it calculates the how many number of bit that each tone can carry and how many number of tones is required according to the current channel condition. Then, RR assigns the number of bits per tone and number of tones in this symbol to the current MS. The last scheduled MS's ID is recorded at the end of the current scheduling event. The

next scheduling cycle (next frame) starts with the user whose ID is the next one to the currently recorded ID. And the round robin continues. All users with non-empty queues must be serviced once before re-servicing this user.

It should be clarified that if the number of users is smaller than the available number of symbols in one frame, a user will be selected more than one time in the same frame. In other words, a user can be assigned with multiple symbols in one frame but only one symbol in each round.

6.3.2 OFDM implementation of RR

Figure 6-1 is the flow diagram of implementation of RR in OFDM system.

6.3.3 Assumptions for this algorithm

Besides the common assumptions for all algorithms, normally the number of user is smaller than the number of symbols in each frame. So that some users are assigned more than one symbols in each frame.

6.3.4 Scenario descriptions

- Two-dimension frame allocation pattern

In the whole time-frequency frame, the RR will allocate the resource in the figure below. This shows the resource allocation is symbol-by-symbol based, or, it goes along the time and this could be regarded as time domain scheduler.

Figure 6-2 RR resource allocation pattern

- Scenario one: when it approaches the last symbol, but is just in the middle of the round.

In this case, the next scheduling procedure starts with the next user for the first available symbol in the new frame. Till a particular scheduling symbol wherein the last user has been served, or equivalently, every user has been served at least once, the service continues by turning around to the first user.

- Scenario two: in one symbol, the required number of tones is less than the available number of tones

In this case, the next MS will jump into the next symbol, in this way, no two users are allowed to share the same symbol. This is a waste of resource; the system throughput can be improved if more than one user can access the same symbol.

- Scenario three: Redundancy in time

Under light traffic load, after all the users' service requests have been fulfilled, there may still be some residual symbols that remain unused at the end of a scheduling procedure. In this case, the channel capacity is more than the service requests.

6.4 MaxSNR DCA

6.4.1 MaxSNR algorithm criterion and description

- **Criterion:** In MaxSNR, at every scheduling event, all MSs with non-empty source queues are ranked by their estimated receiving SNR values. The criterion of MaxSNR is to select the MS with the highest SNR among those non-empty-buffered MSs.
- **Description:** In the calculation of SNR, channel frequency response is updated every OFDM frame. Based on how many user packets are waiting for transmission, the scheduler calculates the tone capacity of each tone and the symbol capacity which equal the sum of all tone capacity. As we know from the common assumption, one packet is regarded as the smallest data unit for transmission. In the case of the packet size is larger than the symbol capacity, more than one symbol will be assigned to this user. In the case of packet size is smaller than the symbol capacity, then this symbol will carry as many packets as it can. For the next symbol, the scheduler will repeat again to find the MS with the largest SNR, regardless whether this MS was picked up in the previous symbol.

As may be noticed, for all the algorithms, the packet size may have impact on the system performance. We may do further study on the impact of packet size.

6.4.2 SNR calculation

One point should be emphasized in MaxSNR: the calculation of the estimated received SNR is different in the two steps of the scheduling procedure introduced in section 4.4.3.

- In user selection, the received SNR is based on the average square of frequency response over all tones. For a more detailed description: in equation 4-15, received SNR is calculated in the frequency domain. Each tone has its own channel frequency response. The total number of tones is equal to the FFT size. For each user, we add up the square of all these frequency response and divide it by the FFT size. In this way, we get the average square of frequency response for the whole frequency band, namely, F_{channel} . This F_{channel} is used to calculate the estimated received SNR of each user. The user who has the largest received SNR will be selected for the current symbol.
- In bit allocation and AMC, we need to calculate the capacity for each tone for the selected user from equation 4-16. At this point, the SNR_{rx} is based on the average square of the frequency response for this specific tone, instead of the average square over the whole frequency band.

6.4.3 OFDM implementation of MaxSNR DCA algorithm

Figure 6-3 is the flow diagram of implementation of MaxSNR algorithm in OFDM system.

Figure 6-3 MaxSNR diagram

6.4.4 MaxSNR DCA points of interests

- Two-dimension frame allocation pattern: In the whole time-frequency frame, the MaxSNR DCA will allocate the resource with the same pattern as RR, which is shown in Figure 6-2.
- The scheduler will skip an empty-buffered MS, even if it has the highest SNR level.
- If a MS is selected in the current symbol, it can be re-selected in the following scheduling event provided that all the requirements of the scheduling algorithm are met, (i.e., it has non-empty buffer and highest SNR for that symbol), even if other MSs haven't been served.
- When the required number of tones is less than the available tones, the left tones in the same symbol will keep vacant. In this sense, no more than one MS could share the same symbol. This is a waste of resource. System throughput can be improved if more than one MS can access the same symbol.
- It's also possible that some portion of frame is left unused in some scheduling case when the system load is small.

6.5 OPF DCA

As can be observed, MaxSNR will have better system throughput which will be shown by the simulation result in the following sections. At the same time, RR provides users with fairness. In the effort of giving better system throughput without sacrificing fairness, we start to search for new algorithms. In order to find a

compromising scheduling scheme, it's natural to combine RR and MaxSNR methods into one scheme. Many efforts in seeking the compromise between throughput and fairness are made. PF is one of these algorithms. As shown in equation 3-1 and 3-2, the performance of PF depends on the observation window T_c . We modify this algorithm to eliminate the impact of observation window and provide fairness from user activation to termination. As mentioned in Chapter 3, PF is applied to TDMA and CDMA systems and its performance is been studied in these systems. Since there is little literature of PF in OFDM broadband wireless systems, we tailor the modified version of PF to fit in OFDM system. Hence, we call it OPF to stand for OFDM PF. To go a step further, we adopt IEEE 802.16a standard. In the standard, the MAC scheduling algorithm is open for research. The OPF can be regarded as a candidate for the MAC scheduling of 802.16a to provide better compromise between throughput and fairness.

In this section, we will describe the OPF in detail. In Chapter 7, we will briefly introduce the simulation tool and how the joint MAC and PHY layer simulation is implemented. Preliminary simulation results, performance comparison with RR, MaxSNR, PF with observation window, weighted PF, and discussions will be given in Chapter 8.

6.5.1 PF basic concept

In [2] and [3], the authors proposed PF and applied it to CDMA system. In that paper, the scheduler determines the next MS to be served based on two factors: the current SNR and the average data rate. The current SNR can be translated into the current supportable data rate of this MS, noted as $D_i(t)$. The average data rate that has been experienced by that MS is defined as the amount of data that has already been transmitted to that MS till the referred time. It is denoted by $R_i(t-1)$. The MS with the highest ratio of $D_i(t)/R_i(t-1)$ is selected by the scheduler at decision point for data transmission. Here is the definition of $R_i(t)$:

$$R_i(t) = \begin{cases} 0 & i \neq j \\ (1 - \frac{1}{T_c})R_j(t-1) + \frac{1}{T_c}D_j(t) & i = j \end{cases} \quad \text{Equation 6-3}$$

It is easily noticed, the performance is a function of the observation window T_c . When T_c is small, the PF emphasizes more on the impact of supportable rate. When T_c is large, the PF gives bias on the experienced rate. Hence the name of this algorithm is proportional fair. In reference [3], the PF algorithm was applied in cdma2000-HDR system, focusing purely on the effect of sector deployment upon system throughput.

6.5.2 OPF in broadband wireless OFDM system

We tailor the modified version of PF in OFDM wireless system. The reason why we regard OFDM implementation as one of the contributions lies in the following areas:

- In OFDM system, the resource is two dimensional of both time and frequency. Compared with TDMA which has only one dimension as time, OFDM provides more room of flexibility. Users can access to the resource from time, or frequency domain, or combination of time and frequency. With the flexibility of the two dimensional multiple access, there always exists the complexity on the other side. Two-dimensional resource allocation adds more complication to the scheduling algorithm. At the same time, it leaves room for both time and frequency domain scheduler. Further studies on frequency domain scheduler will be one of the directions in our future works.
- For the OPF implementation, we regard the scheduler as time domain scheduler. At present, frame-by-frame scheduling is an area open for research. OPF is our approach to allocate resource frame-by-frame. In other words, the OPF updates the $\overline{R_i(t)}$ at each symbol time, thus it selects one MS on a symbol by symbol basis for the whole frame. To the best of our knowledge, this is the first thesis to apply PF as time domain scheduler in OFDM wireless system. For the frequency domain scheduler in OFDM system, the only paper we found is paper [44]. In the future, we may consider implementing OPF as frequency domain scheduler and evaluate its performance.
- From frequency domain, AMC is applied to each OFDM sub-carrier, or each tone, after OPF works as time domain scheduler. Namely, AMC dynamically allocate different bits to each tone based on the specific channel response of this tone for

the selected MS. The objectives of this part are: 1) maintain the channel errors at a sufficiently low level to guarantee the QoS requirement. 2) Transmit more bits on good tones to improve the overall system throughput.

- OPF is applied to broadband system with 20MHz of bandwidth. In previous research works of PF, it is evaluated in CDMA or WCDMA with a typical bandwidth of 5MHz. With multi-media such as video service, bandwidth is in high demand. Compared to 5MHz of bandwidth, we evaluate OPF in 20MHz bandwidth to meet high data rate requirement of high data rate communication system.
- OPF can be a candidate of MAC scheduling algorithm of IEEE 802.16a standard. The frame structure and OFDM parameters are set according to the IEEE 802.16a to fully comply with the standard. It may also be considered for the upcoming 802.16e which is the mobility extension of 802.16a.
- OPF is evaluated in realistic wireless channel. Both time varying caused by motion of MS and frequency selective fading introduced by multi-path is simulated. And we choose a typical wireless mobile environment for urban and suburban areas outside the high-rise core, with BS antenna height of 15m above average rooftop, and cell radius is several kilometers.

6.5.3 OPF criterion

To provide fairness from user activation to termination, in other words, to eliminate the impact of T_c , we define the OPF criterion as:

$$j = \arg \max_{1 \leq i \leq N} \frac{D_i(t)}{\overline{R_i(t)}} \quad \text{Equation 6-4}$$

where $D_i(t)$ is the current supportable data rate to user i . $\overline{R_i(t)}$ is the time-average data rate experienced by user i , which is calculated as the total number of bits received by this user divided by the time from its activation to the current instant, under the condition that this user keeps active from its activation till the current time. This may not true for all scenarios but it is good for some cases such as the downloading process. In the case of user termination, it de-registers from the BS and the scheduler will not count it as one of its users.

The criterion of OPF indicates that it selects the MS not only based on its estimated received SNR values but also based on its time-average experienced rate. When the SNR is large, the numerator $D_i(t)$ is large, thus it optimizes the system throughput. At the same time, it also gives enough fairness to MSs whose channel is not so good in the past. In other words, if a user does not receive service at previous frame, e.g., due to its bad channel condition, the denominator $\overline{R_i(t)}$ becomes less, thus the ratio of

$\frac{D_i(t)}{\overline{R_i(t)}}$ is larger, then its precedence will increase at the next frame. This equals to that

the served user will have less chance to access the next available resource. Therefore, a user will not “starve” too long even if its channel condition is not so good.

Path loss, shadowing, time varying, frequency selective fading and background noise are included in the computation of $D_i(t)$. QoS requirements and external interference are included in the calculation of tone capacity in the bit allocation and AMC after the user is selected by OPF. Even though we are assuming single cell environment, but to consider the future work of multi-cell scenario, we still set an SNR margin for external interference. It is expected that the throughput of OPF should perform better than RR, but not exceed MaxSNR. User satisfaction rate and average user rate should be the best among the 3 described algorithms. The performance of the 3 algorithms will be revealed by simulation results in the following sections.

6.5.4 OFDM implementation of OPF algorithm

Figure 6-4 is the flow diagram of implementation of OPF algorithm in OFDM system. This diagram reflects the scheduling procedure as an integration of time domain scheduler and the AMC in frequency domain. In other words, the criterion of OPF controls the order of user to be served to provide guaranteed performance and fairness, meanwhile, AMC is applied to allocate different bits to each tone.

Figure 6-4 OPF diagram

6.5.5 OPF DCA procedure

Even though we describe the common scheduling procedure for all algorithms in section 4.4.3, but it is clearer to give one detailed explanation. We take OPF as an example.

The OPF procedure can be divided into two phases. First, user is selected at each symbol time by the criterion described in equation 6-4. Second, OPF allocates different bits to each tone of the current symbol for the selected user. It is mentioned as bit allocation and AMC. The two steps are repeated from the first available symbol till the last symbol of the frame. This procedure is shown in Figure 6-4, here we give the detailed description:

- User selection: In OPF scheduler follows the same user selection procedures as MaxSNR except that OPF ranks MSs by the ratio of supportable rate and average experienced rate. As we know, MaxSNR selects the MS with the largest estimated received SNR value. Specifically,
 - At every scheduling event, OPF first calculate the current supportable rate for each user. Here is the detail of how the supportable rate is calculated. As shown in the two-dimensional resource diagram, from the frequency domain point of view, each symbol is composed of numbers of tones. This number is equal to the FFT size. For the whole frequency band, the

estimated received SNR is obtained from equation 4-15 based on the average square of channel frequency response, F_{channel} , of all tones. Then, with equation 4-16, average tone capacity can be obtained. Average symbol capacity is the sum of all tone capacity and the unit is in bits. Finally, supportable data rate is obtained from symbol capacity divided by symbol time.

- The average rate experienced for each user is the total number of bits received by this user divided by the time from this user activation to the current time.
- The ratio of supportable rate and the average experienced rate is easily obtained. OPF selects the user with the largest ratio.
- Bit allocation and AMC: when user is selected, OPF allocates different bits to each tone of the current symbol for the selected user.
 - Starting from the first tone index, the number of bits allocated to this tone equals to its tone capacity. At this point, we use the specific square of channel frequency response of this tone in equation 4-15 to calculate the received SNR. Then this tone capacity is obtained from equation 4-16.
 - Suppose the tone capacity is M and QAM is chosen to be the modulation scheme. Then, 2^M QAM modulation scheme is applied to this tone. We set the maximal number of M to 6 to ensure that it is compliant with the IEEE 802.16a standard.

- Because we are trying to comply OPF with IEEE 802.16a standard, so the combination of modulation and coding scheme should follow the parameters in Table 4-2. Actually, M is possible to take any integer number from 1 to 6. But this is not compliant with the standard. So, we use the following transform table to make OPF fit to the standard.

Table 6-1 Tone capacity and corresponding modulation scheme

M (Tone capacity)	Modulation scheme
1	Skip this tone
2	QPSK
3	QPSK
4	16 QAM
5	16 QAM
6	64 QAM
>6	64 QAM

- Proper coding scheme is applied to fight with the erroneous wireless channel. After modulation scheme is determined, the corresponding coding rate is chosen according to Table 4-2.

6.5.6 OPF points of interest

- As shown in Figure 4-5, the scheduler is called at each beginning of the frame. $\overline{R_i(t)}$ in the criterion is updated at each OFDM symbol.
- The OPF scheduler only selects among those non-empty-buffered MSs. Any empty buffered MS will be skipped.
- We regard one packet as the smallest unit waiting to be transmitted. And the scheduling is done frame by frame. In the current frame, if channel condition is too bad to support even one packet for any user, then we skip this frame and mark this frame as “bad”. Otherwise, the frame is marked as “good”.
- For a “good” frame, there are two cases: 1). packet size is larger or equal than one symbol capacity 2) packet size is smaller than the symbol capacity.
 - If the packet size is larger, the scheduler will assign more than one symbol for this MS to support one of its packets.
 - If the packet size is smaller, the scheduler will pack as many numbers of packets as possible for this MS on all tones in the current symbol.

It can be observed that the packet size has impacts on the system performance.

This leaves an area for further investigation.

- As listed in the common assumptions for all algorithms, no more than one MS is allowed to share the same symbol. Based on the number of packets waiting to be scheduled in the MS's buffer, if the number of tones required by the selected MS

is less than the total number of tones in this symbol, the left tones will be vacant.

So, this is not the most efficient uses of resource.

- In OPF, adaptive number of symbols, ranging from one to maximum number of available symbols in the frame, can be assigned to a particular selected MS depending on the criterion and packet size. It is possible that the whole frame is assigned to only one MS, or be shared among multiple MSs. It is also possible that some portion of frame is left unused in some scheduling case when the system load is extremely small. Similar to MaxSNR, the OPF algorithm permits a previously selected MS to be re-scheduled in the following adjacent symbol and frame, so long as that MS meets the criterion at the decision point in that symbol or frame.

6.6 PF with observation window and weighted PF

We also implement other two versions of PF in the thesis for comparison purpose. They are PF with observation window and PF with DRC exponent rule, which we call weighted PF for short. As mentioned in Chapter 3, there are several versions of PF and they are evaluated in TDMA/CDMA systems. To compare the performance of OPF with the previous works, we implement those two algorithms in our 802.16a OFDM system.

6.6.1 PF with observation window

Proposed in [3], the performance of PF is a function of the observation window T_c as described in equation 3-1 and 3-2. In [2], this algorithm is implemented in CDMA system. Here we modify it and apply it to our 802.16a OFDM system. In [44], it is shown that the system throughput is affected by T_c , and larger T_c will result in better system throughput. The maximal setting of T_c is 7000 in that paper. Under this setting, system reaches its highest throughput. In our simulation we choose the T_c to be 9000 and compare the performance with OPF. In other words, we would like to compare OPF with the best performance of PF. If the OPF throughput is better than PF with T_c equals to 9000, we could conclude that OPF outperforms PF with T_c . Simulation results will be shown in Chapter 8.

6.6.2 Weighted PF

In [31], an exponent term is added to the part of data rate in equation 3-1. Through this approach, some weighting is added to the channel condition. The criterion becomes equation 3-3 for this weighted PF algorithm, where n is the control parameter. If n is lower than 1, the supportable data rate becomes smaller. It means that the algorithm favors the users with bad channel conditions. In this case, fairness is reached to the most extent with the sacrifice of system throughput. If n is greater than 1, supportable data rate is increased. Users in good channel condition get more benefits. In this case, throughput is improved while it is not fair to the users in bad channel condition.

Paper [31] considers the case of n is equal or higher than 1 under the consideration of improving the system throughput. In this thesis, we implement this algorithm in 802.16a OFDM system. We are addressing the fairness and hence we set n to be smaller than 1. The performance of this weighted PF is evaluated and compared with OPF in Chapter 8.

7 Description of Implementation with Simulation Tool

For the implementation of the OFDM broadband wireless system, we use OPNET [69] as the system level software package. In this single cell, there is one BS located at the cell center. Several MSs are uniformly distributed within the cell radius. Only downlink is simulated.

We will give a brief description on OPNET implementation. In the description, we show how to use OPNET to simulate the OFDM based wireless communication system fully compliant with IEEE 802.16a standard. We choose a typical wireless mobile environment for urban and suburban areas. It is outdoor, with high-rise core, BS antenna height of 15m above average rooftop, and cell radius is several kilometers.

7.1 OPNET implementation

OPNET is a popular software tool for system level simulation. It is event-driven and built on Pro-To C, which is an OPNET language. It is quite similar and fully compatible to C and C++. The system structure is based on level construction. Usually one project includes several subnets; each subnet is composed of nodes; each node contains many modules which can implement any functions according to design.

For each module, it is presented by different state, and in each state, a special code is created to simulate various function according to requirements. States, modules, nodes, subsets are linked together through variously interruptions, remote access and traffic flows. In the following sections, we'll give a brief description on implementation of the joint PHY and MAC layers which were described in previous sections.

7.1.1 OPNET single cell description

Figure 7-1 shows a single cell network that consists of one base station (BS) node and four mobile stations (MS). As mentioned in the previous section, the number of MS could be set to other number to fit for real scenario. The maximal number of MS is configurable.

Figure 7-1 Single cell topology of the OFDMA OPNET simulator

7.1.2 BS node description

Now let us look inside at the BS. Figure 7-2 is the block diagram of the BS node. In the figure, the lines are used to show the relations among the modules.

Figure 7-2 Block diagram of the base station (BS) node

One of the important design goals is to provide flexibility in the simulated system. According to the BS node diagram, we will describe the feature of each module. In our design, almost all the control parameters are configurable. They can be easily changed to meet other system requirements for any research purpose.

- Call Generator

The `Call_Generator` is used to simulate the call arrival and specify the call duration.

- Call admission control

The “`Call_Admission_Ctrl`” process is used to handle the call arrival event, which might be from a new call, or from the hand-over (for the future use in the multiple cell environments). Currently, this process handles the call arrival event from the “`Call_Generator`”. The maximal number of calls it accepts is equal to the total number of users in our current setting. Or, one user can only accept one call.

- BS node control

The “`BS_Node_Ctrl`” is used to supervise the overall BS node and handle mobile node registration, call arrival, and call termination.

- Traffic generator

The “`Traffic_Generator`” is used to simulate the traffic source. Note that this is useful in the downlink from the BS to the MS only. As previously mentioned in the “`BS_Node_Ctrl`”, the “`Traffic_Generator`” will dynamically create/kill a child

process named “ofdma_dynamic_traffic_source” (the name of the child process in this module) for the MS corresponding to any newly admitted/terminated call.

Please note that the call generator is handling the activation of calls, while traffic generator is to generate different kind of traffic, or type of calls. This is the difference between call generator and traffic generator.

- BS data base

The “BS_DataBase” is used as a database to record the status of the mobile, channel information and resource allocation information. The resource means the time and frequency resource, more specifically, the number of OFDM symbols in one frame, the number of tones in one OFDM symbol, and for each tone, how many number of bits are transmitted. The allocation will be done in the dynamic channel allocation (DCA) process node. As previously mentioned in the “BS_Node_Ctrl”, it can allocate/release the memory for the MS corresponding to any newly admitted/terminated call. Please note, we set the number of tones the same as the number of the FFT size. The reason we do so is as follows: In the DCA, we will do allocation on frame by frame basis. The total number of tones per frame would be number of time slot * number of tones, if within one time slot only one OFDM symbol is transmitted. However, in practice, within one time slot, we might transmit more than one OFDM symbols, resulting in the total available tones per frame = number of time slots * number of OFDM symbols per time slot

* number of tones per OFDM symbol. The resulting number might be too large to handle. Therefore, we decide to do allocation (as further explained in the DCA) symbol by symbol. It should be clarified that in the context of this thesis, the time slot is the same as OFDM symbol duration. Through this approach, number of available tones is set to FFT size only.

- Queue manager

The “Queue_Manager” is used to handle the unmatched rate between the data link layer and the physical layer. The incoming packets generated by the child processes of the “Traffic_Generator” will be queued up here and wait for the dynamic channel allocation to serve them. Each MS has its own queue. We set this queue length as infinite but actually the traffic could not go to that extreme because we have the mean traffic rate to control the traffic generation. Each packet is stamped with the time of its generation. When this packet arrives at the MS receiver, the arriving time is recorded. The arriving time minus the time of its generation is the packet delay. Because of the cell radius is only several kilometers, the transmission time of the electromagnetic wave in the air is neglected, i.e., the propagation delay is neglected. Hence, the mean packet delay is equal to the queuing delay. This is how we do the delay measurements.

- dynamic channel allocation (DCA)

The “DCA” is for dynamic channel allocation, which is used to look into the service request in “Queue_Manager” and the channel information in the

“BS_DataBase” and try to do the best allocation”. Please note, we do allocation on symbol by symbol basis and the “DCA” self interruption period is symbol interval. The allocation information will be recorded for each MS in the database at the “BS_DataBase” process. “DCA” will directly write information to the database.

As DCA plays an important role in our system. It is the center part of this thesis. Several parallel DCA algorithms are implemented in this module based on the description of each algorithm in Chapter 6.

- BS physical transmitter

The “BS_Phy_Tx” is used to take care of the other physical layer characteristics, such as transmit power, antenna gain, radio frequency etc, and it will directly modify the record in the database of the “BS_DataBase”. Currently, we have not considered the antenna gain yet and therefore the “BS_Phy_Tx” is not actually affecting our simulation.

- BS meter

The “BS_Meter” is created for the statistics collection of all the mobile stations (MSs) in the current cell. As we know, in the OPNET, one can choose the individual statistics to record the simulation results. Unfortunately, the individual statistics have to be done by setting node by node. When the number of the MS nodes becomes large, to set and examine the individual statistics one by one is

becoming tedious and time-consuming. Therefore, we created this “BS_Meter” to automatically collect the statistics of all the MS nodes in the BS. We currently set the number of mobiles to be monitored to 20.

7.1.3 MS node description

For detailed implementation of the MS, Figure 7-3 is the block diagram of the MS node. In the figure, the lines are used to show the relations among the modules. According to the MS node diagram, we will describe the function of each module. Under the same design as BS, almost all the control parameters are configurable. They can be easily changed to meet other system requirements for any research purpose.

Figure 7-3 Block diagram of the MS node

- MS node control

The “MS_Node_Ctrl” is used to supervise and control the operation of the MS node. It does the following tasks: In the initial state, it sends a remote interruption to the “BS_Node_Ctrl” to register this MS. After that, this MS enters to the non-in-call state. In the “non-in-call” state, the MS will wait for the notification from the “BS_Node_Ctrl” as previously mentioned for call activation. When the notification from the “BS_Node_Ctrl” to the “MS_Node_Ctrl” is received, the state is transitioned to the “in_call” state. Immediately, a self-interruption event will be scheduled according to the characteristic of the call duration specified in

the “Call_Generator” and the OPNET timer will start to count. When the self-interruption for this “MS_Node_Ctrl” is received, it will check if this interruption is for call termination. If it is, an interruption request will be sent to the “BS_Node_Ctrl” to notify the call termination and the state of the “MS_Node_Ctrl” will be back to “non_in_call” state.

- MS data base

The “MS_DataBase” is used to hold the channel information and the mobile position. The channel information consists of three parameters: number of paths, channel fading model, and path loss model. The channel response means both time and frequency. We generate the channel time response according to the channel description in section 4.2.5. Frequency response is calculated through Fourier Transform. The mobile position is represented by the pair of “distance” and “angle” between this MS and the BS. Inside the “update” state of the “MS_DataBase”, the mobile position can be changed.

- MS channel generation

The “MS_Chan_Gen” is used to handle the channel information from the BS to this MS. For the simple way, random variables are generated for channel impulse response which has already considered frequency selective (as there are more than one path) and time-varying too as each call of “update” will generate a new set of random variables for channel impulse response. For a more complicated case, time-varying channel of Jake’s is also implemented as described in section 4.2.5.

For each interruption, the channel information will be updated and recorded in the “MS_DataBase”.

- MS physical receiver

The “MS_Phy_Rx” is the actual physical layer receiver. It will Access the “BS_DataBase” to see the bit allocation information, and access the mobile position and channel information in “MS_DataBase” to calculate the path loss and fading effect. Calculate the received SNR per subcarrier (tone) and simulate the number of bits in each sub-carrier (tone). Collect the statistics to record the number of bits received and number of bits in error. Please note all those statistics can be accessed by the “BS_Meter” as previously mentioned. Therefore, we can use the “BS_Meter” to access those statistics.

7.2 C++ implementation

Since OPNET is multi-thread based, it is hard for debug. After building the OPNET simulation, it is not difficult to transform it into C++. So, we also construct C++ platform. Compared with OPNET, C++ has these advantages:

- C++ is very popular, but OPNET is not accessible for everyone;
- The flexibility of C++ make it easy to transform this system to almost all wireless systems. From this point of view, it can provide potential research value;
- C++ is single thread based, it is easier to debug;

- Visual C++ provides very rich debugging environment.

All simulation results can be obtained both from OPNET and C++ program; we compare the results to guarantee the coding is correct. At the same time we do the verification by theoretical calculation. The output numerical simulation results are written to data files. We import the data files to Matlab and plot all the figures.

8 Preliminary Simulation Results and Discussions

Both the MAC and PHY layer are jointly simulated. As mentioned before, the simulation is implemented using OPNET and C++. OPNET does all the drawings in an automated fashion, but in our case, we have some special requirements such as user satisfaction rate and average user rate. These statistics are not part of the OPNET automatic generated program. So, we output the scalar file containing the data we are interested in, and use Matlab to draw the figure. Here in this section, we first list the parameter setting in the simulation, and then some discussions and the preliminary simulation results will follow.

8.1 Simulation parameters

Here are all the parameter settings.

- System level assumption:

Cell radius: 1.5 kilometer

Number of MSs: 10

MS distribution: uniformly distributed within cell radius.

- Frame duration and OFDM parameters: Table 4-1.
- Packet size: 180bits/packet.

- Path loss parameters:

Pass loss model:

$$L = 40(1 - 4 \times 10^{-3} \Delta h_b) \log_{10} R - 18 \log_{10} \Delta h_b + 21 \log_{10} f + 80 [dB]$$

Antenna height: 15m.

BS and MS distance: MS uniformly distributed within cell radius which is 1.5 kilometer.

Shadowing model: lognormal distribution with 10dB standard deviation.

- Channel parameters:

Channel model: time varying multi-path channel with Rayleigh distribution for each path;

Number of multipath components: 3.

Number of arriving angle in Jake's model for each path: 34 [55].

- Traffic model: all MS have the same traffic model, which is exponential distribution.

- IEEE 802.16a standard parameters:

Radio frequency: 2.500-2.520 GHz

Bandwidth: 20MHz

FFT size: 256

CP percentage: 25% of symbol duration

Frame overhead percentage: 25% of total number of symbols

Symbol duration: $16\mu\text{s}$

Frame length: 2ms

Coding scheme: not implemented yet.

Modulation scheme: adaptive, with 1 to 6 bits per symbol according to channel condition.

- Threshold of user satisfaction rate: 50%.
- Other parameters:

Input PSD (power spectrum density): -145dBm/Hz

Background noise floor PSD: -170dBm/Hz

SNR margin (to combat any unexpected interference): 3dB

SNR required (to guarantee symbol error rate of 10^{-4}): 7dB

- Weighted PF:

n: 0.5

- PF with observation window:

$T_c=9000$

8.2 Evaluation of PF with observation window

As described in section 6.6, we implement the original PF which depends on observation window in OFDM system. In this section, we will evaluate its performance under different settings of the observation window T_c . We will choose a specific value of T_c in the next section to compare the performance with other algorithms.

With parameters listed in section 8.1, we obtain the throughput of PF with observation window under different T_c . As shown in Figure 8-1, OPF provides the best throughput when traffic load is heavy, and almost the same with PF with different T_c when traffic is light.

Compared with paper [44] which implements PF with observation window as frequency domain scheduler, there is an interesting phenomena. In figure 2 of that paper, it shows system throughput is increased with the value of T_c . While when we implement PF with observation window as time domain scheduler, the simulation results do not agree with the results in reference [44]. We cannot observe mono increase or decrease of throughput with T_c as shown in Figure 8-1. The reason of this phenomena needs further investigation in the near future. But we could see that our modified OPF which eliminate the impact of T_c provides the highest system throughput.

In the following section, we choose $T_c=9000$ for comparison purpose.

Figure 8-1 Throughput of OPF and PF with observation window

(Reference file name: julie012105code)

8.3 Preliminary Simulation results of OPF and known algorithms

In this section, we compare the performance of our OPF and known algorithms. Known algorithms include conventional ones, namely RR and MaxSNR, and two versions of PF, namely PF with observation window and weighted PF.

Performance is evaluated from two aspect: efficiency and fairness. Efficiency is presented in terms of system throughput versus traffic load, delay mean and variance versus throughput. Fairness is presented in terms of user average rate and user satisfaction rate versus throughput.

Figure 8-2 System throughput of the 5 algorithms

(Reference file name: julie011205code)

In Figure 8-2, throughput vs. traffic load is shown. Here the traffic load is defined as the newly generated traffic, not including the accumulated traffic. Exponential distribution traffic model is applied to all users. As expected, MaxSNR gives the upper limit of system throughput, with RR as the lower limit. OPF improves the system throughput by 10Mbps compared to RR. Compared with weighted PF and PF with observation window, OPF has the highest throughput when traffic load is larger than 40 Mbps. In other word, OPF has its throughput advantage over PF in heavy traffic.

It is noticed that in the light traffic end, the throughput is slightly higher than the traffic load. This is due to the processing technique of raw data. Simulation result is saved as data file and then is imported to Matlab. We use the Curve Fitting properties in Matlab. We choose polynomial curve fitting with degree of 2, thus there might be slight difference between the curve and the raw data. The trend and shape of the curve reflects the nature of the raw data. The raw data is available upon request.

Figure 8-3 Mean delay of the 5 algorithms

(Reference file name: julie011205code)

Figure 8-3 is the mean delay of the 5 algorithms. Compared with RR and MaxSNR, OPF gives better delay performance when throughput is under 35Mbps. Even in the heavy traffic, the delay is considerable improved compared with RR. Compared with PF, OPF gives less delay when traffic is heavy but not as good as PF in light traffic. Figure 8-4 is the delay variance and the curve of OPF has the same trend as the mean delay.

Figure 8-4 Delay variance of the 5 algorithms

(Reference file name: julie011205code)

Figure 8-5 User satisfaction rate of the 5 algorithms

(Reference file name: julie011205code)

To evaluate the user satisfaction rate which is defined in Chapter 4, we need to define the thresholds. Now the threshold is set to 0.5, this means when a user receives 50% of its generated traffic, this user is classified as satisfied. In the above figure, user satisfaction rate is evaluated.

Compared with RR, OPF provides higher user satisfaction rate all the time. Compared with MaxSNR, OPF performs better when throughput is smaller than 32 Mbps as

shown in Figure 8-5. The criterion of OPF indicates that the scheduler not only favors those MSs with high supportable rate, but also its experienced rate. It optimizes the system throughput, at the same time, it also gives enough fairness to MSs whose channel is not so good in the past. In other words, if a user does not receive service at previous frame, e.g., due to its bad channel condition, then its precedence will increase at the next frame. This equals to that the served user will have less chance to access the next frame. Therefore, a user will not “starve” too long even if its channel condition is not so good.

Compared with other two versions of PF, OPF is not as good as them when throughput is less than 30 Mbps and it performs better when in heavy throughput. This can be seen from the criterion described in equation 3-3: light throughput means smaller supportable rate. With an exponent smaller than 1, the value of equation 3-3 gets smaller. In other word, this favors the users with bad channel condition and hence user satisfaction rate is high. On the opposite, when traffic is heavy, user in good channel condition gets more benefit and the user satisfaction rate is low compared to OPF. The same explanation can elaborate why OPF outperforms weighted PF in terms of average user rate in Figure 8-6.

As shown, the fairness performance of RR is the worst. Previously, when we mentioned that RR provides better fairness is from the resource point of view. While

currently, we evaluate the fairness based on the definition of user satisfaction rate, i.e., from the performance point of view.

Figure 8-6 Average user rate of the 5 algorithms

(Reference file name: julie011205code)

In Figure 8-6, compared with RR and MaxSNR, OPF gives highest average user rate in all range of the throughput observed. From this feature, OPF outperforms RR and MaxSNR by providing fairness among all users in terms of highest average user rate.

Compared with PF with observation window, OPF gives better average user rate when throughput is larger than 30 Mbps. Compared with weighted PF, OPF is better when throughput is larger than 27 Mbps.

In summary, we may conclude that OPF outperforms the two versions of PF in terms of higher efficiency and fairness when system throughput is larger than around 30 Mbps. But OPF is slightly worse than PF when throughput is under 30 Mbps.

9 Proposed Algorithms for Multimedia and Analysis

So far, we implemented OPF in OFDM system and showed its performance is better than PF. In this section, we are going to propose some new algorithms and evaluate their performance.

9.1 New algorithm for multimedia services

New criterion will lead to new algorithm. As multimedia is one of the key features of proposed 3G/4G wireless system, we set our goal to develop new algorithms which are suitable for multimedia services. Because the scheduling procedure and its OFDM implementation are the same for all algorithms and are described in Chapters 4 and 6, here we only describe the criteria of the newly proposed algorithms. Simulation results will follow.

9.1.1 Adaptive OPF: AOPF

Inspired by the APF algorithm proposed in [32], we may add an adaptive exponent to the numerator of our OPF algorithm and name it AOPF.

- The design goal is:
 - This adaptive exponent is changing at each symbol for each user, to capture the fast changing wireless channel and also to provide further measure of fairness as the rate is updated at each symbol;

- For each user, we try to equalize the value of j (in equation 6-4) of the current symbol to the previous symbol, under the consideration that the algorithm performs well enough in the past;
- Give some bias on the supportable rate to improve the system throughput.

We will have the simulation results to see if these goals are reached and to show the tradeoff between throughput and fairness.

- AOPF criterion:

$$G_I(t) = \max_{1 \leq i \leq N} \{G_i(t)\} \quad \text{Equation 9-1}$$

where I is the selected user index, i is the user index, and the total number of users is N . It is a function of time t :

$$G_i(t) = \alpha_i \frac{D_i^{n_i}(t)}{\overline{R_i(t)}} \quad \text{Equation 9-2}$$

where $D_i(t)$ and $\overline{R_i(t)}$ are supportable rate and average received rate respectively, they are the same definition as in OPF. $D_i(t)$ is updated on a frame-by-frame basis, while $\overline{R_i(t)}$ is updated on a symbol-by-symbol basis. Thus $G_i(t)$ is calculated symbol-by-symbol. In other words, based on the result of $G_i(t)$, at each symbol, a user is selected. α_i equals to 1 when this user is active and equals to 0 when it is not, that is:

$$\alpha_i = \begin{cases} 0 & \text{non-active} \\ 1 & \text{active} \end{cases} \quad \text{Equation 9-3}$$

The n_i is the newly introduced adaptive exponent, it is also a function of time t , and it is updated symbol by symbol. To simplify the problem, we use another index of j to represent time. Because in OFDM system, as shown in Figure 4-5, the time axis is composed of frames and OFDM symbols, j could be regarded as symbol index. Under the assumption that all users are active, now we can simplify the criterion as:

$$I = \arg \max_{\substack{1 \leq i \leq N \\ 1 \leq j \leq M}} \frac{D_{i,j}^{n_{i,j}}}{R_{i,j}} \quad \text{Equation 9-4}$$

where I is the selected user, i is the user index, j is the symbol index, N is the total number of users, and M is the total number of symbols in each frame.

- Mathematical model:

In AOPF, the crucial part is to obtain the adaptive exponent $n_{i,j}$ for each user at every symbol. Here we give step by step deduction of obtaining this $n_{i,j}$. Suppose there are total N users and there are M symbols in each frame, we can represent the criterion matrix as:

$$\begin{bmatrix} \frac{D_{1,0}^{n_{1,0}}}{R_{1,0}} & \frac{D_{1,1}^{n_{1,1}}}{R_{1,1}} & \dots & \frac{D_{1,j}^{n_{1,j}}}{R_{1,j}} & \dots & \frac{D_{1,M}^{n_{1,M}}}{R_{1,M}} \\ \frac{D_{2,0}^{n_{2,0}}}{R_{2,0}} & \frac{D_{2,1}^{n_{2,1}}}{R_{2,1}} & \dots & \frac{D_{2,j}^{n_{2,j}}}{R_{2,j}} & \dots & \frac{D_{2,M}^{n_{2,M}}}{R_{2,M}} \\ \vdots & \vdots & & \vdots & & \vdots \\ \frac{D_{i,0}^{n_{i,0}}}{R_{i,0}} & \frac{D_{i,1}^{n_{i,1}}}{R_{i,1}} & \dots & \frac{D_{i,j}^{n_{i,j}}}{R_{i,j}} & \dots & \frac{D_{i,M}^{n_{i,M}}}{R_{i,M}} \\ \vdots & \vdots & & \vdots & & \vdots \\ \frac{D_{N,0}^{n_{N,0}}}{R_{N,0}} & \frac{D_{N,1}^{n_{N,1}}}{R_{N,1}} & \dots & \frac{D_{N,j}^{n_{N,j}}}{R_{N,j}} & \dots & \frac{D_{N,M}^{n_{N,M}}}{R_{N,M}} \end{bmatrix}$$

Equation 9-5

In the current frame, based on the assumption that the channel remains constant during the frame time, $D_{i,j}$ remains constant for the i^{th} user. While $\overline{R_{i,0}} \dots \overline{R_{i,j}}$ varies after each symbol allocation. Now, the only unknown parameter in the criterion matrix becomes the adaptive exponent matrix:

$$\begin{bmatrix} n_{1,0} & n_{1,1} & \dots & n_{1,M} \\ n_{2,0} & n_{2,1} & \dots & n_{2,M} \\ \vdots & \vdots & & \vdots \\ n_{N,0} & n_{N,1} & \dots & n_{N,M} \end{bmatrix}$$

Equation 9-6

When regarding OPF as a special case of AOPF when the exponents in equation 9-6 equals to 1, we set the initial value of the exponent for each user as 1. That is:

$$\begin{bmatrix} n_{1,0} \\ n_{2,0} \\ \vdots \\ n_{N,0} \end{bmatrix} = \begin{bmatrix} 1 \\ 1 \\ \vdots \\ 1 \end{bmatrix}$$

Equation 9-7

Let us take row 1 of equation 9-5 as an example to see how to obtain the value of the adaptive exponent $[n_{1,0} \ n_{1,1} \ \dots \ n_{1,M}]$ for user 1.

If we let $c_{1,0} = \frac{D_{1,0}^{n_{1,0}}}{R_{1,0}} = \frac{D_{1,0}}{R_{1,0}}$ to be the initial value for this user in symbol 0, then

for the next symbol, we are trying to obtain $n_{1,1}$ based on this $c_{1,0}$:

$$c_{1,0} = \frac{D_{1,1}^{n_{1,1}}}{R_{1,1}} \quad \text{Equation 9-8}$$

$$\log c_{1,0} = n_{1,1} \log D_{1,1} - \log \overline{R_{1,1}}$$

$$n_{1,1} = \frac{\log c_{1,0} + \log \overline{R_{1,1}}}{\log D_{1,1}} \quad \text{Equation 9-9}$$

Similarly, for this user at symbol j , we have:

$$n_{1,j} = \frac{\log c_{1,(j-1)} + \log \overline{R_{1,j}}}{\log D_{1,j}} \quad \text{Equation 9-10}$$

In general, for user i in symbol j , we have:

$$n_{i,j} = \frac{\log c_{i,(j-1)} + \log \overline{R_{i,j}}}{\log D_{i,j}} \quad \text{Equation 9-11}$$

Thus, we determine the adaptive exponent of each user, and it is updating symbol by symbol. In the next scheduling event or in other words, in the next frame, except updating the $R_{i,j}$ and $n_{i,j}$ at each symbol time, $D_{i,j}$ is also updated with the

new channel information. The same method is used to calculate all $n_{i,j}$ for the new

frame and we do not need to reset the $\begin{bmatrix} n_{1,0} \\ n_{2,0} \\ \vdots \\ n_{N,0} \end{bmatrix}$ to all “1”.

In AOPF, the purpose of adjusting $n_{i,j}$ is to try to equalize $\frac{D_{i,j}^{n_{i,j}}}{R_{i,j}}$. This

consideration will provide higher accuracy from the sense of location, channel and experienced rate. Simulation results and performance comparisons with OPF will be presented in the next section.

- Algorithm description:

Step 1) initialize $\overline{R_{i,j}} = 0$, $n_{i,j}=1$;

Step 2) initialize symbol index=1;

Step 3) calculate $D_{i,j}$;

Step 4) if $\overline{R_{i,j}} = 0$, $I = \arg \max_{\substack{1 \leq i \leq N \\ 1 \leq j \leq M}} D_{i,j}$;

Step 5) if $\overline{R_{i,j}} \neq 0$, $I = \arg \max_{\substack{1 \leq i \leq N \\ 1 \leq j \leq M}} \frac{D_{i,j}^{n_{i,j}}}{R_{i,j}}$;

Step 6) choose the user with the largest selection index I ;

Step 7) do bit allocation for the selected user;

Step 8) update the selected user's $\overline{R_{i,j}}$;

Step 9) update symbol index;

Step 10) if symbol index < total number of available symbol, go to Step 4);

Step 11) if symbol index $\geq$ total number of available symbol, go to Step 2).

9.1.2 Multimedia Adaptive OPF: MAOPF

- The design goal:

In OPF and AOPF, we only consider the supportable data rate and experienced data rate. So far, we did not take the data rate requirement of each user into account. This may be fair enough when all users have the same data rate requirement. This may not be fair when users have different data rate requirements which is an important feature for multimedia services. The design goal of MAOPF is to define a new criterion which takes data rate requirements into consideration.

- MAOPF criterion:

Considering the requirement of different data rates, at the same time the need to provide fairness without significant reduction in system throughput, we incorporate another parameter $R_{l,r}$, which is the required data rate in the

scheduling criterion. We name it Multimedia Adaptive OPF (MAOPF). Here its criterion:

$$G_I(t) = \max_{1 \leq i \leq N} \{G_i(t)\} \quad \text{Equation 9-12}$$

$$G_i(t) = \alpha_i \frac{D_i^{n_i}(t)}{\left(\frac{R_i(t)}{R_{i_r}(t)} \right)} = \alpha_i \frac{D_i^{n_i}(t) \cdot R_{i_r}(t)}{R_i(t)} \quad \text{Equation 9-13}$$

where I , i , N , $D_i(t)$, $\overline{R_i(t)}$, α_i , n_i have the same definition as in AOPF.

$R_{i_r}(t)$ is the required rate of user i and is updated at each symbol time.

Repeating equation 9-4, under the assumption that all users are active (i.e., $\alpha_i=1$ for all users), now we can simplify the criterion as:

$$I = \arg \max_{\substack{1 \leq i \leq N \\ 1 \leq j \leq M}} \frac{D_{i,j}^{n_{i,j}} \cdot R_{i_r,j}}{R_{i,j}} \quad \text{Equation 9-14}$$

The user selection is now based not only on supportable rate and average experienced rate, but also on user required rate. This means that when a user requires more, it has more chance to be selected. This would provide fairness to users with different rate requirement in multimedia environment. This novel algorithm is a new contribution in scheduling for multimedia services in OFDM broadband wireless systems.

- Mathematical model:

Following the same procedure as in AOPF and similar to equation 9-11, we have:

$$n_{i,j} = \frac{\log c_{i,(j-1)} + \log \overline{R_{i,j}} - \log R_{i-r,j}}{\log D_{i,j}} \quad \text{Equation 9-15}$$

where the initial value is: $c_{1,0} = \frac{D_{1,1}^{n_{1,1}} \cdot R_{1-r,1}}{R_{1,1}} = \frac{D_{1,1} \cdot R_{1-r,1}}{R_{1,1}}$, under the setting of

$n_{i,0} = 1$ for all users.

- Algorithm description: is the same as AOPF except the criteria.

9.1.3 Normalized Multimedia Adaptive OPF: NMAOPF

- The design goal:

In AOPF and MAOPF, we noticed that there exists two problems: 1) The value of $G_i(t)$ is very large. As shown in equation 9-2 and equation 9-13, it is proportional to the exponent of $D_i(t)$ which is several dozens of Mbps. 2) The simulation results show (in the next section) that both AOPF and MAOPF cannot provide better users satisfaction rate and average user rate, even though they do give better throughput and delay statistics. Trying to solve the two problems, lead us to:

- Introduce normalization factor that would reduce the value of $G_i(t)$;
- To improve the user satisfaction rate.

- NMAOPF criterion:

We name this new criterion Normalized MAOPF: NMAOPF. The criterion is:

$$G_I(t) = \max_{1 \leq i \leq N} \{G_i(t)\} \quad \text{Equation 9-16}$$

$$G_i(t) = \alpha_i \frac{\left[\frac{D_i(t)}{\sum_{i=1}^N D_i(t)} \right]}{\left[\frac{\overline{R_i(t)}}{R_{i-r}(t)} \right]^{n_i(t)}} = \alpha_i \left[\frac{R_{i-r}(t)}{\overline{R_i(t)}} \right]^{n_i(t)} \left[\frac{D_i(t)}{\sum_{i=1}^N D_i(t)} \right] \quad \text{Equation 9-17}$$

where $I, i, N, D_i(t), \overline{R_i(t)}, R_{i-r}(t), \alpha_i, n_i(t)$ have the same definition as in MAOPF. As in equations 9-4 and 9-14, under the assumption that all users are active, now we can simplify the criterion as:

$$I = \arg \max_{\substack{1 \leq i \leq N \\ 1 \leq j \leq M}} \left[\frac{R_{i-r,j}}{\overline{R_{i,j}}} \right]^{n_{i,j}} \left[\frac{D_{i,j}}{\sum_{i=1}^N D_{i,j}} \right] \quad \text{Equation 9-18}$$

Compare equation 9-17 with equation 9-13, there are two changes: 1) We introduced $\sum_{i=1}^N D_i(t)$ as the denominator of $D_i(t)$. This could be regarded as normalization process which solves the large magnitude problem in AOPF and

MAOPF. Since the $\sum_{i=1}^N D_i(t)$ is the same to all users, it will not have any impact on user selection. 2) We moved the adaptive exponent to the ratio of required rate and average rate instead of to the supportable rate. This is an effort to solve the second problem raised in AOPF and MAOPF.

- Mathematical model:

Aiming at improving user satisfaction rate which is defined in equation 4-22 and

equation 4-23, we let $\left[\frac{\overline{R_i(t)}}{R_{i_r}(t)} \right]^{n_{i,t}}$ equals to the predefined threshold σ_i . That is:

$$\left(\frac{R_{i_r,j}}{R_{i,j}} \right)^{n_{i,j}} = \frac{1}{\sigma_i}$$

Through this approach, we can reach our second design goal. Here is how to obtain $n_{i,j}$:

$$n_{i,j} = \frac{\log \frac{1}{\sigma_i}}{\log R_{i_r,j} - \log R_{i,j}} \quad \text{Equation 9-19}$$

where σ_i is the predefined threshold, the same definition as in equation 4-23. It should be stated that the threshold is not the same for different type of traffic, e.g., in voice communication, even 10% of the traffic are dropped, the quality of voice is still acceptable, while in file transfer, 100% of successful transfer is required. Hence in equation 9-19, the adaptive exponent is a function of σ_i . Here, the user

index i represents the traffic type. This NMAOPF is expected to give the best performance among all the existing algorithms for multimedia services.

- Algorithm description: is the same as AOPF except the criteria.

9.2 Simulation results and discussions under multiple rate traffic

9.2.1 Simulation parameters of different rate

The goal of the 3 proposed algorithms is to support multimedia services. To evaluate the performance, we start with the simplest form of multimedia traffic: same traffic model but with two different data rates.

Compared with the parameter setting in section 8.1 for preliminary simulation, we keep all the parameters the same except for two changes:

- System level assumption: Now the number of users is changed from 10 to 15. The 15 users are divided into two groups: high data rate group of 8 users and low data rate group of 7 users.
- Traffic model: all MS have the same exponent distribution traffic model, but with two different rates. The high data rate is 10 times the low data rate.

9.2.2 Simulation results and discussions

In this section, we only compare the performance of RR, MaxSNR, OPF and the 3 proposed algorithms. We omit the PF and weighted PF in this part.

Figure 9-1 System throughput comparison

(Reference file name: julie032605code)

In this figure, it shows that AOPF, MAOPF outperforms OPF when traffic load is above 70 Mbps. NMAOPF is not only better than OPF within all the observation range of traffic load, but also better than the conventional MaxSNR which is supposed to be the upper limit of throughput. This may be the case when the ratio of required

rate and average rate is larger than 1, and also the adaptive exponent is large than 1. So, NMAOPF exceeds the conventional upper bound of throughput. It provides the highest efficiency compared with all existing algorithms implemented in this thesis.

Figure 9-2 Mean delay comparison

(Reference file name: julie032605code)

Compared with OPF this figure shows that all of the 3 proposed algorithms improve the mean delay when throughput is smaller than 32 Mbps and larger than 47 Mbps. They are worse than OPF in between. Their improvement in the sequence from best

to good is: NMAOPF, MAOPF, AOPF. Especially, NMAOPF is almost as good as the lower bound MaxSNR when throughput is larger than 47 Mbps.

Figure 9-3 Delay variance comparison

(Reference file name: julie032605code)

The delay variance has the same feature as the mean delay, with NMAOPF has almost the same performance as MaxSNR.

Figure 9-4 User satisfaction rate comparison

(Reference file name: julie032605code)

From this figure, we observe that even AOPF and MAOPF improve the throughput, mean delay and delay variance, but they cannot give better user satisfaction rate. On the contrary, their user satisfaction rate is worse than RR, MaxSNR and OPF. In other word, AOPF and MAOPF improve system throughput with the cost of fairness.

While NMAOPF outperforms RR and MaxSNR, and it is almost as good as OPF when throughput is under 43 Mbps. Unfortunately it is worse than OPF when

throughput is larger than 43 Mbps. So far, we can see that NMAOPF improves the throughput and delay statistics without sacrificing the user satisfaction rate.

It is noticed that within some range the user satisfaction rate is slightly higher than 1. This is due to the processing technique of raw data. Simulation result is saved as data file and then is imported to Matlab. We use the Curve Fitting properties in Matlab. We choose polynomial curve fitting with degree of 2, thus there might be slight difference between the curve and the raw data. The trend and shape of the curve reflects the nature of the raw data. The raw data is available upon request.

Figure 9-5 Average user rate comparison

(Reference file name: julie032605code)

From Figure 9-5 Average user rate comparison, we can see that AOPF and MAOPF cannot provide good average user rate, but even worse. Their average rate performance is inferior. Fortunately, NMAOPF not only outperforms RR and MaxSNR when throughput is larger than 35 Mbps, but is also slightly better than OPF between 35 and 55 Mbps of throughput. It is almost the same as OPF when throughput is above 55 Mbps. In other words, NMAOPF gives the highest average user rate among all existing algorithms in multi-rate simulation environment.

For the performance of the 3 proposed algorithms, we may conclude that: AOPF and MAOPF improve system throughput, mean delay and delay variance, but they have worse user satisfaction rate and average user rate. As expected, NMAOPF refreshes new upper bound of system throughput, improves delay statistics at considerable range of throughput, as well as offers higher user satisfaction rate compared with RR and MaxSNR, and gives highest average user rate among all existing methods discussed. In summary, NMAOPF improves the efficiency without sacrificing fairness. It provides better tradeoff between throughput and user fairness. It can be a competitive candidate for 3G/4G OFDM based broadband wireless system.

10 Evaluation of the Proposed Algorithms

In this section, we evaluate and compare all the algorithms under real multimedia traffic. At the same time, we show the impact of the wireless channel on the performance of all the algorithms.

10.1 System performance under multimedia traffic

In Chapter 8, the traffic model is the simplest one: all users have the same type of traffic and the rates are the same. In section 9.2 we go a step further, all users have the same type of traffic but two different rates. This is the simple form of multimedia traffic. Multimedia service is one of the features in 3G/4G wireless networks. Systems should give good performance for all types of traffic. Now, in this section, we evaluate the algorithms in real multimedia traffic.

10.1.1 Traffic model

Typical applications for multimedia traffic are web browsing, MPEG streams and voice applications. We choose these three types of traffic as multimedia traffic. Here we describe each of the traffic types as follows:

- Voice source: constant bit rate (CBR) voice source is considered with 64 kb/s Pulse Code Modulation (PCM), as adopted in [70].
- Web traffic: as in [71], web traffic behavior is modeled by a sequence of activity periods that reproduces the entire Web page download. In each of the activity

periods, a sequence of Web sessions is also identified that involves the transfer of the elements from the downloaded page, e.g., the HTML body and related images. It is Variable Bit Rate (VBR). The distribution of the interarrival time between two consecutive packets is uniform distribution and the mean data rate is 1 Mbps.

- Video traffic: MPEG-2 data streams model the transmission from a video source, e.g., a DVD or video CD playing an action movie. It is VBR, the interarrival time between two consecutive packets is exponential [72]. It's peak rate is 4.74 Mbps and mean rate is 2 Mbps as stated in [73]

The traffic characteristics and simulation parameters are summarized in the following table. We change the number of users to get different traffic load.

Table 10-1 Traffic parameters employed in the simulation

	CBR/VBR	Interarrival distribution	Mean rate	Peak rate	Total number of users	Probability of user to have this traffic type
Voice	CBR	constant	64kbps	N/A	1~121	33.333%
Web	VBR	uniform	1Mbps	N/A		33.333%
Video	VBR	exponential	2Mbps	4.74Mbps		33.333%

10.1.2 System performance definition: packet drop ratio

In section 4.5, we outline the system performance definition. Here in this section, we add another definition, namely, packet drop ratio to further evaluate the performance of various scheduling algorithms.

Packet loss ratio: the ratio of the number of dropped packets and the number of total generated packet. In [74], it is stated that the compressed video can tolerate a maximum delay of 100ms while preserving good real-time interactivity. To have better video quality, we set the maximum tolerable delay to be 80ms in our simulation.

10.1.3 Simulation results and discussions on performance

Figure 10-1 System throughput comparison

(Reference file name: julie072805code.m)

We are aiming at improving the system performance at high traffic load, e.g., when traffic load is above 55Mbps. As can be seen from Figure 10-1, compared with OPF, all three of our proposed algorithms largely improve the system throughput. Compared with the expected upper bound, namely MaxSNR, AOPF is slightly lower than MaxSNR, and MAOPF and NMAOPF slightly outperform MaxSNR.

Figure 10-2 Packet drop ratio comparison

We study the packet drop ratio in this section. Compared with OPF, it is seen from Figure 10-2 that all of our three proposed algorithms have lower packet drop ratio. Compared with MaxSNR, AOPF and MAOPF are better, and NMAOPF is slightly worse than MaxSNR.

Figure 10-3 Mean Delay vs. throughput comparison

In Figure 10-3 presenting mean delay versus throughput, compared with OPF, all of our three proposed algorithms are better. Compared with MaxSNR, when throughput is under 50 Mbps, our three algorithms are better with AOPF the best, MAOPF better and NMAOPF good. But when throughput is larger than 50 Mbps, the three algorithms are all worse than MaxSNR with NMAOPF slightly inferior, MAOPF the worst and AOPF the worst.

Figure 10-4 Mean delay vs. traffic load comparison

In Figure 10-4 showing mean delay versus traffic load, compared with OPF, our three algorithms are better under all traffic. Compared with MaxSNR, when traffic is larger than around 45 Mbps, MAOPF and NMAOPF are almost the same as MaxSNR, or we can say slightly worse than MaxSNR, while AOPF is worse than MaxSNR.

Figure 10-5 Delay variance vs. throughput comparison

Figure 10-6 Delay variance vs. traffic load comparison

For the performance of delay variance, the trend is the same as mean delay performance.

Figure 10-7 User satisfaction rate vs. traffic load comparison

In Figure 10-7 depicting user satisfaction rate versus traffic load, compared with MaxSNR, our proposed AOPF and NMAOPF are always better than MaxSNR. MAOPF is better than MaxSNR when traffic load is under 50 Mbps but it becomes worse when traffic gets larger. It should be mentioned that in this figure, OPF always has the best performance.

Figure 10-8 Average user rate vs. traffic load comparison

In Figure 10-8 showing average user rate versus traffic load, compared with MaxSNR, our proposed NMAOPF is slightly better than MaxSNR, and AOPF and MAOPF are a little bit worse than MaxSNR. When compared with OPF, three of our proposed algorithms are better when traffic is under 50 Mbps but not as good as OPF when traffic becomes heavy.

10.1.4 Tradeoff between efficiency and fairness

One of the key purposes is to seek tradeoff between efficiency and fairness. As in previous sections, throughput is a key indicator of system efficiency. We use user satisfaction rate and the average user rate to evaluate the fairness. Thus, in this section, we take a look at the simulation results of user satisfaction rate and average user rate versus throughput. This shows the tradeoff between efficiency and fairness.

Figure 10-9 User satisfaction rate vs. throughput comparison

In Figure 10-9 depicting user satisfaction rate versus throughput, when throughput is under 50 Mbps, compared with MaxSNR, all our three algorithms greatly improve the performance in terms of fairness. When throughput is larger than 50 Mbps, AOPF and NMAOPF are almost the same as MaxSNR, but MAOPF is worst.

Figure 10-10 Average user rate vs. throughput comparison

In Figure 10-10 showing average user rate versus throughput, our proposed NMAOPF is always the best, and MaxSNR is slightly worse than NMAOPF. Inferior to NMAOPF and MaxSNR, are AOPF and MAOPF. All of the above 4 algorithms are better than OPF.

It is noticed that in Figure 10-9 and Figure 10-10, there are no results presented for RR beyond a throughput of about 40 Mbps. This is because the maximal throughput of RR cannot go beyond that throughput. As shown, the fairness performance of RR is the worst. Previously, when we mentioned that RR provides better fairness is from the resource point of view. While currently, we evaluate the fairness based on the definition of user satisfaction rate, i.e., from the performance point of view.

From Figure 10-9 and Figure 10-10, we could see that for every algorithm, when throughput goes high, user satisfaction rate and average user rate are dropping. This trend is in consistent with our theoretical prediction from section 5.2. As a conclusion, we may say that our proposed NMAOPF provides better tradeoff than conventional RR and MaxSNR when throughput is under 50 Mbps.

To study the performance of each individual traffic in the multimedia is another interesting topic. This may be one of the directions in our future study.

10.1.5 Performance summary under multimedia traffic

Based on the above simulation result, it is hard to conclude which algorithm is the best. In this section, we try to develop an evaluate method to give an overall performance rate of each algorithm.

First, we discuss and compare the performance under different traffic range. We will separate the traffic load in three parts: medium traffic is defined as traffic between 40 and 50Mbps, high traffic is defined between 50 Mbps and 70 Mbps, and very high traffic is between 70 Mbps and 90Mbps. Second, within each range, for each performance indication, we give a “score” to every algorithm. Because we compare 6 algorithms, so we define there are 6 levels of score, from good to bad are 6, 5, 4, 3, 2, 1. If two or more algorithms have almost the same performance, we will give them the same score.

Based on the simulation results in section 10.1.3, we summarize the performance of all algorithms in terms of score in Table 10-2. In this table, we use “R” to stand for RR, “Ma” for MaxSNR, “O” for OPF, “A” for AOPF, “M” for MAOPF, “N” for NMAOPF.

Table 10-2 Performance score in different traffic range

	40 <Traffic load <50 (Mbps)						50 <Traffic load <70 (Mbps)						70 <Traffic load <90 (Mbps)					
	R	Ma	O	A	M	N	R	Ma	O	A	M	N	R	Ma	O	A	M	N
Throughput	1	3	2	3	3	3	1	4	2	3	6	5	1	4	2	3	6	5

Drop ratio	1	4	2	4	4	3	1	4	2	6	5	3	1	4	2	6	5	3
Delay mean	1	4	2	3	4	4	1	5	2	3	6	4	1	6	2	3	5	4
Delay variance	1	5	2	3	6	4	1	5	2	3	6	5	1	4	2	3	6	5
Satisfaction rate	3	1	6	4	2	5	3	2	6	4	1	5	4	2	6	3	1	4
Average user rate	1	5	2	3	4	6	1	5	4	2	3	6	1	4	6	3	2	5
Total score	8	22	16	20	23	25	8	25	18	21	27	28	9	24	20	21	25	26

In the above table, finally, for each individual algorithm, we add up its performance score of each performance term, such as throughput, packet drop ratio and so on. Hence, we get the total score of this algorithm under the specific traffic range.

From Table 10-2, we may observe:

- When traffic load is between 40-50 Mbps, the overall performance of the implemented 6 algorithms from good to bad are: NMAOPF, MAOPF, MaxSNR, AOPF, OPF and RR.

- When traffic load is between 50-70 Mbps, the overall performance of the implemented 6 algorithms from good to bad are: NMAOPF, MAOPF, MaxSNR, AOPF, OPF and RR.
- When traffic load is between 70-90 Mbps, the overall performance of the implemented 6 algorithms from good to bad are: NMAOPF, MAOPF, MaxSNR, AOPF, OPF and RR.

We noticed that from the three traffic range, the performance of the 6 algorithms has the same sequence from good to bad. Hence, we may conclude that, at traffic range from 40-90Mbps, the performance of the 6 algorithms from good to bad are: NMAOPF, MAOPF, MaxSNR, AOPF, OPF and RR.

10.2 System performance under the impact of distance

User location in the cell is another important issue in wireless communications. Especially, from the fairness point of view, the scheduler should give enough fairness to users at different distances from the BS. In this section, we evaluate the algorithms and compare their performance under the impact of MS to BS distance.

10.2.1 Grouping strategy

Figure 10-11 User grouping based on distance

In all previous sections, at the beginning, all users are uniformly distributed in a sector. In this section, we evaluate the impact of distance. Thus, all users remain static. Let ' R ' denote the cell radius, ' d ' denotes the random variable of each user's position, and ' r ' denotes a certain distance, we have [75]:

$$\text{Prob}(d < r) = \frac{r^2}{R^2}, \quad \text{where } r < R; \quad \text{Equation 10-1}$$

According to the cumulative distribution function (cdf) inverse technique, we can derive the formula to generate the random variable d , whose cdf conforms to equation 10-1. Let

$$\frac{r^2}{R^2} = u, \quad \text{Equation 10-2}$$

where u is a random variable uniformly distributed from 0 to 1, i.e., $u \sim U[0,1]$. We get:

$$r = \sqrt{u} \times R, \quad \text{where } u \sim U[0,1] \quad \text{Equation 10-3}$$

This is the equation used to generate the position variable of each user so as to make them evenly distributed within the cell radius, i.e., all MS's are distributed along a line of length of R . Here in this thesis, we do not consider the angular distribution.

We now divide all users into two groups according to their distance so that the number of users in each group will be the same. As illustrated in Figure 10-11 User grouping based on distance, let 'a' denote the radius of the first group, and 'R' denotes the radius of the second group, which is also the cell radius. To find out the values of 'a', we have:

$$\frac{a^2}{R^2} = \frac{1}{2} \quad \text{Equation 10-4}$$

In all previous sections, we set the cell radius to 1.5km. For convenience to compare the results we have in previous sections, now let $a=1.5km$, by solving the above equations, we have

$$R = 2.21Km .$$

Thus we have two groups of users, i.e., near group and far group with their distances to the BS are 1~1.5km and 1.5~2.12km respectively.

10.2.2 Simulation results and discussions of known and proposed algorithms

We evaluate the performance under multimedia traffic as described in section 10.1.1.

Here we show the performance of the far group users. The same evaluation method is

used as in section 10.1.4, i.e., we give each algorithm a score for each term of performance and summarize it in a table.

Figure 10-12 System throughput comparison for far group users

(Reference file name: julie081305code.m)

Figure 10-13 User satisfaction rate comparison for far group users

Figure 10-14 Average user rate comparison for far group users

Based on the simulation results in this section, we summarize the performance of all algorithms in terms of the scores in Table 10-3. In this table, we use “R” to stand for RR, “Ma” for MaxSNR, “O” for OPF, “A” for AOPF, “M” for MAOPF, “N” for NMAOPF.

Table 10-3 Performance score in different traffic range for far group users

	45 <Traffic load <50 (Mbps)						50 <Traffic load <70 (Mbps)						70 <Traffic load <90 (Mbps)					
	R	Ma	O	A	M	N	R	Ma	O	A	M	N	R	Ma	O	A	M	N
Throughput	3	4	5	2	2	6	3	4	5	1	2	6	3	5	4	1	2	6
Drop ratio	4	3	2	6	6	1	4	2	3	6	6	1	4	2	3	6	6	1
Delay mean	4	2	3	6	6	2	4	2	3	6	6	2	4	2	3	6	6	2
Delay variance	4	1	3	6	6	2	4	1	3	6	6	2	4	1	3	6	6	2
Satisfaction rate	5	4	6	2	2	4	5	4	6	2	2	4	5	4	6	2	2	4
Average user rate	5	3	6	2	2	4	5	3	6	2	2	4	3	4	6	2	2	5
Total score	25	17	25	24	24	19	25	16	26	23	24	19	23	18	25	23	24	20

In the above table, finally, for each individual algorithm, we add up its performance score of each performance term, such as throughput, packet drop ratio and so on. Hence, we get the total score of this algorithm under the specific traffic range.

From Table 10-3, we may observe:

- When traffic load is between 45-50 Mbps, for far group users, the overall performance of the implemented 6 algorithms from good to bad are: OPF and RR have the same score and rank the highest, AOPF and MAOPF have the same score and rank the second, then is NMAOPF, MaxSNR has the smallest score.
- When traffic load is between 50-70 Mbps, for far group users, the overall performance of the implemented 6 algorithms from good to bad are: OPF, RR, MAOPF, AOPF, NMAOPF, MaxSNR.
- When traffic load is between 70-90 Mbps, the overall performance of the implemented 6 algorithms from good to bad are: OPF, MAOPF, AOPF and RR have the same score, NMAOPF, MaxSNR.

In summary, we may conclude the performance of the 6 algorithms for far group users as following: within traffic range from 45-70Mbps, the performance of the 6 algorithms from good to bad are: OPF, RR, MAOPF, AOPF, NMAOPF, and MaxSNR. At traffic range from 70-90Mbps, the performance of the 6 algorithms from good to bad are: OPF, MAOPF, AOPF and RR are the same, NMAOPF and MaxSNR. In other words, for far group users, OPF always has the best performance and MaxSNR the worst. Under medium traffic, RR is not as good as OPF, and our proposed algorithms are between RR and MaxSNR with from good to bad sequence as MAOPF, AOPF and NMAOPF. Under heavier traffic, our proposed MAOPF and

AOPF outperform RR but not as good as OPF, NMAOPF is worse than RR but better than MaxSNR.

10.3 System performance summary

We evaluated the performance of the 6 algorithms under multimedia traffic and the impact of distance. Here in this section, we summarize the performance in a table. That is, we put the total score in Table 10-2 and Table 10-3 in one table, and re-score it from 6 to 1 to represent the overall performance from good to bad respectively.

Table 10-4 Performance summary of all algorithms

	45<traffic<50(Mbps)						50<traffic<60(Mbps)					
	R	Ma	O	A	M	N	R	Ma	O	A	M	N
Multimedia	1	4	2	3	5	6	1	4	2	3	5	6
Far group	6	1	6	4	4	2	5	1	6	3	4	2
	60<traffic<70(Mbps)						70<traffic<90(Mbps)					
	R	Ma	O	A	M	N	R	Ma	O	A	M	N
Multimedia	1	4	2	3	5	6	1	4	2	3	5	6
Far group	5	1	6	3	4	2	4	1	6	4	5	2

From Table 10-4 we may conclude:

- Under multimedia traffic, for near group users, in traffic load between 45 to 90 Mbps, the performance from good to bad is: NMAOPF, MAOPF, MaxSNR, AOPF, OPF and RR. In other words, our proposed NMAOPF and MAOPF outperform MaxSNR, OPF and RR, while AOPF is not as good as MaxSNR, but better than OPF and RR.
- For the impact of user location, when traffic load is between 50-70 Mbps, the performance from good to bad is: OPF, RR, MAOPF, AOPF, NMAOPF, MaxSNR. When traffic load is between 70-90 Mbps, the performance from good to bad is: OPF, MAOPF, AOPF and RR, NMAOPF, MaxSNR. In other words, for far group users, OPF has the best performance and MaxSNR the worst. Our proposed three algorithms are not as good as OPF and RR in medium traffic but better than MaxSNR. Under high traffic, our proposed MAOPF and AOPF is not as good as OPF but better than RR and MaxSNR, In short, our modified version of PF performs the best for far group users, but the three proposed algorithms are not as good as OPF.

11 Summary and Conclusions

11.1 Summary

In this thesis, a modified version of Proportional Fair (PF) algorithm named OPF is defined. Round Robin (RR) and Maximal SNR (MaxSNR) are two well known scheduling algorithms. RR has the feature to provide fairness among users and MaxSNR can provide highest system throughput. The PF algorithm aims at providing a tradeoff between system throughput and fairness. The performance of PF depends on the observation window and it is well discussed in TDMA/CDMA systems. In our modification, the impact of the observation window is eliminated. In addition, we apply OPF to OFDM broadband wireless systems. OPF is further tailored to comply with IEEE 802.16a standard as a MAC scheduler, which is an area open for research. For comparison purpose, we also implement 4 algorithms in 802.16a OFDM system. Two of them are conventional MaxSNR and RR. The others are two versions of PF: PF with observation window and weighted PF. Joint MAC and PHY layer simulation results show that, compared with PF and conventional methods, OPF gives better compromise between system throughput and fairness by providing the highest user satisfaction rate under wireless channel conditions of both time varying and frequency selective fading channel.

Combined with adaptive modulation and coding scheme, compared with RR and MaxSNR, simulation results show that OPF increases the system throughput by 10Mbps compared with RR, and it gives the highest user satisfaction rate as well as highest average user rate before the system is saturated. Compared with PF, OPF outperforms PF with observation window and weighted PF in terms of system throughput, average delay and average user rate. OPF is not as good as PF with observation window and weighted PF under low throughput but it becomes competitive at high throughput. In summary, OPF algorithm gives the better compromise between throughput and fairness among known algorithms.

Aiming at developing new algorithms for multimedia services, we proposed 3 algorithms based on our previous work. They are Adaptive OPF (AOPF), Multimedia AOPF (MAOPF), and Normalized MAOPF (NMAOPF). Simulation is implemented to evaluate the three proposed algorithms under two data rate requirements which is the simplest form of multimedia service. Results show that AOPF and MAOPF improve system throughput, mean delay and delay variance, but they have worse user satisfaction rate and average user rate. While NMAOPF provides new upper bound of system throughput, improves delay statistics at considerable range of throughput, as well as offers higher user satisfaction rate compared with RR and MaxSNR, and gives highest average user rate among all existing methods discussed.

In addition, the theoretical framework is developed by presenting the mathematical model of the system performance. To further evaluate the performance of OPF and proposed algorithms, simulation is implemented in real multimedia traffic. The impact of user location is also evaluated. Packet drop ratio is added as another metric of performance. We discuss each performance metric under different traffic range and the overall performance is defined by performance score.

11.2 Conclusions

In conclusion, we have:

- Under multimedia traffic, for near group users, in traffic load between 45 to 90 Mbps, the performance from good to bad is: NMAOPF, MAOPF, MaxSNR, AOPF, OPF and RR. In other words, our proposed NMAOPF and MAOPF outperform MaxSNR, OPF and RR, while AOPF is not as good as MaxSNR, but better than OPF and RR.
- For the impact of user location, when traffic load is between 50-70 Mbps, the performance from good to bad is: OPF, RR, MAOPF, AOPF, NMAOPF, MaxSNR. When traffic load is between 70-90 Mbps, the performance from good to bad is: OPF, MAOPF, AOPF and RR, NMAOPF, MaxSNR. In other words, for far group users, OPF has the best performance and MasSNR the worst. Our proposed three algorithms are not as good as OPF and RR in traffic range from 50 to 70 Mbps but better than MaxSNR. Under higher traffic from 70 to 90 Mbps range, our proposed MAOPF and AOPF is not as good as OPF

but better than RR and MaxSNR, In short, our modified version of PF performs the best for far group users, but the three proposed algorithms are not as good as OPF.

In summary, Under multimedia traffic, for near group users, our proposed NMAOPF and MAOPF outperform conventional MaxSNR and RR. The modified version of PF, namely OPF, performs the best for far group users. All the three proposed algorithms and OPF improve the efficiency without sacrificing fairness under certain conditions. They provide better tradeoff between throughput and user fairness. This conclusion can be extended to any OFDM systems without constraints of IEEE 802.16a. They can be competitive candidate for 3G/4G OFDM based broadband wireless system.

11.3 Future work

During the study, I find this area is very interesting and important in overall system performance. It is becoming a hot research area. Based on our current system model, future work can be done from several aspects.

- From the algorithm point of view, we do not implement priority, and all users can only have one type of traffic. In the future, we may implement different level of priority, and also we may let the same user to have more than one type of traffic, which is a realistic scenario.
- From the evaluation point of view, we only evaluate the performance under slow fading channel. Next, we may evaluate the algorithms under fast fading channel.

Also, in the PHY layer, we may implement the Bit Error Rate which is an important feature in PHY layer performance.

- From the traffic point of view, we may evaluate the performance of each individual traffic in the multimedia. For example, we may compare the performance of voice traffic and the web traffic to find out the best algorithm for a specific kind of traffic.
- From access point of view: our current access method can be regarded as TDMA, so, all schedulers are time domain schedulers. In the future, we may develop frequency domain scheduler, which means the algorithm criteria remains the same while the scheduling is done sub-band by sub-band or frequency by frequency. Thus we could compare the performance of OFDM-TDMA system and OFDMA system. Or, to go a step further, we could develop more flexible Time-frequency schemes to take advantage of the two dimensional OFDM systems.
- From the joint layer point of view: joint layer optimization is another hot area in current research organization. Our current system model is actually joint MAC and PHY layers. We may develop mathematical model for optimizing a specific metric of performance with respect to some physical layer parameters. Thus, the joint layer optimization could be reached.
- From the analysis point of view: further study on the distribution of system throughput is another key point in the optimization problem mentioned in the above item.

- From network point of view: currently, we only consider the single cell. It is possible to build up multi-cell based on our current model, and the scheduling algorithm could be moved to the Core Network (CN), so that the performance of the network can be evaluated.

References

- [1] 3GPP; Technical Specification Group Radio Access Network; Physical Layer Aspects of UTRA High Speed Downlink Packet Access (Release 2000), Serial Code: 3G TR25.848 V0.6.0 (2000-05).
- [2] A.Jalali, R Padovani, R.Pankaj, "Data Throughput of CDMA-HDR, a High Efficiency High Data Rate Personal Communication Wireless System", IEEE 51st Vehicular Technology Conference, Tokyo, Japan - May - 2000, page 1854-1858.
- [3] Bender, Paul; Black, Peter; Grob, Matthew; Padovani, Roberto; Sindhushayana, Nagabhushana; Viterbi, Andrew, "CDMA/HDR: A bandwidth-efficient high-speed wireless data service for nomadic users," IEEE Communications Magazine, v 38, n 7, Jul. 2000, p 70-77.
- [4] R. W. Chang, "Synthesis of Band Limited Orthogonal Signals for Multichannel Data Transmission," Bell Syst. Tech. J., vol. 45, pp. 1775-1796, Dec. 1996.
- [5] Salzberg, B. R., "Performance of an efficient parallel data transmission system," IEEE Trans. Comm., vol, COM-15, pp. 805-813, Dec, 1967.

-
- [6] Mosier, R. R., and R. G. Clabaugh, "Kineplex, a Bandwidth Efficient Binary Transmission System," AIEE Tran., vol. 76, pp.723-728, Jan, 1958.
 - [7] "Orthogonal Frequency Division Multiplexing," U.S. Patent No. 3, 488,4555, filed November 14, 1966, issued Jan. 6, 1970.
 - [8] Weinstein, S. b., and P. m. Ebert, "Data Transmission by Frequency Division Multiplexing Using the Discrete Fourier Transform," IEEE Trans. Comm., vol, COM-19, pp. 628-634, Oct. 1971.
 - [9] Porter, G. C., "Error Distribution and Diversity Performance of a Frequency Differential PSK HF modem," IEEE Trans. Comm., vol. COM-16, pp. 567-575, Aug. 1968.
 - [10] Hirosaki, B., "An 19.2 kbits Boice Band Data Modem Based on Orthogonality Multiplexed QAM Techniques," Proc. of IEEE ICC'85, pp. 21.1.1-5, 1985.
 - [11] Chow, P. S., J. C. Tu and J. M. Cioffi, "Performance Evaluation of a Multichannel Transceiver system for ADSL and VHDSL services," IEEE J. Selected Area, vol., SAC-9, No. 6, pp. 909-9191, Aug. 1991.

-
- [12] Paiement, R. V., "Evaluation of Single Carrier and Multicarrier Modulation Techniques for Digital ATV Terrestrial Broadcasting," CRC Report, No. CRC-RP-004, Ottawa, Canada, Dec. 1994.
- [13] Sari, H., G. Karma and I. Jeanclaude, "Transmission Techniques for Digital Terrestrial TV Broadcasting," IEEE Comm. Mag., vol. 33, pp. 100-109, Feb. 1995.
- [14] Oppenheim, A. V., and R. W. Schaffer, Discrete-time Signal Processing, Prentice-Hall International, ISBN 0-13-216771-9, 1989.
- [15] Hara, S., M. Mouri, M. Okada and N. Morinaga, "Transmission Performance Analysis of Multi-Carrier Modulation in Frequency Selective Fast Rayleigh Fading Channel," in Wireless Personal Communications, Kluwer Academic Publishers, vol. 2, pp. 335-356, 1996.
- [16] Blahut, R. E., "Fast Algorithm for Digital Signal Processing," Reading, MA: Addison-Wesley, 1985.
- [17] Pollet, T., M. Van Bladel and M. Moeneclaey, "BER Sensitivity of OFDM Systems to Carrier Frequency Offset and Wiener Phase Noise," IEEE Trans. On Comm., vol. 43, no. 2/3/4, pp. 191-193, Feb-Apr. 1995.

-
- [18] Pauli, M., and H. P. Kuchenbecker, "Minimization of the Intermodulation Distortion of a Nonlinearly Amplified OFDM Signal," *Wireless Personal Communications*, vol. 4, no. 1, pp. 93-101, Jan. 1997.
- [19] Grunheid, Rainer; Chen, Bing; Rohling, Hermann, "Joint layer design for an adaptive OFDM transmission system," *IEEE Vehicular Technology Conference*, v 58, n 1, 2003
IEEE 58th Vehicular Technology Conference, VTC2003-Fall, 2004, p 542-546.
- [20] Laroia, Rajiv; Uppala, Sathyadev; Li, Junyi, "Designing a mobile broadband wireless access network," *IEEE Signal Processing Magazine*, v 21, n 5, September, 2004, p 20-28.
- [21] Cao Y, Li VOK., "Scheduling Algorithms in Broad-band Wireless Networks", *Proceedings of the IEEE 2001*; 89 (1), p. 76-87.
- [22] Bhagwat, Pravin; Bhattacharya, Partha; Krishna, Arvind; Tripathi, Satish K., "Enhancing throughput over wireless LANs using channel state dependent packet scheduling," *Proceedings - IEEE INFOCOM*, v 3, 1996, p 1133-1140.

-
- [23] Fragouli, Christine; Sivaraman, Vijay; Srivastava, Mani B., "Controlled multimedia wireless link sharing via enhanced class-based queuing with channel-state-dependent packet scheduling," *Proceedings - IEEE INFOCOM*, v 2, 1998, p 572-580.
- [24] Demers, Alan; Keshav, Srinivasan; Shenker, Scott, "Analysis and simulation of a fair queuing algorithm," *SIGCOMM 89 Symp Commun Archit Protoc*, 1989, p 1-12.
- [25] Lu, Songwu; Bharghavan, Vaduvur; Srikant, R., "Fair scheduling in wireless packet networks," *IEEE/ACM Transactions on Networking*, v 7, n 4, 1999, p 473-489.
- [26] Ng, T.S. Eugene (Carnegie Mellon Univ); Stoica, Ion; Zhang, Hui, "Packet fair queueing algorithms for wireless networks with location-dependent errors," *Proceedings - IEEE INFOCOM*, v 3, 1998, p 1103-1111.
- [27] Gomez, Javier; Campbell, Andrew T.; Morikawa, Hiro, "Havana framework for supporting application and channel dependent QOS in wireless networks," *International Conference on Network Protocols*, 1999, p 235-244.
- [28] Holtzman, J.M., "Asymptotic analysis of proportional fair algorithm," *IEEE International Symposium on Personal, Indoor and Mobile Radio Communications, PIMRC*, v 2, 2001, p F33-F37.

-
- [29] Yamaguchi, A., Takeuchi, Y., "Forward link packet scheduler for high-speed packet data system," IEEE International Symposium on Personal, Indoor and Mobile Radio Communications, PIMRC, v 2, 2001, p F21-F24.
- [30] Rentel, Carlos H.; Krzymien, Witold A.; Darian, Bahman; Vanghi, Vieri; Elliott, Robert, "Comparative forward link traffic channel performance evaluation of HDR and 1XTREME systems," IEEE Vehicular Technology Conference, v 1, 2002, p 160-164.
- [31] Hoon Kim; Keunyoung Kim; Youngnam Han; Jiwoong Lee, "An efficient scheduling algorithm for QoS in wireless packet data transmission," 13th IEEE International Symposium on Personal Indoor and Mobile Radio Communications Proceedings, 2002, p 2244-8 vol.5.
- [32] Aniba, Ghassane; Aissa, Sonia, "Fast packet scheduling assuring fairness and quality of service in HSDPA," Canadian Conference on Electrical and Computer Engineering, v 4, Canadian Conference on Electrical and Computer Engineering; Technology Driving Innovation, 2004, p 2243-2246.
- [33] Soohyun Oh, Yo Huh, Beshan Kulapala, Goran Konjevod, Andrea W. Richa and Martin Reisslein, "A Modular Algorithm-Theoretic Framework for the Fair and Efficient

- Collaborative prefetching of Continuous Media," IEEE Transactions on Broadcasting, vol. 51, No. 2, June 2005, pp. 200-215.
- [34] Chuang, Justin; Sollenberger, Nelson, "Beyond 3G: Wideband wireless data access based on OFDM and dynamic packet assignment," IEEE Communications Magazine, v 38, n 7, Jul, 2000, p 78-87.
- [35] Enright, Michael A.; Kuo, C.-C. Jay, "Generalized OFDM method for multimedia transmission over wireless fading channels," Proceedings of SPIE - The International Society for Optical Engineering, v 4529, 2001, p 125-136.
- [36] Nassar, Carl R., "Discovery of uniform multi-carrier frameworks for multiple access technologies," Proceedings of SPIE - The International Society for Optical Engineering, v 4529, 2001, p 81-93.
- [37] I. Koffman and V. Roman, "Broadband wireless access solutions based on OFDM access in IEEE802.16", IEEE Commun. Mag., pp.96-103, Apr.2002.
- [38] K. D. Shirali and D. D. Dabiri and T. Y. Chen, "An OFDM based transceiver for broadband wireless access", IEEE 802.16pc-99/30, IEEE, Nov. 1999.

-
- [39] Eriksson, M., "Dynamic single frequency networks," IEEE Journal on Selected Areas in Communications, v 19, n 10, October, 2001, p 1905-1914.
- [40] Cai, Jun; Shen, Xuemin; Mark, Jon W., "Downlink Resource Management for Packet Transmission in OFDM Wireless Communication Systems," Conference Record / IEEE Global Telecommunications Conference, v 6, 2003, p 2999-3003.
- [41] Cai, Jun; Shen, Xuemin; Mark, Jon W., "Downlink resource management with adaptive modulation and dynamic scheduling for OFDM wireless communication systems," IEEE Wireless Communications and Networking Conference, WCNC 2004, v 4, 2004, p 2143-2147.
- [42] Zhang, Ying Jun; Letaief, Khaled Ben, "Adaptive resource allocation and scheduling for multiuser packet-based OFDM networks," IEEE International Conference on Communications, v 5, 2004 IEEE International Conference on Communications, 2004, p 2949-2953.
- [43] Classon, Brian; Sartori, Philippe; Nangia, Vijay; Zhuang, Xiangyang; Baum, Kevin, "Multi-dimensional adaptation and multi-user scheduling techniques for wireless OFDM systems," IEEE International Conference on Communications, v 3, 2003, p 2251-2255.

-
- [44] Wang, Anchun; Xiao, Liang; Zhou, Shidong; Xu, Xibin; Yao, Yan, "Dynamic resource management in the fourth generation wireless systems," International Conference on Communication Technology Proceedings, ICCT, v 2, 2003, p 1095-1098.
- [45] Wongthavarawat, Kitt; Ganz, Aura, "Packet scheduling for QoS support in IEEE 802.16 broadband wireless access systems," International Journal of Communication Systems, v 16, n 1 SPEC., February, 2003, p 81-96.
- [46] Wongthavarawat, Kitt; Ganz, Aura, "IEEE 802.16 based last mile broadband wireless military networks with quality of service support," Proceedings - IEEE Military Communications Conference MILCOM, v 2, MILCOM 2003 - 2003 IEEE Military Communications Conference, 2003, p 779-784.
- [47] IEEE Draft Standard for local and metropolitan area networks – part 16, IEEE P802.16/D5-2001.
- [48] IEEE draft amendment to IEEE standard for local and metropolitan area networks-MAC modification and additional physical layer specification for 2-11GHz, P802.16a/D3-2002.

- [49] Draft Amendment to IEEE Standard for Local and Metropolitan Area Networks, Part 16: Air Interface for Fixed Broadband Wireless Access Systems-Medium Access Control Modifications and Additional Physical Layer Specifications for 2-11GHz, P802.16a/D5-2002.
- [50] IEEE Computer Society Distinguished Visitors Program Lecture, IEEE New Jersey Coast Section, 8 April 2003 and IEEE Baltimore Section, 9 April 2003. (Roger Marks, 03/04/16) IEEE 802.16-02/52. <http://wirelessman.org>
- [51] J.B. Andersen, I. S. Rappaport, and S. Yoshida, "Propagation Measurements and Models for Wireless Communications Channels," IEEE Communications Mag., vol. 33, no. 1, Jan, 1995, pp.42-49.
- [52] Eklund, Carl; Marks, Roger B.; Stanwood, Kenneth L., "IEEE standard 802.16: A technical overview of the wirelessMAN air interface for broadband wireless access," IEEE Communications Magazine, v 40, n 6, June, 2002, p 97-107.
- [53] Michel C. Jeruchim, Philip Balaban, and K.Sam Shanmugan, "Simulation of Communication Systems, Modeling, Methodology, and Techniques", 2nd edition, Kluwer Academic/Plenum Publishers, 2003.

-
- [54] G. Proakis, "Digital Communications", fourth edition, McGraw Hill, 2001, p. 813.
- [55] William C. Jakes, "Microwave Mobile Communications", IEEE communications society, 1994.
- [56] B. Sklar, "Digital Communications: fundamentals and applications," Prentice Hall PTR, New Jersey, U.S., 2000
- [57] T1E1.4 T1E1.4/2001-002, "Draft proposed American National Standard, Spectrum Management for Loop Transmission Systems, Issue 2", Project: T1E1-38, Spectral Compatibility Aspects for Facilities between a Central Office and the Network-to-Customer Interface (Twisted Pair Transmission System), p. 53.
- [58] Fu, Jianyu; Karasawa, Yoshio, "Fundamental analysis on throughput characteristics of orthogonal frequency division multiple access (OFDMA) in multipath propagation environments," IEEE Vehicular Technology Conference, v 56, n 2, 2002, p 1235-1239.
- [59] Keller, Thomas; Hanzo, Lajos, "Adaptive modulation techniques for duplex OFDM transmission," IEEE Transactions on Vehicular Technology, v 49, n 5, Sep, 2000, p 1893-1906.

- [60] Dobre, Octavia Adina; Yao, Yu-Dong, "An adaptive data transmission scheme for OFDM systems," IEEE Vehicular Technology Conference, v 56, n 3, 2002, p 1398-1403.
- [61] 3rd Generation Partnership Project, Technical Specification Group Radio Access Network, Physical Layer Aspects of UTRA High Speed Downlink Packet Access (Release 2000), 3G TR25.848 v0.6.0 (2000-05) TSG-RAN Working Group 1 meeting #19.
- [62] Hua Sun, "Fast Scheduling strategies for WCDMA-High speed downlink packet access (HSDPA) system", Graduate Thesis in System and Computer Engineering Department, Carleton University, April 2003.
- [63] Canpolat, B.; Tanik, Y., "Performance of adaptively loaded OFDM under Rayleigh fading with diversity combining," IEEE Vehicular Technology Conference, v 2, n 54ND, 2001, p 957-961.
- [64] ETSI, TR UMTS 30.03, "Selection procedures for the choice of radio transmission technologies of the UMTS", version 3.2.0, April, 1998.

-
- [65] Guowang Miao and Zhisheng Niu, "Profit Oriented Multichannel Resource Management of Integrated Communication and Broadcast Networks", IEEE Trans. Broadcasting, vol 51, no. 4, Dec. 2005.
- [66] Guocong Song and Ye Geoffrey Li, "Cross-Layer Optimization for OFDM Wireless networks----Part I: Theoretical Framework," IEEE Transaction on Wireless Communications, Vol. 4, No. 2, March 2005, pp. 614-624.
- [67] Guocong Song and Ye Geoffrey Li, "Cross-Layer Optimization for OFDM Wireless networks----Part II: Algorithm Development," IEEE Transaction on Wireless Communications, Vol. 4, No. 2, March 2005, pp. 625-634.
- [68] Mustafa Ergen, Sinem Coleri and Pravin Varaiya, "QoS Aware Adaptive Resource Allocation Techniques for Fair Scheduling in OFDMA Based Broadband Wireless Access Systems," IEEE Transaction on Broadcasting, vol. 49, No. 4, Dec. 2003, pp. 362-370.
- [69] OPNET modeler, OPNET Online documentation, 2003. available at:
<http://www.opnet.com>

-
- [70] Roberto Cusani, Francesco Delli Priscoli, Gaetano Ferrari and Massimiliano Torregiani, "A Novel MAC and Scheduling Strategy to Guarantee QoS for the New-Generation WIND-FLEX Wireless LAN," *IEEE Wireless Communications*, v 9, n 3, June 2002, p. 46-56.
- [71] E. Anderlind and J. Zander, "A Traffic Model for Non-Real-Time Data Users in a Wireless Radio Network," *IEEE Commun. Lett.*, vol. 1, no. 3, Mar. 1997, pp. 37-39.
- [72] F. Yegenoglu, B. Jabbari and Ya-Qin Zhang, "Motion-Classified Autoregressive Modeling of Variable Bit Rate Video," *IEEE Trans. Circ. Sys. Video Tech.*, vol. 3, no. 1, Feb. 1993, pp. 42-53.
- [73] Steven Gringeri, Khaled Shuaib, Roman Egorov, Arianne lewis, Bhumip Khasnabish and Bert Basch, "Traffic Shaping, Bandwidth Allocation, and Quality Assessment for MPEG Video Distribution over Broadband Networks," *IEEE Networks*, vol. 12, no. 6, Nov-Dec, 1998, p. 94-107.
- [74] D. Dubios, N.D. Geoganas, and E. Horlait, "A QoS selector for multimedia applications on ATM Networks", *IEEE ICC'94, conf. record*, pp. 160-164, 1994.

-
- [75] Hua Sun, "Fast Scheduling Strategies for WCDMA-High Speed Downlink Packet Access (HSDPA) System," Graduate Thesis in System and Computer Engineering Department, Carleton University, April 2003.