Cross-layer Optimization

for High Speed Wireless Communication Networks

by

Yanfei Qi

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Abstract

Orthogonal frequency division multiplexing (OFDM) techniques, a key technology for 3G and 4G wireless communications, are used to meet the transmission requirements of the wireless communication system nowadays. The adaptive cross-layer design with packet dependent (PD) scheduling [1] is used to maximize the weighted sum capacity of the downlink multiuser multi-tasking OFDM systems. The weight design considers the delay, packet size and quality of service (QoS) priority packet level in scheduling and brings much better performance on system throughput, average packet delay and packet drop rate compared with other designs. Its lower complexity with correct choice of packet number in weight design utilizes the spectrum and energy much more efficiently and achieves the maximum system stability region. The proposed design which modifies the weight design by considering channel state information additionally can increase the system throughput while maintain all the other characteristics.

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Abbreviations

3G	Third Generation
4G	Fourth Generation
ADSL	Asymmetric Digital Subscriber Line
AWGN	Additive White Gaussian Noise
BE	Best Effort
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
CIR	Channel Impulse Response
CNR	Channel to Noise Ratio
CP	Cyclic Prefix
CSI	Channel State Information
DAB	Digital Audio Broadcasting
Delay-EDD	delay- Earliest Due Data
DRR	Deficit Round-robin
DVB	Digital video Broadcasting
EDF	Earliest Deadline First
ETSI	European Telecommunications Standards Institute
FDD	Frequency Division Duplexing
FDE	Frequency Domain Equalisation
FDM	Frequency Division Multiplexing
FFT	Fast Fourier Transform
FIFO	First in First out
FQ	Fair Queueing
HDTV	High-Definition Television
HOL	Head of Line
ICI	Inter-Carrier Interference
ISO	International Standard Organization

IFFT	Inverse Fast Fourier Transform
ISI	Inter-Symbol Interference
LOS	Line of Sight
MAC	Medium Access Control
MC	Maximum Capacity
MCM	Multi-Carrier Modulation
MF	Matched Filter
M-LWDF	Modified Largest Weight Delay First
MWC	Maximum Weighted Capacity
M-MWC	Modified Maximum Weighted Capacity
OFDM	Orthogonal Frequency Division Multiplexing
OSI	Open System Interconnection
P/S	Parallel to Serial
PD	Packet Dependent
MPD	Modified Packet Dependent
pdf	probability distribution function
PF	Proportional Fairness
PHY	Physical Layer
psd	power spectral density
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RMS	Root Mean Square
RR	Round-robin
S/P	Serial to Parallel
SCFQ	Self-Clocked Fair Queueing
SCM	Single-Carrier Modulation
SISO	Single Input Single Output
SNR	Signal to Noise Ratio

TDD	Time Division Duplexing
VC	Virtual Clock
WF	Water-filling
WLAN	Wireless Local Area Network
WWF	Weighted Water-filling
WFQ	Weighted Fair Queueing
WiMAX	Worldwide Interoperability for Microwave Access
WSS	Wide Sense Stationary
ZF	Zero Forcing

1. Introduction

With increasing demands of mobile wireless network communication such as high capacity, reliability, low delay, and diverse quality of service (QoS) requirements, orthogonal frequency division multiplexing (OFDM) has been identified as a very promising technique to utilize the limited radio resources efficiently. It also gets the advantages on power transmission in the wideband transmission over time-varying multi-path channels.

Cross-layer optimization for interactions and communications among different layers in Open System interconnection (OSI) reference model can significantly improves the system performance in terms of throughput, fairness, bandwidth efficiency, packet loss rate and so on. Main considerations of the cross-layer design in OFDM system are resource allocation at PHY layer and scheduling strategies at MAC layer in this project.

1.1 Background

The continuous increase in demand for all types of wireless services requires higher data-rate, more reliable and efficient communications in wireless channels. More and more wireless devices are based on the multicarrier technique like OFDM because of their frequency-selective nature. As the key technologies for 3G and 4G wireless communications, OFDM techniques win by the parallelization principle compared with the single-carrier techniques [2]. OFDM divides the entire channel into many orthogonal narrow band sub channels (i.e. subcarrier) instead of transmitting symbols sequentially over the communication channel. High rate data communications are significantly limited by inter symbol interference (ISI) whose impact decreases in OFDM system. OFDM not only has high spectral efficiency and high tolerance to multi-path fading, but also is adaptive for providing the diverse transmission power and modulation modes for each subcarrier. Modern digital audio and video broadcasting systems rely on OFDM system such as some well-known wireless local area network (WLAN) standards (e.g., IEEE 802.11a/g) and Worldwide Interoperability for Microwave Access (WiMax) system (e.g., IEEE 802.16e).

On the other hand, conventional wireless communication networks based on the OSI model are decomposed into seven layers which are used for performing the variable and complex tasks [3]. The layer-specific protocols can be developed and optimized independently in the conventional approach while there are some limitations on the flexibility that only the information from adjacent layers can be processed. The transmission performance varies over time and frequency due to the frequency selective and time variant broadband channel. This brings the high requirements of flexibility for multiuser multi-tasking services with variable system parameters (e.g., bit rate, QoS). The system performance can be optimized by utilizing interactions between different layers, which is called cross-layer optimization. The OFDM transmission techniques are very flexible in adapting the transmission parameters to the current channel situation and to the application–specification requirements [4]. OFDM systems not only provide excellent physical layer properties, but also offer the detailed current channel information which can be used for the adaption procedure in the cross layer approach. The adaptive modulation types or the transmitted power for each sub channel increases the spectral efficiency. The diversity of the system (such as frequency, time and space) can be achieved by dynamically assigning different sets of subcarriers to different terminals. The principle of the cross-layer optimization approaches is dynamically matching the requirements to the available resources among different layers to improve the system performance remarkably [2]. Apparently a large degree of interaction among layers leads to a considerable capacity enhancement.

1.2 Previous Research

Recently, quite a lot of research effort has been spent on cross-layer wireless OFDM system design. Utility theory has been used for downlink resource allocation and scheduling for OFDM broadband wireless networks since 2005 [3]. In [5], the joint design of the multiuser subcarrier, bit and power allocation in the physical layer and the packet scheduling in the data link layer achieve significant gains in both throughput and average delay compared with the traditional single-layer ones. The integrated design of power control, subcarrier allocation and packet scheduling in the proposed algorithm brings the bandwidth and power allocation optimizations [6]. Compared with the wired channel, it achieves the same QoS and fairness. In [7], choosing best channel gain for each user is the principle of the resource allocation and the power allocation is based on the water-filling algorithm. Adaptive resource allocation with proportional rate constraints which achieves the optimal capacity is proposed in [8]. For the significant performance improvement for downlink OFDM with heterogeneous traffic, a new fairness criterion based on QoS satisfaction of users is developed in [7].

The proposed resource allocation is designed depending on channel state information (CSI), QoS requirements of users, random packet arrivals and queue states. In [10], the serial scheduling in OFDM network where the voice traffic is scheduled first and then data traffic is put forward. This design is good for decreasing the delay of voice packets while the total system data traffic throughput is maximized. The adaptive cross layer design based in the packet scheduling for the downlink multiuser OFDM system not only takes account for packet delay and channel quality, but also maximize the weighted sum capacity of all users [1]. From recent developments on this topic, it can be seen that most of the joint designs are quality of service (QoS) awareness. In order to satisfy the rapid growth demands there are some development like from single-layer design to cross-layer design, from single carrier system to multi-carrier system, from queue based scheduling scheme to packet dependent one and from homogeneous traffic to heterogeneous one.

1.3 Project Objectives and Methodology

This project aims to create an adaptive cross-layer design for wireless communication systems, with a focus on physical and link layers [1]. The OFDM technique will be employed. The adaptive cross-layer design is used to maximize the weighted sum capacity of downlink multiuser multi-tasking OFDM system.

Wireless communication systems should support multiple users with multiple heterogeneous traffic queues simultaneously for each, with the growing demands of high capacity and multi-tasking services. Compared with the conventional communication network which is based on the independent layer design, lower complexity cross-layer design utilizes the spectrum and energy much more efficiently and achieves the maximum system stability region.

The design followed by [1] can be implemented in following four main steps. First, a general downlink time division duplex OFDM system model will be built, which has multiple heterogeneous traffic queues at the same time for each user. Second, a packet dependent (PD) scheduling scheme which is employed at the MAC layer for determining the packet transmission order by assigning different weights should be proposed. The delay, packet size and QoS priority packet level need to be considered in the weight design. Third, the user based resource allocation at the PHY layer will be done by maximizing the weighted sum capacity of the users. The number of selected packets is required to be chosen properly to reduce the complexity for further weight summing up. Software Matlab will be used for simulation for the complexity and performance analysis of the proposed cross-layer design for the system with some proper assumptions. In summary, the project can be simply described two parts, packet dependent scheduling scheme at MAC layer which provides suitable weights for the resource allocation at PHY layer. The optimization can be achieved by the joint designs with low complexity.

1.4 Thesis Organisation

This thesis is organized as follows. First, the basic knowledge of channel models and OFDM techniques are introduced with some simulation results in Section 2 and 3. Then in Section 4 the concept of cross-layer design will be presented. In this part, different existent resource allocation and scheduling schemes are discussed not only in theory but also with simulation results. Advantages and disadvantages of all these designs will be analyzed. Moreover, a modified cross-layer design based on the design in [1] will be proven to achieve much better performance results. Subsequently, applications of OFDM technique and dynamic resource management in real world will be given. Final Section will draw the conclusions and talk about future work.

2. Channel Models

Complex wireless communication channel models are statistical models which are based on the measured values on specific frequency bands. Not only the surrounding environment influences the channel, the mobility of the channel itself also restricts the performance of the communication system. The correct channel model selection is the first step of the communication design.

2.1 AWGN Channel

Additive white Gaussian noise (AWGN) is a typical idea channel model which does not consider fading, interference and nonlinearity. White noise with a constant spectral density and a Gaussian distribution of amplitude is added to the original signal linearly [3].All the frequency components appear with equal power if the noise is white. Probability distribution function (pdf) of random variable X with Gaussian distribution which relates to mean and variance is expressed as following:

$$f_X(x) = \frac{1}{\sqrt{2\pi\sigma}} \exp[-\frac{(x-\mu)^2}{2\sigma^2}] \qquad (2.1)$$

where μ is the mean value of X and σ is variance.

A random process can be defined as Wide sense stationary (WSS) process due to its constant mean and one dimensional time difference dependent autocorrelation function [3]. One example of a zero mean Gaussian WSS process can be considered as thermal noise which comes from thermal agitations of electrons in a conductor. The power spectral density (psd) of AWGN channel is flat so noise bandwidth is infinite:

$$S_X(f) = \frac{N_0}{2}\ watts/Hz \qquad (2.2)$$

AWGN channel is usually considered as the basis of analysis of communication system performance due to its representation and simplicity. In AWGN channel, the received signal *y* comes from the linear addition of the original signal *x* and AWGN noise *n*.

$$\mathrm{y} = \mathrm{x} + \mathrm{n} \qquad (2.3)$$

Signal detection [3] in AWGN channel can be done by decision device. It makes a decision at a sampling point (i.e.T) according to decision threshold (i.e.V_T). For example, the threshold can be set as halfway of two symbol values. If the received signal is larger than threshold, it implies the signal representing “1” (i.e. $S_1(t)$) is transmitted. Otherwise, the transmitted signal represents “0” (i.e. $S_0(t)$). Bit error rate (BER) is defined as the ratio of the error bit numbers at the receiver and the total number of transmitted bits. It occurs when the detected signal represents “1” or “0” with transmitted signal representing “0” or “1”. The probability of error (i.e. P_e)measures the performance of the digital communication systems and should be minimized. For equally-likely binary signals, the impulse response (i.e.$h_{opt}(t)$) of Matched Filer (MF) which is the optimum receiver for minimizing the error probability:

$$h_{opt}(t) = S_1(T - t) - S_0(T - t) \tag{2.4}$$

The optimum threshold V_{Topt} which minimizes error probability can be achieved by setting the derivation of error probability and threshold as zero:

$$V_{Topt} = \frac{E_1 - E_0}{2} = \frac{\int_0^T S_1(t)^2 dt - \int_0^T S_0(t)^2 dt}{2} \tag{2.5}$$

The achieving minimum error probability which is also called minimum BER is:

$$BER = Q\left[\sqrt{\frac{E_g}{2N_0}}\right] = Q[\sqrt{\frac{\int_0^T (S_1(t) - S_0(t))^2 dt}{2N_0}}] \tag{2.6}$$

2.2 Fading Channel

As mentioned in the previous part, in the ideal free space, there is no energy loss because of ideal assumptions of atmosphere. So the received energy can be predicted in such ideal propagations. However, in the real situations, the problems of diffraction, reflection and scattering should be faced. The performance of communication system such as capacity and bit error rate (BER) can be affected by these channel effects like multipath fading, shadowing and propagation loss.

According to the covering distance, channel fading can be divided into two classifications large-scale path loss and small-scale fading [3].

Large-scale path loss is characterised by the long distance and the average density of field prediction and highly depends on the location and environment. The reduction of signal strength is the results of the long distance between transmitter and receiver.

Another signal attenuation cause *small-scale fading* is more focus on the variations of the density of field on the specific locations. The changes of amplitude and phase of the signals are the most important considering factors in fading channels. Different from path loss, rapid fluctuation of channel gain and phase over a short time or distance results in fading. Large scale path loss can be seen as an example of “space average” of the small-scale fading.

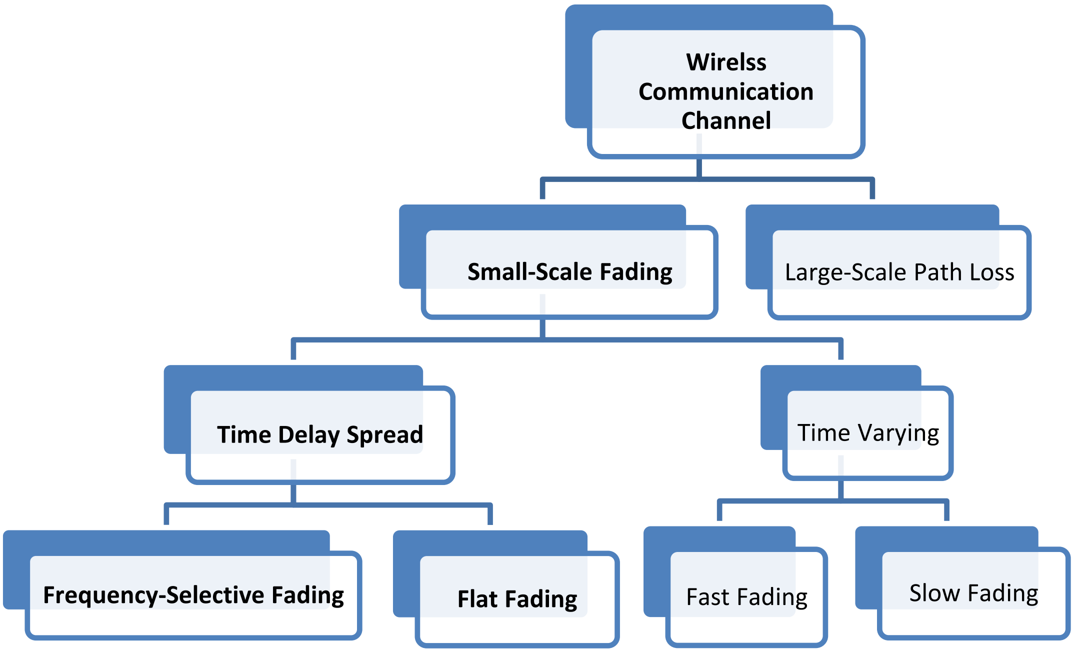


Figure 2.1 Classification of Wireless Communication Channel

Above figure presents the classification of wireless communication channel. Due to the time delay spread caused by the multipath there are two types of fading frequency selective fading and flat fading. The relative movements between the base station and mobile objects in the environment cause the time-varying nature of fading channels. Doppler shift for measuring the rate of variation of the channel characteristics and coherent time determine the fading channel is fast or slow.

In this project, a Single-Input Single-Output (SISO) system with small-scale fading which caused by multipath are considered. Considering a quasi-static fading channel, the channel impulse response (CIR) between transmitter and receiver [3] is given by

$$f(t) = \sum_{n=0}^{L-1} f_n \delta(t - \sigma_n) \tag{2.7}$$

where L is the total number of paths, f_n and σ_n are the channel gain and delay for the $n^{th} (n = 0, \dots, L)$ path, respectively.

Let g (t) represents the total effects of the transmit filter and receive filter, the convolution of the physical CIR f (t) with g(t) gives the overall CIR h(t):

$$h(t) = \sum_{n=0}^{L-1} f_n g(t - \sigma_n) \tag{2.8}$$

Signals with different attenuations, delays and phase shifts arrive at the receiver as a result of multiple copies of transmitted signal through multipath. So the users undergo the signal distortion and longer delay spread is, larger effects of multipath are. The root-mean-square (RMS) delay spread σ for measuring the time dispersion of multipath channels to measure multipath channel is expressed as

$$\sigma = \sqrt{\frac{\sum_n f_n^2 \sigma_n^2}{\sum_n f_n^2} - \left(\frac{\sum_n f_n^2 \sigma_n}{\sum_n f_n^2}\right)^2} \qquad (2.9)$$

Coherence bandwidth B_c is used to describe the channel delay spread in frequency domain. It is defined as frequencies separated by less than this width have their fades highly correlated (i.e. channel bandwidth) [3]. It is in the inverse ratio of RMS delay spread. For frequency correlation higher than 0.5, their relations are:

$$B_c = \frac{1}{5\sigma} \qquad (2.10)$$

The differences between flat and frequency selective fading channel depend on the characteristics of the transmitted signal such as signal period *Ts* and signal bandwidth *Bs* and the nature of multipath like the RMS delay spread σ as well. The conditions are shown as follows.

Table 2.1 Conditions of fading classification caused by Multipath

	Frequency Domain	**Time Domain**
Frequency Selective Fading	$Bc<Bs$	$\sigma > Ts$
Flat Fading	$Bc>Bs$	$\sigma < Ts$

2.2.1 Flat Fading Channel

If coherence (channel) bandwidth is greater than message signal bandwidth, (i.e. $Bc > Bs$) which also means the RMS delay spread is smaller than the signal period (i.e.$\sigma < Ts$), all the frequency components in the message will arrive at the receiver with little or no distortion [3]. Flat fading means the all the frequencies over the transmission bandwidth have similar channel gain and linear phase. Within the coherence bandwidth the frequency components of the signals have highly correlation on the amplitude. Fading is the same for different frequencies and the received signal undergoes a single channel fading. The received signal can't be decomposed so there is no ISI and the delay spread can't caused the overlapping of the adjacent signals. There are some examples of fading models for the distribution of the attenuation such as Nakagami fading, Weibull fading, Rayleigh fading and Rician fading.

Rayleigh fading channel is used as the wireless communication channel simulation model in this project. The pdf of channel gain $|f_n|$, the amplitude envelope of complex channel response is:

$$p(|f_n|) = \frac{2|f_n|}{\Omega}\exp(-\frac{|f_n|^2}{\Omega}) \qquad (2.11)$$

where Ω is mean power of the channel response. This model is suitable for using in the urban environments because there is no dominant propagation along a line of sight (LOS) between the transmitter and receiver due to around trees and buildings. Instead, there are some reflection, diffraction and scattering of the transmitted signals.

In Rayleigh fading channel of SISO system, the received signal y is the addition of AWGN n and the multiplication of the transmitted signal x and Rayleigh channel gain h.

$$y = x * h + n \qquad (2.12)$$

The channel effect can be reduced by the equaliser *1/h* at the receiver z. Then the received signal looks like to be transmitted in AWGN channel.

$$z = \frac{1}{h} * y = x + \frac{n}{h} \qquad (2.13)$$

Average BER in Rayleigh fading channels is:

$$P_e = \int_0^\infty P_e(\gamma)f(\gamma)d\gamma \qquad (2.14)$$

Where γ stands for the ratio of the received signal energy per bit to noise spectral density $f(\gamma)$ is probability density function.

2.2.2 Frequency Selective Fading Channel

If coherence (channel) bandwidth is smaller than message signal bandwidth, (i.e. *Bc* <*Bs*) which also means the RMS delay spread is larger than the signal period (i.e.$\sigma > Ts$), different frequency components of the signal go through uncorrelated fading [3]. In frequency selective fading channel, the signal energy of each symbol is spread out in time. Fading is different for different frequencies and the received signal undergoes multiple fading channels. The long delays of previous transmitted symbol and short delay of the current one may cause the overlap at the receiver, which is called inter-symbol interference (ISI). In other words, the frequency components of the received signal have different amplitudes. It introduces errors in the detection of the receiver so that its effects should be minimised. Equalizers are often used for compensation.

2.3 Simulation results

Following simulation results show both the characteristics of AWGN channel and Rayleigh fading channel. Rayleigh fading channel CIR and Rayleigh distribution are shown in Figure 2.3 and 2.4. “Equal gain” delay profile is assumed for Rayleigh channel generation. In figure 2.4, the mean power of the channel response Ω is set as 0.5W, 1W and 3W and 10000 samples are used for plotting Rayleigh fading distribution in theory. It can be seen that more distributed on small x with lower mean power while there is homogeneous distribution with enough power. The approximation of Rayleigh fading distribution with 1W mean power is also shown.

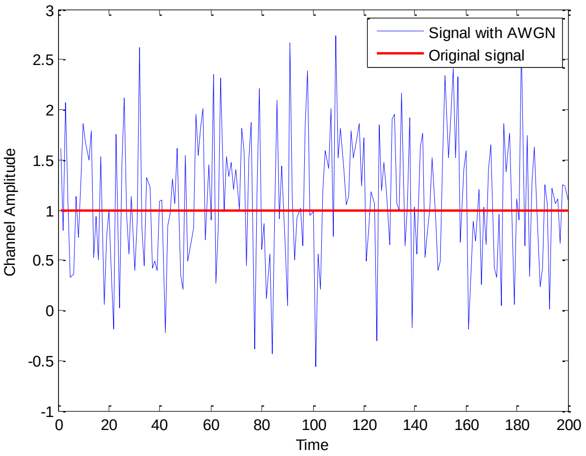


Figure 2.2 Signal through AWGN channel

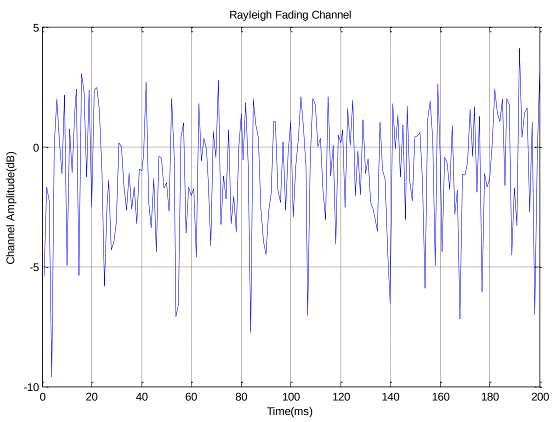


Figure 2.3 Rayleigh fading channel CIR

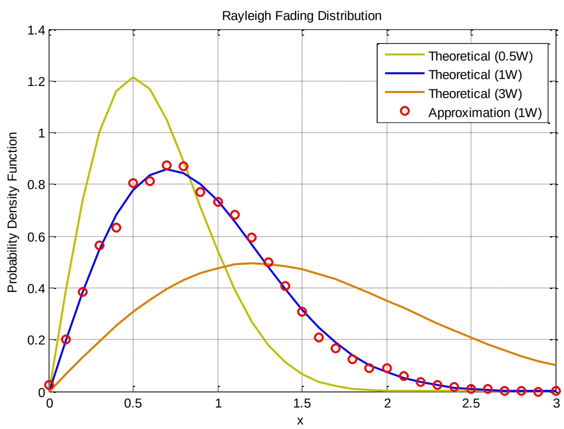


Figure 2.4 Rayleigh fading distribution

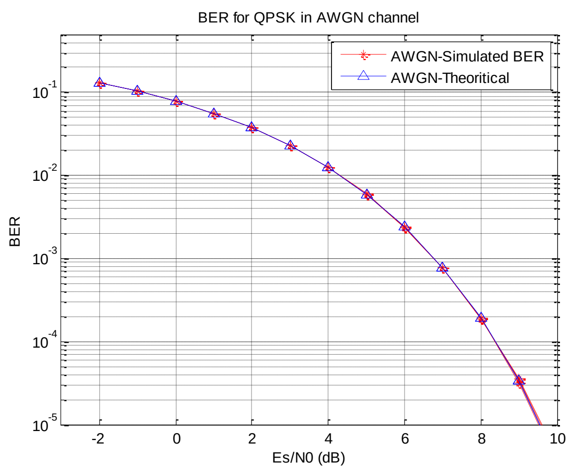


Figure 2.5 BER for QPSK in AWGN channel

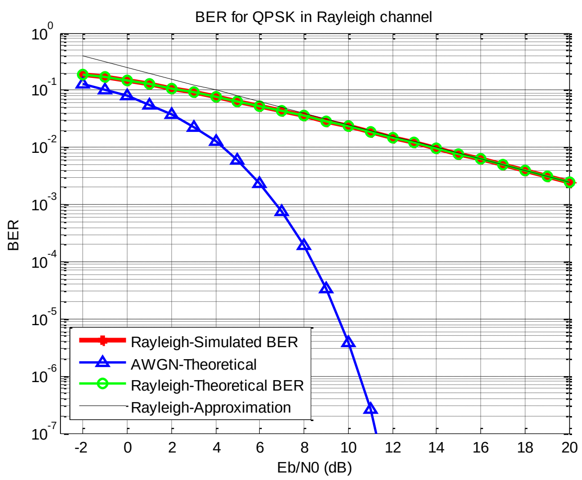


Figure 2.6 BER for QPSK in Rayleigh channel

Quadrature Phase Shift Keying (QPSK) is one of digital modulation technique which uses four phases of carriers to represent the transmitted signal information. Two branches in QPSK are coherent *I* branch and non-coherent *Q* branch. The advantages of QPSK such as high frequency spectrum utilization ratio, simple implementations and good anti-jamming nature make it becomes one of most important modulation method in the wireless communication.

Figure 2.5 and 2.6 show the performance of system using QPSK in both AWGN and Rayleigh channels. Following Table gives the BER equations of QPSK with both channels in theory.

Table 2.2 Theoretical BER of QPSK in AWGN and Rayleigh channels

	AWGN Channel	**Rayleigh fading Channel**	
	Theoretical	Theoretical	Approximation
QPSK_BER	$\frac{1}{2}\mathrm{erfc}(\sqrt{\frac{E_b}{N_0}})$	$\frac{1}{2}(1-\sqrt{\frac{E_b/N_0}{1+E_b/N_0}})$	$\frac{1}{4 * E_b/N_0}$

In the above equations, $\frac{E_b}{N_0}$ stands for the signal to noise ratio (SNR) which is an important factor to determine the maximum data rate for error-free communication over a noisy channel [3] in digital communication system where E_b is bit energy and N_0 is the power spectral density of AWGN. 10000 symbols and 1000 channels are set in the simulation and N_0 is 1 for simplicity. From the equation of BER approximation for Rayleigh channel, it can be seen that BER is in the inverse ratio of $\frac{\mathrm{E_b}}{\mathrm{N_0}}$. In the other words, probability of errors decreases almost linearly with the increase of SNR. Lower BER brings much better performance.

Maximum data rate over a channel which is called channel capacity is affected by channel bandwidth and noise level. Shannon' theorem gives the theoretical channel capacity for error-free communications over an AWGN channel [3]:

$$C = B\log_2\left(1+\frac{S}{N}\right)\ bps \qquad (2.15)$$

Where B means bandwidth, S/N is SNR. Only when the data rate R is less than channel capacity, there is probability for the error-free transmission. Then it can be deduced that the minimum $\frac{\mathrm{E_b}}{\mathrm{N_0}}$ required for reliable communication is larger than -1.6 dB. So the start point of $\frac{\mathrm{E_b}}{\mathrm{N_0}}$ is set as -2dB in the simulation.

From the above BER simulation results in AWGN and Rayleigh channel, it can be seen that BER in Rayleigh channel is much higher than that in AWGN channel. It is obvious that the performance of communication system is much better in the ideal case. In real communication system, $\mathrm{BE}R < 10^{-5}$ is acceptable. In ideal AWGN channel, it can be achieved when $\frac{\mathrm{E_b}}{\mathrm{N_0}} > 10dB$ while for Rayleigh channel, the linear reduction of BER with the increase of the SNR is not so much. The gap becomes larger for higher SNR. The lowest BER in Rayleigh channel is larger than 10^{-3} with 20dB SNR which is much worse.

3. OFDM Systems

3.1 System Model

A general downlink time division duplexing (TDD) OFDM system with K users which meets QoS requirements is considered as shown in Figure 3.1 [1]. TDD which is used to separate the transmitting and receiving channels (i.e. uplink and downlink) is one of the duplexing techniques in mobile communication system. The information can be transmitted or received in the same frequency band of channel (i.e. allocate different slots separated by a guard time). Accordingly, frequency division duplexing (FDD) should guarantee different frequencies. Due to the asynchronism of TDD, it has a big advantage on spectrum efficiency especially in the case with asymmetry uplink and downlink data rates [13]. The interconnections between MAC and PHY layer utilize the spectrum and energy efficiently due to the limited bandwidth, the resource competitions among users and the time-varying channels. Channel state information (CSI) which measures the channel quality can be acquired through the uplink from the mobile station to base station for the further resource allocation at the PHY layer. QoS information and resulted subcarrier and power information are used for the scheduling by the traffic controller at the MAC layer. Then it feeds back to the PHY layer for the system optimization. All these implementations are benefit from plenty of subcarriers by the OFDM techniques.

With the time-slotted OFDM signaling, a quasi-static fading channel where during each slot the channel gain is constant and independent with other slots [3]. It is assumed that in such system a total bandwidth B is shared by K users with independent N subcarriers, Ω_{k} represents the index set of the subcarriers allocated to user k (k=1…K), $\mathrm{p}_{\mathrm{k,n}}$ is the power allocated to user k on subcarrier $n \epsilon \Omega_{\mathrm{k}}$, $\mathrm{h}_{\mathrm{k,n}}$ denotes the corresponding channel gain, and N_0 stands for the power spectral density of AWGN [8][1]. With perfect channel estimation, the achievable total data rate of user k can be reduced from mentioned Shannon Theorem:

$$R_k = \sum_{n\epsilon\Omega_{\mathrm{k}}} R_{k,n} = \sum_{n\epsilon\Omega_{\mathrm{k}}} \frac{B}{N} log_2\left(1 + p_{k,n}\gamma_{k,n}\right) \qquad (3.1)$$

where
$$\gamma_{k,n} = \frac{|\hslash_{k,n}|^2}{N_0 * (\frac{B}{N})} \qquad (3.2)$$

is the channel-to-noise power ratio (CNR) for user k on subcarrier n.

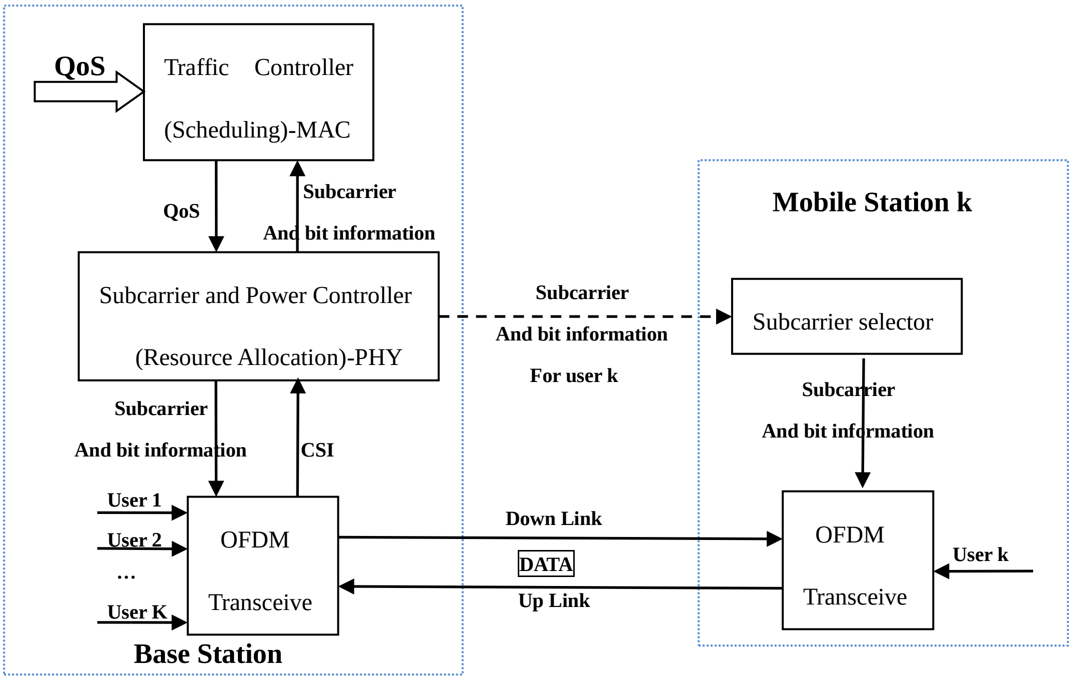


Figure 3.1 Multiuser OFDM System Model [8]

3.2 Frequency Domain Equalisation

The shortages of frequency selective fading like ISI have been introduced in part 2.2.2. Frequency domain equalisation (FDE) is one way to solve this problem. OFDM technique, which belongs to *multi-carrier* modulation (MCM), can overcome it [3]. It divides the whole signal band into many orthogonal narrow bands with sufficient large number of subcarriers. All the subcarriers are transmitted in parallel and orthogonal signals can be separated by the related technology at the receiver.

As one of the promising high data rate transmission technology for future wireless communication, the history of OFDM is more than 40 years. With the development of Fast Fourier Transform (FFT) technique for multicarrier modulation the restriction that high complexity of OFDM system implementation has been removed.

OFDM becomes more and more practical for its variety of advantages. Figure 3.2 illustrates a typical realization of orthogonality among all subcarriers. It can be seen that at the peak of the spectral centered at the frequency fc, all the neighboring spectra are zero so there is no “inter-carrier interference” (ICI) due to the orthogonal subcarriers. It also leads to high spectrum efficiency. In the time domain, the insertion of cyclic prefix (CP) avoids the ISI if the length of CP is larger than the maximum delay caused by the multipath. CP is to extend the OFDM symbol by copying the last samples into its front. On the others, energy will be lost if the length of CP is too long especially in the system with few subcarriers. So OFDM is robust against frequency selective fading and multipath. Each sub-carrier is modulated with a conventional modulation scheme at a low symbol rate, maintaining total data rates similar to conventional *single-carrier* modulation (SCM) schemes in the same bandwidth. Furthermore, MCM does not require a high-complexity equalizer as SCM while it is suitable for high data rate transmission. Compared with conventional frequency division multiplexing (FDM) which does not overlap the carriers with guard band, OFDM system can achieve higher spectrum efficiency.

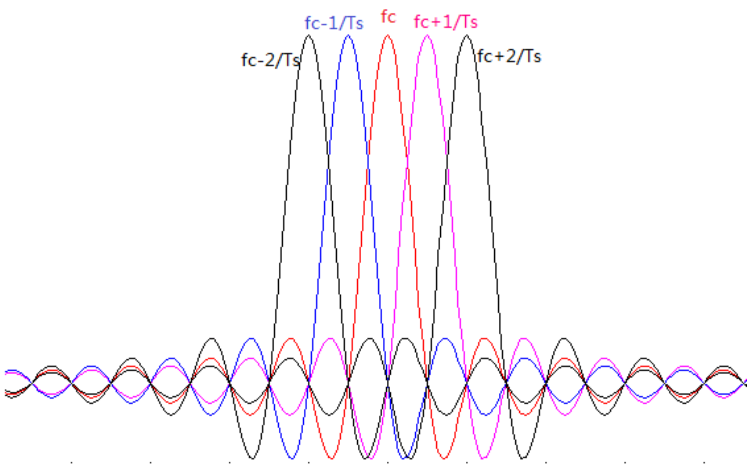


Figure 3.2 Five subcarriers waveform in OFDM system

The principles of Modulation and demodulation in the OFDM system are shown in the following block diagram Figure 3.3 [12][13]. OFDM transmitter maps the message bits into a sequence of symbols (i.e. BPSK, Quadrature Amplitude Modulation (QAM)) which will be converted into N parallel streams through serial to parallel (S/P) conversion, one for each subcarrier. N symbols are carried by different subcarriers and each symbol period is extended as $T_{sym} = NT_s$. Correspondingly in the frequency domain, the frequency of each symbol is narrowed into $f_{sym} = B_s/N$ which eliminates the ISI problems in the multipath environment. As a result, the signal undergoes the flat fading. CP is added to prevent inter-block interference. Simple FDE can be performed on each subcarrier independently, using methods like zero forcing (ZF). Its complexity increases linearly with number of IFFT.

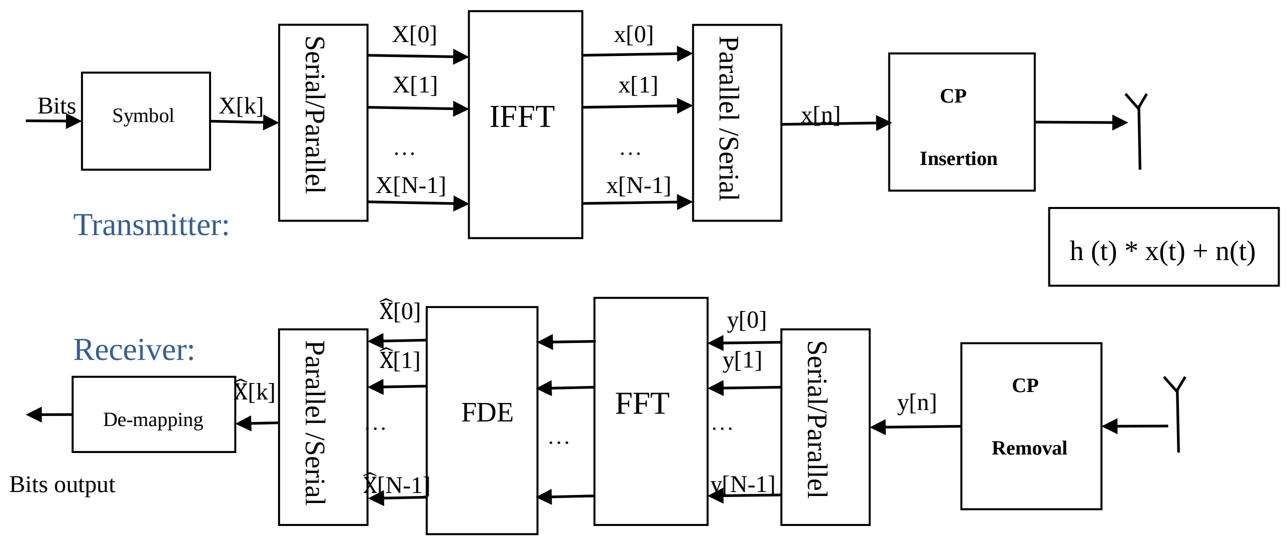


Figure 3.3 Block diagram of the OFDM System

Let X[k] denote the transmit symbol at the k^{th} subcarriers, k = 0, 1, 2… N-1. After the inverse Fast Fourier Transform (IFFT), the corresponding OFDM symbol in time domain is:

$$x[n] = \sum_{k=0}^{N-1} X[k]e^{j2\pi kn/N} \quad \text{for } n = 0,1, \dots N-1 \qquad (3.3)$$

In this case, the OFDM system channel is Rayleigh multipath channel with AWGN, so the received signal can be represented as the convolution of the transmitted signal and CIR of Rayleigh channel h[n] and then add with the AWGN n[n]:

$$y[n] = h[n] * x[n] + n[n] \quad \text{for } n = 0,1, \dots N-1 \qquad (3.4)$$

The received OFDM symbol can be reconstructed by the orthogonality among the subcarriers as follows:

$$\hat{X}[k] = \sum_{n=0}^{N-1} y[n]e^{-j2\pi kn/N} \quad \text{for } n = 0,1, \dots N-1 \qquad (3.5)$$

Furthermore, from equation 3.3, the frequency domain symbol X[k] modulates the subcarrier with a frequency of $f_k = f_c + k/T_s$.

3.2 Simulation Results

Table 3.1 Simulation Parameters Setting

Parameter	Value
Multipath	6
FFT Size	64
Number of subcarriers	64
Number of bits per symbol	64
Total number of symbols	10000
Length of CP	3 or 16
Total length of OFDM signal	67 or 80

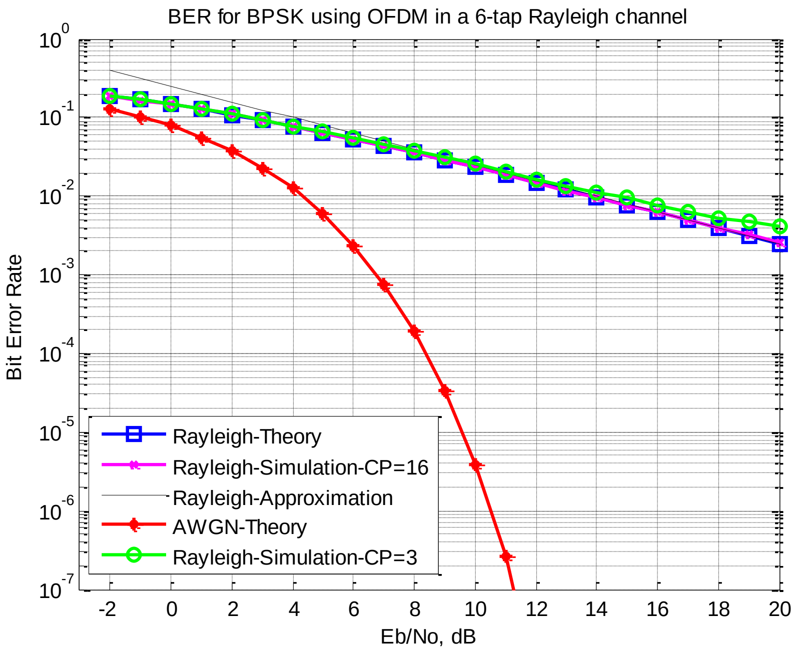


Figure 3.4 BER for BPSK in OFDM system (Rayleigh)

Binary Phase Shift Keying (BPSK) is used as the modulation method for BER calculation in OFDM system. From the BER results of QPSK (Figure 2.5, 2.6) and BPSK in OFDM (Figure 3.4) with AWGN and Rayleigh channel, it can be seen there is no difference. This is because BER for

BPSK and QPSK are the same in theory. Seen from Table 3.1, parameters are set similar to the standard of IEEE 802.11a without zero padding which is 64-point FFT, 64 subcarriers (as well as the number of bits per symbol), 0.8 us Cyclic prefix duration, 3.2 us Data symbol duration (16-bit CP) so total symbol size is 80. 1000 channels are repeated for the average simulation results. This OFDM system is simulated with flat fading without ISI. When CP is small (i.e. CP=3->frequency selective fading), the decrease of BER with the increase of Eb/No (above 14dB) is not as much as that with large CP (i.e. CP=16->flat fading). This implies that the OFDM system is just subject to a flat fading channel as long as CP is larger than the delay. The length of CP has no influence in the AWGN channel because there is no multipath delay. It is obvious that OFDM system takes advantages on the Rayleigh multipath channel from frequency selective fading to flat fading because the effect of ISI on the BER performance becomes significant as the length of guard interval decreases.

4. Cross-layer Resource Management for OFDM systems

4.1 Cross-layer Design

Nowadays the requirements of infinite variety service types such as broadband data service, low-delay voice or video service keep growing. The real-time media transfer should guarantee the low time delay but can tolerance some errors. While reliability is the most important factor for the non-real-time transfer such as web access or download files which tolerances some time delay. The conventional layer-separated design no longer satisfies the current network environment to respond with flexibility and speed of ever-changing customer demand.

Table 4.1 OSI Reference Model

Application Layer	
Presentation Layer	
Session Layer	
Transport Layer	
Network Layer	
Data Link Layer (DLL)	Logical Link Control (LLC)
	Medium Access Control (MAC)
Physical Layer (PHY)	

The famous OSI Reference Model was proposed by International Standard Organization (ISO) for network communication in 1983 [12]. It is s seven-layer model shown in Table 4.1. Upper layer achieves data transformation with the services provided by the lower layer. Each layer contains its own independent protocol. The bottom PHY layer is used to transmit the raw bits from one machine to another. Error-free data transformation between several entities connected by a single hop or link is enabled at data link layer (DDL). Its MAC sublayer controls access to shared channel. Above network layer is in charge of controlling the intra-network operation. QoS, the technique for solving problems of delay and congestion is controlled at this layer. During communication process, some main problems such as dropping data packets, delay and other transmission error exist. QoS can be guaranteed by the suitable broadband, jitter, delay and packet drop rate reductions. With the limitation of the broadband, high resource efficiency can be achieved by correct resource management depending on the characteristics of different services. Next transport layer is responsible for the correct transformation from source to destination. And the top application layer contains the protocols and directly serves users. In the conventional layer-separated design which is suitable for the wired network, each layer is designed and optimized independently. The communication only exists between the adjacent layers. Such kind of structure simplifies the network design significantly with stability.

Different from wired network, wireless communication has higher BER because the time-varying wireless channel which is affected by multipath fading, shadowing and other interference is not stable. Furthermore, the limitation of transmitted power and bandwidth also lead to signal fading. That is the reason of using CSI to measure the states of diverse channels. Under this background, cross layer design, the concept of joint optimization over multiple layers in system design has been put forward [2]. The mentioned dynamics require the corporations among layers to optimize the total quality of network. For instant, MAC layer can obtain information like types of service or related QoS parameters from upper layers to adjust itself to the dynamic traffic.

The interactions among layers should be considered in the cross layer design. This project focuses on the corporations of PHY layer and MAC layer as mentioned in section 3.1. The role of PHY layer is providing the parameters of CSI for upper layers. The system throughput and acquired power can be improved because of the real-time adjustments at the upper layers. MAC layer which is the bottom sublayer of DLL controls if transmit the data or not. Data with added control information will be transmitted to PHY layer in the specific format. The adaptability at MAC and PHY layer in response to application services is a very important factor in a simple cross layer design [2]. MAC should distinguish the type of service for supporting the heterogeneous traffics and map it to appropriate PHY configurations. The PHY parameters should be adjusted for best system performance. The cross layer design transfers and coordinates the information among layers to satisfy variety demands of services. Sharing information among layers brings high quality of network and brings the suitability of wireless communication environment.

The objective of the adaptive cross layer design for OFDM systems in this project is to maximize the weighted sum capacity of all users [12][1]. As mentioned as in section1.3, at the MAC layer, a PD scheduling scheme determines the packet transmission order by the weight assignment for each packet. The summation of the assigned weights of packets for each user is used in the resource allocation at the PHY layer. The considerations of the weight design deeply influence the performance of the system. More details will be given in the following part. QoS, one of important factor of the performance measurement is considered by jointly channel-aware and queue-aware data scheduling in [15].

4.2 Resource Allocation at the PHY Layer

Resource allocation is used to assign the available resources in an economic way which plays a very important role at PHY layer in OFDM wireless communication systems [16][8][1][13][7]. A subset of subcarriers is allocated to each user and the number of subcarriers should be scheduled by the system. A group of subcarriers as a basic unit of resource allocation in OFDM system is defined as *subchannel*. Since the independent channel responses vary on different subcarriers for different user, the subcarrier allocation for multiple users improves the system performance. Subcarrier allocation means allocating the subcarrier based on the static or dynamic allocation for the highest spectrum efficiency. For example, some channels are deep fading while they are in good condition for the other users. The system can always choose the subcarrier with the best channel-to-noise ratio (CNR) for each user. The increased system capacity is benefit from the plenty subcarriers in OFDM system. On the other side, power allocated on the independent subcarriers can vary for better system performance. In a word, the resource allocation consists of adaptive subcarrier allocation and power allocation.

The dynamic resource allocation in the multiuser OFDM system has been attracted enormous research interests. In [16], the sum capacity can be maximized if allocate resource according to the best channel gain. Water filling algorithm is used in the power allocation while it is not practical due to the loss of the fairness. And fairness can be guaranteed by the proportional fairness coefficients while not considered QoS [7]. Margin Adaptive [17] achieve the minimum overall transmit power with the data rate and BER constraints. Rate adaptive aims to maximize each user’s error free capacity with the constraints of total transmit power. The max-min optimization proposed in [18] can maximize the worst user’s capacity with similar data rate for each user. The optimization only provides maximum fairness among the users while data rate requirements are different for different users in practical wireless networks. The resource allocation and scheduling which is not practical due to only one connection is based on the utility function which reflects QoS [3].

Power allocation plays a very important role in the resource allocation. There are three typical classifications of power allocation. The simplest static one is equal power allocation for each subcarrier. It benefits from its low complexity and simple implementation while it can’t adapt to the changing environment and not contribute the improvement of system performance. Another one is the joint one which means both two factors CSI and power should be considered simultaneously for resource allocation. Generally, such kind of optimization problem of resource allocation is very hard to solve due to the non-linear constraints [7][8][1]. Furthermore, it is computationally complex to find the optimal solution with a large number of subcarriers. The last one substep method is more implementable and used widely in current research. Subcarrier allocation and power allocation are implemented separately. Equal power is allocated for each subcarrier first for subcarrier and then power is allocated again according to the results of subcarrier allocation. It takes advantage of its low complexity while the solution is just suboptimal while not optimal. Whatever such low-complexity suboptimal scheme which separates the subcarrier and power allocation is desired for cost-effective and delay-sensitive implementations. Without loss of the generality and for simplicity, each subcarrier is assigned for only one user. Three developed suboptimal resource allocation schemes will be reviewed first and a proposed one will be then presented in this part.

4.2.1 Subcarrier Allocation

4.2.1.1. Maximum Capacity (MC) based Subcarrier Allocation

In Maximum Capacity (MC) based resource allocation [16][8], each subcarrier is allocated to the only one user with the best channel gain on that subcarrier, and the power allocation is distributed over subcarrier by the water-filling algorithm. The optimization formulation for MC with some constraints can be expressed as following:

$$\max\ J = \sum_{k=1}^{K} R_k \tag{4.1}$$

$$s.t. \begin{cases} p_{k,n} \geq 0 \\ \sum_{k=1}^{K} \sum_{n \in \Omega_k} p_{k,n} \leq P_t \\ \Omega_i \cap \Omega_j = \phi (i \neq j) \\ \Omega_1 \cup \Omega_2 \cup \dots \Omega_K \subseteq \{1,2,\dots,N\} \end{cases}$$

where P_t is the total power which is transmitted from the base station.

The constraints show that power allocated to each user in each subcarrier is guaranteed to be positive and the summation of the allocated power should be equal to the total transmitted power. Each subcarrier is allocated to one user which deduces different user should not be allocated the same subcarrier and total number of allocated subcarriers for K users should be N.

The limitation of the MC scheme is that only when users have enough data to send, the system capacity can be maximized. And the resource allocation does not depend on the QoS requirements. That means more resources may be occupied by the users with not so urgent traffic demands. The shortage of delay sensitive data is also impracticable. The implementation method of MC based methods is very simple that each subcarrier is allocated to the user who has the highest CNR.

4.2.1.2. Proportional Fairness (PF) based Subcarrier Allocation

A proportional fairness (PF) scheme was proposed for resource allocation to guarantee the fairness among users with proportional fairness coefficients employed at the cost of system capacity [7][8]. The method balances the tradeoff between capacity and fairness by adding the constraint $R_1 : R_2 : \dots : R_K = \eta_1 : \eta_2 : \dots : \eta_K$ for setting the data rate of each user by the predetermined rate ratios. Such constraint can guarantee the fairness no matter how much better channel gains for users there are. The related optimization formulation is shown as following, and in this project, all the users are assumed to have equal data rates (i.e. $\eta_k = 1$):

$$\max\ J = \sum_{k=1}^{K} R_k \tag{4.2}$$

$$s.t. \begin{cases} p_{k,n} \geq 0 \\ \sum_{k=1}^{K} \sum_{n \in \Omega_k} p_{k,n} \leq P_t \\ \Omega_i \cap \Omega_j = \phi (i \neq j) \\ \Omega_1 \cup \Omega_2 \cup \dots \Omega_K \subseteq \{1,2,\dots,N\} \\ \boldsymbol{R_1 : R_2 : \dots : R_K = \eta_1 : \eta_2 : \dots : \eta_K = 1} \end{cases}$$

Equal power distribution (i.e. $p_{k,n} = P_t/N$) is assumed across all the subcarriers for suboptimal subcarrier allocation algorithm [7] which can be shown in Figure 4.1. L and Ω_k are the sets of unallocated and allocated subcarrier respectively. Their elements stand for N subcarriers. For each user k=1, 2…K, the best subcarrier n with highest CNR should be allocated (i.e. add n into Ω_k and remove from L). Simultaneously, the data rate of user k is update according to the equation (3.1). As long as there is still some unallocated subcarrier, the user with lowest date rate should be found and assigned with best subcarrier among the rest ones. Update the data rate and the set of allocated subcarrier for this user. The allocation process finishes when L is empty. This algorithm makes sure each user can use the subcarrier with high CNR and the user with minimum capacity has the priority to pick the subcarrier which guarantees the fairness.

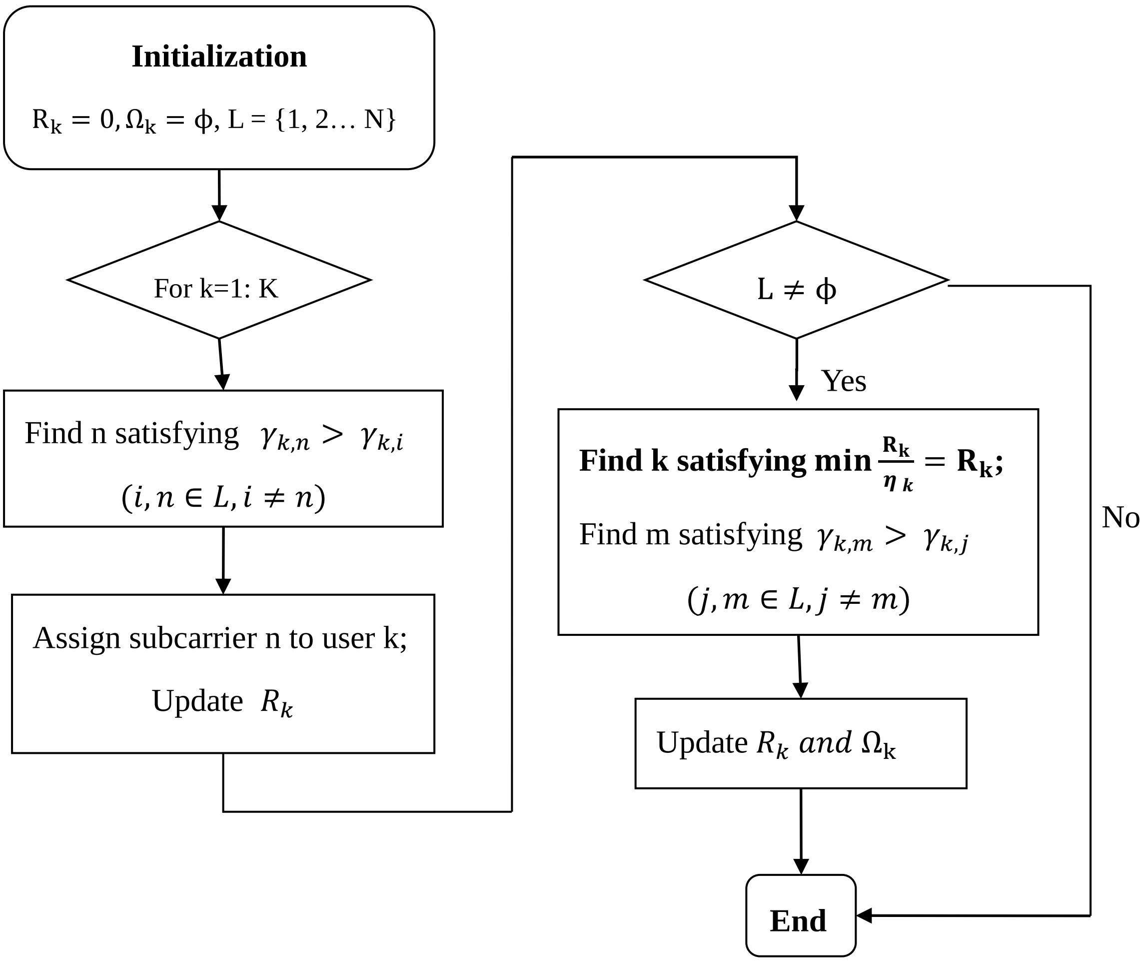


Figure 4.1 Flow chart of PF based subcarrier allocation

4.2.1.3. Weighted Maximum Capacity (MWC) based Subcarrier Allocation

Both schemes based on MC and PF aim to maximize the total capacity while not consider the QoS information. In weighted maximum capacity (MWC) based scheme [8][1], the weighted sum of all user's capacity is required to be maximized. The formulation of optimization with constraints is shown in following. There is no limitation of data rate for each user:

$$\max \ J = \sum_{k=1}^{K} \boldsymbol{W_k} R_k \tag{4.3}$$

$$s.t. \begin{cases} p_{k,n} \geq 0 \\ \displaystyle\sum_{k=1}^{K} \sum_{n \in \Omega_k} p_{k,n} \leq P_t \\ \Omega_i \cap \Omega_j = \phi (i \neq j) \\ \Omega_1 \cup \Omega_2 \cup \ldots \Omega_K \subseteq \{1, 2, \ldots, N\} \\ R_k T \leq Q_k \end{cases}$$

Where W_k denotes the weight for user k which contains QoS information and it can be achieved by the data scheduling at MAC layer through weight design. The length of each time slot is set as T and Q_k is the queue length of user k. Last constraint increases the resource efficiency because resource is allocated only when it is not sufficient to send all data in one slot. The resource which may be wasted can be allocated to other users.

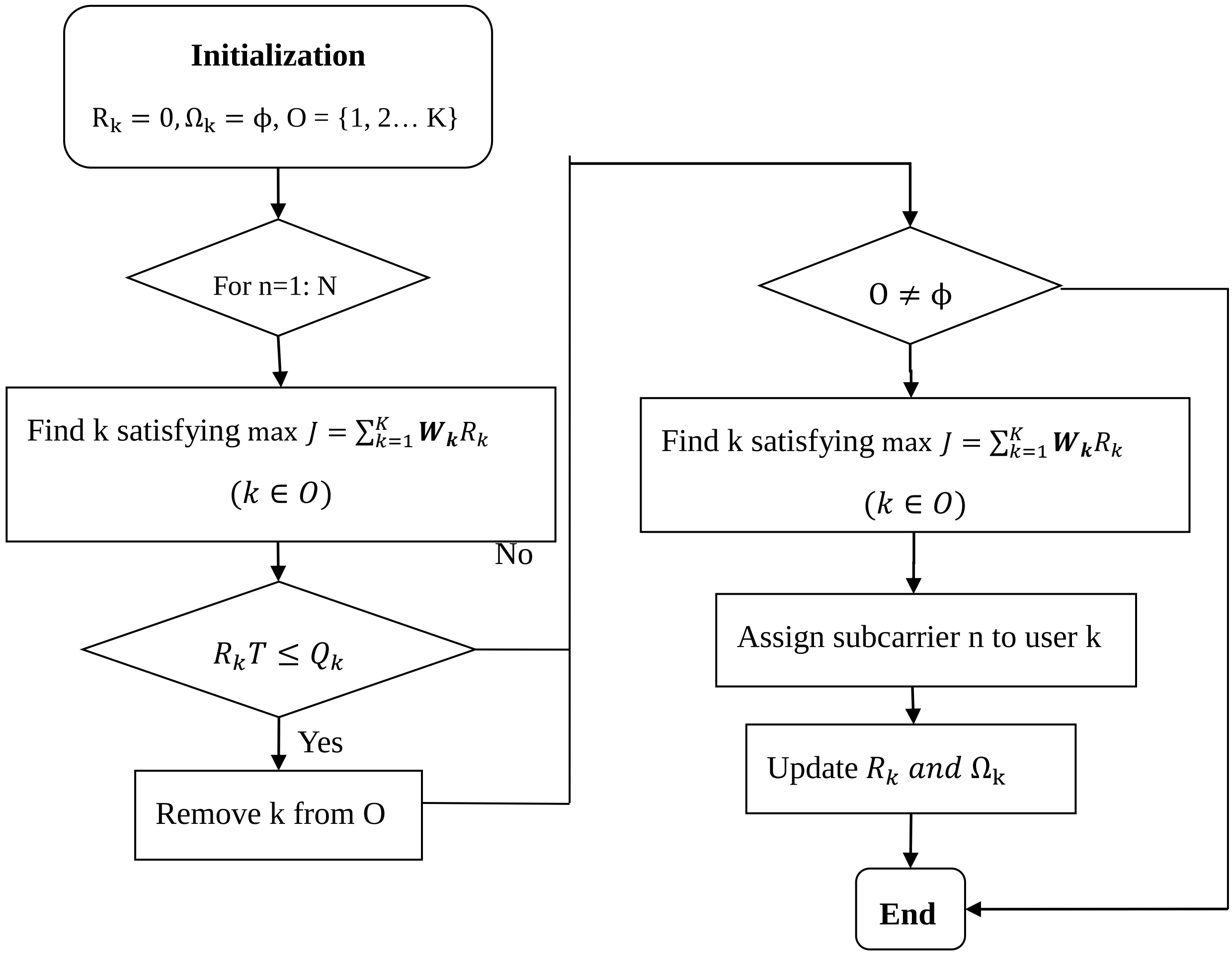


Figure 4.2 Flow chart of MWC based subcarrier allocation

Compared with that of MC based subcarrier allocation, the weight in the cost function should be considered jointly. As mentioned before, suboptimal allocation is considered for optimization for low complexity especially in real time system. Equal power distribution (i.e. $p_{k,n} = P_t/N$) is assumed across all the subcarriers for suboptimal subcarrier allocation algorithm [8] which can be described in Figure 4.2. Ω_k is the sets of allocated subcarrier and O is the index set of user which requires resource. Their elements stand for N subcarriers and K users respectively. Each subcarrier is assigned to the user who satisfies the optimized condition (i.e. max $J = \sum_{k=1}^{K} \boldsymbol{W_k} R_k$). Once there is enough resource for one user, the user will be removed and will not be assigned with subcarriers any more. The data rate and the set of allocated subcarrier for this user should be update in each loop. The allocation process finishes when O is empty (i.e. each user contains sufficient resource). This algorithm also makes sure each user can use the subcarrier with high CNR but not as much as MC based one. It considers QoS information which is much more practical for heterogeneous traffic data. The improvement is that the weights based on QoS information are used to determine the subcarrier pick order because a larger weight demands a higher data rate. It can be deduced to MC based allocation by setting $W_k = 1$ and a special case in PF based allocation by setting equal data rate.

Above discussed MWC based subcarrier allocation is based on the user number, it can be easily to be extended into queue based one [1] by replacing user k with queue i. Each user is assumed to have I queues. There are IK queues in total. The cost function can be changed into:

$$\max \ J = \sum_{i=1}^{IK} W_i R_i \tag{4.4}$$

$$s.t. \begin{cases} p_{i,n} \geq 0 \\ \sum_{i=1}^{IK} \sum_{n \in \Omega_i} p_{i,n} \leq P_i \\ \Omega_i \cap \Omega_j = \phi (i \neq j) \\ \Omega_1 \cup \Omega_2 \cup \dots \Omega_{IK} \subseteq \{1,2, \dots, N\} \\ R_i T \leq Q_i \end{cases}$$

Because each user has more than one queues generally (i.e. $IK \geq K$), the complexity of conventional queue based MWC is much higher.

4.2.1.4. Modified Weighted Maximum Capacity (M-MWC) based Resource Allocation

Compared with MWC based scheme, the proposed modified weighted maximum capacity (M-MWC) considers the weight not only for each user but also for each subcarrier (i.e. $W_{k,n}$ denotes the weight for user k and subcarrier n) which maximize the capacity in higher degree. Its implementation is the same as that of MWC. The optimization formulation maximizes the summation of weighted data rates for each user and each subcarrier to get more capacity is the modified as follows:

$$\max \ J = \sum_{k=1}^{K} \sum_{\boldsymbol{n \in \Omega_k}} \boldsymbol{W_{k,n} R_{k,n}} \tag{4.3}$$

$$s.t. \begin{cases} p_{k,n} \geq 0 \\ \sum_{k=1}^{K} \sum_{n \in \Omega_k} p_{k,n} \leq P_t \\ \Omega_i \cap \Omega_j = \phi (i \neq j) \\ \Omega_1 \cup \Omega_2 \cup \dots \Omega_K \subseteq \{1,2, \dots, N\} \end{cases}$$

4.2.2 Power Allocation

4.2.2.1. Water-filling (WF) Algorithm

With the limitation of total transmitted power, the best power allocation method for maximising the data rate is water-filling algorithm which is described as follows [13]. There is a power threshold which is decided by the environment (i.e. fading or SNR). Worse the subchannel is, the value of the subchannel CNR is closer to that threshold. Power allocation depends on the difference between CNR and threshold. That means more power will be allocated to the better subchannel. It is just like the situation of filling the tank by water. The distributions of the bottom of the tank are presented by CNR. The threshold is the height of water level. The allocated powers are the depth of water. That is why it called a water-filling algorithm. It is note that if the bottom of the tank is higher than the water surface, there is no allocated power for that subchannel. It implies that the poor enough channel will not be used to transmit data. During the process, the summation of total signal and noise power on each subchannel should be a constant which matches the fluidity of water.

Power allocation for maximizing the cost function in this project can be obtained by using the Lagrange multiplier $\{\lambda_i\}_{i=1}^{K}$:

$$J = \sum_{k=1}^{K} R_k + \lambda\left(\sum_{k=1}^{K} \sum_{n\epsilon\Omega_k} p_{k,n} - P_t\right) \qquad (4.4)$$

$$\mathrm{s.t.}\begin{cases} \displaystyle\sum_{k=1}^{K} \sum_{n\epsilon\Omega_k} p_{k,n} = P_t) \\ p_{k,n} \geq 0 \end{cases}$$

The optimal solution is obtained by:

$$p_{k,n} = \begin{cases} \dfrac{P_t + \sum_{i=1}^{K} \sum_{q\epsilon\Omega_j} \frac{1}{\gamma_{i,q}}}{size\ of\ (\Omega_k)} - \dfrac{1}{\gamma_{k,n}} \\ 0 \end{cases} \qquad (4.5)$$

The allocated power should be positive, otherwise the corresponding subcarrier will be removed and all the related parameters will be updated. Then power allocation will be implemented again and again until there is no negative one. The removed subcarriers are allocated without power. This proves that the poor channel is not used for transmission. The total data rate in OFDM system varies by the power allocation to the subcarrier. There are connections between the total transmitted power and data rate.

The above equations give the suboptimum solutions of power allocation which can be combined with MC and PF based subcarrier allocation. Although the suboptimum solutions can be achieved by the water-filling algorithm in multiuser system, the complexity of implementation increases with the increase of the user numbers. So the key of the resource allocation is the efficient subchannel allocation algorithm.

4.2.2.2. Weighted Water-filling (WWF) Algorithm

Based the classical water filling power allocation, the optimal power allocation [1] for maximizing the cost function with weight for MWC scheme can be obtained by using the Lagrange multiplier $\{\lambda_i\}_{i=1}^{K}$:

$$J = \sum_{k=1}^{K} W_k R_k + \lambda\left(\sum_{k=1}^{K} \sum_{n\epsilon\Omega_k} p_{k,n} - P_t\right) \qquad (4.6)$$

$$\text{s.t.} \begin{cases} \sum_{k=1}^{K} \sum_{n\epsilon\Omega_k} p_{k,n} = P_t) \\ p_{k,n} \geq 0 \end{cases}$$

The optimal solution is obtained by:

$$p_{k,n} = \begin{cases} \frac{W_k}{\sum_{i=1}^{K}(W_i\, size\ of\ (\Omega_i))}\left(P_{total} + \sum_{i=1}^{K} \sum_{q\epsilon\Omega_j} \frac{1}{\gamma_{i,q}}\right) - \frac{1}{\gamma_{k,n}}) \\ 0 \end{cases} \qquad (4.7)$$

There is no enough power for the subcarrier with small weight or low CNR when the allocation power is negative. Such subcarriers will be removed and re-allocated power until all allocated ones are positive. For guaranteeing the power allocated for each user is positive, the weight should satisfy:

$$W_k \geq \frac{\sum_{i=1}^{K}(W_i\, size\ of\ (\Omega_i))}{\gamma_{k,n}(P_{total} + \sum_{i=1}^{K} \sum_{q\epsilon\Omega_j} \frac{1}{\gamma_{i,q}})} \qquad (4.8)$$

By considering two users i, and j $(i, j \in \{1,2, \dots, K\})$ with subcarriers allocation m and n $(m \in \Omega_i\ and\ n \in \Omega_j)$, it can be derived that:

$$\frac{W_i}{W_j} = \frac{\gamma_{j,n}(1+p_{i,m}\gamma_{i,m})}{\gamma_{i,m}(1+p_{j,n}\gamma_{j,n})} \qquad (4.9)$$

From equation 4.9, it can be implied that more allocated power for better channel gain with equal gain and furthermore, more allocated power if there are higher weights.

4.3 Scheduling at the MAC Layer

OFDM system not only has quite good characteristics of physical layer but also provides the chance for the improvement on the data link layer. The requirements of the resource at the end can be satisfied well by the division of the channels. The dynamic adaption of modultaion types and allocated power for each subcarrier increases the spectrum efficeincy. With the development multimedia resource should be shared by multiple users in communication network. Both voice and video streaming are transmitted through the same network while they require different delays and jitters. Find the solution of meeting different requirements of different service scheduling are desirable [19][20]. The total larger channel capacity and high spectral efficiency can be acquired from the adaptive scheduling design which copes with the dynamics of channel and traffic at MAC layer.

The scheduling algorithm has three main functions [2]. First one is the allocation of the bandwidth. The scheduler can guarantee the minimum bandwidth for the specific type of service. Second one is controlling the delay. The requirements of delay control for real-time and non-real-time service are quite different. The last one is the application of the priority. So QoS parameter which determines the priority plays quite an important role on the scheduling for distinguishing different services. The design of scheduling algorithms influences the quality of the MAC layer. There are some considered parameters such as QoS, throughputs, fairness, power control and complexity in the scheduling design. QoS of diverse traffics include the information like maximum delay or jitter tolerance. The maximization of throughput is good for the current shortage of wireless communication resources. Fairness should be considered with the satisfaction of QoS requirements. The implementation of the scheduling should be simple, fast and less resource consumed.

There are a lot of scheduling algorithms for wired networks which can also be applied at the MAC layer of the wireless communication systems after the adaptive subcarrier allocation and power allocation. CSI of users and characteristics of the data packets determine the rate requirements and packet transmission priority by specific algorithms.

The basic principle of First-in First-out (FIFO) algorithm is quite straight forward [21]. The scheduling depends on the data arriving order, which is first in first out. It is simple and fast but not considers fairness and QoS. Round-robin (RR) is another simplest method for allocating the bandwidth [22]. It transmits data among different queues in a circle. The grouping which needs to be transmitted will be checked and sent. All the queues are checked repeated until it finds the send-ready grouping. Weighted Round-robin (WRR) was proposed because RR does not satisfy the QoS requirements. The dynamic weight is used to represent the length of queue, delay of packet or number of slots to adjust the throughputs and delays As a variation of WRR, deficit round robin (DRR) compensates the unselected queues caused by over length in the next round [12][24]. It is simple and not cares about the mean packet size of each connection while there is no fairness in a short time. In real-time system, when the waiting time of the packet is close to the delay constraint, higher priority will be given [21]. However, it will be given to the users with better channel quality when packets are non-real or not urgent. The proposed algorithm does not work well if there are many kinds of services and not so efficient because of no considerations of different packets' characteristics. The most well known fair queueing (FQ) schemes [12][26] has been proposed to provide fair services to user in wired network. Network capacity is divided into equal fractions and each of them is consumed by an active queue. The FQ scheduling scheme can achieve the max-min fairness due to same capacity assigned to the each active queue. Weighted fair queueing (WFQ) as the most complex conventional scheduling method is different from FQ in that the allocated capacity is proportional to the weights of queues [27]. The fairness fair among the traffic, the system throughputs and the end-to-end delays can be guaranteed by WFQ. There are some other well know scheduling strategies like virtual clock (VC), delay- earliest due data (delay-EDD) , self-clocked fair queueing (SCFQ) and earliest deadline first (EDF) scheduling [28], ect. Serving the backlogged users in proportion to the share of bandwidth is the principle of most scheduling methods.

In this project, the subcarrier and power controller at the PHY layer performs the subcarrier and power allocation. Then the traffic controller at the MAC layer performs the data scheduling which provides the QoS information for the resource allocation. The function of scheduling is determine the received data sending order according to the factors such as delay, size, and QoS priority level of packets. Quite different from the conversional traffic queue based scheduling, the proposed packet depend (PD) scheduling [1][12] overcomes the inefficiency when some packets are more urgent in the unselected queue than those in the selected one. Furthermore, it not only considered about the packet delay, channel quality but also the system capacity with heterogeneous traffic for each user. By considering about the weights of packets, the corresponding transmission order can be optimize the whole system performance. The complexity of the whole cross-layer design can be reduced by choosing the proper number of the selected packets for the weight calculation.

4.3.1 Modified Largest Weighted Delay First (M-LWDF) Scheduling

Modified Largest Weighted Delay First (M-LWDF) satisfies QoS requirements by guaranteeing the system throughput. The delay of most users can be reduced based on the pre-set probabilities [15][1]. It uses the state information of current channel and queue to optimize the system throughput. The basic idea of M-LWDF is setting three parameters which describe the different service natures, delay and queue length. The queue with the maximum multiplication of three parameters will be served first. The queue based M-LWDF scheduling [15] can be extended by combining with queue based MWS resource allocation in the OFDM systems. The weight of each queue *i* can be calculated by:

$$W_i = -\frac{\log(\delta_i) D_i}{U_i \overline{R_i}} \tag{4.10}$$

Where for queue i, D_i and U_i denote head of line (HOL) packet delay and delay threshold, $\overline{R_i}$ stands for the average data rate over slots and δ_i is defined by the maximum probability than the delay exceeds delay threshold. The higher priority is given to the queue which have relative high data rate or delay (i.e. $\frac{R_i}{\overline{R_i}}$ or $\frac{D_i}{U_i}$) or have a larger outage probability (i.e. $-\log(\delta_i)$). M-LWDF scheduling can control flow delays and provides the minimum throughput guarantees. It is suitable for the multiuser system which shares the wireless link with different QoS requirements.

4.3.2 Packet Dependent (PD) Scheduling

Each queue consists of some packets. Compared with traditional queue based scheduling like M-LWDF scheduling, the packet dependent (PD) scheduling is based on smaller unit. That is, PD scheduling is more flexible. In the M-LWDF scheduling the weight is set queue by queue, so all the packets in a queue have the same weight. However data is sent packet by packet according to different weights in PD scheduling. It takes advantages in the situation that some packets in the queue are very urgent. With the queue based scheduling, a whole selected queue should be served once and maybe at the same time some packets in the unselected queue are much more urgent which can't be transmitted immediately. On that hand, PD scheduling is much more efficient.

The mechanism of PD scheduling is to assign different weights to different packets in one queue [1]. Three traffic queues (i.e. Voice, Video and Best Effort (BE)) as three typical services are set for each user (seen Figure 4.3). The order of serving BE stream depends on the arriving order of the data. The network gives the best effort to serve the stream but not guarantees the reliability or delay. The BE service is the default Internet service model and is suitable for most network applications such as Email. The characteristics of these three queues are compared in Figure 4.4.

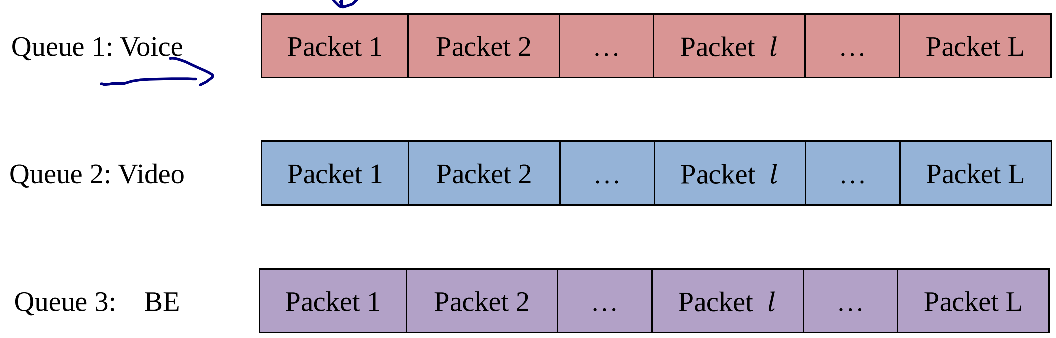


Figure 4.3 Data structure of each user

Table 4.2 Comparisons of Three traffic queues

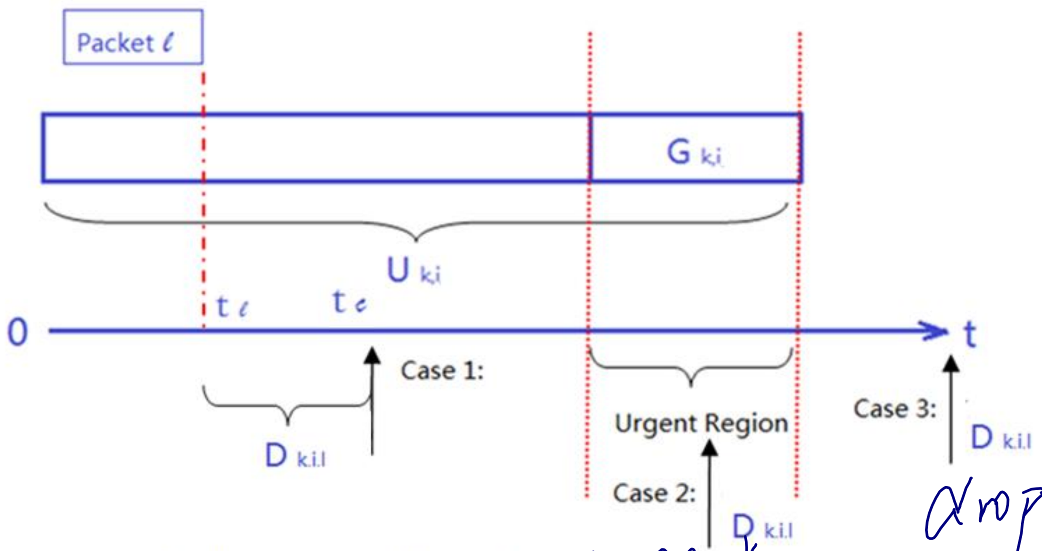
	Delay Tolerance	**QoS Priority Lever**
Voice	Low	High
Video	Medium	Medium
BE	High	Low

The information provided by the above table matches the experience in the real network. Short delay tolerance requires high QoS priority lever. Voice traffic queue mostly applied on the online telephone or broadcast can't tolerant long delay due to its instantaneity, while it's not necessary to service BE traffic queue as quickly as voice or video traffic queues.

The weight design of PD scheduling depends on QoS priority level of packets, packet size and delay [1]. The delay issues with the mentioned data structures are discussed as follows (seen Figure 4.5). All the related parameters in PD scheduling are shown in Table 4.3.

Table 4.3 Parameters used in the PD scheduling

Parameter	**Mean**
$i\ (i \in (1,2,\dots,I)$	Number index of *I* queues (I=3 in this project)
$k\ (k \in (1,2,\dots,K)$	Number index of *K* users
$U_{k,i}$	Deadline for queue *i* of user *k*
$G_{k,i}$	Guard interval for queue *i* of user *k*
$t_c(t_c \in [0,+\infty))$	The current time
$t_l(t_l \in [0,t_c])$	The arriving time of packet *l* in queue *i* of user *k*
$D_{k,i,l} = t_c - t_l$	The delay time of packet *l* in queue *i* of user *k*
$Q_{k,i}(Q_{k,i} \in [1,+\infty))$	The QoS priority levels of queue *i*
$S_{k,i,l}$	The packet size of packet *l* in queue *i* of user *k* (in bits)
$X_{k,i}$	Number of to-be-served packets in queue *i* of user *k*
$x_{k,i}$	Number of to-be-served packets (out of guard interval)
$x'_{k,i}$	Number of to-be-served packets (in the guard interval)



Note: Transmission of packet *l* in queue *i* for user *k*

urgent (high priority)

dropped

Figure 4.4 Diagram of delay issue in the packet transmission

Seen from Figure 4.4, it can be seen that there are three cases according to the current time, the delay tolerance and guard interval [1]. Once the waiting time of packets exceeds the corresponding deadline (i.e.$D_{k,i,l} > U_{k,i}$), the packets will be dropped (Case 3). The guard interval added to the deadline is introduced to reduce the packet drop rate. When the waiting time comes into guard interval (i.e.$U_{k,i} - G_{k,i} \le D_{k,i,l} \le U_{k,i}$) which is treated as an urgent region, Case 2 will be given a higher priority level for distinguishing from Case 1. It can be reduced that more weight should be assigned to the packet whose delay time is much closer to the start bound of urgent region. And the packets within the urgent region have maximum weights. The packets with higher weights will be transmitted first no matter which queues they are belong to. This reflects its advantages on flexibility and efficiency over queue based M-LWDF scheduling. The weight equation in PD scheduling of packet *l* in queue *i* of user *k* [1] is expressed by:

$$W_{k,i,l} = \begin{cases} \frac{Q_{k,i} S_{k,i}}{U_{k,i} - G_{k,i} - D_{k,i,l} + 1} & (\, U_{k,i} - G_{k,i} - D_{k,i,l} \ge 0 \,) \\ Q_{k,i} S_{k,i} & (\, U_{k,i} - G_{k,i} - D_{k,i,l} < 0 \,) \end{cases} \qquad (4.11)$$

The designed weight according to equation 4.11 is not only determined by the packet delay as mentioned before, but also directly related to the QoS priority and packet size. In a word, a packet with larger data amount, higher QoS priority and longer delay will be given a higher weight and be served earlier. The weight is set as two values according to different conditions.

✧ *Case 1:*

$U_{k,i} - G_{k,i} - D_{k,i,l} \ge 0$ (i.e. $D_{k,i,l} \le U_{k,i} - G_{k,i}$) means the delay time has not fallen into the urgent region. So the weight is designed to be increased with the increase with delay. Larger the packet delay is, less time left for the packet to be urgent. The packet's demand to be served as soon as possible becomes stronger. The degree of weight variation with the delay can be shown by taking the derivation of the equation 4.11 to the delay time:

$$\frac{\partial W_{k,i,l}}{\partial D_{k,i,l}} = \frac{Q_{k,i} S_{k,i}}{(U_{k,i} - G_{k,i} - D_{k,i,l} + 1)^2} \qquad (4.12)$$

When delay is very small, the influence of delay variation on weight variation is not obvious. At this situation, packet size and QoS priority level are the more important factor of weight design. On the contrary, increased degree of weight is higher when the delay is larger because delay plays a more important role.

✧ *Case 2:*

$U_{k,i} - G_{k,i} - D_{k,i,l} \le 0$ (i.e. $D_{k,i,l} \ge U_{k,i} - G_{k,i}$) means the packet delay has fallen into the urgent region. The packet will be dropped if it is not served during the last Guard interval. Under this condition higher priority should be assigned to the packet, so maximum designed weight is just based on QoS priority level and packet size. The packets with higher QoS priority level or larger data amount will be served first in this case.

The number of to-be-served packets should be selected correctly for balancing the complexity and performance [1]. The first $P_{k,i}$ selected packets which are most urgent in queue i of user k are chosen. Among these packets, the relation between the number of packets inside (i.e.$x_{k,i}$) or outside (i.e.$x'_{k,i}$) the guard interval is:

$$x_{k,i}+x'_{k,i} = X_{k,i} \qquad (4.13)$$

The weight of user k in current slot then can be derived from equation 4.11 as:

$$W_k = \sum_{i=1}^{I} Q_{k,i} \left(\sum_{l\in x_{k,i}} S_{k,i,l} + \sum_{l\in x'_{k,i}} \frac{S_{k,i,l}}{U_{k,i}-G_{k,i}-D_{k,i,l}+1}\right) \qquad (4.14)$$

For combining with the proposed M-MWC resource allocation, the weight for each user achieved from above PD scheduling should consider one more factor than is channel state information. In the existing weight design, the weight of one user on all the channel are the same which is not optimal, the proposed new weight based on PD scheduling (i.e. Modified-PD scheduling(MPD)) is given by:

$$W_{k,n} = \gamma_{k,n} \sum_{i=1}^{I} Q_{k,i} \left(\sum_{l\in x_{k,i}} S_{k,i,l} + \sum_{l\in x'_{k,i}} \frac{S_{k,i,l}}{U_{k,i}-G_{k,i}-D_{k,i,l}+1}\right) \qquad (4.15)$$

The different weight assign to different users in different channel provides more information to PHY layer for resource allocation. The system throughput can be maximized because more weight will be assigned if the channel is much better.

4.4 Simulation Results and Analysis

4.4.1 Complexity Analysis

The complexity analysis of all kinds of mentioned cross-layer optimization design is summarised in the Table 4.4 [1]. Five subcarrier allocation schemes, two power allocation schemes and three scheduling schemes are combined together for comparison in this project. The overall complexity of cross-layer design depends on the highest order complexity of resource allocation and scheduling.

Table 4.4 Complexity of different Cross-layer Optimization Design

Resource Allocation		Scheduling	Overall
Subcarrier Allocation	**Power Allocation**		
MC: O(N)	*WF:* O(N)	*PD:* $O(IX_{k,i}K)$	$\begin{cases} O(N) & I \le N/(KX_{k,i}) \\ O(IX_{k,i}K) & I > N/(KX_{k,i}) \end{cases}$
PF: O(KN)	*WF:* O(N)	*PD:* $O(IX_{k,i}K)$	$\begin{cases} O(KN) & I \le N/X_{k,i} \\ O(IX_{k,i}K) & I > N/X_{k,i} \end{cases}$
Queue Based MWC: O(IKN)	*WWF:* O(N)	*M-LWDF:* O(IK)	O(IKN)
User Based MWC: O(KN)	*WWF:* O(N)	*PD:* $O(IX_{k,i}K)$	$\begin{cases} O(KN) & I \le N/X_{k,i} \\ O(IX_{k,i}K) & I > N/X_{k,i} \end{cases}$
User Based MMWC: O(KN)	*WWF:* O(N)	*MPD:* $O(IX_{k,i}K)$	$\begin{cases} O(KN) & I \le N/X_{k,i} \\ O(IX_{k,i}K) & I > N/X_{k,i} \end{cases}$

Among all subcarrier allocation schemes, it can be seen that MC based scheme have the lowest complexity of O (N). This is because its principle is to allocate the best channel to users by N comparisons. PF and user based MWC subcarrier allocation described in section 4.2.1 both require a complexity of O (KN) because they requires N (or K) comparisons in each of K (or N) iterations for allocation. The number of queues is I times of that of users. IK comparisons in each of N iteration are required in queue based MWC scheme so its complexity is O (IKN) which is higher than that (i.e. O (KN)) of user based one.

Power allocation based on water-filling algorithm (WF and WWF) for N subcarriers has a complexity of O (N) which is low enough to be ignored.

The complexity of scheduling can be acquired from the weight design. The number of to-be-served packets $X_{k,i}$ is a very important factor. Larger number value leads to higher complexity while smaller packet number can't provide enough QoS information to PHY layer. In order to get the balance for an improved system performance, the value of $P_{k,i}$ is set below 100 by experiment in [1]. The complexity of PD scheduling is O $(IX_{k,i}K)$ according to equation 4.14. The mechanism of M-LWDF is much easier with a complexity of O (IK) according to equation 4.10.

The highest order complexity of resource allocation and scheduling determines the overall complexity. So the number of queues per user I in heterogeneous traffic OFDM system should be selected to decide the highest order complexity as the overall one. From above table, there are three designs (i.e. PF+WF+PD, user based MWC+WWF+PD and MMWC+WWF+MPD) with the same complexity. Compare with queue based MWC+WWF+ M-LWDF design, it can be seen that the complexity difference mainly comes from different scheduling.

✧ *Case 1:*

With $1 \leq \mathrm{I} \leq \mathrm{N}/\mathrm{X}_{\mathrm{k,i}}$, which is very practical, the overall complexity mainly depends on the complexity of resource allocation $\mathrm{O(KN)}$ for MWC+WWF+PD design or $\mathrm{O(IKN)}$ for MWC+WWF+ M-LWDF design. In the specific case that each user has a single traffic queue (i.e. I=1), both design have the same complexity. It implies that design with PD scheduling have more superiority on complexity especially when I increases. I times complexity reduction of MWC+WWF+PD design is more suitable for the high data rate heterogeneous traffic transmission.

✧ *Case 2:*

Under the condition $I > \mathrm{N}/\mathrm{X}_{\mathrm{k,i}}$, the complexity reduction of the design with PD scheduling over that with M-LWDF is about $\mathrm{N}/\mathrm{X}_{\mathrm{k,i}}$.

The correct selections of parameters such as number of queues per user I, number of subcarrier N and number of selected packets $\mathrm{X}_{\mathrm{k,i}}$ in queue i for user k, the design with PD scheduling has lower complexity in general. The proposed modified MWC (MMWC) subcarrier allocation and modified PD (MPD) scheduling has the same complexity with the original ones.

The normalised numerical complexity of all the designs are summarised in the following table. The complexity is normalised to the lowest complexity for easy comparisons.

Table 4.5 The Normalised Numerical Complexity of different Cross-layer Optimization Design

Resource Allocation		Scheduling	Overall
Subcarrier Allocation	**Power Allocation**		
MC: O(1)	*WF:* O(1)	*PD:* $O(X_{k,i})$	$\begin{cases} O(1) & I \leq N/(KX_{k,i}) \\ O(X_{k,i}) & I > N/(KX_{k,i}) \end{cases}$
PF: O(K)	*WF:* O(1)	*PD:* $O(X_{k,i})$	$\begin{cases} O(K) & I \leq N/X_{k,i} \\ O(X_{k,i}) & I > N/X_{k,i} \end{cases}$
Queue Based MWC: O(IK)	*WWF:* O(1)	*M-LWDF:* O(1)	O(IK)
User Based MWC: O(K)	*WWF:* O(1)	*PD:* $O(X_{k,i})$	$\begin{cases} O(K) & I \leq N/X_{k,i} \\ O(X_{k,i}) & I > N/X_{k,i} \end{cases}$
User Based MMWC: O(K)	*WWF:* O(1)	*MPD:* $O(X_{k,i})$	$\begin{cases} O(K) & I \leq N/X_{k,i} \\ O(X_{k,i}) & I > N/X_{k,i} \end{cases}$

4.4.2 Resource Management Results and Analysis

4.4.2.1 Simulation Parameters Setting

In this section, simulation results to illustrate the mentioned cross layer design performance. The advantages and disadvantages of different designs can be demonstrated by comparing their resulted targets. All setting parameters in [1] are summarised in the following table.

Table 4.6 (1) Parameters setting in the Heterogeneous traffic Cross Layer Design

Parameter	Value
Total transmit power	P_t=1W
Slot Duration	T_{slot}=2ms
Packet Duration	T_{packet}=1ms
Number of slots	2000
Total bandwidth	B =5MHz

Table 4.6 (2) Parameters setting in the Heterogeneous traffic Cross Layer Design

Parameter	Value
Number of subcarriers	N =512
Number of subcarriers	N =512
Number of queues per user	I =3
Deadline for voice traffic queue	$U_{k,i}^{voice}$=100ms
Deadline for video traffic queue	$U_{k,i}^{video}$=400ms
Deadline for BE traffic queue	$U_{k,i}^{BE}$=1000ms
Guard interval in queue *i* of user *k* for all traffic queues	$G_{k,i}$=1ms
Packet arriving data rate for video traffic queue	$V_{k,i}^{video\_max}$=420Kbps
	$V_{k,i}^{video\_mean}$=239Kbps
	$V_{k,i}^{video\_min}$=120Kbps
Packet arriving data rate for BE traffic queue	$V_{k,i}^{BE}$=500Kbps
Number of to-be-served packets for voice traffic queue	$X_{k,i}^{voice}$=100
Number of to-be-served packets for video traffic queue	$X_{k,i}^{video}$=75
Number of to-be-served packets for BE traffic queue	$X_{k,i}^{video}$=50
QoS priority levels for voice traffic queue	$Q_{k,i}^{voice}$=1024
QoS priority levels for video traffic queue	$Q_{k,i}^{video}$=512
QoS priority levels for BE traffic queue	$Q_{k,i}^{BE}$=1
Multipath Number	6

The characteristics of three traffic queues (i.e. voice, video and BE) have been described in section 4.3.2. The noteworthy point is that video traffic queue has variable bit rate as show in above table. It follows a truncated exponential distribution in each state and the duration of each state follows an exponential distribution with mean 160 ms [1]. The average packet sizes for three traffic queues which can be calculated from the known setting parameters are 64 bits, 239 bits and 500 bits. Number of to-be-served packets is considered according to both complexity and performance. SNR is defined as the ratio of the average signal energy to the noise power spectral density:

$$\mathrm{SNR} = \frac{\mathrm{E_s}}{\mathrm{N_o}} = \frac{\mathrm{P_t T_{slot}/N}}{\mathrm{N_o}} \qquad (4.16)$$

Take an example, for 20dB SNR the noise power spectral density should be set as -74dBW/Hz with the mentioned parameter setting. Six independent Rayleigh fading paths with an equal delay profile are used to represent the channel state information in the system model.

4.4.2.2 Summary of Schemes

Before demonstrating the simulation results, the theoretical comparisons of different schemes will be summarized in this section.

Table 4.7 Summary of different Cross Layer Design

<table>
<tr><th colspan="3">Resource Allocation</th><th rowspan="2">Scheduling</th><th rowspan="2">Cost Function</th><th rowspan="2">Considering Targets</th></tr>
<tr><th colspan="2">Subcarrier</th><th>Power</th></tr>
<tr><td colspan="2">MC</td><td rowspan="2">WF</td><td rowspan="3">PD</td><td>$\max\ J = \sum_{k=1}^{K} R_k$</td><td>Max Throughput</td></tr>
<tr><td colspan="2">PF</td><td>$\max\ J = \sum_{k=1}^{K} R_k$
$R_1 : R_2 : \dots : R_K = \eta_1 : \eta_2 : \dots : \eta_K = 1$</td><td>Throughput & Fairness</td></tr>
<tr><td rowspan="2">MWC</td><td>User based</td><td rowspan="3">WWF</td><td rowspan="2">$\max\ J = \sum_{k=1}^{K} W_k R_k$</td><td rowspan="2">Throughput & QoS</td></tr>
<tr><td>Queue based</td><td>M-LWDF</td></tr>
<tr><td colspan="2">MMWC</td><td>MPD</td><td>$\max\ J = \sum_{k=1}^{K} \sum_{n \in \Omega_k} W_{k,n} R_{k,n}$</td><td>More Throughput & QoS</td></tr>
</table>

From the optimization equations of each design, the considering factors can be easily compared.

In MC based scheme [8], the subcarrier is allocated to the user with the best channel gain. When all users have enough data to sent, the maximized capacity can be achieved. It does not consider fairness because some urgent users can't be allocated resource with pool CNR while some with good channel gain may have more than enough data to be sent. This leads to the waste of resource. Furthermore, QoS information including delay is also not guaranteed. Although MC based design achieve maximization of throughput, it is not practical.

Compared with MC based scheme, an additional data rate constraint is added in the PF based scheme [8]. The ratio of data rate of each user limits the subcarrier allocation. The user with lowest capacity has the priority to choose the best subcarrier to guarantee the fairness among users at the cost of system throughput. PF based scheme also does not consider QoS information.

MWC based resource allocation combined with scheduling uses QoS information to determine the weights of different packets or queues. Only the weighted data units can achieve the maximum capacity so the system throughput is not as much as that of MC based one. Delay-sensitive data is sent in the order which is determined by the weight.

4.4.2.3 Simulation Results and Analysis

A desired cross layer design for high data rate wireless communications requires a higher capacity, lower power consummation, more flexible system coverage and better QoS [12]. The system performance in the simulation can be reflected in some factors like *system throughput, average packet delay and packet drop rate*. In this section, the simulation results of different resource allocation and scheduling will be presented. The performance of system with different users or SNR will be analyzed.

4.4.2.3.1. Fairness Issue (MC & PF & MWC)

Shannon capacity is the maximum capacity that the system can achieve in theory by considering the channel states. System throughput is sum of data rates which considers the real situations like the data drop rate and delay at the terminals of the network. For easy comparisons of different resource allocation schemes, Shannon capacities are first used to demonstrate their characteristics on fairness.

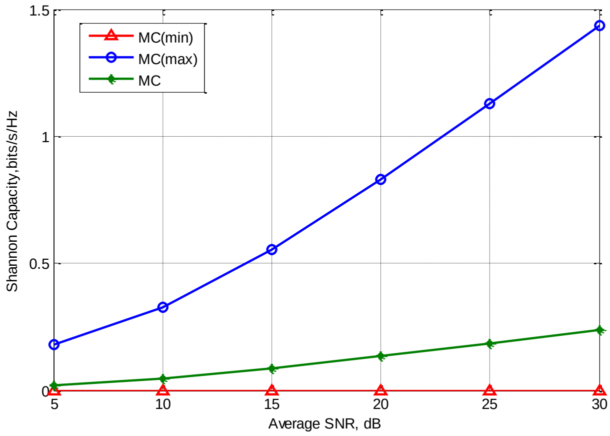


Figure 4.5 (1) MC [16] based Scheme

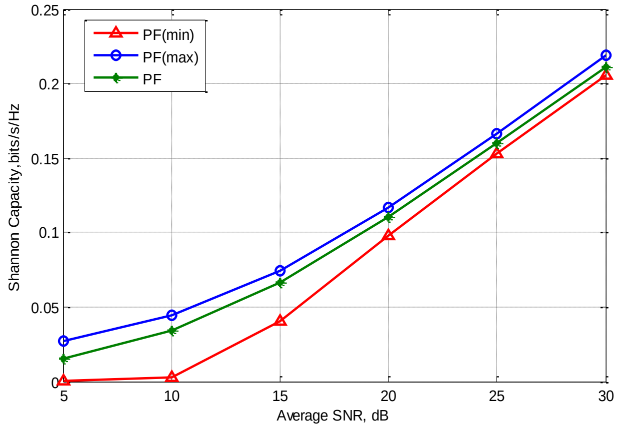


Figure 4.5 (2) PF (ratio =1) [7] based Scheme

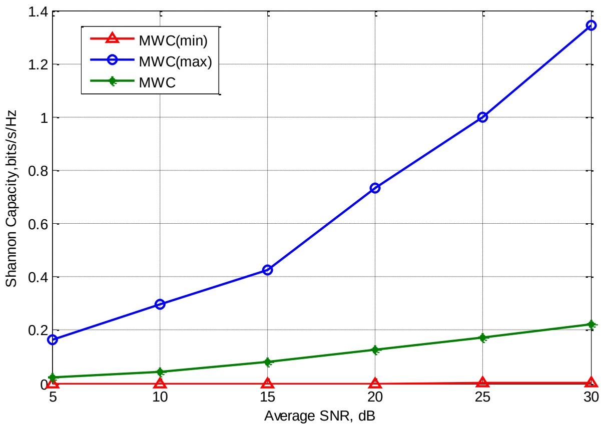


Figure 4.5 (3) MWC [1] based Scheme

Above three figure gives the system Shannon capacity of each user over different with MC, PF and MWC based resource allocation. Maximum Shannon capacity comes from the best case and on the contrary, minimum one results from the worst case. The rest Shannon capacity is the average Shannon capacity of each user in system with 32 users. First, the Shannon capacity with all schemes increases with the increase of SNR which matches the equation 2.8. More signal energy brings more capacity. In both MC and MWC based schemes, the minimum capacity is about 0 bps/Hz and maximum one is quite large with the same SNR. Take an example, at SNR=30dB the minimum capacity for each user is 0 bps/Hz while minimum one is 1.7 bps/Hz. The average capacity is 0.23 bps/Hz. It implies that in MC and MWC based schemes the user who has good channel gain is allocated with a lot of resource and may be allocated with nothing when the channel is bad enough. The capacity is only depends on the users' channel states while not the fairness. Seen from the results of PF based schemes, the gaps among three capacities are quite small. In PF based resource allocation, the added limitation of data rate for each user guarantees the capacities of all users are quite similar (i.e. ratio=1). Its capacity not only depends on channel states but also the acquired data of each user. Furthermore, compared with all three figures, it can be seen that the differences among minimum, maximum and average capacities of each user increase with the increase of SNR. It is caused by the principles of three schemes. Better optimal results can be achieved if there is more signal energy. That is, for MC and MWC based schemes in the best case each user has more capacity while in the worst case user is always allocated nothing. And for PF based scheme, the resource will be allocated more fairly so that the difference between best and worst case reduces with more energy.

The above results prove that except the throughput, PF based resource allocation also consider fairness compared with MC and MWC based schemes.

4.4.2.3.2. Resource Allocation Issues (MC & PF & MWC)

In this part PD based scheduling scheme are combined with three different resource allocation schemes to compare their characteristics according the system throughput with different users over a variety of SNR.

As discussed before, MWC based scheme can only maximize the capacities of the weighted packets while MC based scheme maximizes the capacities of all packets. It can be deduced that MWC based scheme is MC based scheme when all the weights are the same (i.e. set as 1). This can be show in Figure 4.6 that their Shannon capacities are exactly the same. When all the weights which represent QoS information are set as one, all the packets have the same probability to be selected for resource allocation. It looks like that there is no QoS information considered which is as same as the MC based scheme. On the other side, if the weights are different due to PD scheduling at MAC layer, the system throughput of MC based scheme should be the largest among these three one. It can be checked in Figure 4.7. Compared with Figure 4.6, it’s noted that the system throughput which considers delay is lower than Shannon capacity. When the number of user is above 24, the system throughput differences among three schemes become obvious. The top one which reaches the maximization of throughput comes from MC based scheme. MWC based scheme maximizes weighted capacities which should be lower. The bottom one from PF cased scheme guarantees fairness as mentioned in last section at the cost of capacity. Furthermore, the throughput variation becomes very small in the system with more than 24 users. MC, MWC and PF based system design have around 4.5bps/Hz, 4bps/Hz and 3.6bps/Hz maximized throughput in this simulation.

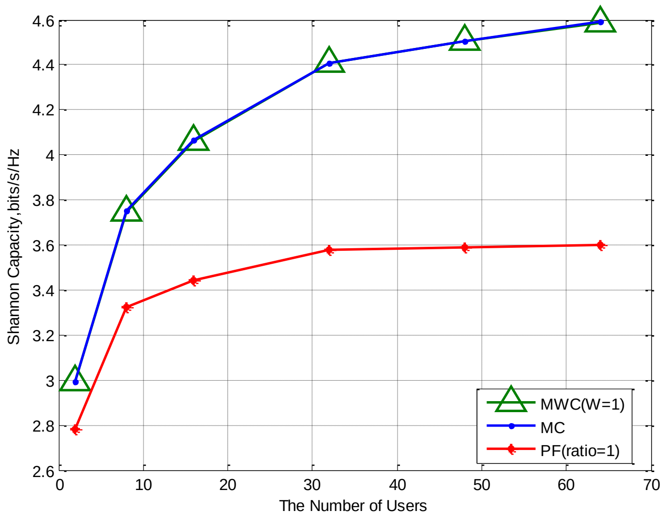


Figure 4.6 Impact of User number on Shannon Capacities in MC [16], PF (ratio=1) [7] and MWC (W=1) [1] based scheme when SNR=20dB

Seen from Figure 4.8 which shows the impact of SNR on system throughput with 32 users, MC based scheme also outperforms MWC based and PF based ones at low SNR when it is below 25dB. As an example, at SNR=15, the system throughput achieved by MC based scheme is 20% and 25% higher than that achieved by MWC based and PF based schemes.

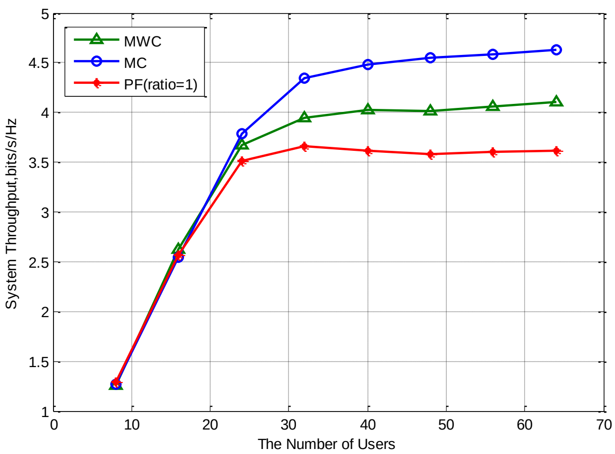


Figure 4.7 Impact of User number on system throughput in MC, PF (ratio=1) and MWC based scheme when SNR=20dB

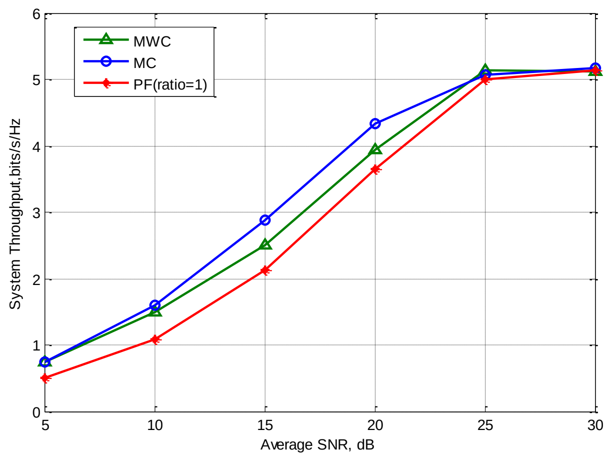


Figure 4.8 Impact of SNR on system throughput in MC, PF (ratio=1) and MWC based scheme with 32 users

4.4.2.3.3. Scheduling Issues (MWC_user based + PD & MWC_queue based + M-LWDF & MMWC + MPD)

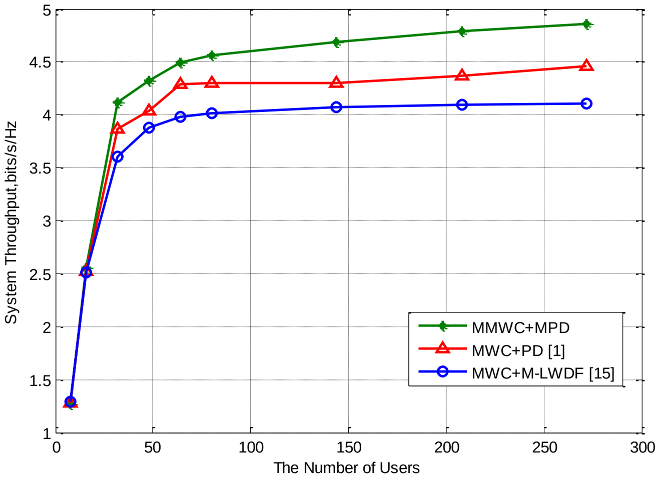


Figure 4.9 Impact of User number on System Throughput with MMWC+MPD, MWC+PD and MWC+M-LWDF designs when SNR=20dB

Although in the previous part MC based resource allocation scheme achieves the highest system throughput, it is not practical especially for delay sensitive data. Consider the trade off between QoS requirements and throughput, MWC based resource allocation is selected for comparing different scheduling methods. The performance of proposed MMWC+MPD design is also demonstrated in this section. With a wide range of the number of users (K=8～272), PD scheduling shows its significant advantages on system throughput over M-LWDF in Figure 4.9. Multiuser diversity which is enhanced by higher resource allocation degree causes the increase of system throughput when user number increases. When the user number is above 64, the increasing degree of throughput is not obvious. Because when user number increases, the weight differences among different packets of different queues increase which leads to the system throughput reduction. This effect is more significant with M-LWDF which serves the queue with largest weight first than with PD which serves the packet with largest weight first. As a result, PD performs better than M-LWDF on system throughput due to multiuser diversity. The improvement of MMWC+MPD over MWC+MPD benefits from the modified weight design which also considers CNR. In 48-user system, system throughput for MC+PD (Figure 4.7), MWC +PD and MMWC+MPD (Figure 4.9) based designs are 4.5 bps/Hz, 4.3 bps/Hz and 4 bps/Hz. Compared with MWC+PD design, the proposed one not only achieves higher system throughput but also higher throughput of each traffic queue as shown in following.

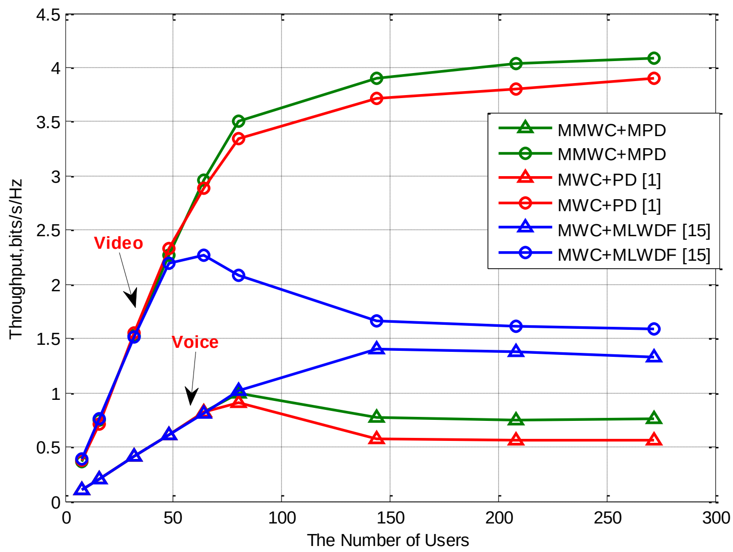


Figure 4.10 Impact of User number on voice and video traffic throughput with MMWC+MPD, MWC+PD and MWC+M-LWDF designs when SNR=20dB

Figure 4.10 and 4.11 demonstrate the throughput with MMWC+MPD, MWC+PD and MWC+M-LWDF designs for three traffic queues of each user. The differences for voice and video traffic queues become significant when there are more than 64 users (Figure 4.10). It seems like that the total throughput of voice and video traffic queues is fixed because for both designs, if the voice traffic throughput is large then there is less throughput for video traffic and vice versa. Both these two traffic are real time traffic which have much higher QoS priority levels. The throughput of video traffic is more than that of voice traffic in general due to its high data rate requirements and larger packet size. For BE traffic queue, MMWC+MPD, MWC+PD outperform MWC+M-LWDF in the small user number range (K=16～48 or K=16～56). With users more than 80, there is almost no throughput for BE traffic with MMWC+MPD, MWC+PD designs which means they are only suitable for the real time service in the system with quite a large number of users.

MMWC+MPD and MWC+PD have the same basic design principles and only have the significant differences on throughput. So in the following discussions of system performance, the comparisons are focus on PD and M-LWDF scheduling.

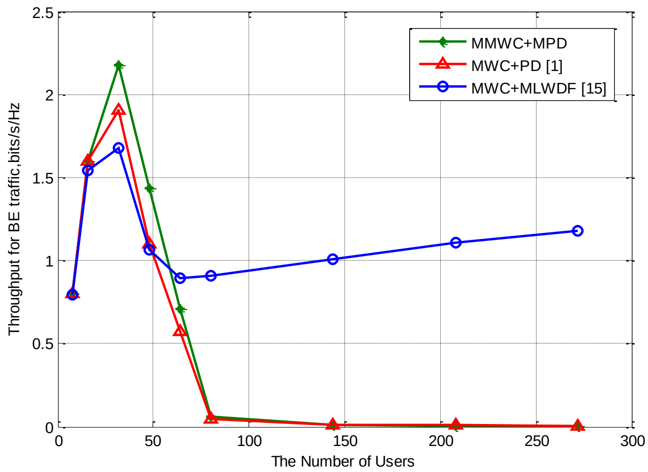


Figure 4.11 Impact of User number on BE traffic Throughput with MMWC+MPD, MWC+PD and MWC+M-LWDF designs when SNR=20dB

For real time traffic, there is another target called average packet delay to show the system performance. Figure 4.12 and 4.13 presents the impact of the user number on the average packet delay for voice and video traffic with different designs. For voice traffic, designs with PD performs better than that of M-LWDF in the user number range K=48～64. With 64 users, average packet delays are 8ms and 16ms for PD and M-LWDF respectively. PD based design benefits from more allocated resources to QoS traffic and gets about two times lower delays. While when user number increases, PD and MPD have higher delay than M-LWDF. This is because M-LWDF serves all the packets in the select queue which leads to lower average packet delayer especially for quite large data (i.e. increase of user number).

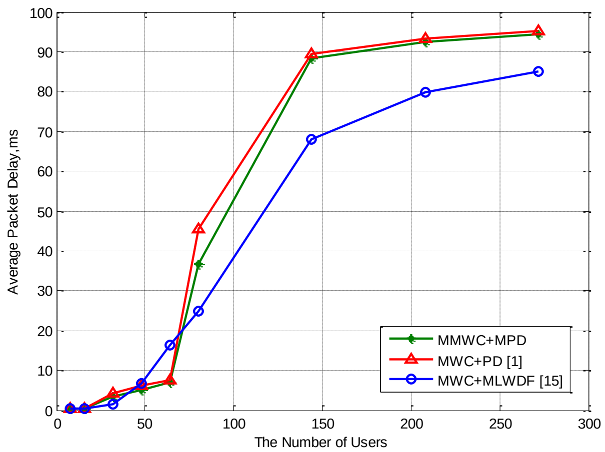


Figure 4.12 Impact of User number on average packet delay for voice traffic with MMWC+MPD, MWC+PD and MWC+M-LWDF designs when SNR=20dB

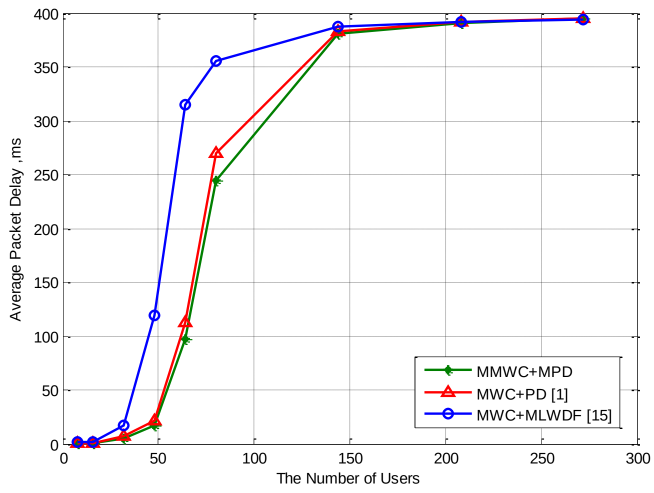


Figure 4.13 Impact of User number on average packet delay for video traffic with MMWC+MPD, MWC+PD and MWC+M-LWDF designs when SNR=20dB

The average packet delay of video traffic versus user number in Figure 4.13 shows the advantages of PD over M-LWDF on video traffic. For system with more than 32 and less than 144 users, PD has much lower delays. For example, the delays for PD and M-LWDF are about 20ms and 100ms in 64-user system. That is during this user number range, PD outperforms M-LWDF with up to 5 times lower delay. It also shows that when the user number is larger enough, the delay goes into the saturation region with the value which is close to the deadline of video traffic (400 ms) and there is no difference between two designs.

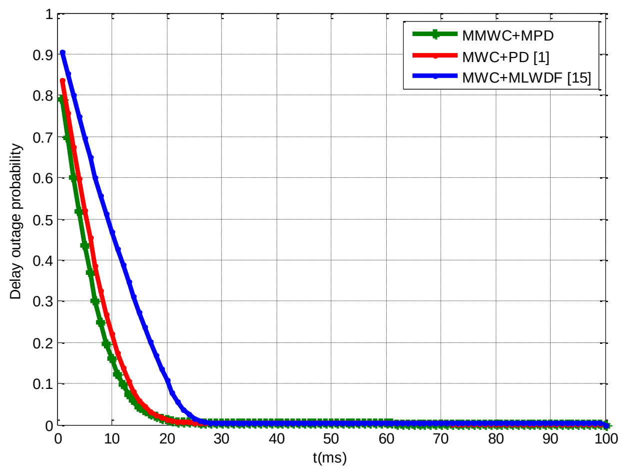


Figure 4.14 Delay outage probability for voice traffic with MMWC+MPD, MWC+PD and MWC+ M-LWDF designs with 48 users when SNR=20dB

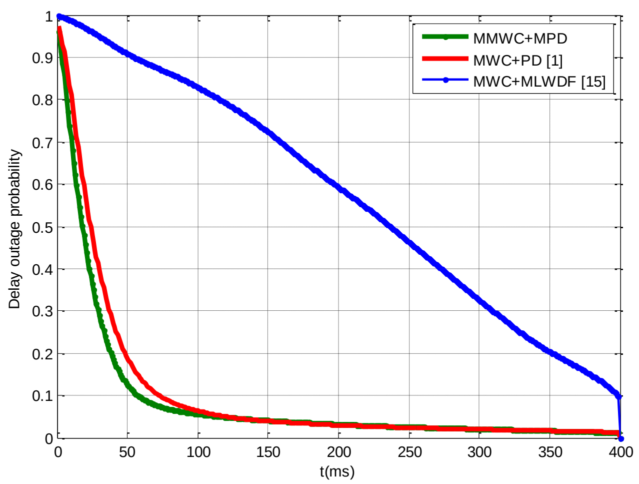


Figure 4.15 Delay outage probability for video traffic with MMWC+MPD, MWC+PD and MWC+ M-LWDF designs with 48 users when SNR=20dB

Figure 4.14 and 4.15 gives the delay outage probability of voice traffic and video traffic which represents their packet drop rates. It has been mentioned that the deadline of these two traffics are 100ms and 400ms. The outage probability is defined as the probability of packet delay larger than the current time (i.e. delay > t). So it implies that if the packet delay is greater than the deadline (i.e. delay_voice > 100ms and delay_video > 400ms), the packets will be dropped. For voice traffic in 48-user system, the delay outage probability is 0 when it is 20ms and 30ms for PD and M-LWDF respectively. It is shown that PD spends 10ms less than M-LWDF to guarantee all the voice traffic packets to be sent totally. For video traffic which has lower QoS propriety levels, the drop rate is higher. It can be seen that at 400 ms, PD has almost zero probability while it is 10% for M-LWDF. Furthermore, the delay outage probability of PD decreases much faster. After 60ms, it is below 10% for PD while it is still above 50% for M-LWDF at 250ms.

Figure 4.16 to 4.19 following show the impact of SNR in different cross layer design on throughput and average packet delay.

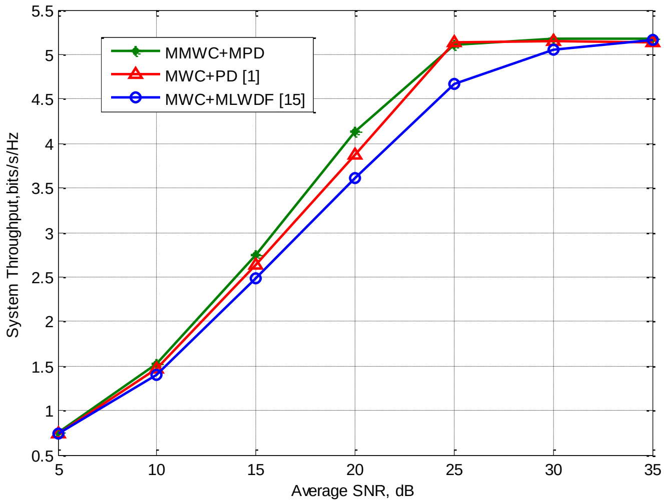


Figure 4.16 Impact of SNR on system throughput with

MMWC+MPD, MWC+PD and MWC+M-LWDF designs with 32 users

Seen from Figure 4.16, the system throughput in different designs has difference under various SNR. Below 30dB, MMWC+MPD, MWC+PD and MWC+M-LWDF designs perform best, medium and worst respectively on system throughput. Take an example, at SNR=20dB, MMWC+MPD achieves system throughput of 4.2 bps/Hz which is 10.5% and 16.7% higher than that of MWC+PD and MMWC+M-LWDF designs. Throughput for BE traffic (Figure 4.17) has the similar trend at the range SNR=15～30dB. When SNR is high, there is enough resource to send all the traffic data so that all the designs get the similar performance on throughput.

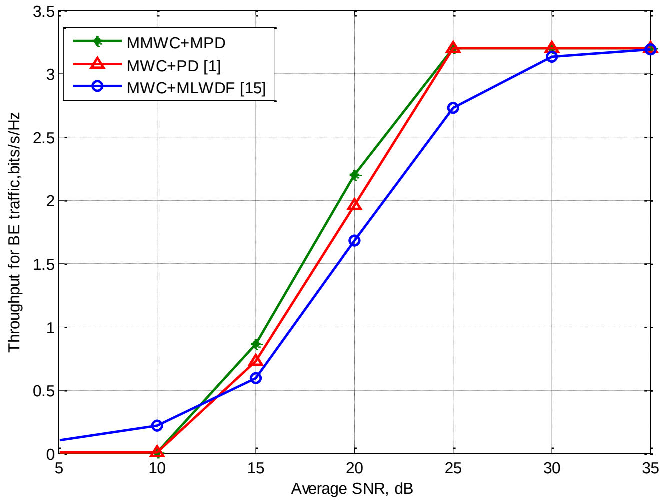


Figure 4.17 Impact of SNR on BE traffic throughput with

MMWC+MPD, MWC+PD and MWC+M-LWDF designs with 32 users

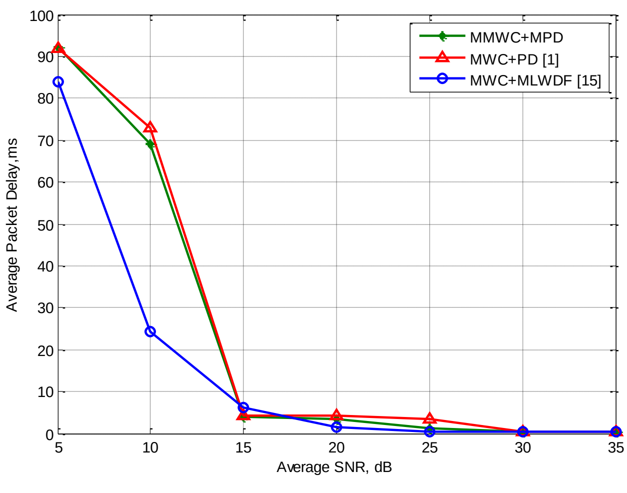


Figure 4.18 Impact of SNR on average packet delay for voice traffic with MMWC+MPD, MWC+PD and MWC+M-LWDF designs with 32 users

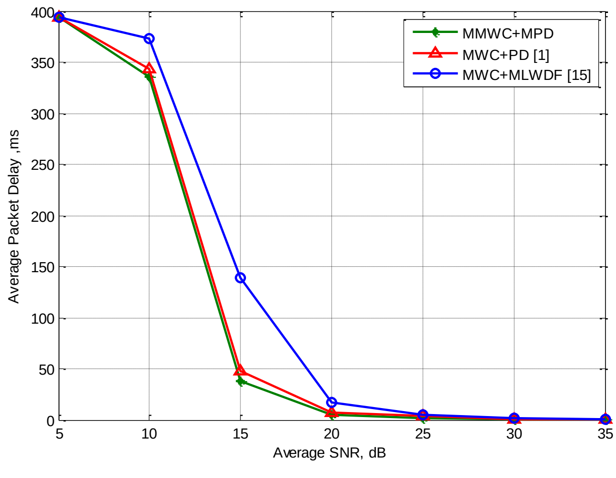


Figure 4.19 Impact of SNR on average packet delay for video traffic with MMWC+MPD, MWC+PD and MWC+M-LWDF designs with 32 users

Comparing the impact results (Figure 4.18 and 4.19) of SNR on average packet delay for voice and video traffics with difference designs, it can be seen that MMWC+MPD and MWC+PD designs performs better for video traffic at lower SNR. For voice traffic, lower packet delay with M-LWDF results from its queue based scheduling. Limited resources make all the packets in the selected queues to be sent first. In general, average packet delay is resulted. For video traffic, during the SNR range from 10 to 25 dB, PD obtains lower video packets. This advantage becomes more with the increase of SNR until 15dB. Same as the previous discussed throughput, the average packet delays with different designs are similar at high SNR (i.e. above 15dB for voice traffic and above 25dB for video traffic). On the other side, PD design has much lower packet drop rate at high SNR.

4.4.2.3.4. Weight Design (MMWC + MPD)

The key of scheduling is weight design at MAC layer for providing QoS information which affects the resource allocation at PHY layer. In [1], the weight calculation depends on the packet delay, packet size and packet QoS priority level which discussed in section 4.3.2. The equations of weight calculation are not unique so in this part the performance of different weight designs in scheduling with MMWC will be analyzed. Following table summarises three different weight designs.

Table 4.8 Summary of different Weight Design Equations

Weight Design	Calculation Equation
W	$\begin{cases} \dfrac{Q_{k,i}S_{k,i}}{U_{k,i}-G_{k,i}-D_{k,i,l}+1} & (\, U_{k,i}-G_{k,i}-D_{k,i,l} \geq 0 \,) \\ Q_{k,i}S_{k,i} & (\, U_{k,i}-G_{k,i}-D_{k,i,l} < 0 \,) \end{cases}$
W1	$\begin{cases} \dfrac{Q_{k,i}S_{k,i}}{\log(U_{k,i}-G_{k,i}-D_{k,i,l}+10)} & (\, U_{k,i}-G_{k,i}-D_{k,i,l} \geq 0 \,) \\ Q_{k,i}S_{k,i} & (\, U_{k,i}-G_{k,i}-D_{k,i,l} < 0 \,) \end{cases}$
W2	$\begin{cases} \dfrac{Q_{k,i}\log(S_{k,i}+1)}{U_{k,i}-G_{k,i}-D_{k,i,l}+1} & (\, U_{k,i}-G_{k,i}-D_{k,i,l} \geq 0 \,) \\ Q_{k,i}\log(S_{k,i}+1) & (\, U_{k,i}-G_{k,i}-D_{k,i,l} < 0 \,) \end{cases}$

Weight W comes from the original equation used in PD scheduling [1]. Weight W1 and W2 can be achieved by some modifications (i.e. add *log* on delay or packet size) from W. Three designs place their own extra emphasis on different factors. Weight design providing weight according to QoS information is sensitive to the real time traffic. The impact of these modifications on throughput, average packet delay and packet drop rate for voice and video traffics are shown in following figures.

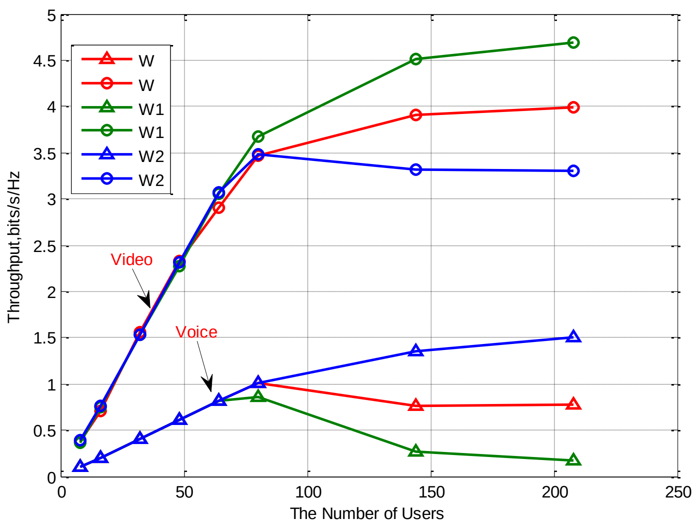


Figure 4.20 Impact of user number on voice and video traffic throughput with three weight designs when SNR=20dB

Seen from Figure 4.20, it can be found that when user number increases in the system with more than 80 users, design with weight W1 have the largest video traffic throughput and at the same time have the least voice traffic throughput. On the contrary, design with weight W2 performs best on throughput for voice traffic and worst for video traffic. In the design with weight W1, *log* is added on the packet delay (i.e.$\log(U_{k,i} - G_{k,i} - D_{k,i,l} + 10)$) which enhances the importance of QoS priority level and packet size in the weight. That is the reason why video traffic throughput is much larger than that of voice traffic due to its large data amount. For weight W2, the weight reflects the QoS priority level and packet delay more. Then voice traffic which requires shorter deadline benefits from it. Both voice and video traffic throughputs are medium with weight design W in [1].

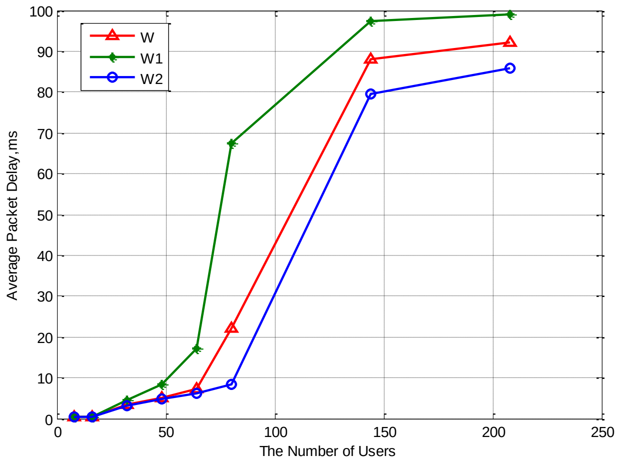


Figure 4.21 Impact of user number average packet delay for voice traffic with three weight design when SNR= 20dB

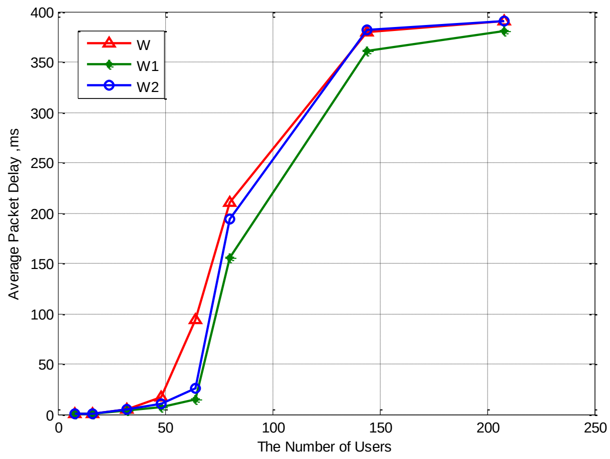


Figure 4.22 Impact of user number on average packet delay for video traffic with three weight designs when SNR =20dB

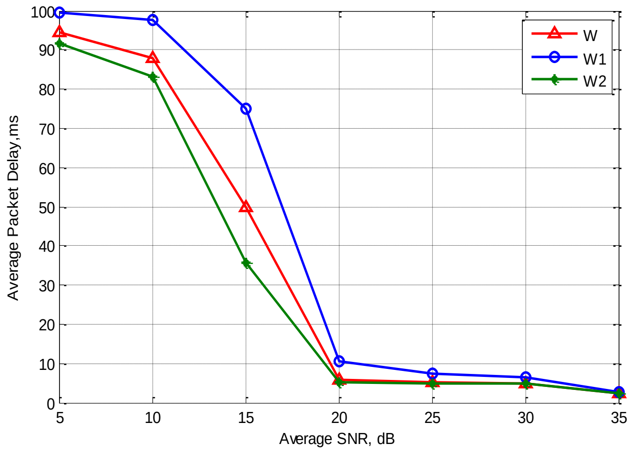


Figure 4.23 Impact of SNR on average packet delay for voice traffic with three weight design with 32 users

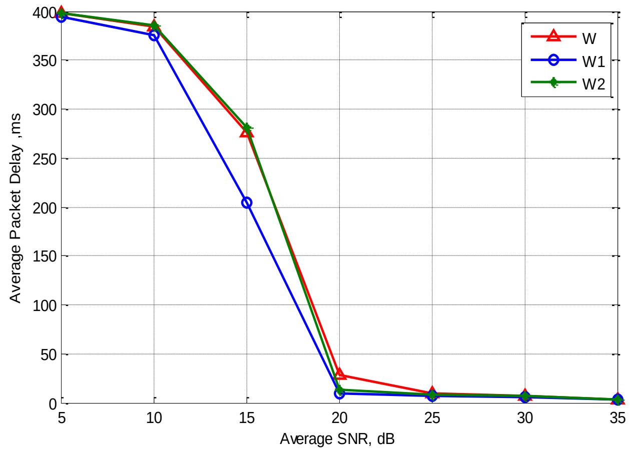


Figure 4.24 Impact of SNR on average packet delay for video traffic with three weight designs with 32 users

Figure 4.21 to 4.24 demonstrate the impact of SNR and user number on average packet delay for voice and video traffic with three weight designs. Generally speaking, there is trade off between average packet delay of voice and video traffics. The design with W1 has largest voice packet delay while smallest one with the increase of user number. At low SNR, it performs best for voice traffic while worst for video traffic. Similar situations are for the design with W2. The design with W looks like to be selected to balance the performance between voice and video traffics.

On the aspect of packet drop rate, the design with weight W2 generally performs best according to the results for 54 users in Figure 4.25 and 4.26. The design with W1 performs quite distinctly for voice and video traffics again. It spends 40ms (worst) and 50ms (best) to guarantee no packet dropping for voice and video respectively. The original weight design requires relative more time for zero probability for video traffic. By considering all three factors throughput, delay and drop rate on balance, the design with weight W may have much more stable and fair performance in heterogeneous traffic system with acceptable delay output probability.

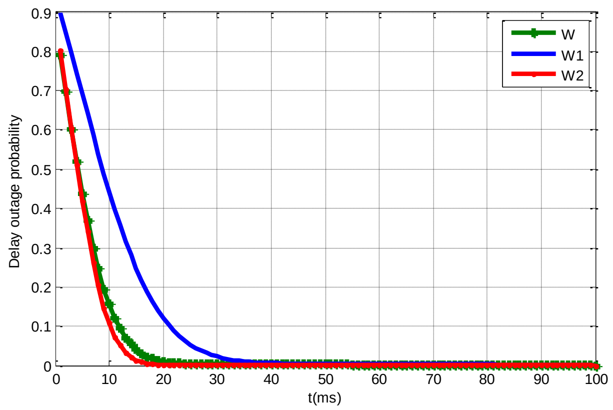


Figure 4.25 Delay outage probability for voice traffic with three weight deigns with 54 users when SNR=20dB

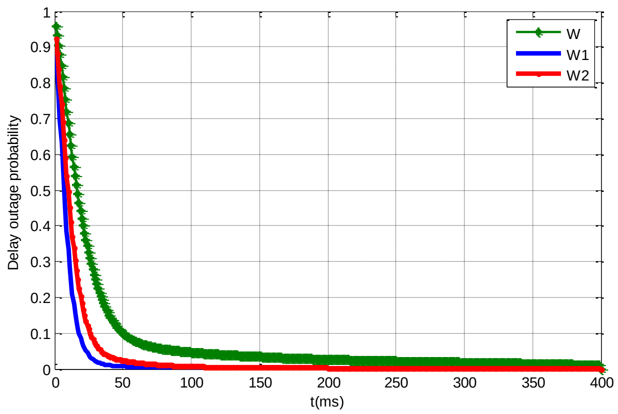


Figure 4.26 Delay outage probability for video traffic with three weight deigns with 54 users when SNR=20dB

4.4.3 Cross-layer Design Applications

After giving theoretical and analytical discussions of different Cross Layer designs in multi-user OFDM system, this section will simply present some related applications in real world [2].

The promising OFDM technology now is widely applied in the field of WLAN and is developed to be used in mobile telecommunication filed. Its high bandwidth efficiency, good ability of anti-interference or anti-multipath-delay makes it meet the requirements of television systems. It can be applied in Asymmetric Digital Subscriber Line (ADSL), High-Definition Television (HDTV), Digital Audio Broadcasting (DAB) and Digital video Broadcasting (DVB) according to European Telecommunications Standards Institute (ETSI). IEEE 802.11a and IEEE802.11g based on OFDM technique are defined on frequency band of 5.8GHz and 2.5GHz for high data rate which is up to 54 Mbps.

The dynamic resource management in the Cross Layer design also plays a very important role in current wireless communication network. It allocates and schedules the obtained resource efficiently to increase the capacity or get the effects of increasing the number of stations. It has some typical applications. First is subcarrier allocation used in different areas during busy or free time periods. It also can be applied at the places (i.e. a public square, conference and exhibition centre, sports field, etc.) where amount of service may increase largely in some periods. Increased subcarrier utilization ratio in free areas solves the coverage of wireless signal. By increasing the signal covering range, the number of base stations can be reduced which is economical and easier for management. Last, the problem of continuous signal cover for high speed moving users can be solved by taking advantage of up-link time slot gain control. For instance, the service in the train or car which is running at high speed may be stopped without exchange in time.

In summary, cross layer design in the multiuser OFDM system is the output that conforming times. It meets the requirements of the high data rate wireless transmission with heterogeneous traffic and considers both limited resources and increasing service demands.

5. Conclusions and Future Work

This thesis has been investigated different adaptive cross layer designs for downlink multi-user OFDM system with heterogeneous traffic. The resource management consists of resource allocation at PHY layer and scheduling at MAC layer. The superiority of MWC [1] based resource allocation scheme is both QoS information and system throughput considerations compared with MC [16] and PF [7] based schemes. For scheduling, packet based PD [1] scheme performance better than queue based M-LWDF [15] scheme on system throughput, packet drop rate and average packet delay in specific user number or SNR range. The correct choice of packet number in weight design brings three times deduction of overall complexity for PD based cross layer design over M-LWDF based one. The proposed MMWC + MPD design which modifies MWC + PD design [1] by considering CSI in weight design additionally. It is proven that the proposed design achieves much more throughput while keeps the performance on packet drop rat and average packet delay as MWC + PD design.

The proposed MMWC +MPD design only considers system throughput maximization and QoS information without the fairness among multiple users. Fairness should be considered in future research and the weight design can focus on more factors for better performance. On the other side, the current cross layer design is implemented at PHY and MAC layers. Further research can achieve optimized cross layer design which can meet the increased requirements of customers better by more layers corporation such as combing with network layer.

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Appendix A_Project Specifications

Project objective:

This project aims to create an adaptive cross-layer design for wireless communication systems, with a focus on physical and link layers [1]. The multiple input multiple output (MIMO) based physical layer and orthogonal frequency division multiplexing (OFDM) techniques, a key technology for 3G and 4G wireless communications, will be employed. The adaptive cross-layer design is used to maximize the weighted sum capacity of the downlink multiuser multi-tasking OFDM systems.

Overall objective:

Wireless communication systems should support multiple users with multiple heterogeneous traffic queues simultaneously for each, with the growing demands of high capacity and multi-tasking services. Compared with the conventional communication network which is based on the independent layer design, lower complexity cross-layer design utilizes the spectrum and energy much more efficiently and achieves the maximum system stability region.

Project Methodology:

The design can be implemented by following four main steps. First, a general downlink time division duplex OFDM system model will be built, which has multiple heterogeneous traffic queues at the same time for each user. Second, a packet dependent (PD) scheduling scheme which is employed at the medium access control (MAC) layer for determining the packet transmission order by assigning different weights should be proposed. The delay, packet size and quality of service (QoS) priority packet level need to be considered for the weight design. Third, the user based resource allocation at the physical (PHY) layer will be done by maximizing the weighted sum capacity of the users. The number of selected packets is required to be chosen properly to reduce the complexity for further weight summing up. Finally, software Matlab will be used for simulation for the complexity and performance analysis of the proposed cross-layer design for the system with some proper assumptions.

[1] Nan Zhou, Xu Zhu, Yi Huang & Hai Lin, "Low complexity cross-layer design with packet dependent scheduling for heterogeneous traffic in multiuser OFDM systems", IEEE Transactions on Wireless Communications, vol. 9, no. 6, pp. 1912-1923, June 2010.

Appendix B_Revised Time Table

: Milestones

Task / Week	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
1. Literature Review	✓														
2. System Building		✓	✓												
2.1 Channel Model		✓													
2.2 OFDM System Model		✓	✓												
2.3 Problem Formulation			✓												
3. Resource Allocation at PHY			✓	✓	✓	✓									
3.1 MC Method				✓	✓	✓									
3.2 PF Method				✓	✓	✓									
3.3 MWC Method (theory)			✓	✓	✓	✓									
4. Scheduling at MAC 							✓	✓	✓	✓					
4.1 M-LWDF							✓	✓							
4.2 PD								✓	✓	✓					
5. Complexity Analysis										✓					
6. New Design 										✓	✓	✓			
7. Performance Analysis											✓	✓			
8. Matlab Simulation		✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓			
9. Simulation Results Analysis		✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓			
10. Draw Conclusion													✓		
11. Thesis Writing 													✓	✓	✓

Appendix C_Partial Matlab Codes

```
% MWC (user based) subcarrier allocation
function
[index_CNR,index,sumweight]=subcarrierallocation_
MWC(Pt,B,N,CNR,K,currentslot,Q,W,T)
            ptmp=Pt/N; sumweight=0;
            RT=zeros(K,1);
            Rtmp=(B/N).*log2(1+ptmp.*CNR);
            for k=1:K
                J(k,:)=Rtmp(k,:)*W(k,currentslot);
            end
             for n=1:N % One subcarrier-> One
user

k=find(J(:,n)==max(J(:,n)));k=k(1);
                while (RT(k)>=Q(k,currentslot))
                    J(k,:)=zeros(1,N);
                    if ~isempty(find(J(:,n),1))

k=find(J(:,n)==max(J(:,n))); k=k(1);
                    else
                        break;
                    end
                end

sumweight=sumweight+W(k,currentslot);
                index(k,n)=1;
                index_CNR(n)=CNR(k,n);

RT(k)=RT(k)+(B/N)*log2(1+ptmp*CNR(k,n))*(T*1e-
3);
            end
```

```
% MMWC subcarrier allocation
function
[index_CNR,index,sumweight]=subcarrierallocation_
M(Pt,B,N,CNR,K,currentslot,Q,W,T)
            ptmp=Pt/N; sumweight=0;
            RT=zeros(K,1);
            Rtmp=(B/N).*log2(1+ptmp.*CNR);
            for k=1:K

Wn(k,:)=W(k,currentslot)*CNR(k,:);
            end
             J=Rtmp.*Wn;
             for n=1:N % One subcarrier-> One
user

k=find(J(:,n)==max(J(:,n)));k=k(1);
                while (RT(k)>=Q(k,currentslot))
                    J(k,:)=zeros(1,N);
                    if ~isempty(find(J(:,n),1))

k=find(J(:,n)==max(J(:,n))); k=k(1);
                    else
                        break;
                    end
                end

sumweight=sumweight+W(k,currentslot);
                index(k,n)=1;
                index_CNR(n)=CNR(k,n);

RT(k)=RT(k)+(B/N)*log2(1+ptmp*CNR(k,n))*(T*1e-
3);
            end
```

voice | P1 | P2 | . . . | PL1 |
video | P1 | P2 | . . . | PL2 |
BE | P1 | . . . | PL3 |

```
% MWC (queue based) subcarrier allocation
function
[index_CNR,index,sumweight]=subcarrierallocation_MLWDF(Pt,B,N,CNR,K,currentslot,Q,W,T,QoS)

ptmp=Pt/N;sumweight=0;

RT=zeros(K*length(QoS),1);

Rtmp_u=(B/N)*log2(1+ptmp.*CNR);

Rtmp_q=zeros(K*length(QoS),N);
            for k=1:K

Rtmp_q((k-1)*length(QoS)+1,:)=Rtmp_u(k,:)/3;

Rtmp_q((k-1)*length(QoS)+2,:)=Rtmp_u(k,:)/3;

Rtmp_q((k-1)*length(QoS)+3,:)=Rtmp_u(k,:)/3;
            end
            for q=1:K*length(QoS)

J(q,:)=Rtmp_q(q,:)*W(q,currentslot);
            end
             for n=1:N

q=find(J(:,n)==max(J(:,n)));q=q(1);
                while (RT(q)>=Q(q,currentslot))

J(q,:)=zeros(1,N);
                    if ~isempty(find(J(:,n),1))

q=find(J(:,n)==max(J(:,n)));q=q(1);
                    else

break;
                    end
                end

sumweight=sumweight+W(q,currentslot);
                index(q,n)=1;

index_CNR(n)=CNR(ceil(q/3),n);

RT(q)=RT(q)+(B/N)*log2(1+ptmp*CNR(ceil(q/3),n))*(T*1e-3);
            end
```

```
%% Weight Calculation of 3 queues By Packet Dependent Scheduling
function
[w_Voice,w_Video,w_BE,Data,rcdslot]=scheduling_PD(data,currentslot,U,K,QoS,slots,T)
V=[100 75 50]; % #.of Packets for weight caculation

w_Voice=zeros(K,U(1));w_Video=zeros(K,U(2));w_BE=zeros(K,U(3));
Data=zeros(K, slots); % Data to be send at each slot for each user
G=1; % Guard Time
tc=currentslot*T;
rcdslot=ones(K*length(QoS),1)*tc;

for k=1:K

v_Voice=V(1);v_Video=V(2);v_BE=V(3); % left    #.of Packets for weight caculation
    %% Case A <=100ms
    if tc<=U(1)
        % lth packet arrives at tl belongs to (0,tc)
        for tl=1:tc
            %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
            % For Voice: 2 situations
            if tc-tl>U(1)-1-G % S_k,i,l > U_k,i-G_k,i (falling into guardtime-urgent)
                if v_Voice>0
                    % W_k,i,l=B_k,i*D_k,i,l

w_Voice(k,tl)=QoS(1)*data(length(QoS)*(k-1)+1,tl);
                end
            else      % out of guradtime
                if v_Voice>0
                    % W_k,i,l=B_k,i*D_k,i,l/(C_k,i,l+1)=B_k,i*D_k,i,l/(U_k,i-S_k,i,l-G_k,i+1)

w_Voice(k,tl)=QoS(1)*data(length(QoS)*(k-1)+1,tl)/(U(1)+tl-tc-G);
                end
            end
            % For Video and BE, out of guradtime
            if v_Video>0

w_Video(k,tl)=QoS(2)*data(length(QoS)*(k-1)+2,tl)/(U(2)+tl-tc-G);
            end
            if v_BE>0
```

```
w_BE(k,tl)=QoS(3)*data(length(QoS)*(k-1)+3,tl)/(U(3)+tl-tc-G);
            end
            %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
            % Accumulated summation of the data (3 queues)
            if v_Voice>0 Data(k,currentslot)=Data(k,currentslot)+data(length(QoS)*(k-1)+1,tl); v_Voice=v_Voice-1;end
            if v_Video>0 Data(k,currentslot)=Data(k,currentslot)+data(length(QoS)*(k-1)+2,tl); v_Video=v_Video-1;end
            if v_BE>0 Data(k,currentslot)=Data(k,currentslot)+data(length(QoS)*(k-1)+3,tl); v_BE=v_BE-1;end

            if v_Voice==0 rcdslot(length(QoS)*(k-1)+1)=tl;end
            if v_Video==0 rcdslot(length(QoS)*(k-1)+2)=tl;end
            if v_BE==0 rcdslot(length(QoS)*(k-1)+3)=tl;end
        end
    end

    %% Case B (100ms-400ms)
    if (U(1)<tc)&&(tc<=U(2))
        % Voice
        for tl=tc+1-U(1):tc % Previous Voice Packets has been droped, new voice caculation
            if tc-tl>U(1)-G-1
                if v_Voice>0
                    w_Voice(k,U(1)+tl-tc)=QoS(1)*data(length(QoS)*(k-1)+1,tl);
                end
            else
                if v_Voice>0
                    w_Voice(k,U(1)+tl-tc)=QoS(1)*data(length(QoS)*(k-1)+1,tl)/(U(1)+tl-tc-G);
                end
            end
            if v_Voice>0
                Data(k,currentslot)=Data(k,currentslot)+data(length(QoS)*(k-1)+1,tl);
                if w_Voice(k,U(1)+tl-tc)>0
                    v_Voice=v_Voice-1;
                    if v_Voice==0 rcdslot(length(QoS)*(k-1)+1)=tl; end
                end
            end
        end
        % Video & BE traffic
        for tl=1:tc
            %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
            if tc-tl>U(2)-1-G
                if v_Video>0
                    w_Video(k,tl)=QoS(2)*data(length(QoS)*(k-1)+2,tl);
                end
            else
                if v_Video>0
                    w_Video(k,tl)=QoS(2)*data(length(QoS)*(k-1)+2,tl)/(U(2)+tl-tc-G);
                end
            end
            if v_BE>0
                w_BE(k,tl)=QoS(3)*data(length(QoS)*(k-1)+3,tl)/(U(3)+tl-tc-G);
            end
            %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
            if v_Video>0
                Data(k,currentslot)=Data(k,currentslot)+data(length(QoS)*(k-1)+2,tl);
                if w_Video(k,tl)>0
                    v_Video=v_Video-1;
                    if v_Video==0 rcdslot(length(QoS)*(k-1)+2)=tl; end
                end
            end
            if v_BE>0
                Data(k,currentslot)=Data(k,currentslot)+data(length(QoS)*(k-1)+3,tl);
                if w_BE(k,tl)>0
                    v_BE=v_BE-1;
                    if v_BE==0 rcdslot(length(QoS)*(k-1)+3)=tl; end
                end
            end
        end
    end
    %% Case C (400ms-1000ms)
    if (U(2)<tc)&&(tc<=U(3))
        % Voice
```

```
            for tl=tc+1-U(1):tc
                if tc-tl>U(1)-1-G
                    if v_Voice>0

w_Voice(k,U(1)+tl-tc)=QoS(1)*da
ta(length(QoS)*(k-1)+1,tl);
                    end
                else
                    if v_Voice>0

w_Voice(k,U(1)+tl-tc)=QoS(1)*da
ta(length(QoS)*(k-1)+1,tl)/(U(1)+t
l-tc-G);
                    end
                end
                if v_Voice>0

Data(k,currentslot)=Data(k,current
slot)+data(length(QoS)*(k-1)+1,tl)
;
                    if
w_Voice(k,U(1)+tl-tc)>0

v_Voice=v_Voice-1;
                        if
v_Voice==0
rcdslot(length(QoS)*(k-1)+1)=tl;e
nd
                    end
                end
            end
            % Video
            for tl=tc+1-U(2):tc
                if U(2)-1-G<=tc-tl
                    if v_Video>0

w_Video(k,U(2)+tl-tc)=QoS(2)*da
ta(length(QoS)*(k-1)+2,tl);
                    end
                else
                    if v_Video>0

w_Video(k,U(2)+tl-tc)=QoS(2)*da
ta(length(QoS)*(k-1)+2,tl)/(U(2)+t
l-tc-G);
                    end
                end
                if v_Video>0

Data(k,currentslot)=Data(k,current
slot)+data(length(QoS)*(k-1)+2,tl)
;
                    if
w_Video(k,U(2)+tl-tc)>0

v_Video=v_Video-1;
                        if
v_Video==0
rcdslot(length(QoS)*(k-1)+2)=tl;
end
                    end
                end
            end
            % BE
            for tl=1:tc
                if tc-tl>U(3)-G-1
                    if v_BE>0

w_BE(k,tl)=QoS(3)*data(length(Q
oS)*(k-1)+3,tl);
                    end
                else
                    if v_BE>0

w_BE(k,tl)=QoS(3)*data(length(Q
oS)*(k-1)+3,tl)/(U(3)+tl-tc-G);

                    end
                end
                if v_BE>0

Data(k,currentslot)=Data(k,current
slot)+data(length(QoS)*(k-1)+3,tl)
;
                    if
w_BE(k,tl)>0

v_BE=v_BE-1;
                        if
v_BE==0

rcdslot(length(QoS)*(k-1)+3)=tl;
                        end
                    end
                end
            end
        end
        %% Case D (>1000ms)
        if tc>U(3)
            % Voice
            for tl=tc+1-U(1):tc
                if tc-tl>U(1)-1-G
                    if v_Voice>0

w_Voice(k,U(1)+tl-tc)=QoS(1)*da
ta(length(QoS)*(k-1)+1,tl);
                    end
                else
                    if v_Voice>0

w_Voice(k,U(1)+tl-tc)=QoS(1)*da
ta(length(QoS)*(k-1)+1,tl)/(U(1)+t
l-tc-G);
                    end
                end
                if v_Voice>0

Data(k,currentslot)=Data(k,current
slot)+data(length(QoS)*(k-1)+1,tl)
;
                    if
w_Voice(k,U(1)+tl-tc)>0

v_Voice=v_Voice-1;
                        if
v_Voice==0
rcdslot(length(QoS)*(k-1)+1)=tl;e
nd
                    end
                end
            end
            % Video
```

```
        for tl=tc+1-U(2):tc
            if U(2)-1-G<=tc-tl
                if v_Video>0
                    w_Video(k,U(2)+tl-tc)=QoS(2)*data(length(QoS)*(k-1)+2,tl);
                end
            else
                if v_Video>0
                    w_Video(k,U(2)+tl-tc)=QoS(2)*data(length(QoS)*(k-1)+2,tl)/(U(2)+tl-tc-G);
                end
            end
            if v_Video>0
                Data(k,currentslot)=Data(k,currentslot)+data(length(QoS)*(k-1)+2,tl);
                if w_Video(k,U(2)+tl-tc)>0
                    v_Video=v_Video-1;
                    if v_Video==0 rcdslot(length(QoS)*(k-1)+2)=tl;end
                end
            end
        end
        % BE
        for tl=tc+1-U(3):tc
            if U(3)-1-G<=tc-tl
                if v_BE>0
                    w_BE(k,U(3)+tl-tc)=QoS(3)*data(length(QoS)*(k-1)+3,tl);
                end
            else
                if v_BE>0
                    w_BE(k,U(3)+tl-tc)=QoS(3)*data(length(QoS)*(k-1)+3,tl)/(U(3)+tl-tc-G);
                end
            end
            if v_BE>0
                Data(k,currentslot)=Data(k,currentslot)+data(length(QoS)*(k-1)+3,tl);
                if w_BE(k,U(3)+tl-tc)>0
                    v_BE=v_BE-1;
                    if v_BE==0 rcdslot(length(QoS)*(k-1)+3)=tl;end
                end
            end
        end
    end
end
```

```
 %% M-LWDF Scheduling (queue based)
function [Data, weight]=scheduling_MLWDF(data,U,Data,K,currentslot, Cap_avg, weight, T)
tc=currentslot*T;

a1=-log10(0.05)/(U(1)*1e-3);
a2=-log10(0.05)/(U(2)*1e-3);
a3=-log10(0.77)/(U(3)*1e-3);
%a3=0.26;
%% For Queue1: Voice
for k=1:K
    if tc<=U(1)
        for ptr=1:tc % record the time for the delayed packet
            if data((k-1)*length(U)+1, ptr)>0
                break; end
        end
        Data((k-1)*length(U)+1, currentslot)=Data(length(U)*(k-1)+1, currentslot)+sum(data(length(U)*(k-1)+1,1:tc));
    else
        for ptr=tc+1-U(1):tc
            if data((k-1)*length(U)+1, ptr)>0
                break; end
        end
        Data((k-1)*length(U)+1, currentslot)=Data(length(U)*(k-1)+1, currentslot)+sum(data(length(U)*(k-1)+1,tc+1-U(1):tc));
    end

weight((k-1)*length(U)+1,currentslot)=a1*(tc-ptr+1)/Cap_avg((k-1)*length(U)+1)*(T*1e-3);
end
 %% For Queue 2: Video
for k=1:K
    if tc<=U(2)
        for ptr=1:tc
            if data((k-1)*length(U)+2, ptr)>0
                break; end
```

```
            end
            Data((k-1)*length(U)+2,
currentslot)=Data(length(U)*(k-1)+2,
currentslot)+sum(data(length(U)*(k-1)+2,1:tc));
        else
            for ptr=tc+1-U(2):tc
                if data((k-1)*length(U)+2, ptr)>0
                    break;end
            end
            Data((k-1)*length(U)+2,
currentslot)=Data(length(U)*(k-1)+2,
currentslot)+sum(data(length(U)*(k-1)+2,tc+1-U(2):tc
));
        end

weight((k-1)*length(U)+2,currentslot)=a2*(tc-ptr+1)/
Cap_avg((k-1)*length(U)+2)*(T*1e-3);
end
 %% For Queue 3: BE
for k=1:K
        if tc<=U(3)
            for ptr=1:tc
                if data((k-1)*length(U)+3, ptr)>0 break;
end
            end
            Data((k-1)*length(U)+3,
currentslot)=Data(length(U)*(k-1)+3,
currentslot)+sum(data(length(U)*(k-1)+3,1:tc));
        else
            for ptr=tc+1-U(3):tc
                if data((k-1)*length(U)+3, ptr)>0 break;
end
            end
            Data((k-1)*length(U)+3,
currentslot)=Data(length(U)*(k-1)+3,
currentslot)+sum(data(length(U)*(k-1)+3,tc+1-U(3):tc
));
        end

weight((k-1)*length(U)+3,currentslot)=a3*(tc-ptr+1)/
Cap_avg((k-1)*length(U)+3)*(T*1e-3);
end

 %%   Power Allocation
function
[PA]=powerallocation(N,Pt,W,index_CNR,index,sum
W,currentslot)
                PA=zeros(1,N);
                for n=1:N
                    k=find(index(:,n));

PA(n)=(Pt+sum(1./index_CNR)).*W(k,currentslot)/su
mW-1/index_CNR(n);
                end

                while(~isempty(find(PA<0)))
                    neg= find(PA <= 0);
                    pos= find(PA >   0);

                    PA(neg) = 0;
                    Newindex_CNR =
index_CNR(pos);

                    for t=1:length(neg)
                        k=find(index(:,neg(t)));

sumW=sumW-W(k,currentslot);
                    end

                    for t=1:length(pos)
                        k=find(index(:,pos(t)));

PA(pos(t))=(Pt+sum(1./Newindex_CNR))*W(k,curren
tslot)/sumW-1/index_CNR(pos(t));
                    end
                end
```

```
%% Capacity or Delay VS. SNR (MWC+PD & MMWC+MPD & MWC+MLWDF)
K=32;                    % Number of Users
N=512;                   % Number of Subcarriers
B=5e6;
Pt=1;
T=2;slots=2000;
N_Loop=1;
gamma=ones(K,1);
QoS=[1024 512 1];            % QoS priority level for
Voice, VBR video and BE traffic (3 queues)
U = [100 400 1000];        % Delay tolerance for 3
queues - (slots)

SNR_dB=[5 10 15 20 25 30 35];
No_=Pt*(T*1e-3/N)./10.^(SNR_dB/10);

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%%%%%%%%%%%%%%%%%%%%    Initialization
Capacity_Voice_PD=zeros(1,length(SNR_dB));Capaci
ty_Voice_MLWDF=zeros(1,length(SNR_dB));Capacit
y_Voice=zeros(1,length(SNR_dB));
Capacity_Video_PD=zeros(1,length(SNR_dB));Capac
ity_Video_MLWDF=zeros(1,length(SNR_dB));Capaci
ty_Video=zeros(1,length(SNR_dB));
Capacity_BE_PD=zeros(1,length(SNR_dB));
Capacity_BE_MLWDF=zeros(1,length(SNR_dB));
Capacity_BE=zeros(1,length(SNR_dB));

Delay_avg_Voice_PD=zeros(1,length(SNR_dB));Dela
y_avg_Voice_MLWDF=zeros(1,length(SNR_dB));Del
ay_avg_Voice=zeros(1,length(SNR_dB));
Delay_avg_Video_PD=zeros(1,length(SNR_dB));Dela
y_avg_Video_MLWDF=zeros(1,length(SNR_dB));De
lay_avg_Video=zeros(1,length(SNR_dB));
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%%%%%%%%%%%%%%%%%%%%    Initialization

for n_loop=1:N_Loop

capacity_PD=zeros(K*length(QoS),length(SNR_dB));
capacity_MLWDF=zeros(K*length(QoS),length(SNR
_dB));
capacity=zeros(K*length(QoS),length(SNR_dB));
    for i=1:length(SNR_dB)
        No=No_(i)
        Data_PD=zeros(K, slots);
Data_MLWDF=zeros(K*length(QoS), slots);
Data=zeros(K, slots);
        weight_PD=zeros(K,slots);
weight_MLWDF=zeros(K*length(QoS),slots);
weight=zeros(K,slots);
        Cap_avg=ones(K*length(QoS),1);
        %% Data(Packet size) and delay generation
of 3 queues with specific #. users

[data,delay]=Data_Delay_Generation(K,QoS,T*slots,
U);

[data_PD,delay_PD]=Data_Delay_Generation(K,QoS,
T*slots,U);

[data_MLWDF,delay_MLWDF]=Data_Delay_Genera
tion(K,QoS,T*slots,U);

        for currentslot=1:slots
            %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%    Initialization

shannoncapacity_PD=zeros(K,1);shannoncapacity_M
LWDF =
zeros(K*length(QoS),1);shannoncapacity=zeros(K,1);
            %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%    Initialization
            %% Scheduling-> Weight Caculation
            [w_Voice_PD,w_Video_PD,w_BE_PD,
Data_PD,
rcdslot_PD]=scheduling_PD(data_PD,currentslot,U,K,
QoS,slots,T);
            [w_Voice,w_Video,w_BE, Data,
rcdslot]=scheduling_PD(data,currentslot,U,K,QoS,slot
s,T);
            [Data_MLWDF,
weight_MLWDF]=scheduling_MLWDF(data_MLWD
F,U,Data_MLWDF,K,currentslot, Cap_avg,
weight_MLWDF,T);
            for k=1:K
```

```
weight_PD(k,currentslot)=sum(w_Voice_PD(k,:))+sum(w_Video_PD(k,:))+sum(w_BE_PD(k,:));

weight(k,currentslot)=sum(w_Voice(k,:))+sum(w_Video(k,:))+sum(w_BE(k,:));
            end
            %% Generate channel information
            CSI=channel(K,N);chh=abs(CSI).^2;
noise=No*B/N;CNR=chh/noise;
            %% Subcaarier Allocation

[Index_CNR,Index,sumweight]=subcarrierallocation_M(Pt,B,N,CNR,K,currentslot,Data,weight,T);

[Index_CNR_PD,Index_PD,sumweight_PD]=subcarrierallocation_MWC(Pt,B,N,CNR,K,currentslot,Data_PD,weight_PD,T);

[Index_CNR_MLWDF,Index_MLWDF,sumweight_MLWDF]=subcarrierallocation_MLWDF(Pt,B,N,CNR,K,currentslot,Data_MLWDF,weight_MLWDF,T,QoS);
            %% Power Allocation

PA=powerallocation(N,Pt,weight,Index_CNR,Index,sumweight,currentslot);

PA_PD=powerallocation(N,Pt,weight_PD,Index_CNR_PD,Index_PD,sumweight_PD,currentslot);

PA_MLWDF=powerallocation(N,Pt,weight_MLWDF,Index_CNR_MLWDF,Index_MLWDF,sumweight_MLWDF,currentslot);
            %% Shannon Capacity of each user
            for n=1:N
                user=find(Index(:,n));
                user_PD=find(Index_PD(:,n));

queue_MLWDF=find(Index_MLWDF(:,n));

shannoncapacity(user)=shannoncapacity(user)+(B/N)*log2(1+PA(n)*Index_CNR(n));

shannoncapacity_PD(user_PD)=shannoncapacity_PD(user_PD)+(B/N)*log2(1+PA_PD(n)*Index_CNR_PD(n));

shannoncapacity_MLWDF(queue_MLWDF)=shannoncapacity_MLWDF(queue_MLWDF)+(B/N)*log2(1+PA_MLWDF(n)*Index_CNR_MLWDF(n));
            end
            %% Real Capacity Caculation
            % New
            for k=1:K
                if Data(k, currentslot)>
(shannoncapacity(k)*(T*1e-3))

[data,delay,capacity]=realdata_shannon(currentslot, k, QoS, U, shannoncapacity(k)*(T*1e-3), data, w_Voice, w_Video, w_BE, delay,capacity, i,T);
                    Data(k, currentslot) = Data(k,
currentslot) - shannoncapacity(k)*(T*1e-3);
                else

[data,delay,capacity]=realdata_all(currentslot, k, QoS, U, data, delay, rcdslot, capacity, i,T);
                    Data(k, currentslot) =0;
                end
            end
            % PD
            for k=1:K
                if Data_PD(k, currentslot) >
(shannoncapacity_PD(k)*(T*1e-3))

[data_PD,delay_PD,capacity_PD]=realdata_shannon(currentslot, k, QoS, U,
shannoncapacity_PD(k)*(T*1e-3), data_PD,
w_Voice_PD, w_Video_PD, w_BE_PD,
delay_PD,capacity_PD, i,T);
                    Data_PD(k, currentslot) =
Data_PD(k, currentslot) -
shannoncapacity_PD(k)*(T*1e-3);
                else

[data_PD,delay_PD,capacity_PD]=realdata_all(currentslot, k, QoS, U, data_PD, delay_PD, rcdslot_PD,
capacity_PD, i,T);
                    Data_PD(k, currentslot) = 0;
```

```
            end
        end
        % MLWDF
        for q=1:(K*length(QoS))
            if Data_MLWDF(q,
currentslot)>(shannoncapacity_MLWDF(q)*(T*1e-3))

[data_MLWDF,delay_MLWDF,capacity_MLWDF]=realdata_shannon_MLWDF( currentslot,q,U,shannoncapacity_MLWDF(q)*(T*1e-3),data_MLWDF,delay_MLWDF,capacity_MLWDF,i,T);

Cap_avg(q)=0.9*Cap_avg(q)+0.1*shannoncapacity_MLWDF(q)*(T*1e-3);
            else

[data_MLWDF,delay_MLWDF,capacity_MLWDF]=realdata_all_MLWDF(currentslot,q,U,data_MLWDF,delay_MLWDF,capacity_MLWDF,i,T);

Cap_avg(q)=0.9*Cap_avg(q)+0.1*Data_MLWDF(q,
currentslot);
            end
        end
    end
    for k=1:K

Capacity_Voice(i)=Capacity_Voice(i)+capacity((k-1)*length(QoS)+1,i);

Capacity_Voice_PD(i)=Capacity_Voice_PD(i)+capacity_PD((k-1)*length(QoS)+1,i);

Capacity_Voice_MLWDF(i)=Capacity_Voice_MLWDF(i)+capacity_MLWDF((k-1)*length(QoS)+1,i);


Capacity_Video(i)=Capacity_Video(i)+capacity((k-1)*length(QoS)+2,i);

Capacity_Video_PD(i)=Capacity_Video_PD(i)+capacity_PD((k-1)*length(QoS)+2,i);

Capacity_Video_MLWDF(i)=Capacity_Video_MLWDF(i)+capacity_MLWDF((k-1)*length(QoS)+2,i);


Capacity_BE(i)=Capacity_BE(i)+capacity((k-1)*length(QoS)+3,i);

Capacity_BE_PD(i)=Capacity_BE_PD(i)+capacity_PD((k-1)*length(QoS)+3,i);

Capacity_BE_MLWDF(i)=Capacity_BE_MLWDF(i)+capacity_MLWDF((k-1)*length(QoS)+3,i);


Delay_avg_Voice(i)=Delay_avg_Voice(i)+sum(delay((k-1)*length(QoS)+1, 1:slots*T-U(1)))/(slots*T-U(1));

Delay_avg_Voice_PD(i)=Delay_avg_Voice_PD(i)+sum(delay_PD((k-1)*length(QoS)+1,
1:slots*T-U(1)))/(slots*T-U(1));

Delay_avg_Voice_MLWDF(i)=Delay_avg_Voice_MLWDF(i)+sum(delay_MLWDF((k-1)*length(QoS)+1,
1:slots*T-U(1)))/(slots*T-U(1));


Delay_avg_Video(i)=Delay_avg_Video(i)+sum(delay((k-1)*length(QoS)+2, 1:slots*T-U(2)))/(slots*T-U(2));

Delay_avg_Video_PD(i)=Delay_avg_Video_PD(i)+sum(delay_PD((k-1)*length(QoS)+2,
1:slots*T-U(2)))/(slots*T-U(2));

Delay_avg_Video_MLWDF(i)=Delay_avg_Video_MLWDF(i)+sum(delay_MLWDF((k-1)*length(QoS)+2,
1:slots*T-U(2)))/(slots*T-U(2));
    end
    Delay_avg_Voice(i)=Delay_avg_Voice(i)/K;

Delay_avg_Voice_PD(i)=Delay_avg_Voice_PD(i)/K;

Delay_avg_Voice_MLWDF(i)=Delay_avg_Voice_MLWDF(i)/K;
```

```
Delay_avg_Video(i)=Delay_avg_Video(i)/K;

Delay_avg_Video_PD(i)=Delay_avg_Video_PD(i)/K;

Delay_avg_Video_MLWDF(i)=Delay_avg_Video_MLWDF(i)/K;
    end
end
Capacity=Capacity_Voice+Capacity_Video+Capacity_BE;
Capacity_PD=Capacity_Voice_PD+Capacity_Video_PD+Capacity_BE_PD;
Capacity_MLWDF=Capacity_Voice_MLWDF+Capacity_Video_MLWDF+Capacity_BE_MLWDF;

figure;
plot(SNR_dB,Capacity/N_Loop/B/slots/(T*1e-3),'g*-');
hold on;
plot(SNR_dB,Capacity_PD/N_Loop/B/slots/(T*1e-3),'r^-');
hold on;
plot(SNR_dB,Capacity_MLWDF/N_Loop/B/slots/(T*1e-3),'o-');
grid on
legend('Modified-PD','PD','MLWDF');
xlabel('Average SNR, dB')
ylabel('System Throughput,bits/s/Hz')

figure;
plot(SNR_dB,Capacity_BE/N_Loop/B/slots/(T*1e-3),'g*-');
hold on;
plot(SNR_dB,Capacity_BE_PD/N_Loop/B/slots/(T*1e-3),'r^-');
hold on;
plot(SNR_dB,Capacity_BE_MLWDF/N_Loop/B/slots/(T*1e-3),'o-');
grid on
legend('Modified-PD','PD','MLWDF');
xlabel('Average SNR, dB')
ylabel('Throughput for BE traffic,bits/s/Hz')

figure;
plot(SNR_dB,Capacity_Voice/N_Loop/B/slots/(T*1e-3),'g^-');
hold on;
plot(SNR_dB,Capacity_Video/N_Loop/B/slots/(T*1e-3),'go-');
hold on;
plot(SNR_dB,Capacity_Voice_PD/N_Loop/B/slots/(T*1e-3),'r^-');
hold on;
plot(SNR_dB,Capacity_Video_PD/N_Loop/B/slots/(T*1e-3),'ro-');
hold on;
plot(SNR_dB,Capacity_Voice_MLWDF/N_Loop/B/slots/(T*1e-3),'^-');
hold on;
plot(SNR_dB,Capacity_Video_MLWDF/N_Loop/B/slots/(T*1e-3),'o-');
grid on
legend('Modified-PD-Voice','Modified-PD-Video','PD-Voice','PD-Video','MLWDF-Voice','MLWDF-Video');
xlabel('Average SNR, dB')
ylabel('Throughput,bits/s/Hz')

figure;
plot(SNR_dB,Delay_avg_Voice/N_Loop, 'g*-');
hold on;
plot(SNR_dB,Delay_avg_Voice_PD/N_Loop, 'r^-');
hold on;
plot(SNR_dB,Delay_avg_Voice_MLWDF/N_Loop, 'o-');
grid on;
legend('Modified-PD-Voice','PD-Voice','M-LWDF-Voice');
xlabel('Average SNR, dB');
ylabel('Average Packet Delay,ms');

figure;
plot(SNR_dB,Delay_avg_Video/N_Loop, 'g*-');
hold on;
plot(SNR_dB,Delay_avg_Video_PD/N_Loop, 'r^-');
hold on;
plot(SNR_dB,Delay_avg_Video_MLWDF/N_Loop, 'o-');
grid on;
```

```
legend('Modified-PD-Video','PD-Video','M-LWDF-Video');
xlabel('Average SNR, dB');
ylabel('Average Packet Delay ,ms');

%% Capacity or Delay VS. User number (MWC+PD & MMWC+MPD & MWC+MLWDF)
N_Loop=1;               % Simulation times
N=512;                  % Number of Subcarriers
B=5e6;                  % Bandwidth
T=2;slots=2000;    % number of total slot in one simulation
Pt=1;                   % Total Power
SNR_dB=20;No=Pt*(T*1e-3/N)./10.^(SNR_dB/10);    % AWGN Noise Density of 20dB SNR
K_=[ 8 16:16:64 80:64:272];               % User Number Array - Bata
QoS=[1024 512 1];                              % QoS priority level for Voice, VBR video and BE traffic (3 queues)
U = [100 400 1000];         % Delay tolerance for 3 queues - (slots)

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%    Initialization
Capacity_Voice_PD=zeros(1,length(K_));Capacity_Voice_MLWDF=zeros(1,length(K_));
Capacity_Video_PD=zeros(1,length(K_));Capacity_Video_MLWDF=zeros(1,length(K_));
Capacity_BE_PD=zeros(1,length(K_));
Capacity_BE_MLWDF=zeros(1,length(K_));

Delay_avg_Voice_PD=zeros(1,length(K_));Delay_avg_Voice_MLWDF=zeros(1,length(K_));
Delay_avg_Video_PD=zeros(1,length(K_));Delay_avg_Video_MLWDF=zeros(1,length(K_));

Capacity_Voice=zeros(1,length(K_));
Capacity_Video=zeros(1,length(K_));
Capacity_BE=zeros(1,length(K_));

Delay_avg_Voice=zeros(1,length(K_));
Delay_avg_Video=zeros(1,length(K_));
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%    Initialization
for j=1:length(K_)
        K=K_(j)
        gamma=ones(K,1);
        %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%    Initialization

capacity_PD=zeros(K*length(QoS),length(K_));capacity_MLWDF=zeros(K*length(QoS),length(K_));capacity=zeros(K*length(QoS),length(K_));
        delay_PD=zeros(K*length(QoS), slots*T);
delay_MLWDF=zeros(K*length(QoS), slots*T);delay=zeros(K*length(QoS), slots*T);
        data_PD=zeros(K*length(QoS), slots*T);
data_MLWDF=zeros(K*length(QoS), slots*T);
data=zeros(K*length(QoS), slots*T);
        %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%    Initialization
        for n_loop=1:N_Loop
                Data_PD=zeros(K, slots);
Data_MLWDF=zeros(K*length(QoS), slots);
Data=zeros(K, slots);
                weight_PD=zeros(K,slots);
weight_MLWDF=zeros(K*length(QoS),slots);weight=zeros(K,slots);
                Cap_avg=ones(K*length(QoS),1);
                %% Data(Packet size) and delay generation of 3 queues with specific #. users

[data_PD,delay_PD]=Data_Delay_Generation(K,QoS,T*slots,U);

[data,delay]=Data_Delay_Generation(K,QoS,T*slots,U);

[data_MLWDF,delay_MLWDF]=Data_Delay_Generation(K,QoS,T*slots,U);
                for currentslot=1:slots
                        %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%    Initialization

shannoncapacity_PD=zeros(K,1);shannoncapacity_MLWDF =
```

```
zeros(K*length(QoS),1);shannoncapacity=zeros(K,1);
            %%%%%%%%%%%%%%%%%%%%%
%%%%%%%%%%%%%%%%   Initialization
            %% Scheduling-> Weight Caculation
            [w_Voice_PD,w_Video_PD,w_BE_PD,
Data_PD,
rcdslot_PD]=scheduling_PD(data_PD,currentslot,U,K,
QoS,slots,T);
            [w_Voice,w_Video,w_BE, Data,
rcdslot]=scheduling_PD(data,currentslot,U,K,QoS,slot
s,T);
            [Data_MLWDF,
weight_MLWDF]=scheduling_MLWDF(data_MLWD
F,U,Data_MLWDF,K,currentslot, Cap_avg,
weight_MLWDF,T);
            for k=1:K

weight_PD(k,currentslot)=sum(w_Voice_PD(k,:))+su
m(w_Video_PD(k,:))+sum(w_BE_PD(k,:));

weight(k,currentslot)=sum(w_Voice(k,:))+sum(w_Vid
eo(k,:))+sum(w_BE(k,:));
            end
            %% Generate channel information
            CSI=channel(K,N);
chh=abs(CSI).^2;noise=No*B/N; CNR=chh/noise;
            %% Subcaarier Allocation

[Index_CNR_PD,Index_PD,sumweight_PD]=subcarri
erallocation_MWC(Pt,B,N,CNR,K,currentslot,Data_P
D,weight_PD,T);

[Index_CNR,Index,sumweight]=subcarrierallocation_
M(Pt,B,N,CNR,K,currentslot,Data_PD,weight,T);

[Index_CNR_MLWDF,Index_MLWDF,sumweight_M
LWDF]=subcarrierallocation_MLWDF(Pt,B,N,CNR,
K,currentslot,Data_MLWDF,weight_MLWDF,T,QoS)
;
            %% Power Allocation

PA_PD=powerallocation(N,Pt,weight_PD,Index_CNR
_PD,Index_PD,sumweight_PD,currentslot);

PA=powerallocation(N,Pt,weight,Index_CNR,Index,su
mweight,currentslot);

PA_MLWDF=powerallocation(N,Pt,weight_MLWDF,
Index_CNR_MLWDF,Index_MLWDF,sumweight_M
LWDF,currentslot);
            %% Shannon Capacity of each user
(how many bits at each subcarrier)
            for n=1:N
                user_PD=find(Index_PD(:,n));
                user=find(Index(:,n));

queue_MLWDF=find(Index_MLWDF(:,n));

shannoncapacity_PD(user_PD)=shannoncapacity_PD(
user_PD)+(B/N)*log2(1+PA_PD(n)*Index_CNR_PD(
n));

shannoncapacity(user)=shannoncapacity(user)+(B/N)*
log2(1+PA(n)*Index_CNR(n));

shannoncapacity_MLWDF(queue_MLWDF)=shannon
capacity_MLWDF(queue_MLWDF)+(B/N)*log2(1+P
A_MLWDF(n)*Index_CNR_MLWDF(n));
            end
            %% Real Data Caculation
            % PD
            for k=1:K
                if Data_PD(k, currentslot)>
(shannoncapacity_PD(k)*(T*1e-3))

[data_PD,delay_PD,capacity_PD]=realdata_shannon(c
urrentslot, k, QoS, U,
shannoncapacity_PD(k)*(T*1e-3), data_PD,
w_Voice_PD, w_Video_PD, w_BE_PD,
delay_PD,capacity_PD, j,T);
                    Data_PD(k, currentslot) =
Data_PD(k, currentslot) -
shannoncapacity_PD(k)*(T*1e-3);
                else

[data_PD,delay_PD,capacity_PD]=realdata_all(current
slot, k, QoS, U, data_PD, delay_PD, rcdslot_PD,
capacity_PD, j,T);
```

```
                Data_PD(k, currentslot) =0;
            end
        end
        % New
        for k=1:K
            if Data(k, currentslot)>
(shannoncapacity(k)*(T*1e-3))

[data,delay,capacity]=realdata_shannon(currentslot, k,
QoS, U, shannoncapacity(k)*(T*1e-3), data, w_Voice,
w_Video, w_BE, delay,capacity, j,T);
                Data(k, currentslot) = Data(k,
currentslot) - shannoncapacity(k)*(T*1e-3);
            else

[data,delay,capacity]=realdata_all(currentslot, k, QoS,
U, data, delay, rcdslot, capacity, j,T);
                Data(k, currentslot) =0;
            end
        end
        % M-LWDF
        for q=1:(K*length(QoS))
            if Data_MLWDF(q,
currentslot)>(shannoncapacity_MLWDF(q)*(T*1e-3))

[data_MLWDF,delay_MLWDF,capacity_MLWDF]=r
ealdata_shannon_MLWDF( currentslot,q,U,shannonca
pacity_MLWDF(q)*(T*1e-3),data_MLWDF,delay_M
LWDF,capacity_MLWDF,j,T);

Cap_avg(q)=0.9*Cap_avg(q)+0.1*shannoncapacity_M
LWDF(q)*(T*1e-3);
            else

[data_MLWDF,delay_MLWDF,capacity_MLWDF]=r
ealdata_all_MLWDF(currentslot,q,U,data_MLWDF,d
elay_MLWDF,capacity_MLWDF,j,T);

Cap_avg(q)=0.9*Cap_avg(q)+0.1*Data_MLWDF(q,
currentslot);
            end
        end
    end % end all the slots
    for k=1:K
                %% calculate throughput

Capacity_Voice(j)=Capacity_Voice(j)+capacity((k-1)*
length(QoS)+1,j);

Capacity_Voice_PD(j)=Capacity_Voice_PD(j)+capaci
ty_PD((k-1)*length(QoS)+1,j);

Capacity_Voice_MLWDF(j)=Capacity_Voice_MLW
DF(j)+capacity_MLWDF((k-1)*length(QoS)+1,j);


Capacity_Video(j)=Capacity_Video(j)+capacity((k-1)*
length(QoS)+2,j);

Capacity_Video_PD(j)=Capacity_Video_PD(j)+capaci
ty_PD((k-1)*length(QoS)+2,j);

Capacity_Video_MLWDF(j)=Capacity_Video_MLW
DF(j)+capacity_MLWDF((k-1)*length(QoS)+2,j);


Capacity_BE(j)=Capacity_BE(j)+capacity((k-1)*lengt
h(QoS)+3,j);

Capacity_BE_PD(j)=Capacity_BE_PD(j)+capacity_P
D((k-1)*length(QoS)+3,j);

Capacity_BE_MLWDF(j)=Capacity_BE_MLWDF(j)+
capacity_MLWDF((k-1)*length(QoS)+3,j);

                %% calculate the average packet delay

Delay_avg_Voice(j)=Delay_avg_Voice(j)+sum(delay(
(k-1)*length(QoS)+1, 1:slots*T-U(1)))/(slots*T-U(1));

Delay_avg_Voice_PD(j)=Delay_avg_Voice_PD(j)+su
m(delay_PD((k-1)*length(QoS)+1,
1:slots*T-U(1)))/(slots*T-U(1));

Delay_avg_Voice_MLWDF(j)=Delay_avg_Voice_ML
WDF(j)+sum(delay_MLWDF((k-1)*length(QoS)+1,
1:slots*T-U(1)))/(slots*T-U(1));
```

```
Delay_avg_Video(j)=Delay_avg_Video(j)+sum(delay((k-1)*length(QoS)+2, 1:slots*T-U(2)))/(slots*T-U(2));

Delay_avg_Video_PD(j)=Delay_avg_Video_PD(j)+sum(delay_PD((k-1)*length(QoS)+2,
1:slots*T-U(2)))/(slots*T-U(2));

Delay_avg_Video_MLWDF(j)=Delay_avg_Video_MLWDF(j)+sum(delay_MLWDF((k-1)*length(QoS)+2,
1:slots*T-U(2)))/(slots*T-U(2));
            end

            Delay_avg_Voice(j)=Delay_avg_Voice(j)/K;

Delay_avg_Voice_PD(j)=Delay_avg_Voice_PD(j)/K;

Delay_avg_Voice_MLWDF(j)=Delay_avg_Voice_MLWDF(j)/K;


Delay_avg_Video(j)=Delay_avg_Video(j)/K;

Delay_avg_Video_PD(j)=Delay_avg_Video_PD(j)/K;

Delay_avg_Video_MLWDF(j)=Delay_avg_Video_MLWDF(j)/K;
      end
end

Capacity=Capacity_Voice+Capacity_Video+Capacity_BE;
Capacity_PD=Capacity_Voice_PD+Capacity_Video_PD+Capacity_BE_PD;
Capacity_MLWDF=Capacity_Voice_MLWDF+Capacity_Video_MLWDF+Capacity_BE_MLWDF;

figure;
plot(K_,Capacity/N_Loop/B/slots/(T*1e-3),'g*-');
hold on;
plot(K_,Capacity_PD/N_Loop/B/slots/(T*1e-3),'r^-');
hold on;
plot(K_,Capacity_MLWDF/N_Loop/B/slots/(T*1e-3),'o-');
```

```
grid on;
legend('New','PD','M-LWDF');
xlabel('The Number of Users')
ylabel('System Throughput,bits/s/Hz')

figure;
plot(K_,Capacity_BE/N_Loop/B/slots/(T*1e-3),'g*-');
hold on;
plot(K_,Capacity_BE_PD/N_Loop/B/slots/(T*1e-3),'r^-');
hold on;
plot(K_,Capacity_BE_MLWDF/N_Loop/B/slots/(T*1e-3),'o-');
grid on
legend('New','PD','MLWDF');
xlabel('The Number of Users')
ylabel('Throughput for BE traffic,bits/s/Hz')

figure;
plot(K_,Capacity_Voice/N_Loop/B/slots/(T*1e-3),'g^-');
hold on;
plot(K_,Capacity_Video/N_Loop/B/slots/(T*1e-3),'go-');
hold on;
plot(K_,Capacity_Voice_PD/N_Loop/B/slots/(T*1e-3),'r^-');
hold on;
plot(K_,Capacity_Video_PD/N_Loop/B/slots/(T*1e-3),'ro-');
hold on;
plot(K_,Capacity_Voice_MLWDF/N_Loop/B/slots/(T*1e-3),'^-');
hold on;
plot(K_,Capacity_Video_MLWDF/N_Loop/B/slots/(T*1e-3),'o-');
grid on
legend('New-Voice','New-Video','PD-Voice','PD-Video','MLWDF-Voice','MLWDF-Video');
xlabel('The Number of Users')
ylabel('Throughput,bits/s/Hz')

figure;
plot(K_,Delay_avg_Voice/N_Loop, 'g*-');
```

```
hold on;
plot(K_,Delay_avg_Voice_PD/N_Loop, 'r^-');
hold on;
plot(K_,Delay_avg_Voice_MLWDF/N_Loop, 'o-');
grid on;
legend('New-Voice','PD-Voice','M-LWDF-Voice');
xlabel('The Number of Users');
ylabel('Average Packet Delay,ms');

figure;
```

```
plot(K_,Delay_avg_Video/N_Loop, 'g*-');
hold on;
plot(K_,Delay_avg_Video_PD/N_Loop, 'r^-');
hold on;
plot(K_,Delay_avg_Video_MLWDF/N_Loop, 'o-');
grid on;
legend('New-Video','PD-Video','M-LWDF-Video');
xlabel('The Number of Users');
ylabel('Average Packet Delay ,ms');
```

```
%% Data and Delaytime Generation of 3 queues (Voice, VBR video and BE) for Specific users (size=(K*3)*units)
function
[data,delay]=Data_Delay_Generation(K,QoS,units,U)
Rate_Voice=64;
Rate_Video_max=420;Rate_Video_min=120;Rate_Vi
deo_mean=239;
Rate_BE=500;
%% Data Generation
for k=1:K
    %% Voice (64kbps)
    data(length(QoS)*(k-1)+1, :) = ones(1,units)*
Rate_Voice;
    %% VBR video traffic
(120kbps-239kbps-420-kbps)
    left=units;
    ptr=1;
    while (left>0)
        %duration of each station with exp
distribution, 160ms mean time
        tmp=fix(exprnd(160)); % units distribution
e.g units=slots
        rate=fix(exprnd(Rate_Video_mean)); % rate
distribution
        while
((rate>Rate_Video_max)||(rate<Rate_Video_min))
            rate=fix(exprnd(Rate_Video_mean));
        end
end
```

```
        if tmp<left % distributerd units is smaller
than the left
            data(length(QoS)*(k-1)+2,
ptr:ptr+tmp-1)= ones(1,tmp)*rate;
            left=left-tmp;
            ptr=ptr+tmp;
        else
            data(length(QoS)*(k-1)+2,
ptr:ptr+left-1)=ones(1,left)*rate;
            ptr=ptr+left;
            left=0;
        end
    end
    %% BE (500kbps)
    data(length(QoS)*(k-1)+3,:) =
ones(1,units)*Rate_BE;
end
%% Delay Generation
for k=1:K
    delay(length(QoS)*(k-1)+1,:) = ones(1,units)*
U(1);
    delay(length(QoS)*(k-1)+2,:) = ones(1,units)*
U(2);
    delay(length(QoS)*(k-1)+3,:) = ones(1,units)*
U(3);
```

```
%% Packet drop rate (MWC+PD & MMWC+MPD & MWC+MLWDF)
N_Loop=1;                % Simulation times
N=512;                   % Number of Subcarriers
B=5e6;                   % Bandwidth
T=2;slots=2000;   % number of total slot in one
simulation
Pt=1;                    % Total Power
SNR_dB=20;No=Pt*(T*1e-3/N)./10.^(SNR_dB/10);   %
AWGN Noise Density of 20dB SNR
K_=[ 54];                % User Number Array - Bata
QoS=[1024 512 1];                          % QoS
priority level for Voice, VBR video and BE traffic (3
queues)
U = [100 400 1000];        % Delay tolerance for 3
queues - (slots)

for j=1:length(K_)
    K=K_(j)
    gamma=ones(K,1);
    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%%%%%%%%%%%    Initialization

capacity_PD=zeros(K*length(QoS),length(K_));capaci
ty_MLWDF=zeros(K*length(QoS),length(K_));capaci
ty=zeros(K*length(QoS),length(K_));
    delay_PD=zeros(K*length(QoS), slots*T);
delay_MLWDF=zeros(K*length(QoS),
slots*T);delay=zeros(K*length(QoS), slots*T);
    data_PD=zeros(K*length(QoS), slots*T);
data_MLWDF=zeros(K*length(QoS), slots*T);
data=zeros(K*length(QoS), slots*T);
    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%%%%%%%%%%%    Initialization
    for n_loop=1:N_Loop
        Data_PD=zeros(K, slots);
Data_MLWDF=zeros(K*length(QoS), slots);
Data=zeros(K, slots);
        weight_PD=zeros(K,slots);
weight_MLWDF=zeros(K*length(QoS),slots);weight=
zeros(K,slots);
        Cap_avg=ones(K*length(QoS),1);
        %% Data(Packet size) and delay generation
of 3 queues with specific #. users
[data_PD,delay_PD]=Data_Delay_Generation(K,QoS,
T*slots,U);

[data,delay]=Data_Delay_Generation(K,QoS,T*slots,
U);

[data_MLWDF,delay_MLWDF]=Data_Delay_Genera
tion(K,QoS,T*slots,U);
        for currentslot=1:slots
            %%%%%%%%%%%%%%%%%%%%%%%%%
%%%%%%%%%%%%%%%%%    Initialization

shannoncapacity_PD=zeros(K,1);shannoncapacity_M
LWDF =
zeros(K*length(QoS),1);shannoncapacity=zeros(K,1);
            %%%%%%%%%%%%%%%%%%%%%%%%%
%%%%%%%%%%%%%%%%%    Initialization
            %% Scheduling-> Weight Caculation
            [w_Voice_PD,w_Video_PD,w_BE_PD,
Data_PD,
rcdslot_PD]=scheduling_PD(data_PD,currentslot,U,K,
QoS,slots,T);
            [w_Voice,w_Video,w_BE, Data,
rcdslot]=scheduling_PD(data,currentslot,U,K,QoS,slot
s,T);
            [Data_MLWDF,
weight_MLWDF]=scheduling_MLWDF(data_MLWD
F,U,Data_MLWDF,K,currentslot, Cap_avg,
weight_MLWDF,T);
            for k=1:K

weight_PD(k,currentslot)=sum(w_Voice_PD(k,:))+su
m(w_Video_PD(k,:))+sum(w_BE_PD(k,:));

weight(k,currentslot)=sum(w_Voice(k,:))+sum(w_Vid
eo(k,:))+sum(w_BE(k,:));
            end
            %% Generate channel information
            CSI=channel(K,N);
chh=abs(CSI).^2;noise=No*B/N; CNR=chh/noise;
            %% Subcaarier Allocation

[Index_CNR_PD,Index_PD,sumweight_PD]=subcarri
```

```
erallocation_MWC(Pt,B,N,CNR,K,currentslot,Data_PD,weight_PD,T);

[Index_CNR,Index,sumweight]=subcarrierallocation_M(Pt,B,N,CNR,K,currentslot,Data_PD,weight,T);

[Index_CNR_MLWDF,Index_MLWDF,sumweight_MLWDF]=subcarrierallocation_MLWDF(Pt,B,N,CNR,K,currentslot,Data_MLWDF,weight_MLWDF,T,QoS);
            %% Power Allocation

PA_PD=powerallocation(N,Pt,weight_PD,Index_CNR_PD,Index_PD,sumweight_PD,currentslot);

PA=powerallocation(N,Pt,weight,Index_CNR,Index,sumweight,currentslot);

PA_MLWDF=powerallocation(N,Pt,weight_MLWDF,Index_CNR_MLWDF,Index_MLWDF,sumweight_MLWDF,currentslot);
            %% Shannon Capacity of each user (how many bits at each subcarrier)
            for n=1:N
                user_PD=find(Index_PD(:,n));
                user=find(Index(:,n));

queue_MLWDF=find(Index_MLWDF(:,n));

shannoncapacity_PD(user_PD)=shannoncapacity_PD(user_PD)+(B/N)*log2(1+PA_PD(n)*Index_CNR_PD(n));

shannoncapacity(user)=shannoncapacity(user)+(B/N)*log2(1+PA(n)*Index_CNR(n));

shannoncapacity_MLWDF(queue_MLWDF)=shannoncapacity_MLWDF(queue_MLWDF)+(B/N)*log2(1+PA_MLWDF(n)*Index_CNR_MLWDF(n));
            end
            %% Real Data Caculation
            % PD
            for k=1:K
                if Data_PD(k, currentslot)>(shannoncapacity_PD(k)*(T*1e-3))

[data_PD,delay_PD,capacity_PD]=realdata_shannon(currentslot, k, QoS, U, shannoncapacity_PD(k)*(T*1e-3), data_PD, w_Voice_PD, w_Video_PD, w_BE_PD, delay_PD,capacity_PD, j,T);
                    Data_PD(k, currentslot) = Data_PD(k, currentslot) - shannoncapacity_PD(k)*(T*1e-3);
                else

[data_PD,delay_PD,capacity_PD]=realdata_all(currentslot, k, QoS, U, data_PD, delay_PD, rcdslot_PD, capacity_PD, j,T);
                    Data_PD(k, currentslot) =0;
                end
            end
            % New
            for k=1:K
                if Data(k, currentslot)>(shannoncapacity(k)*(T*1e-3))

[data,delay,capacity]=realdata_shannon(currentslot, k, QoS, U, shannoncapacity(k)*(T*1e-3), data, w_Voice, w_Video, w_BE, delay,capacity, j,T);
                    Data(k, currentslot) = Data(k, currentslot) - shannoncapacity(k)*(T*1e-3);
                else

[data,delay,capacity]=realdata_all(currentslot, k, QoS, U, data, delay, rcdslot, capacity, j,T);
                    Data(k, currentslot) =0;
                end
            end
            % M-LWDF
            for q=1:(K*length(QoS))
                if Data_MLWDF(q, currentslot)>(shannoncapacity_MLWDF(q)*(T*1e-3))

[data_MLWDF,delay_MLWDF,capacity_MLWDF]=realdata_shannon_MLWDF( currentslot,q,U,shannoncapacity_MLWDF(q)*(T*1e-3),data_MLWDF,delay_MLWDF,capacity_MLWDF,j,T);
```

```
Cap_avg(q)=0.9*Cap_avg(q)+0.1*shannoncapacity_MLWDF(q)*(T*1e-3);
                else
[data_MLWDF,delay_MLWDF,capacity_MLWDF]=realdata_all_MLWDF(currentslot,q,U,data_MLWDF,delay_MLWDF,capacity_MLWDF,j,T);
Cap_avg(q)=0.9*Cap_avg(q)+0.1*Data_MLWDF(q,currentslot);
                end
            end
        end % end all the slots

        for i=1:2
p_PD=zeros(1,U(i));p_MLWDF=zeros(1,U(i));p=zeros(1,U(i));
            for t=1:U(i)
                for k=1:K_
B1=size(find(delay_PD((k-1)*length(QoS)+i,:)>t));
                    C1=B1(1,2);
p_PD(t)=p_PD(t)+C1/(T*slots);
B2=size(find(delay_MLWDF((k-1)*length(QoS)+i,:)>t));
                    C2=B2(1,2);
p_MLWDF(t)=p_MLWDF(t)+C2/(T*slots);
B3=size(find(delay((k-1)*length(QoS)+i,:)>t));
                    C3=B3(1,2);
                    p(t)=p(t)+C3/(T*slots);
                end
                p_PD(t)= p_PD(t)/K_;
                p_MLWDF(t)= p_MLWDF(t)/K_;
                 p(t)= p(t)/K_;
            end
            figure;
            plot([1:U(i)],p, 'g*-','LineWidth',2);
            hold on;
            plot([1:U(i)],p_PD, 'b-','LineWidth',2);
            hold on;
            plot([1:U(i)],p_MLWDF,
'r.-','LineWidth',2);
            hold on;
            grid on;
            if i==1
                legend('Modified,voice','PD,
voice','MLWDF, voice');
            else
                legend('Modified,video','PD,
video','MLWDF, video');
            end
            xlabel('t(ms)');
            ylabel('Delay outage probability');
        end
        end

       end
```