# Packet Scheduling for AMC-Based OFDMA Wireless Communication Systems

by

Bai Zhang

A thesis submitted to the Faulty of Graduate Studies

Lakehead University

in partial fulfillment of the requirements for the degree of

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### Abstract

In this thesis, we propose a basic priority and fairness (BPF) and a modified priority and fairness (MPF) based packet scheduling algorithms for adaptive modulation and coding (AMC) scheme based orthogonal frequency division multiple access (OFDMA) wireless communication systems supporting both real time (RT) and non-real time (NRT) traffics. In the AMC based OFDMA systems considered in this thesis, multiple access is achieved by assigning subchannels, each of which consists of a set of subcarriers, to individual users, and the modulation and coding scheme on each subchannel is determined adaptively according to the time-varying channel conditions. In the proposed packet scheduling algorithms, various traffics are transmitted in a sequence determined by their priorities on each subchannel, jointly considering the fairness among users and subchannels. Simulation results show that the proposed algorithms are able to satisfy various quality of service (QoS) requirements, like packet loss rate, delay, or throughput, for a variety of RT and NRT traffics such as voice, video, WWW and email. Also, the proposed BPF and MPF packet scheduling algorithms can support more users than some existing packet scheduling algorithms under the same QoS requirements.

*Index Terms*---Orthogonal frequency division multiple access (OFDMA), Orthogonal frequency division multiplexing (OFDM), adaptive modulation and coding (AMC), packet scheduling, quality of service (QoS).

Thesis Supervisor: Zhiwei Mao

Title: Associate Professor, Department of Electrical Engineering

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# Chapter 1

# Introduction

Orthogonal frequency division multiple access (OFDMA) system is a strong candidate for the next generation wireless communication systems. OFDMA is a multiple access scheme based on orthogonal frequency division multiplexing (OFDM). In an OFDMA system, the broad frequency band is divided into a set of narrowband subchannels, each of which consists of a set of subcarriers [1]. Thus, it allows multiple users to transmit their data simultaneously on different subchannels. Adaptive modulation and coding (AMC) scheme can be employed in OFDMA systems to maximize the system throughput by adapting the modulation and coding scheme on each subchannel according to the time-varying channel conditions of different users [2].

The AMC-based OFDMA wireless communication system is expected to provide a broad range of multimedia services such as real time (RT) voice, RT video, non-real time (NRT) WWW, NRT Email, etc. to wireless users with different quality of service (QoS) requirements, like packet delay and loss rate requirements for the RT traffics and throughput requirement for the NRT traffics [2]. Packet scheduling plays an important role in providing QoS support to multimedia communications in various wireless communication networks. Therefore, designing an effective packet scheduling scheme for AMC-based OFDMA wireless communication systems becomes essential to meet the QoS requirements for individual service while increasing the system capacity. In this thesis, we propose a priority and fairness based wireless downlink packet scheduling algorithms for AMC-based OFDMA wireless communication systems supporting both RT and NRT traffics. The proposed packet scheduling algorithm is based on the urgency factor of the packet, the efficiency factor of the subchannel, and the fairness factors among users and subchannels at each scheduling instant [1]. In the proposed algorithm, the urgency factor of a packet derives from the value of the time-utility function associated with the packet and the efficiency factor of a subchannel is defined as the ratio of the current subchannel state to the average subchannel state. Thus, the priority of a user on a subchannel is determined by the product of the urgency factor of the user's head-of-line (HOL) packet and the efficiency factor of the subchannel to the user. In addition, the fairness factor of a subchannel and fairness factor of a user are introduced to deal with some special situations, such as two or more users having the same priority on the same subchannel or one user having the same priority on two or more subchannels. The main objective of the proposed algorithm is to preferentially transmit the packets from the user with high priority jointly taking care of fairness among users and subchannels.

However, according to the proposed priority and fairness based packet scheduling algorithm, each subchannel is allocated to only one user during one frame interval, which consists of 20 timeslots in this thesis. It is noted that some users having less packets to transmit may not occupy all timeslots in one frame on one subchannel, which results in waste of capacity of the subchannel. In order to avoid this kind of waste, we also revise the proposed priority and fairness based packet scheduling algorithm, which allows for sharing the same subchannel among different users during one frame interval and is referred to as modified priority and fairness (MPF) based packet scheduling algorithm. For the clarity of discussion, the original algorithm will be referred to as the basic priority and fairness (BPF) based packet scheduling algorithm in the following. According to MPF packet scheduling algorithm, the unoccupied timeslots on one subchannel, which is allocated to one user initially, may be reallocated to other users. Simulation results show that our proposed BPF and MPF packet scheduling algorithms can increase the number of users satisfying underlying QoS requirements, as compared to M-LWDF packet scheduling algorithm.

## 1.1 Review of Previous Work

In [1], an urgency and efficiency based packet scheduling algorithm is studied to support both RT and NRT traffic transmissions. The author presents a specific OFDMA system model and four types of traffic models such as RT voice, RT video, NRT WWW and NRT Email traffics. The value of the time-utility function, in which the jitter concept is adopted to represent the timeliness requirements of RT traffics, is used to represent the urgency factor of the HOL packet. The jitter concept is adopted to schedule RT traffics according to its timeliness requirement. The ratio of the current channel state to the average channel state is used to determine the efficiency factor. The incoming packets are scheduled to transmit based on their priority, which is determined by both the urgency factor and efficiency factor. The packet with the highest priority will be scheduled to be transmitted prior to others. The objective of the scheduling algorithm is to maximize the throughput of NRT traffics while the QoS requirements of RT traffics are satisfied. However, there might be more than two users who have the same value of priority on the same subchannel, or the priorities of a user on different subchannels are of the same values. These situations are not considered in the scheduling algorithm in [1] and will affect the performance of the scheduling algorithm.

In [2], the subchannel multiplexing scheme is proposed in AMC-based OFDMA system. The AMC scheme determines the capacity of each subchannel based on the time varying channel condition. The modulation and coding scheme on each subchannel is assigned by the base station according to the received signal-to-noise (SNR) ratio. Packet loss rate-based scheduling algorithm is proposed for RT traffics transmission. The priority of the HOL packet is determined by the HOL packet delay, the channel efficiency, and ratio of the maximum packet loss rate requirement to the current packet loss rate. In addition, the modified minimum bit rate-based packet scheduling algorithm is proposed for NRT traffic transmission. The priority of the algorithm is determined by the channel efficiency and the ratio of the required minimum data rate to the average one. For both RT and NRT traffics, the packet with the highest priority value is transmitted prior to others. In this paper, the author presents a practical OFDMA system model, and various RT and NRT traffics, which are also studied and adopted in this thesis.

In [5], a novel channel allocation and scheduling algorithm is studied. The current channel state is considered as the priority of the algorithm. The packet is scheduled to transmit based on the priority jointly taking care of the alternative fairness factors of channels and users. The channel efficiency is not considered in the algorithm, which may result in waste of wireless resource. Moreover, if a user is in a shadowed area for a long time, then the user will not receive any service according to this algorithm, which is unfair to that user.

In [7], a largest weighted delay first (LWDF) packet scheduling algorithm is presented in detail. The LWDF packet scheduling algorithm is broadly studied and applied in various communication networks. The LWDF algorithm is mostly designed to support single access communication system. It always schedules the longest waiting user, who has the largest weighted delay for transmission. In the LWDF packet scheduling algorithm, the priority of user $j$ at each scheduling instant $t$ is given by

$$P_j(t) = a_j W_j(t) \tag{1.1}$$

where $W_j(t)$ is the HOL packet delay of user $j$ at time $t$, $a_j = -\log[\delta_j / \tau_j]$, with $\tau_j$ being the maximum allowable delay of user $j$ and $\delta_j$ being the maximum acceptable probability of $W_j(t)$ exceeding $\tau_j$. In the LWDF algorithm, the user with the highest value of priority is scheduled to transmit earlier than others. It does not consider the channel condition. The selected user may have bad condition on the channel resulting in decreased channel throughput.

In [8] and [9], a modified largest weighted delay first (M-LWDF) packet scheduling algorithm is studied. The M-LWDF packet scheduling algorithm is a typical packet scheduling algorithm used to support multiple access communication systems and is currently broadly applied in wireless communication systems. The M-LWDF scheduling algorithm was proposed to support RT and NRT traffics simultaneously. The main idea of the M-LWDF scheduling algorithm is to take the maximum delay requirement of each user into account and to jointly consider the channel efficiency. In the M-LWDF scheduling algorithm, the priority of user $j$ on the $i$th subchannel at each scheduling instant $t$ is given by

$$P_{ij}(t) = a_j W_j(t) R_{ij}(t) / \bar{R}_{ij}(t) \tag{1.2}$$

where $W_j(t)$ is the HOL packet delay of user $j$ at time $t$, $R_{ij}(t)$ is the current capacity of user $j$ on the $i$th subchannel, $\bar{R}_{ij}(t)$ is the average value of $R_{ij}(t)$, $a_j = -\log[\delta_j / \tau_j]$ with $\tau_j$ and $\delta_j$ defined as in (1.1). In the M-LWDF algorithm, the user with the highest priority value is scheduled to transmit prior to others. However, the fairness among the users and the subchannels is not considered in the algorithm, which will affect the performance of the system.

In [12], an overview of fundamental real-time concepts and terms is introduced and explained. The deadline approach and the time-utility function are presented to study real-time environment. The soft-deadlined and hard-deadlined utility functions are described particularly. The timeliness concept of RT is also included to study various real-time systems. We adopt the time utility function and deadline approach to determine the urgency factor of the HOL packet, which is considered as one of the two factors to determine the priority of the HOL packet in this thesis.

This thesis is organized as follows. Chapter 2 presents fundamentals of OFDMA wireless communication systems. In Chapter 3, a model of OFDMA wireless systems is described. The proposed BPF and MPF packet scheduling algorithms designed to satisfy the QoS requirements of diverse traffics are introduced in Chapter 4. In Chapter 5, simulation results are presented and discussed. Conclusion is drawn and possible future studies are discussed in Chapter 6.

# Chapter 2

# Fundamentals of OFDMA Wireless Communication Systems

## 2.1 Wireless Multipath Channel

In wireless communication systems, multipath propagation is a common phenomenon. When a signal is transmitted through a wireless channel, there might be multiple paths for a signal to travel from the transmitter to the receiver. The signals arrived at the receiver might contain not only the direct line-of-sight (LOS) radio wave, but also reflected, refracted, or scattered radio waves [3], due to the ground and objects between a transmitter and a receiver such as buildings, vehicles, mountains, plants, etc. The concept of wireless multipath channel is illustrated in Figure 2.1.

The components of all paths superpose at the receiver, which cause inter-symbol interference (ISI) and significant performance degradation if no overcoming measure is adopted. Equalizers are often used to overcome ISI, but it usually has high implementation complexity. OFDM is being used more and more to deal with the multipath problem.

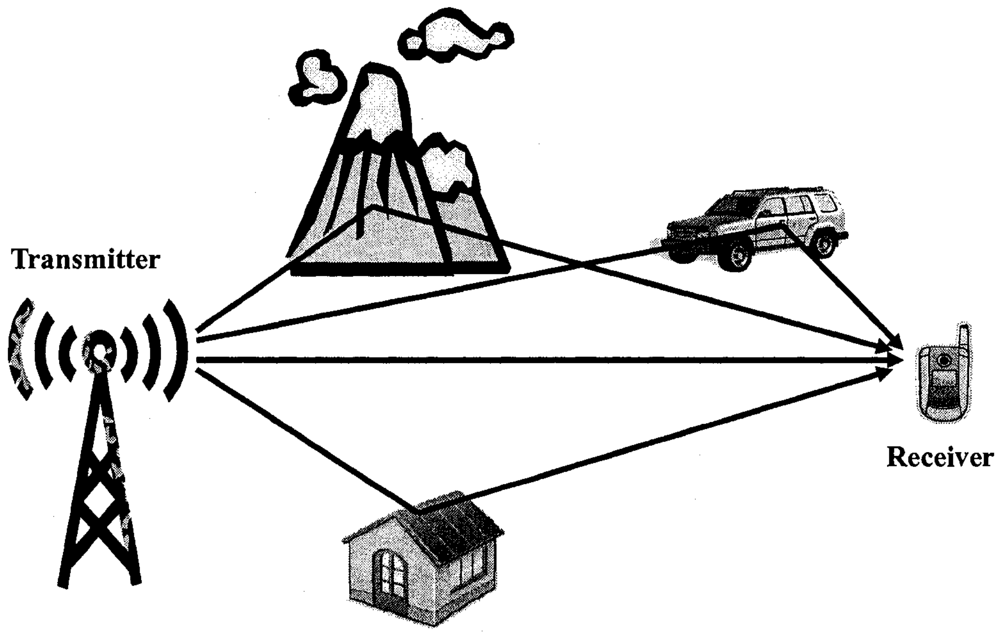



Figure 2.1 Concept of wireless multipath channel

## 2.2 Digital Modulation and Coding

Nowadays, digital modulation is extensively adopted in communication systems. Compared with analog modulation technique, digital modulation has many advantages, including higher data rate, more powerful error correction capability, greater noise immunity and resistance to channel impairments, more efficient multiple access strategies, better security and privacy, etc.

In digital communication systems, the modulating signal can be described as a time sequence of symbols or pulses, where each symbol has $M$ possible finite states. Therefore, each symbol represents $N$ bits of information, where $N = \log_2 M \quad bits/symbol$.

There are two types of digital modulation techniques, which are linear and non-linear modulations. Due to the simplicity, linear modulation methods are broadly used in wireless communication systems. There are three basic types of linear modulation schemes, which are listed as follows.

- Pulse amplitude modulation (PAM), where information is represented by amplitudes of modulated signals only.
- Phase shift keying (PSK), where information is represented by phases of modulated signals only.
- Quadrature amplitude modulation (QAM), where information is represented by both amplitudes and phases of modulated signals.

Therefore, the linearly modulated signal $s(t)$ can be expressed as

$$\begin{aligned} s(t) &= \mathrm{Re}[m(t)\exp(j2\pi f_c t)] \\ &= m_R(t)\cos(2\pi f_c t) - m_I \sin(2\pi f_c t) \end{aligned} \tag{1.1}$$

where $f_c$ denotes the carrier frequency, $m(t) = m_R(t) + jm_I(t)$ denotes the modulating signal in complex form.

In digital communication systems, error control coding is often included. The objective of error control coding is to introduce some controlled redundancy in the transmitted information sequence that can be used at the receiver to detect or/and correct the errors, which are introduced by various noises and interference during transmission through the channel. Adopting error control coding can keep the probability of transmission error at much lower level. Typically, coding rate is expressed as a fractional number form, $k/n$, which means, for every $k$ information bits, the encoder appends $n-k$ redundant bits to generate total $n$ bits of transmitted data. In Figure 2.2, the basic structure of a digital communication system is shown.

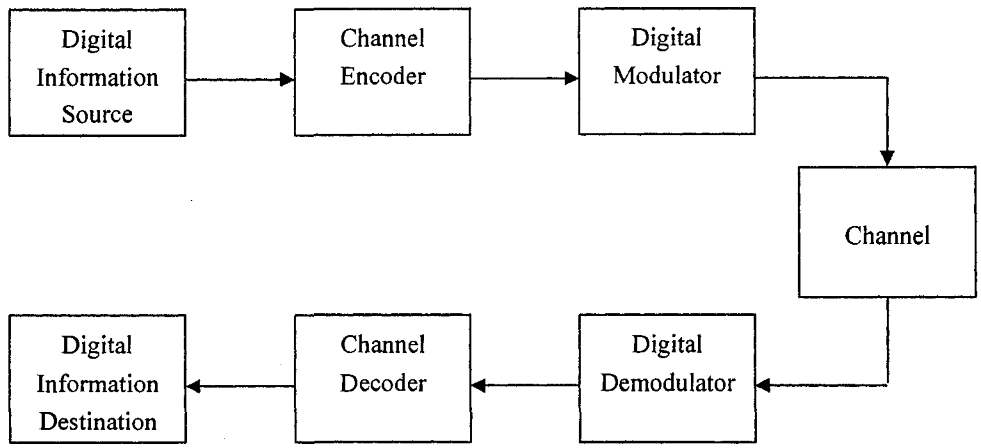



Figure 2.2 Basic structure of a digital communication system

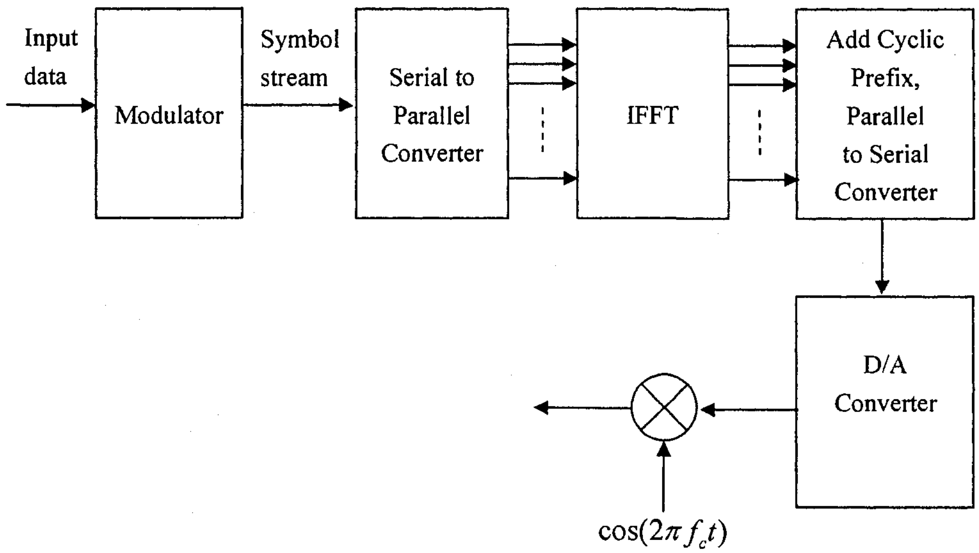
## 2.3 OFDM

It has been more than 40 years since the modulation technique of OFDM was first brought forward. This modulation technology has been becoming a popular scheme for many existing and future wideband digital communication systems, whether wireless or over wirelines, such as asymmetric digital subscriber line (ADSL) broadband internet access systems, digital video and audio broadcasting systems, IEEE Wi-Fi systems, IEEE WiMAX systems, etc.

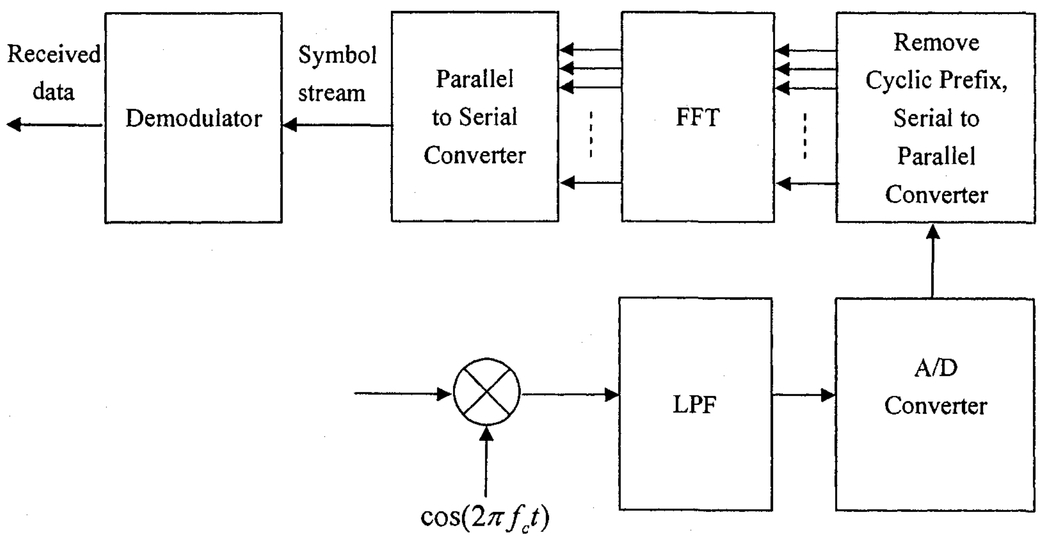
OFDM is a digital transmission technique that uses a large number of orthogonal narrowband subcarriers instead of a single wideband carrier to transmit data. OFDM modulation is essentially the simultaneous transmission over a large number of narrowband subcarriers, each of which transmits a signal at a lower data rate, and all of which are aggregated at receiving side into the original high speed signal. Each narrowband subcarrier can be modulated with various conventional modulation schemes such as PSK or QAM. The frequency spacing of the subcarriers is chosen in such a way that although the subcarriers' spectra overlap in frequency domain, the modulated subcarriers are orthogonal and do not interfere with each other, thus it enables the receiver to detect the signals in environments with multipath and other interference more easily. The block diagram of an OFDM system is shown in Figure 2.3.

At the transmitter side, the input serial high speed data stream is modulated by conventional modulation scheme and then divided into parallel data substreams for transmission by assigning each data substream to one subcarrier. An inverse fast Fourier transform (IFFT) is then used to find the corresponding time domain waveform. The cyclic prefix is appended to the beginning of each symbol to overcome the remaining ISI. After that, the symbols are then converted back to a serial format, which is the baseband signal for OFDM transmission. At the receiver side, after the OFDM baseband signal is recovered, the cyclic prefix is removed. A fast Fourier transform (FFT) is then used to recover the original transmitted data in parallel. Finally, the parallel data substreams are aggregated into serial data stream and demodulated to recover the original high speed information data stream.

Compared with single-carrier modulation schemes, OFDM can cope with severe channel conditions including narrowband interference, attenuation of high frequencies in a long copper wire, and frequency-selective fading due to multipath environments. OFDM has an advantage over single carrier modulation in combating narrowband frequency interference, since this kind of interference will only affect some of the frequency subcarriers, and the other subcarriers will not be affected by the interference. The ability of OFDM to deal with multipath frequency-selective fading is due to the natural character of OFDM modulation scheme. Since the transmission rate on each subcarrier of OFDM signals is much lower compared with that of a single-carrier modulated signal, the digital data symbol period is longer and thus each subcarrier experiences flat fading in multipath environments and is relatively simple to equalize.




(a) Transmitter




(b) Receiver

Figure 2.3 Block diagram of an OFDM system

Another benefit of OFDM is high spectral efficiency. In OFDM, the subcarriers' overlap in frequency domain and the system spectral efficiency is improved. The individual signals on subcarriers can be recovered at receiver so far as the orthogonal relations among subcarriers are all along preserved during transmission even though they overlap in frequency domain.

## 2.4 OFDMA

Since OFDMA technology was initially introduced by Runcom Technologies Ltd. in 2000, it has been used in IEEE 802.16a/d/e standards and has been adopted as a candidate access method for IEEE 802.22 Wireless Regional Area Networks (WRAN) standard [4].

OFDMA is a multi-user version of OFDM modulation scheme that allows multiple access to the same wideband channel, which is divided into narrowband subchannels. Each subchannel consists of a group of subcarriers. In OFDMA systems, each subscriber station or user can be considered independently and subsets of subcarriers are assigned to individual users so that multiple users can transmit and receive at the same time. According to the feedback information about the channel conditions, the system can adaptively assign the subcarriers to multiple users, and the modulation and coding schemes on each subcarrier can be adapted to provide improved coverage and throughput.

OFDMA is a scheme that combines OFDM modulation technique and frequency division multiple access (FDMA) technologies. Moreover, we can also regard OFDMA as a combination of frequency domain and time domain multiple access. From this point of view, channel resource is partitioned in the time-frequency space, and is assigned along the OFDM symbol index as well as OFDM sub-carrier index.

Therefore, OFDMA inherits the advantages of OFDM. In an OFDMA system, users may experience independent fading due to the multipath environment. Thus, for a specific subcarrier or subchannel, if a user is in deep fading, the other users may not be in deep fading at the same time and this subcarrier or subchannel may be allocated to the users with the best channel condition. Thus, the most remarkable feature of OFDMA is its multiuser diversity which enormously improves the system throughput.

## 2.5 AMC Scheme in OFDMA System

Most OFDM systems use a fixed modulation and coding scheme over all subcarriers for simplification. By adopting a fixed modulation and coding scheme, the subcarrier modulation and coding scheme must be assigned in advance to guarantee a prescribed bit-error rate (BER) under the worst channel conditions. This results that most systems have to adopt binary PSK or quadrature PSK, which provides a poor spectral efficiency (1 or $2 bits/s/Hz$).

However, each subcarrier can adopt an adaptive modulation and coding scheme based on the current received signal quality or channel condition. Any coherent or differential, phase or amplitude modulation scheme like BPSK, QPSK, 8PSK, 16QAM, 64QAM, etc., can be employed. The spectral efficiency can be maximized by adapting the modulation and coding scheme with the highest information transmission rate under an acceptable BER. Adopting adaptive modulation and coding scheme, the subscribers can transmit at a much higher information rate when the radio channel is in good condition. Thus, generally, as a subscriber station is close to the base station (BS), the modulation can be increased from 1 $bits/s/Hz$ (BPSK) up to 4-6 $bits/s/Hz$ (16QAM - 64QAM), and the spectral efficiency of the overall system can be increased significantly.

Therefore, the main idea of AMC is to periodically change modulation and coding scheme (MCS) based on the channel condition in succeeding frames to improve the system spectral efficiency. According to the observed channel condition, the decision about selecting appropriate MCS is executed at receiver side, and the decision is fedback to the transmitter via a control channel.

AMC scheme is generally employed in the OFDMA wireless communication system to improve system performance. In an AMC-based OFDMA wireless communication system, the subscriber stations close to the BS are in general assigned higher order modulation and higher code rates (e.g. 64 QAM and coding rate $R = 3/4$), and the modulation order and coding rate will be generally decreased as the distance of the subscriber station from BS increases and the channel condition becomes worse.

The concept of an AMC scheme is illustrated in Figure 2.4

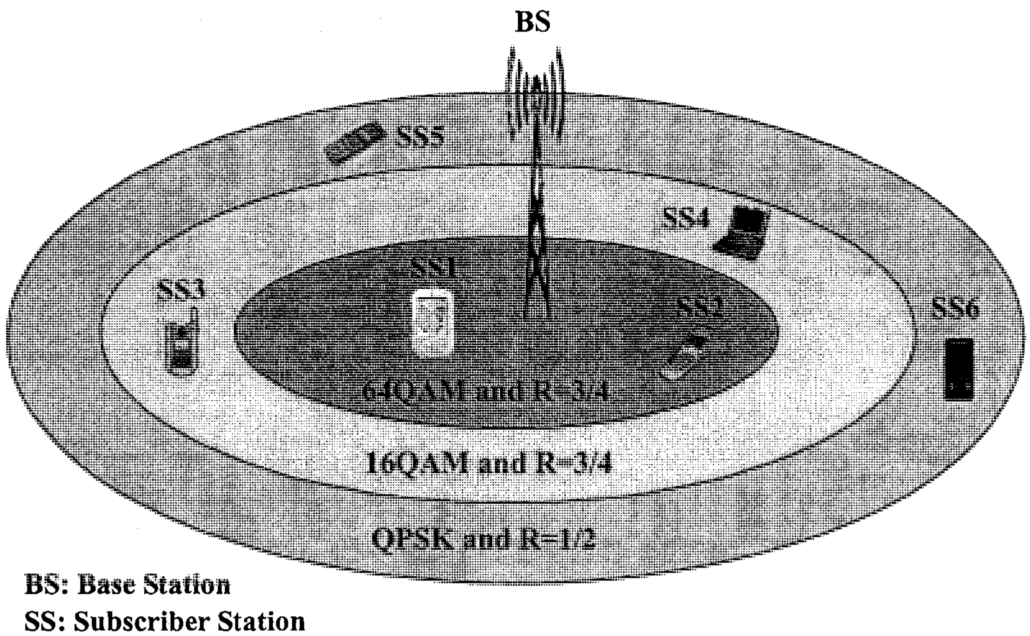



Figure 2.4 Concept of AMC scheme

## 2.6 Packet Scheduling in OFDMA System

Packet scheduling has been playing a very important role in providing QoS support to multimedia services in various wired and wireless communication systems.

In OFDMA wireless communication systems, packet scheduling is very important due to resource sharing, i.e., multiple users need to compete for a common channel at the same time. The communication systems need to support both RT and NRT. They also need to provide QoS guarantees for all users. Typically, in wireless communication systems, the BSs are responsible for packet scheduling in both uplink and downlink transmissions. The packet scheduler manages queues of service requests, allocates resources to individual users, and decides the order of packet transmission for different users. An ideal packet scheduling scheme should be easy to implement, be fair to individual users, and provide good performance.

# Chapter 3

# System Architecture

## 3.1 Basic model of OFDMA wireless system

In this thesis, we consider an OFDMA wireless communication system which consists of a BS and multiple subscriber stations or users. For the convenience of description, we only focus on downlink transmission. We consider an OFDMA wireless communication system with a bandwidth of 20MHz. The frequency channel is divided into 1536 subcarriers, and these subcarriers are grouped into 12 subchannels. Thus, each subchannel consists of a set of 128 subcarriers. All subcarriers are shared among all users in a cell in terms of subchannels. One OFDM symbol has a duration of 100 μs. For the purpose of simplification, we assume that all subcarriers are used for data transmission. The subcarriers in each subchannel are selected by a pre-specified pattern.

In opposite to typical OFDM scheme in which all subcarriers are allocated to a single user in each frame period, OFDMA allows different users to share the same frame by dividing the subcarriers into subchannels, each of which is allocated to one or more users.

The modulation and coding mode of any subchannel is specified according to the subchannel condition for the corresponding user that the subchannel has been allocated to. The representative modulation schemes, which include BPSK, QPSK, 16QAM and 64QAM, are adopted in our discussions according to the instantaneous signal-to-noise ratio (SNR) of each subchannel. The primary parameters of the OFDMA wireless system that we consider in this thesis are summarized in Table 3.1 [1] [2].

Table 3.1 Summary of OFDMA system

| Parameter | Value |
|---|---|
| System | OFDMA |
| Downlink channel bandwidth | 20MHz |
| OFDM symbol duration | 100μs |
| Total number of subcarriers | 1536 |
| Number of subcarriers per subchannel | 128 |
| Number of subchannels | 12 |
| Frame period | 20ms |
| Timeslot period | 1ms |

## 3.2 Architecture of packet scheduling algorithm

In this thesis, we just focus on the downlink transmission for simplification. Figure 3.1 shows the architecture of the packet scheduling algorithm for the downlink packet transmission. We assume that each source data user only generates one type of the traffics and the packets generated by each user only occupy the buffer for that user. The incoming packet arriving at BS are classified according to their traffic type, QoS requirements, and user ID, and sent to their own buffers by a packet classifier, which also records the arrival time and deadline of each packet, as well as the HOL packet delay of each buffer. The system packet scheduler then transmits the specified packets

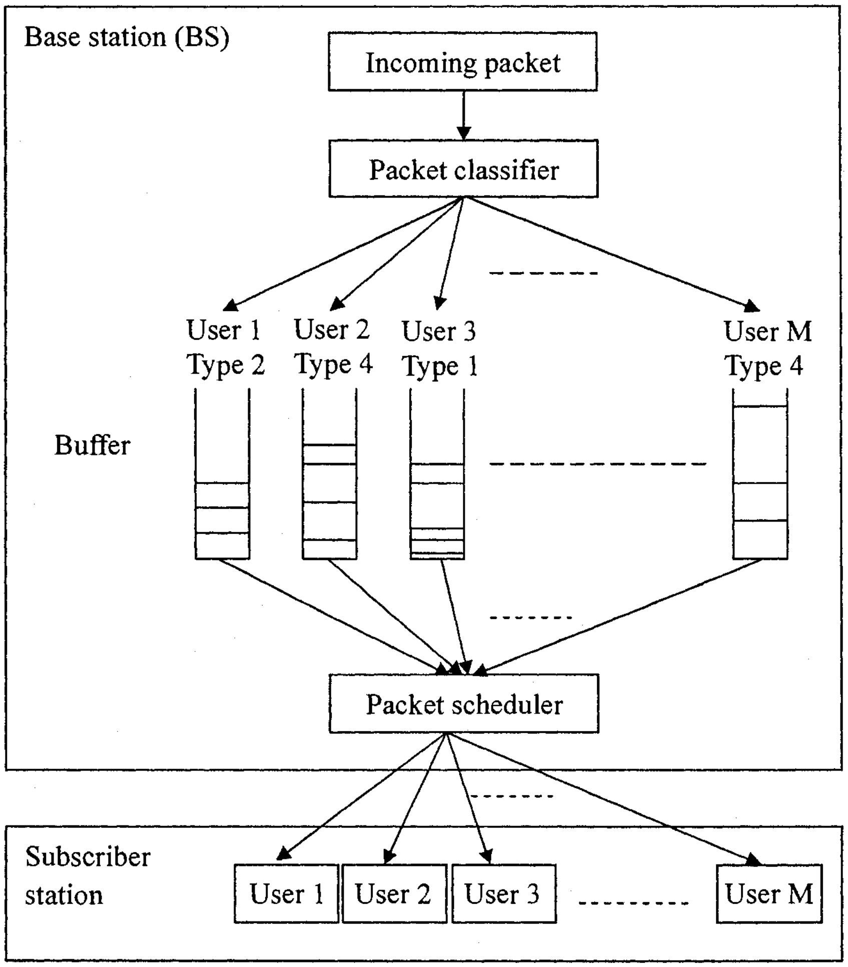



Figure 3.1 Architecture of packet scheduling algorithm

to the corresponding destination users according to the scheduling decision, which is based on the priority of the packet and fairness factors in our discussion. In this thesis, the priority of the packet is determined by the urgency factor of the incoming packet and the efficiency factor of the current channel state. The fairness factors are determined by the priority matrix, which is updated at each scheduling instant.

## 3.3 OFDMA channel model

In OFDMA system, a subchannel is a basic unit of resource allocation, which consists of multiple subcarriers. In this thesis, AMC scheme is employed in order to control the modulation and coding scheme employed on each subchannel adaptively according to the channel conditions. Figure 3.2 shows an OFDMA channel structure. As shown in Figure 3.2, one subchannel on the frequency axis and one transmission time interval called timeslot on the time axis jointly consists one channel-unit (CU), which is the basic resource unit used in packet scheduling. At each frame interval, the CUs of each subchannel may be allocated to one or more users according to the packet scheduling algorithm. Each subchannel consists of $L$ subcarriers and each timeslot consists of $K$ OFDM symbol intervals [5]. From Table 3.1, it is shown that $L=128$ and $K=10$ in our discussion. It is easy to see from Table 3.1 that 20 timeslots make up a frame.

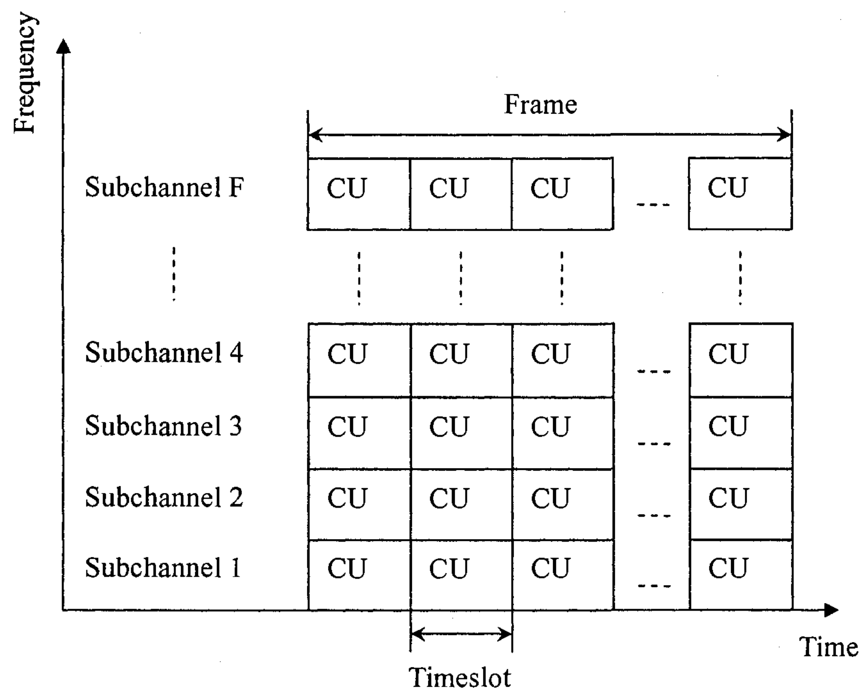



Figure 3.2 OFDMA channel structure

## 3.4 Parameters of AMC mode

AMC scheme adaptively adjust the modulation and coding scheme on each subchannel according to the subchannel condition for the user that the subchannel has been allocated to. With AMC scheme, the power of the transmitted signal on each subchannel is kept invariable during a frame interval, and modulation and coding format is adjusted to match the current received signal quality on that subchannel or subchannel condition. In this thesis, we notice that there are 12 subchannels in each timeslot. The capacity of each subchannel varies with the AMC mode adopted on that subchannel, which is determined by the subchannel state and can be changed from frame to frame. The multiple AMC modes are determined in terms of modulation order and coding rate, according to the SNR on that subchannel. Consequently, the transmission data rate of each subchannel changes periodically according to its AMC mode. Table 3.2 shows the AMC mode that may be adopted by a subchannel [2] [6].

Table 3.2 Summary of AMC mode

| AMC mode | Received SNR values (dB) | Modulation scheme | Coding Rate | Rn (bit/symbol) | Transmission data rate (kbps) |
|---|---|---|---|---|---|
| State 1 | 1.5-4.0 | BPSK | 1/2 | 0.5 | 640 |
| State 2 | 4.0-7.0 | QPSK | 1/2 | 1.0 | 1280 |
| State 3 | 7.0-11.0 | QPSK | 3/4 | 1.5 | 1920 |
| State 4 | 11.5-13.5 | 16 QAM | 9/16 | 2.25 | 2880 |
| State 5 | 13.5-18.5 | 16 QAM | 3/4 | 3.0 | 3840 |
| State 6 | 18.5- | 64 QAM | 3/4 | 4.5 | 5760 |

Let us take AMC mode with State 2 in Table 3.2 as an example. We know that one symbol represents 2 bits with QPSK modulation scheme. With coding rate 1/2, the information bit rate on each subcarrier is thus

$$\begin{aligned} R_n &= 2 \text{ bits/symbol/subcarrier} \times 1/2 \\ &= 1 \text{ bit/symbol/subcarrier} \end{aligned}$$

For simplicity, we assume that all subcarriers belonging to a subchannel undergo the same or similar channel effects. From Table 3.1, we know that symbol duration is $100\,\mu s$ and each subchannel contains 128 subcarriers, so the transmission rate of each subchannel is given by

$$\begin{aligned} T_n &= R_n \times 128 \text{ subcarriers} \times 1 \text{ symbol}/100\mu s \\ &= 1 \text{ bit}/\text{symbol}/\text{subcarrier} \times 128 \text{ subcarriers} \times 1\times 10^4 \text{ symbol}/s \\ &= 1280 \text{ kbps} \end{aligned}$$

Similarly, we can obtain the transmission rate of each subchannel with other AMC modes.

## 3.5 Random distributions used in channel and traffic model

The theory of probability and stochastic processes is an essential mathematic tool in the study of various wireless communication systems. In order to describe the channel model and generate various RT and NRT traffics, some important probability distributions are introduced in this thesis.

**Exponential Distribution:** In probability theory and statistics, the exponential distributions are a class of continuous probability distribution. The probability density function (PDF) and cumulative density function (CDF) of an exponential distribution are defined as

$$p(x;\lambda)=\begin{cases}\lambda e^{-\lambda x} & ,x\geq 0\\ 0 & ,x<0\end{cases} \tag{3.1}$$

$$F(x;\lambda)=\begin{cases}1-e^{-\lambda x} & ,x\geq 0\\ 0 & ,x<0\end{cases} \tag{3.2}$$

respectively, where $\lambda > 0$ is the rate parameter of the distribution. The mean or expected value and variance of an exponentially distributed random variable $X$ are given by

$$E[X]=\frac{1}{\lambda} \tag{3.3}$$

$$Var[X]=\frac{1}{\lambda^2}. \tag{3.4}$$

The PDF and CDF of an exponential distribution are illustrated in Figures 3.3 and 3.4, respectively.

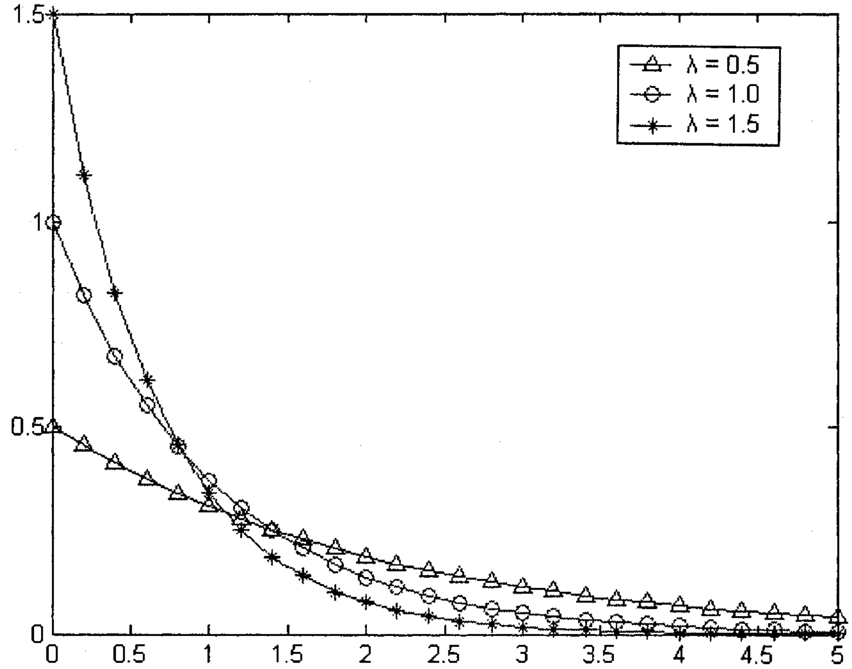



Figure 3.3 PDF of an exponential distribution

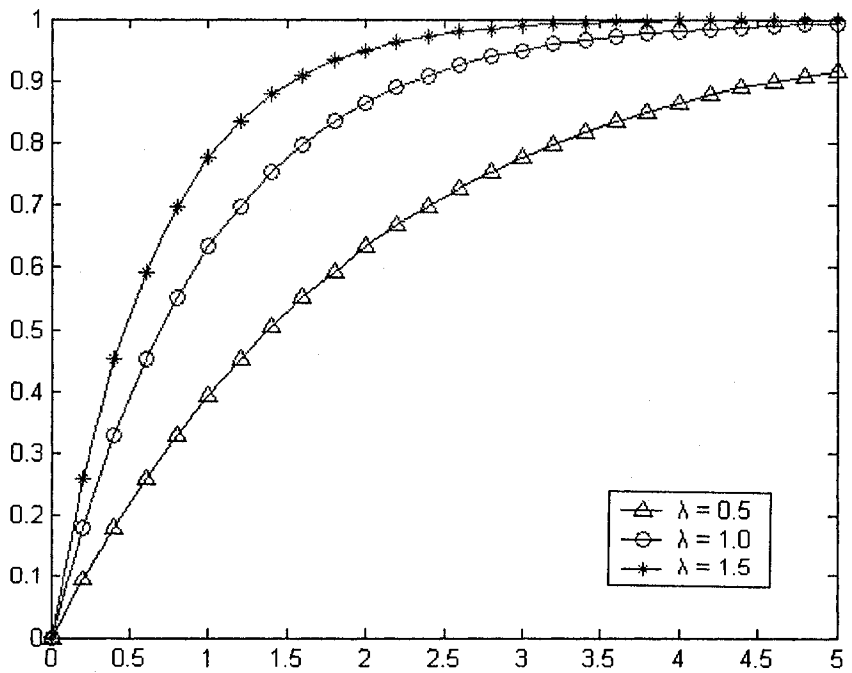



Figure 3.4 CDF of an exponential distribution

**Pareto Distribution and Truncated Pareto Distribution:** We assume that $X$ is a random variable with a Pareto distribution, then the probability that $X$ is greater than some value $x$ is defined as

$$\Pr(X > x) = \left(\frac{x}{x_m}\right)^{-k} \tag{3.5}$$

for all $x \geq x_m$, where $x_m$ is the minimum possible value of random variable $X$, and both $x_m$ and $k$ are positive values. Therefore, the PDF and CDF of a Pareto distribution are given by

$$p(x;k,x_m) = k\frac{x_m^k}{x^{k+1}} \quad \text{for} \quad x \geq x_m \tag{3.6}$$

$$F(x;k,x_m) = 1 - \left(\frac{x_m}{x}\right)^k \quad \text{for} \quad x \geq x_m. \tag{3.7}$$

The expected value and variance of a Pareto distribution are given by

$$E(X) = \frac{kx_m}{k-1} \quad \text{for} \quad k > 1 \tag{3.8}$$

$$Var(X) = \left(\frac{x_m}{k-1}\right)^2 \frac{k}{k-2} \quad \text{for} \quad k > 2 \tag{3.9}$$

The PDF and CDF of a Pareto distribution with $x_m = 1$ are illustrated in Figures 3.5 and 3.6, respectively.

A truncated distribution is a conditional distribution. Suppose we need to know the probability density of a random variable after restricting it between the range of $(a,b]$. That is to say, we need to know how random variable $X$ is distributed with $a < X \leq b$. Therefore, a truncated distribution with PDF $p(x)$ and CDF $F(X)$ is given by

$$Tr(x) = p(X \mid a < X \le b) = \frac{g(x)}{F(b) - F(a)} \tag{3.10}$$

where $g(x) = p(x)$ for all $a < X \le b$, otherwise $g(x) = 0$ for $X \notin (a,b]$. From equations (3.6), (3.7) and (3.10), we can obtain the PDF of a truncated Pareto distribution, which is given by

$$Tr_{pareto}(x) = \frac{k\frac{x_m^k}{x^{k+1}}}{\left(\frac{x_m}{a}\right)^k - \left(\frac{x_m}{b}\right)^k} \quad \text{for } a < x \le b \tag{3.11}$$

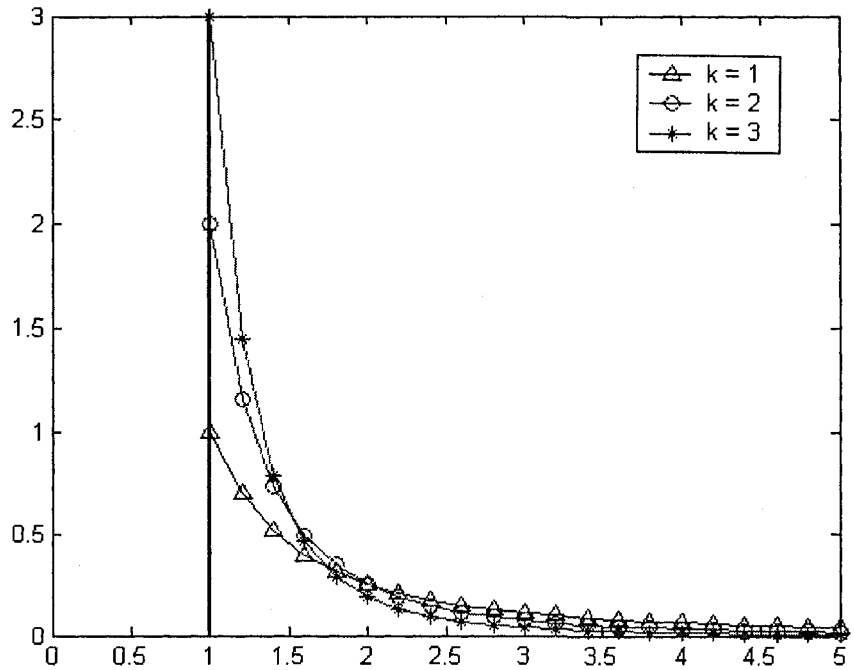



Figure 3.5 PDF of a Pareto distribution ($x_m = 1$)

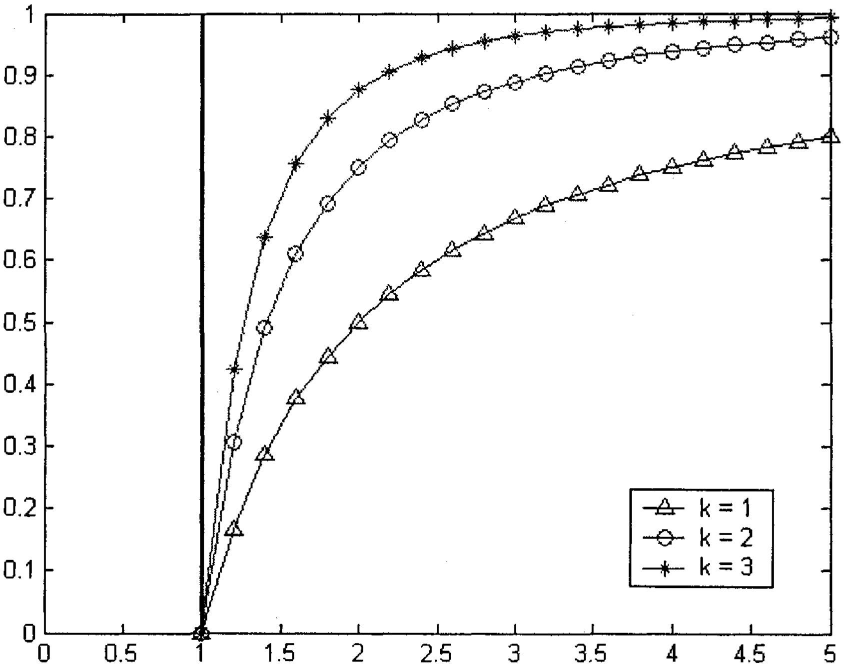



Figure 3.6 CDF of a Pareto distribution ($x_m = 1$)

**Normal Distribution and Truncated Normal Distribution:** A random variable $X$ with a normal distribution, also called Gaussian distribution, is expressed as $X \sim N(\mu, \sigma^2)$, where $\mu$ is the expected value and $\sigma^2$ is the variance of the random variable. The PDF and CDF of a normal distribution are given by

$$p(x; \mu, \sigma^2) = \frac{1}{\sqrt{2\pi}\sigma} e^{-(x-\mu)^2/2\sigma^2} \tag{3.12}$$

$$\begin{aligned} F(x; \mu, \sigma^2) &= \int_{-\infty}^{x} p(u) du \\ &= \frac{1}{\sqrt{2\pi}\sigma} \int_{-\infty}^{x} e^{-(u-\mu)^2/2\sigma^2} du \\ &= \frac{1}{2}\left[1 + erf\left(\frac{x-\mu}{\sqrt{2}\sigma}\right)\right] \end{aligned} \tag{3.13}$$

where $erf(x)$ denotes the error function, which is defined as

$$erf(x) = \frac{2}{\sqrt{\pi}} \int_0^x e^{-t^2} dt \qquad (3.14)$$

The PDF and CDF of a normal distribution are illustrated in Figures 3.7 and 3.8, respectively.

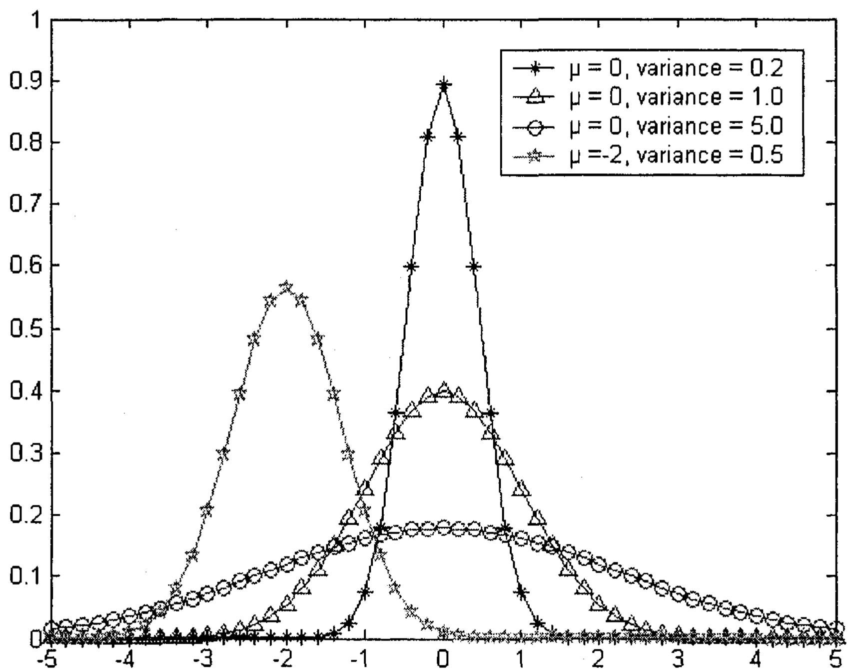



Figure 3.7 PDF of a normal distribution

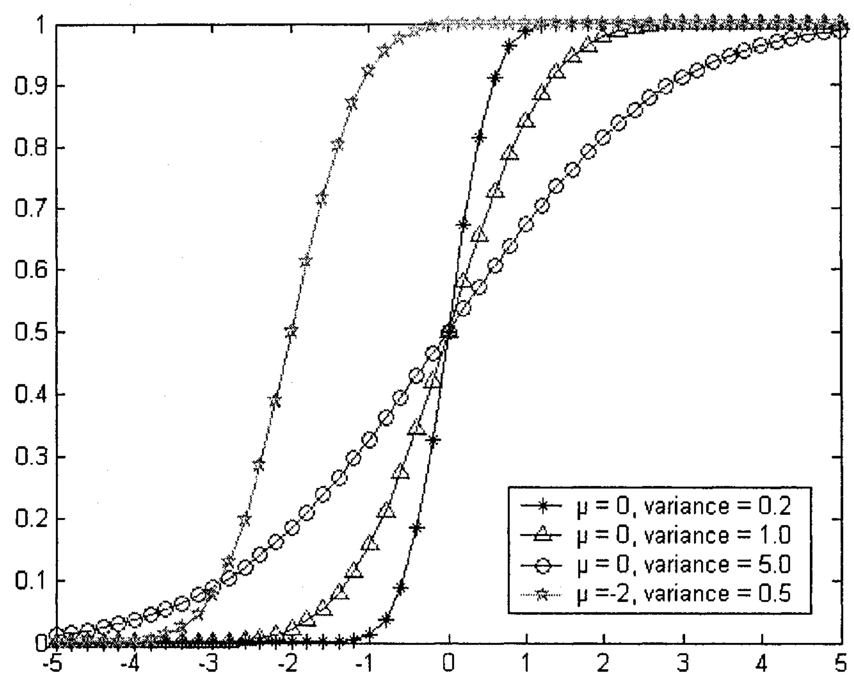



Figure 3.8 CDF of a normal distribution

A truncated normal distribution is also a conditional normal distribution. Similar to the case of the truncated Pareto distribution, from equations (3.12), (3.13) and (3.10), we obtain the PDF of a truncated normal distribution, which is given by

$$Tr_{normal}(x) = \frac{\frac{1}{\sqrt{2\pi}\sigma} e^{-(x-\mu)^2/2\sigma^2}}{\frac{1}{2}\left[ erf\left(\frac{a-\mu}{\sqrt{2}\sigma}\right) - erf\left(\frac{b-\mu}{\sqrt{2}\sigma}\right)\right]} \quad \text{for } a < x \le b \qquad (3.15)$$

**Poisson Distribution:** The occurrences of a sequence of discrete events can often be realistically modeled as a Poisson distribution. Poisson distribution is a discrete probability distribution that expresses the probability of a number of events occurring in a fixed period of unit time. It can be used to describe the number of events in specified intervals. The PDF and CDF of a Poisson distribution is given by

$$p(k,\lambda)=\frac{\lambda^k e^{-\lambda}}{k!} \tag{3.16}$$

$$F(x)=\frac{\Gamma(\lfloor k+1 \rfloor,\lambda)}{\lfloor k \rfloor!} \quad \text{for } k \geq 1 \tag{3.17}$$

where $k$ is the number of occurrences of the event, $\lambda$ is a positive real number, which is equal to the expected number of occurrences of the event during the given interval. The parameter $\lambda$ is both the expected value and the variance. The PDF and CDF of a Poisson distribution are illustrated in Figures 3.9 and 3.10, respectively.

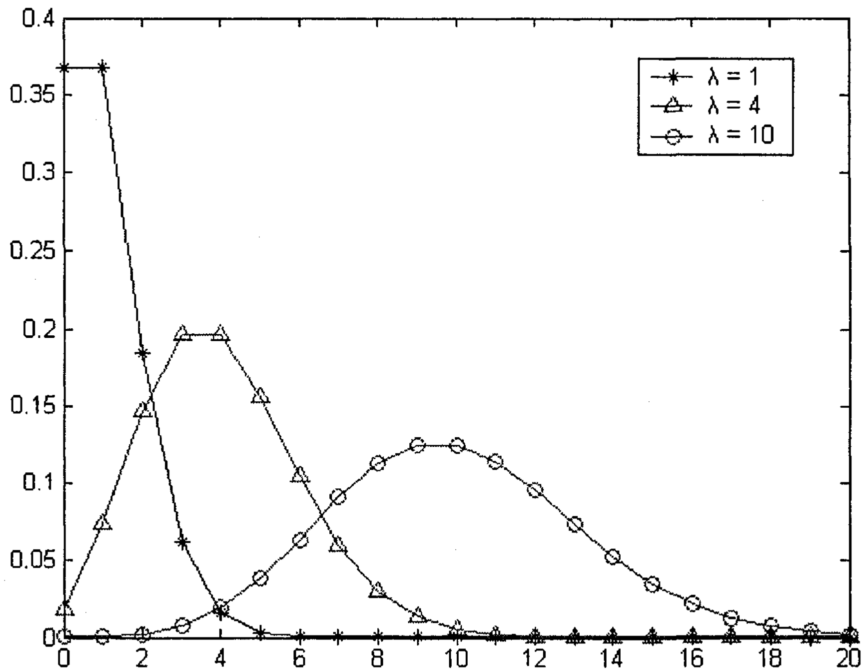



Figure 3.9 PDF of a Poisson distribution

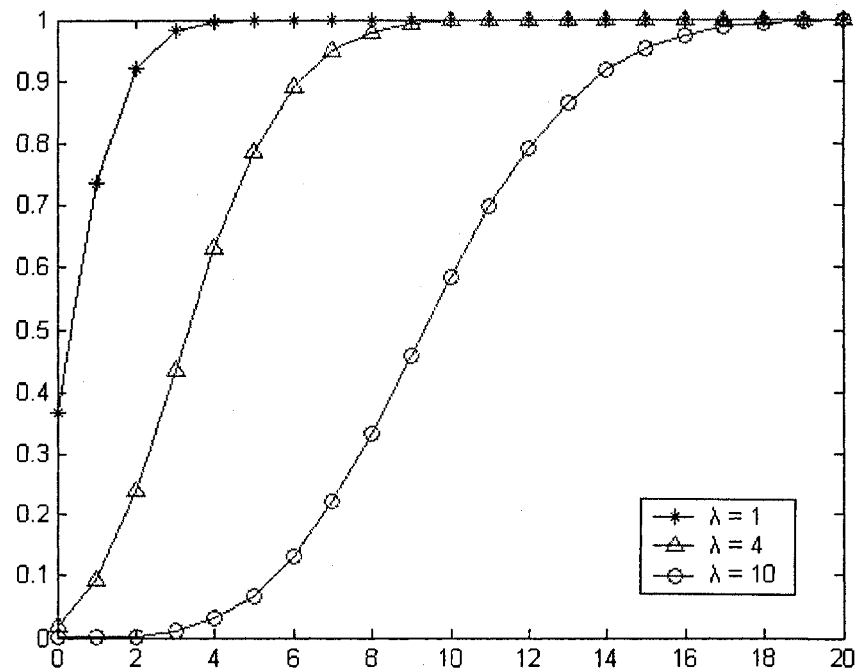



Figure 3.10 CDF of a Poisson distribution

The defining characteristic of a Poisson distribution is that the time intervals between successive event occurrences are exponentially distributed. We suppose that $X$ is a Poisson random variable with parameter $\lambda$. The probability of seeing a particular number $k$ of event occurrences in one unit of time is $p(X=k)=\dfrac{\lambda^k e^{-\lambda}}{k!}$. If we substitute $t>0$ units of time for one unit of time, then we have a new random variable $X_t$ with $\lambda t$ events occurring during $t$ units of time. The probability of a particular number $k$ of events occurrence in $t$ units of time is thus $p(X_t=k)=\dfrac{(\lambda t)^k e^{-\lambda t}}{k!}$. Now we consider the random variable $X_t$ with a Poisson distribution and the random variable $A$, which denote the interval between $(i-1)th$ and $ith$ event occurrence. The inequality $A>t$ denotes that no events occur in [0, t], so the following two events are equivalent: $\{A > t\} \equiv \{X_t = 0\}$. Both events represent that no events occur in the first $t$ units of time. Hence,

$p(A > t) = p(X_t = 0) = \frac{(\lambda t)^0 e^{-\lambda t}}{0!} = e^{-\lambda t}$. Therefore, the CDF of the random variable $A$ is $F(t) = p(A \leq t) = 1 - p(A > t) = 1 - e^{-\lambda t}$. Finally, the PDF of random variable $A$ can be found by differentiating $F$ with respect to $t$. We obtain the function $p(t) = F'(t) = \lambda e^{-\lambda t}$ for $t > 0$, which is the PDF of a typical exponential distribution, with mean $E(A) = 1/\lambda$. Thus, we can say that the time interval between any two successive events following a Poisson distribution follows an exponential distribution.

**Rayleigh Distribution:** Rayleigh distribution is a reasonable model of the random received signal magnitude when there are many objects in the environment that scatter the radio signal before it arrives at the receiver. Suppose $X_1$ and $X_2$ are zero-mean statistically independent Gaussian random variables, each having a variance $\sigma^2$. Now we define a new random variable $R = \sqrt{X_1^2 + X_2^2}$. Then $R$ follows a Rayleigh distribution whose PDF and CDF are given as

$$p_R(r) = \frac{r}{\sigma^2} \exp\left(-\frac{r^2}{2\sigma^2}\right), \quad r \geq 0 \tag{3.19}$$

$$F_R(r) = \int_0^r \frac{u}{\sigma^2} \exp\left(-\frac{u^2}{2\sigma^2}\right) du = 1 - \exp\left(-\frac{r^2}{2\sigma^2}\right), \quad r \geq 0 \tag{3.19}$$

The PDF and CDF of a Rayleigh distribution are illustrated in Figures 3.11 and 3.12, respectively.

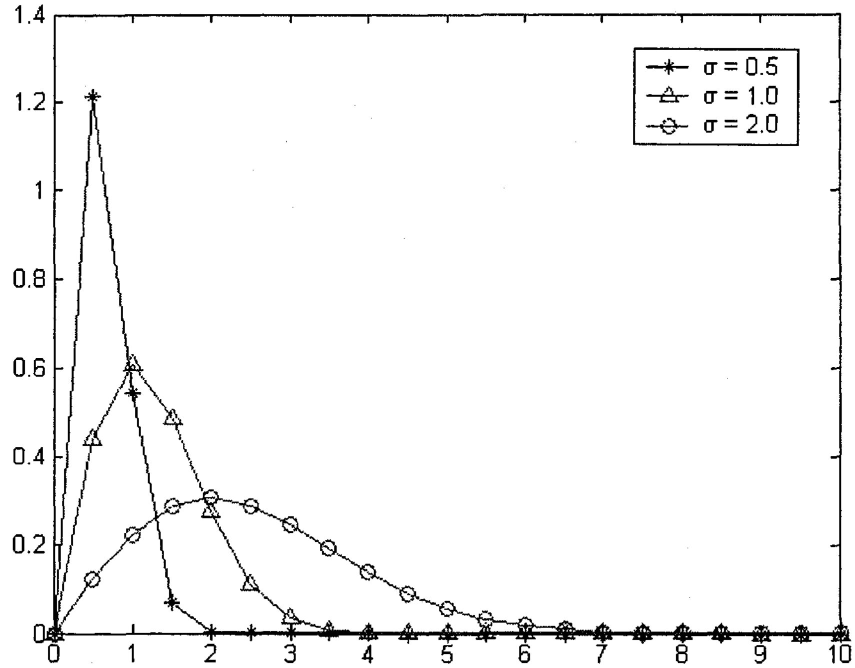



Figure 3.11 PDF of a Rayleigh distribution

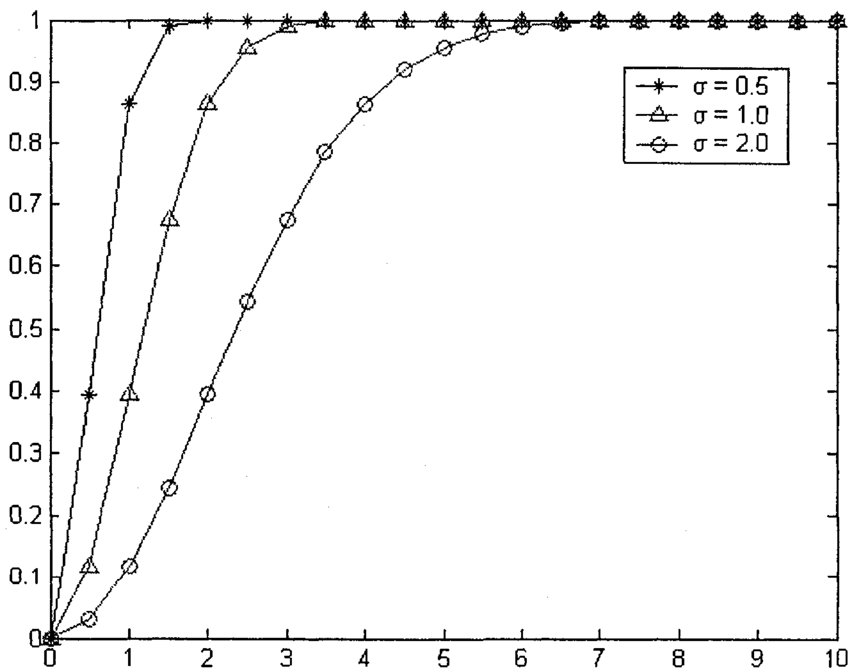



Figure 3.12 CDF of a Rayleigh distribution

# Chapter 4

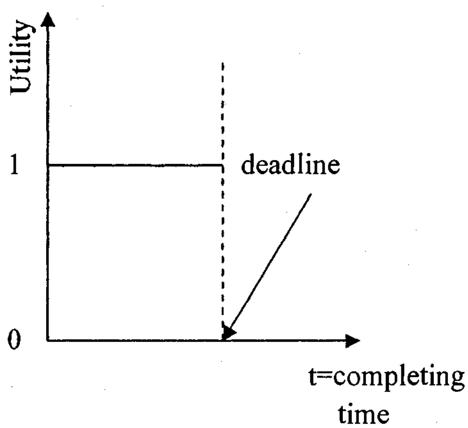
# BPF and MPF based Packet Scheduling

## 4.1 Time-Utility Function and Deadline Approach

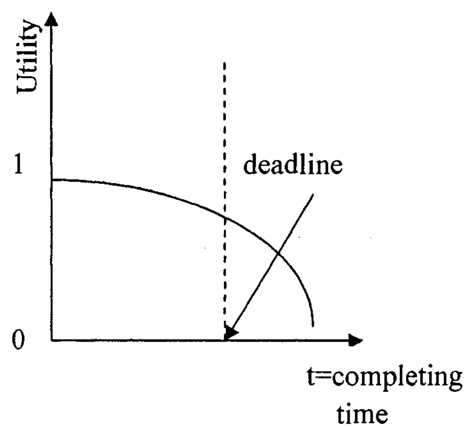
In this thesis, we adopt the time-utility function and deadline approach to schedule RT and NRT traffics in our study. The deadline approach is the most widely studied approach for time constraint in RT applications [10] [11] [12]. A deadline constraint for the task of packet scheduling indicates that completing the task at any time before the deadline introduces some certain utility to the system. Moreover, if the utility has a diminished value when the task is finished after the deadline, then the deadline is called soft deadline. Whereas if the utility has the value of zero when the task is finished after the deadline, then the deadline is called hard deadline.

We introduce the time utility function to indicate the packets transmission activity. For RT traffic, the hard deadline is adopted. The packets will be received correctly from BS to the subscriber station when they are transmitted before the deadline, but the packets will be discarded when they are transmitted after the deadline. By contraries, the soft deadline is adopted for NRT traffic. The packets will be received correctly as long as the utility is none zero, even if they are transmitted after the deadline.

As illustrated in Figure 4.1, a time-related scheduling task with hard deadline is described by a hard-deadlined time-utility function, whose value of utility is 1 before the deadline and 0 after the deadline. On the other hand, a time-related scheduling task with soft deadline is described by a soft-deadlined time-utility function, whose value of utility decreases monotonically with the increasing completing time [1] [5] [6].




(a) hard-deadlined time-utility




(b) soft-deadlined time-utility

Figure 4.1 Time-utility functions

## 4.2 Urgency of Incoming Packets

The proposed packet scheduling algorithm supports both RT and NRT traffics. The time-utility function is used to determine the urgency factor of the incoming packets. For delay-sensitive RT traffic, we use the hard-deadlined time-utility function. The soft-deadlined time-utility function is used for NRT traffic. The packets are transmitted before the deadline to meet the QoS requirements of different traffics. However, it is also noted that we need RT traffic to be transmitted only during a short period of time before the deadline. For example, the RT video signals need to be transmitted just in time, so that we receive and watch the video smoothly. We will feel that the video is overlapped if the video signal is received too early, or the video is discontinuous if the signal is received too late. Therefore, we adopt the concept of jitter to indicate the short period of time for the transmission of RT traffic [1]. The incoming packets of the RT traffic are transmitted only during the time interval $[D_i - J_i, D_i]$, where $D_i$ indicates the deadline of the HOL packet of buffer $i$ and $J_i$ indicates the length of the jitter of RT traffic $i$ before the deadline. The incoming packets of RT traffic are transmitted only during the time interval of $[D_i - J_i, D_i]$. Otherwise, if the delay is less than $D_i - J_i$, the packet will not be scheduled to transmit. If the delay is more than $D_i$, the packet will be discarded. Consequently, we specify that the value of utility is nonzero only during the interval $[D_i - J_i, D_i]$, as shown in Figure 4.2 [1]. Beyond the jitter period of RT traffics, the NRT traffics will be scheduled to be transmitted.

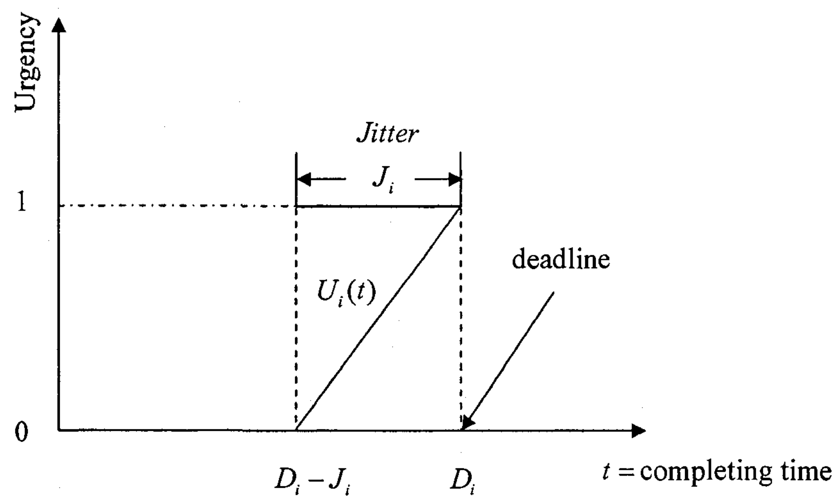



Figure 4.2 Concept of jitter for RT traffic

For RT traffic with a hard-deadlined time-utility function, if the utility value is kept the same during the interval $[D_i - J_i, D_i]$, the HOL packets of different users will have the same urgency factor during the interval $[D_i - J_i, D_i]$. Even though these HOL packets may arrive at different instants. This may result in unexpected packets loss. In order to overcome this problem, we introduce a monotonously increasing function to indicate the urgency of RT traffic. The function value will increase with increasing completing time, which means the urgency increases with the completing time approaching the deadline. Thus, the urgency factor of RT traffic is given by

$$U_i(t) = \frac{t - (D_i - J_i)}{J_i}, \quad t \in [D_i - J_i, D_i] \tag{4.1}$$

The urgency factor of each traffic is used by the packet scheduler as one of the factors to determine the scheduling priority. We assume that there are four types of traffics in our study, which are RT voice, RT video, NRT WWW and NRT E-mail. The HOL packet of each traffic having large value of urgency is considered to be more urgent to be transmitted than others with small values of urgency.

## 4.3 Efficiency of Channel

The channel efficiency of the incoming packets on subchannels is another factor to determine the priority in the proposed packet scheduling algorithms. We use the current channel state and the average channel state in the past $W$ timeslots to determine the efficiency of the channel.

Therefore, based on Table 3.2, we define the current channel state matrix at each scheduling instant as

$$\boldsymbol{R}(t)=\begin{bmatrix} R_{11}(t) & R_{12}(t) & \cdots & R_{1N}(t) \\ R_{21}(t) & R_{22}(t) & \cdots & R_{2N}(t) \\ \vdots & \vdots & \ddots & \vdots \\ R_{F1}(t) & R_{F2}(t) & \cdots & R_{FN}(t) \end{bmatrix} \tag{4.2}$$

Denote $R_{ij}(t)$ $(i \in F,\ j \in N)$ as the current transmission data rate of user $j$ on subchannel $i$ with $N$ denoting the number of users and $F$ denoting the number of subchannels. It is seen that $R_{ij}(t)$ is a good representative of the current channel state of subchannel $i$ experienced by user $j$. Accordingly, we define the average channel state matrix at each scheduling instant as $\overline{\boldsymbol{R}}(t)$ with the $\{i, j\}$ th element as

$$\overline{R_{ij}}(t)=(1-1/W)\overline{R_{ij}}(t-1)+R_{ij}(t)/W \tag{4.3}$$

which denotes the average value of subchannel $i$ 's transmission rate experienced by user $i$ in the past $W$ timeslots [1] [11]. We use the ratio of the current transmission rate to the average transmission rate of user $j$ on subchannel $i$, which is

$$\eta_{ij}(t)=R_{ij}(t)/\overline{R_{ij}}(t) \tag{4.4}$$

as the channel efficiency factor.

## 4.4 Priority and Fairness

Based on Equations (4.1) and (4.4), the priority of the $j$th user on the $i$th subchannel is defined as

$$P_{ij}(t)=U_j(t)\times\eta_{ij}(t) \tag{4.5}$$

The priority $P_{ij}(t)$ $(i \in F,\ j \in N)$ is updated at each scheduling instant.

Therefore, at each scheduling instant, we obtain the priority matrix which is given by

$$\boldsymbol{P}(t)=\begin{bmatrix} P_{11}(t) & P_{12}(t) & \cdots & P_{1N}(t) \\ P_{21}(t) & P_{22}(t) & \cdots & P_{2N}(t) \\ \vdots & \vdots & \ddots & \vdots \\ P_{F1}(t) & P_{F2}(t) & \cdots & P_{FN}(t) \end{bmatrix} \tag{4.6}$$

In the proposed scheduling algorithms, we consider how to allocate the subchannels to the users. The user with larger value of priority will be allocated a subchannel prior to those with smaller values of priority until all of the subchannels are used up. If there are more than two users who have the same value of priority on the same subchannel, or the priorities of a user on different subchannels are of the same values, the fairness factor should be considered in the allocation. Therefore, we adopt two fairness factors, the fairness factor of subchannel $i$, $FF_{sub\_i}$ and the fairness factor of user $j$, $FF_{user\_j}$, which are respectively defined as

$$FF_{sub\_i}=\prod_{j=1}^{N} P_{ij} \tag{4.7}$$

$$FF_{user\_j}=\prod_{i=1}^{F} P_{ij}\,. \tag{4.8}$$

For example, suppose that on the $k$th subchannel, user $l$ and user $m$ have the same value of priority, which is $P_{kl}(t)=P_{km}(t)$, where $k\in N$, $l,m\in F$ and $l\neq m$. Then we look at the fairness factor of the users. If $FF_{user\_l}>FF_{user\_m}$, then user $m$ is scheduled to be transmitted prior to user $l$ and subchannel $k$ is allocated to user $m$ with transmission rate $R_{km}(t)$, which is obtained from the current channel state matrix. That is, in competing the subchannels, the user with smaller value of fairness factor is chosen prior to those with larger value of fairness factor. Or in other words, the user having poorer alternative subchannels is selected to transmit prior to those having better alternative subchannels. In this way, the deviation of delays experience by different users' packets becomes smaller, and thus user fairness will be increased [5].

For another example, suppose that user $m$ has the same value of priority on subchannels $p$ and $q$, which is $P_{pm}(t)=P_{qm}(t)$, where $p,q \in N$, $p \neq q$ and $m \in F$. Looking at the fairness factor of subchannels, if $FF_{sub\_p} > FF_{sub\_q}$, then subchannel $q$ is allocated to user $m$ with transmission rate $R_{qm}(t)$, which is obtained base on the current channel state matrix. Larger value of fairness factor of a subchannel means that on that subchannel, more users have larger values of priority. Hence, when one user has the same highest value of priority on different subchannels, we prefer to allocate the subchannel with smaller value of fairness factor to the user than that with larger value of fairness factor, since the subchannel with larger value of fairness factor has more chance to be allocated to other users, and thus higher fairness among subchannels can be achieved and channel efficiency can be increased.

## 4.5 BPF and MPF Based Packet Scheduling Algorithms

Based on the urgency and efficiency based packet scheduling algorithm, which is studied in [1], we propose the BPF and the MPF packet scheduling algorithms, which jointly consider the urgency factor of different RT and NRT traffics, the efficiency factor of the OFDMA wireless channel, and the fairness factors of the users and the subchannels. The incoming packets of different RT and NRT traffics with the highest level of priority are scheduled to be transmitted prior to those with lower level of priority, while the fairness factors of the users and the subchannels are considered as well. Specifically, the proposed priority and fairness based packet scheduling algorithms at BS consists of three main parts:

**Part 1**. Define the packet arrival event. The incoming packets are classified according to their traffic type and user ID and sent to the corresponding buffers. The buffer of each user records the arrival time of the incoming packets, the packet size of each packet, the sequence of the incoming packet in the buffer, the number of packet in the buffer, buffer size, deadline of each packet, etc.

**Part 2**. Calculate the priority of each user on each subchannel. Calculate the fairness factors of each user and each subchannel

**Part 3**. Define the packet departure event. At each scheduling instant, users are chosen to transmit their packets based on their priority and fairness factors. We calculate the loss rate, HOL packet delay and throughput, and obtain the simulation results.

The proposed BPF and MPF algorithms deal with parts one and two in the above.

### 4.5.1 BPF Packet Scheduling Algorithm

In the proposed BPF packet scheduling algorithm, each subchannel is only allocated to one user at every scheduling instant. Based on the priority and fairness described before, the BPF packet scheduling algorithm is summarized as follows. Note that packet scheduling algorithm needs to be executed at each scheduling instant.

Table 4.1 BPF packet scheduling algorithm

---

For a set $S$, $S_i$ denotes the $i$th element in the set and $size(S)$ denotes the size of the set.

Initialize $C = \{1, \cdots, F\}$ as the available subchannel set, $U = \{1, \cdots, N\}$ as the user set.

Step 1: For $\forall j \in U$, calculate the urgency factor $U_j(t)$. If $U_j(t) = 0$, discard the HOL packet of user $j$. Repeat Step 1 until $U_j(t) > 0$.

Step 2: For $\forall i \in C$ and $\forall j \in U$, calculate the channel efficiency factor $\eta_{ij}(t) = R_{ij}(t) / \overline{R_{ij}}(t)$.

Step 3: Obtain the priority matrix $\boldsymbol{P}(t)$ with the $\{i, j\}$th element as

$P_{ij}(t) = U_j(t) \times \eta_{ij}(t)$

Step 4: For $j \in U$, calculate the fairness factor for each user as $FF_{user\_j}(t) = \prod_{i=1}^{F} P_{ij}(t)$. For $i \in C$, calculate the fairness factor for each subchannel as $FF_{sub\_i}(t) = \prod_{j=1}^{N} P_{ij}(t)$.

Step 5: For $i \in C$, find the maximum priority of each subchannel $P_{i\_\max}(t) = \max(P_{iU_1}(t), P_{iU_2}(t), \cdots, P_{iU_{size(U)}}(t))$

Step 6: For $i \in C$, find the subchannels whose maximum priority is equal to the maximum of $P_{i\_\max}(t)$ $(i \in C)$, i.e., $P_{i^*\_\max}(t) = \max(P_{C_1\_\max}, P_{C_2\_\max}, \cdots P_{C_{size(C)}\_\max})$. Put these subchannels $i^*$ into a set $A$.

Step 7: If the size of the set $A$ is one, then pick up the subchannel $i^*$ in the set $A$ and go to Step 8;

Otherwise, if the size of the set $A$ is greater than one, then find the subchannel in the set $A$ with the minimal subchannel fairness factor, i.e., pick up the subchannel $i^*$ with $FF_{sub\_i^*}(t) = \min(FF_{sub\_A_1}(t), \ldots, FF_{sub\_A_{size(A)}}(t))$.

Step 8: For the subchannel $i^*$ picked up in Step 7, find the users in the set $U$

whose priority on this subchannel is equal to the maximum value, i.e., $P_{i^*j^*}(t) = P_{i^*\_\max}(t)$.

Put these users $j^*$ into a set $B$.

Step 9: If the size of the set $B$ is one, then pick up the user $j^*$ in the set $B$ and go to Step 10;

Otherwise, if the size of the set $B$ is greater than one, then find the user in the set $B$ with the minimal user fairness factor, i.e., pick up the user $j^*$ with $FF_{user\_j^*}(t) = \min(FF_{user\_B_1}(t), \ldots, FF_{user\_B_{size(B)}}(t))$.

Step 10: Allocate the subchannel $i^*$ to the user $j^*$ .

Step 11: Delete the subchannel $i^*$ from the set $C$ and delete the user $j^*$ from the set $U$. If $C \neq \varnothing$ and $U \neq \varnothing$ then go to Step 5;

Otherwise, stop.

In our study, there are $F$ subchannels. From the algorithm in Table 4.1, it can be seen that the system supports $F$ users at each scheduling instant at most. If there are more than $F$ users generating packets at the same time, the scheduler selects just only $F$ users with the highest priority at each scheduling instant. The unselected users have to wait for the next scheduling instant to be scheduled.

### 4.5.2 MPF Packet Scheduling Algorithm

In the BPF scheduling algorithm, one subchannel is only allocated to one user during the whole frame period, which consists of 20 timeslots in the system we studied as an example, even if the user has a very small amount of packets to transmit. In this case, the user may only occupy a few timeslots, and other timeslots are idle, and not used by any other users, which may have a large number of packets waiting to be transmitted. This will result in waste of wireless resource. In order to avoid the waste of wireless resources, we modify the BPF scheduling algorithm, which is referred to as the MPF scheduling algorithm, by allowing other users to share the unoccupied timeslots, which are left by users with light traffic load.

According the proposed MPF algorithm, firstly, a user is selected and allocated a subchannel according to the BPF algorithm. Secondly, we calculate how many timeslots the user will occupy according to that subchannel condition and that user's number of packets in its buffer. Then, we can find out how many timeslots are left. Thirdly, we repeat the previous processes to select user and allocate a subchannel to it until there are no more timeslots left on each subchannels or all the buffers are empty. The MPF packet scheduling algorithm is summarized as follow.

Table 4.2 MPF packet scheduling algorithm

---

For a set $S$, $S_i$ denotes the $i$th element in the set and $size(S)$ denotes the size of the set.

Initialize $C = \{1, \cdots, F\}$ as the available subchannel set, $U = \{1, \cdots, N\}$ as the user set, $M_i$ as the number of unoccupied timeslots on subchannel $i$ $(i = 1, \cdots, F)$ and pointer $h_j$ pointing to the HOL packet in the buffer of user $j$ $(j = 1, \cdots, N)$

Step 1: For $\forall j \in U$, calculate the urgency factor $U_{j,h_j}(t)$ for the packet pointed by $h_j$. If $U_{j,h_j}(t) = 0$, discard the packet pointed by $h_j$ and update $h_j$ pointing to the next packet. Repeat Step 1 until $U_{j,h_j}(t) > 0$.

Step 2: For $\forall i \in C$ and $\forall j \in U$, calculate the channel efficiency factor $\eta_{ij}(t) = R_{ij}(t) / \overline{R_{ij}}(t)$.

Step 3: Obtain the priority matrix $\boldsymbol{P}(t)$ with the $\{i,j\}$ th element as

$P_{ij}(t)=U_{j,h_j}(t)\times\eta_{ij}(t)$

Step 4: For $j\in U$, calculate the fairness factor for each user as $FF_{user\_j}(t)=\prod_{i=1}^{F}P_{ij}(t)$. For $i\in C$, calculate the fairness factor for each subchannel as $FF_{sub\_i}(t)=\prod_{j=1}^{N}P_{ij}(t)$.

Step 5: For $i\in C$, find the maximum priority of each subchannel $P_{i\_\max}(t)=\max(P_{iU_1}(t),P_{iU_2}(t),\cdots,P_{iU_{size(U)}}(t))$

Step 6: For $i\in C$, find the subchannels whose maximum priority is equal to the maximum of $P_{i\_\max}(t)(i\in C)$, i.e., $P_{i^*\_\max}(t)=\max(P_{C_1\_\max},P_{C_2\_\max},\cdots P_{C_{size(C)}\_\max})$.

Put these subchannels $i^*$ into a set $A$.

Step 7: If the size of the set $A$ is one, then pick up the subchannel $i^*$ in the set $A$ and go to Step 8;

Otherwise, if the size of the set $A$ is greater than one, then find the subchannel in the set $A$ with the minimal subchannel fairness factor, i.e., pick up the subchannel $i^*$ with $FF_{sub\_i^*}(t)=\min(FF_{sub\_A_1}(t),\ldots,FF_{sub\_A_{size(A)}}(t))$.

Step 8: For the subchannel $i^*$ picked up in Step 7, find the users in the set $U$ whose priority on this subchannel is equal to the maximum value, i.e., $P_{i^*j^*}(t)=P_{i^*\_\max}(t)$.

Put these users $j^*$ into a set $B$.

Step 9: If the size of the set $B$ is one, then pick up the user $j^*$ in the set $B$ and go to Step 10;

Otherwise, if the size of the set $B$ is greater than one, then find the user in the set

$B$ with the minimal user fairness factor, i.e., pick up the user $j^*$ with $FF_{user\_j^*}(t) = \min(FF_{user\_B_1}(t), \ldots, FF_{user\_B_{size(B)}}(t))$.

Step 10: Calculate the number of timeslots $M_{i^*,j^*,h_{j^*}}$ needed on the subchannel $i^*$ for user $j^*$ to transmit its packet pointed by $h_{j^*}$.

If $M_{i^*} \geq M_{i^*,j^*,h_{j^*}}$, then allocate $M_{i^*,j^*,h_{j^*}}$ timeslots to user $j^*$. Update the pointer $h_{j^*}$ to point to the next packet in the buffer, update the value of $M_{i^*}$, and go to Step 10 until the buffer of user $j^*$ is empty.

Otherwise, if $M_{i^*} < M_{i^*,j^*,h_{j^*}}$, then set $M_{i^*} = 0$.

Step 11: Update $P_{i^*j^*}(t) = U_{j^*,h_{j^*}}(t)\eta_{i^*j^*}(t)$.

If $M_{i^*} = 0$, delete the subchannel $i^*$ from the set $C$, and then go to Step 4.

# Chapter 5

# Simulation Results

## 5.1 Modes of RT and NRT Traffics

In this thesis, we adopt four types of traffics in our simulations: RT voice, RT video, NRT WWW, and NRT Best Effort (BE) traffics. Moreover, each individual user only generates one of the four types of traffics. The four types of traffics models are described as follows.

**RT voice traffic model**: Voice over IP (VoIP) traffic is used as an example of the RT voice traffic. The VoIP traffic follows a 2-state Markov (ON/OFF) model, in which the state when the user talks corresponds to ON state and the state when the user is silent corresponds to OFF state. The user alternates between talking (ON) state and silence (OFF) state. The lengths of ON and OFF periods are assumed to be exponential distribution with mean equal to 1 second and 1.35 seconds, respectively. The packets are generated at a regular rate with a fixed size during ON period. We assume that the generation rate of VoIP traffic follows a bit rate of 48 kbps with packet size 0.4 kbits [2]. Figures 5.1 and 5.2 show the 2-state Markov model and the model of VoIP traffic, respectively.

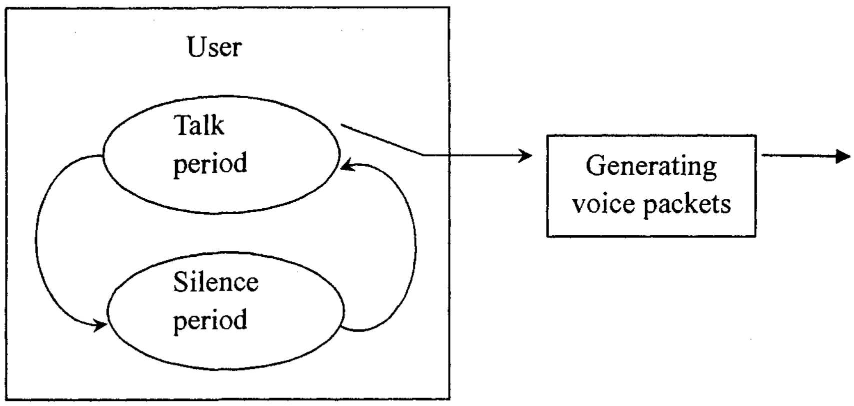



Figure 5.1 2-State Markov (ON/FF) model

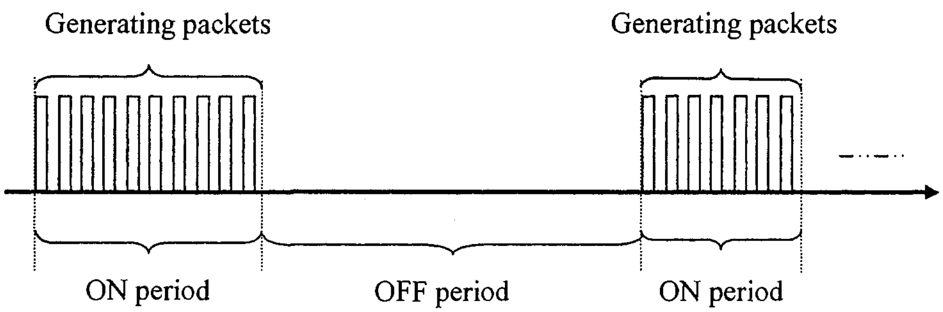



Figure 5.2 VoIP traffic model

**RT video traffic model**: Video streaming is used to simulate the RT video traffic model. The video streaming consists of a group of frames which are generated periodically with interval $T$. Each frame of the video streaming consists of a set of packets with variable sizes, which follow a truncated Pareto distribution. The number of packets is fixed for every video frame. The inter-arrival time between any two consecutive packets in the frame follows a truncated Pareto distribution as well. The model of the video streaming is shown in Figure 5.3 and the distribution parameters of the video streaming are given in Table 5.1 [1] [2].

**NRT WWW traffic model**: A page-oriented World Wide Web (WWW) traffic is used in our simulations. A user starts a WWW session by browsing the web pages, which consists of multiple packets or objects. Within a session, user browses several web pages and reads them. The reading time between two pages follows an exponential distribution with mean equal to 30 seconds. A web page includes one main object and several embedded objects. The inter-arrival between the main object and the first embedded objects is called parsing time, which follows an exponential distribution with mean equal to 0.13 seconds. The main object size and embedded object size follow truncated normal distributions, and the number of embedded object follows a truncated Pareto distribution as shown in Table 5.2. Figure 5.4 shows a WWW traffic model [1].

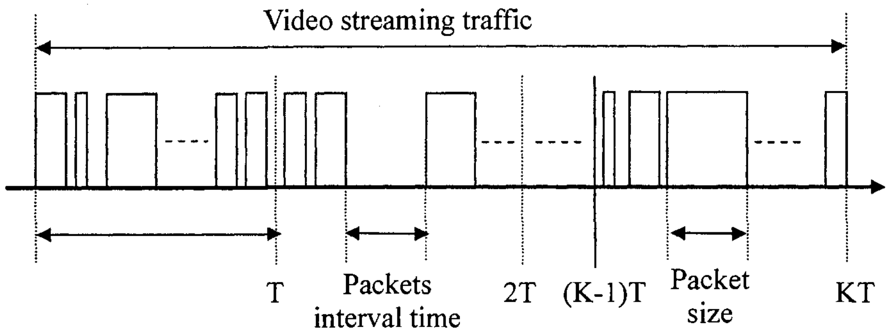



Figure 5.3 Video streaming model

Table 5.1 Distribution parameter of video streaming

| Item | Distribution | Parameters |
|---|---|---|
| Packet size | Truncated Pareto | $x_m = 20$ bytes $k = 1.2$<br>Min:50 bytes<br>Max: 125 bytes |
| Packets interval time | Truncated Pareto | $x_m$=2.5 ms $k = 1.2$<br>Min: 6 ms<br>Max: 12.5ms |
| Frame interval time | Deterministic | $T = 100ms$ |
| Number of packet per frame | Deterministic | 8 |

Table 5.2 Distribution parameter of WWW traffic

| Items | Distribution | Parameters |
|---|---|---|
| Main object size | Truncated Normal | Mean:10710 bytes STD: 25032 bytes Min: 100 bytes Max: 2Mbytes |
| Embedded object size | Truncated Normal | Mean: 7758 bytes STD: 12168 bytes Min: 50 bytes Max: 2Mbytes |
| Number of embedded object per page | Truncated Pareto | Mean: 5.64 STD: 53 |
| Reading time | Exponential | Mean: 30 seconds |
| Parsing time | Exponential | Mean: 0.13 seconds |

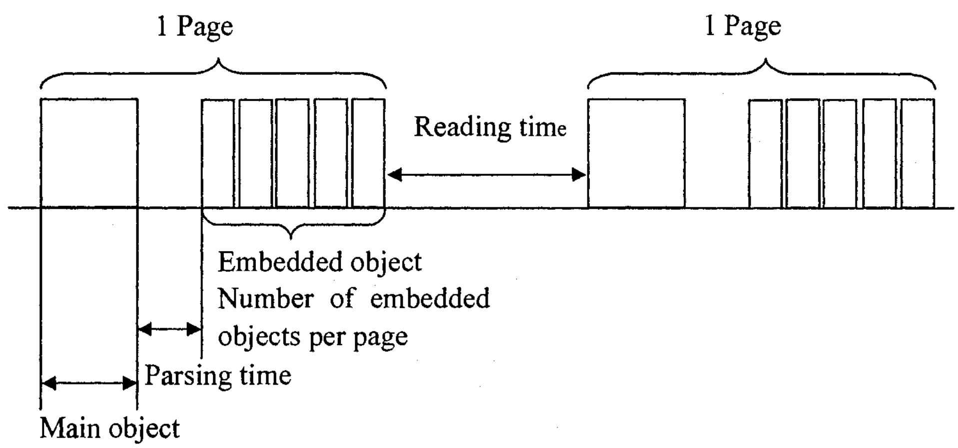



Figure 5.4 WWW traffic model

**NRT BE traffic model**: NRT best effort traffic can tolerate longer delay than RT traffic. The BE traffic can be transmitted as long as the channel capacity remains sufficient over longer time intervals. Due to their sensitivity to delay, RT traffic are usually given higher priority to send packets. Thus BE traffic receive the channel capacity left over by RT traffic. Email traffic is used as an example of the NRT BE traffic in our simulations. We assume that Email messages arrive at the mailboxes randomly, which can be modeled by a Poisson process with constant packet size. From the property of the Poisson process, we notice that if events occurring in time $T$ as a Poisson process with parameter $\lambda$, the time between events are distributed as an exponential random variable with parameter $\lambda$, so we adopt an exponential distribution to describe the interval time of adjacent packet arrivals for Email traffic. Figure 5.5 shows an Email traffic model. The generation rate of Email traffic follows a bit rate of 48 kbps with packet size 1 kbits in our simulations.

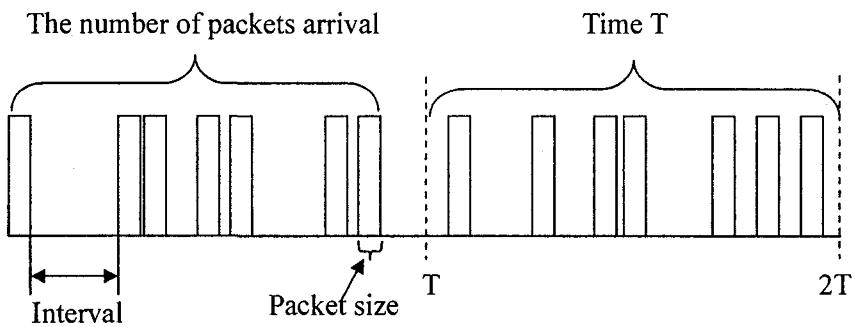



Figure 5.5 Email traffic model

## 5.2 Channel Model

In this study, we consider a hexagonal cell system. We assume the BS transmission power is 12W, and all power is used to transmit the packets. The transmit power is averagely assigned to all 12 subchannels, which means 1W power is assigned to each subchannel for packet transmission. We adopt Rayleigh distribution model to describe the received signal SNR for the downlink transmission in our simulations [6]. The SNR of the received signal $\gamma$ is thus a random variable with a Rayleigh probability density function as given by

$$p(\gamma) = \frac{1}{2\sigma^2}\exp\left(-\frac{\gamma}{2\sigma^2}\right) = \frac{1}{\gamma_0}\exp\left(-\frac{\gamma}{\gamma_0}\right), \quad \gamma \geq 0 \tag{5.1}$$

where $\gamma_0 = \frac{1}{2\sigma^2}$ is the mean of SNR and is assumed to be 7 dB in our simulations.

For simplicity, we assume that the received signal SNR remains the same over the whole frame duration. Table 5.3 shows the parameter of the channel model [2],

Table 5.3 Parameter of the simulation model

| Parameter | Value |
|---|---|
| Channel model | Rayleigh fading channel |
| BS total transmission power | 12 W |
| The power of each subchannel | 1 W |
| Mean of SNR | 7 dB |
| $W$ | 1000 ms |
| $M$ | 20 |

where $W$ is window size used to calculate the average channel state, $M$ is the total timeslots on each subchannel during a frame period.

## 5.3 Simulation Results and Discussion

We evaluate the performance of the proposed algorithms in terms of the QoS requirements of different traffics, including packet loss rate, average packet delay for

RT traffic, average throughput for NRT traffic. In our simulations, we set the QoS requirements for RT voice and RT video traffic [1] [2] as follows.

RT voice: delay < 40 ms, packet loss rate < 3%

RT video: delay < 150 ms, packet loss rate <1%

If the packet loss rate exceeds the boundary of the RT traffic, the quality of the received signal can not be guaranteed.

The average throughput is used to evaluate the performance for all NRT traffics. The number of users generating packets at the same time is considered as the offered load factor in the simulations. We evaluate the performance of the proposed scheduling algorithms under different offered loads by gradually varying the number of users of every type of traffic from 4 to 60.

**Simulation results of RT traffics**: From the model of RT traffic shown before, we know that the length of jitter is a very important factor for the performance of the proposed scheduling algorithms for RT traffic. We set the length of jitter to 10, 20 and 30 ms respectively, for the RT voice traffic and 30 ms for the RT video traffic. The number of users of voice traffic, which is the same as the number of users of video traffic, is varying from 4 to 60.

Figure 5.6 shows the simulation results of the average packet loss rates of voice traffics. For the BPF and MPF algorithms, the average packet loss rate of voice traffic decreases with length of jitter changing from 10, 20 to 30 ms for voice traffic. Increasing the length of jitter of voice traffic makes voice traffic having much time to transmit, which results in more voice packets to be transmitted. Thus, the average packet loss rate will decrease. For MPF algorithm, when the length of jitter is equal to 30 ms for voice traffic, the average packet loss rate of voice traffic is even lower than $10^{-4}$ in all the offered load cases we considered, thus it is not included in Figure 5.6.

For RT voice traffic, the maximum acceptable loss rate is 3%. From Figure 5.6, we notice that 24 users can be supported by the M-LWDF algorithm, 36 users can be supported by the BPF algorithm with the length of jitter equal to 20 ms for voice traffic and 30 ms for video traffic, and 48 users can be supported by the MPF algorithm with the length of jitter equal to 20 ms for voice traffic and 30 ms for video traffic. Thus, the MPF algorithm gives better performance than the BPF and the M-LWDF algorithms in terms of the average packet loss rate of voice traffic. The BPF algorithm is better than the M-LWDF algorithm when the length of jitter is greater than or equal to 20 ms for voice traffic and 30 ms for video traffic.

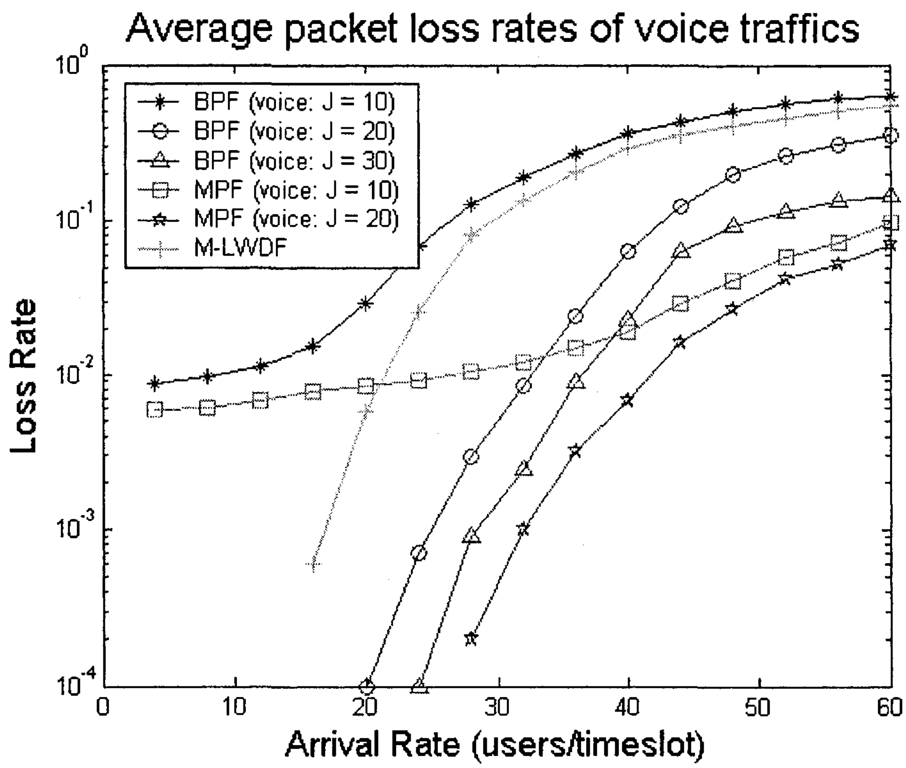



Figure 5.6 Average packet loss rate of voice traffics ( $J_{video} = 30ms$ )

Figure 5.7 shows the simulation result of the average packet loss rates of video traffics. For the MPF algorithm, the average packet loss rate of video traffic increases a little bit with length of jitter changing from 10, 20 to 30 ms for voice traffic. For the BPF algorithm, the average packet loss rate of video traffic increases apparently with length of jitter changing from 10, 20, 30 ms for voice traffic. Increasing the length of jitter of voice traffic makes video traffic having less time to transmit, which results in more video packets waiting in the buffer. Thus, the average packet loss rate will increase accordingly.

For RT video traffic, the maximum acceptable loss rate is 1%. From Figure 5.7, we notice that approximately 24 users can be supported by the M-LWDF and the BPF algorithms with the length of jitter equal to 20 ms for voice traffic, although the BPF is a little bit better than the M-LWDF algorithm, and 40 users can be supported by the MPF algorithm. Thus, the MPF algorithm gives better performance than the BPF and the M-LWDF algorithms in terms of the average packet loss rate of video traffic.

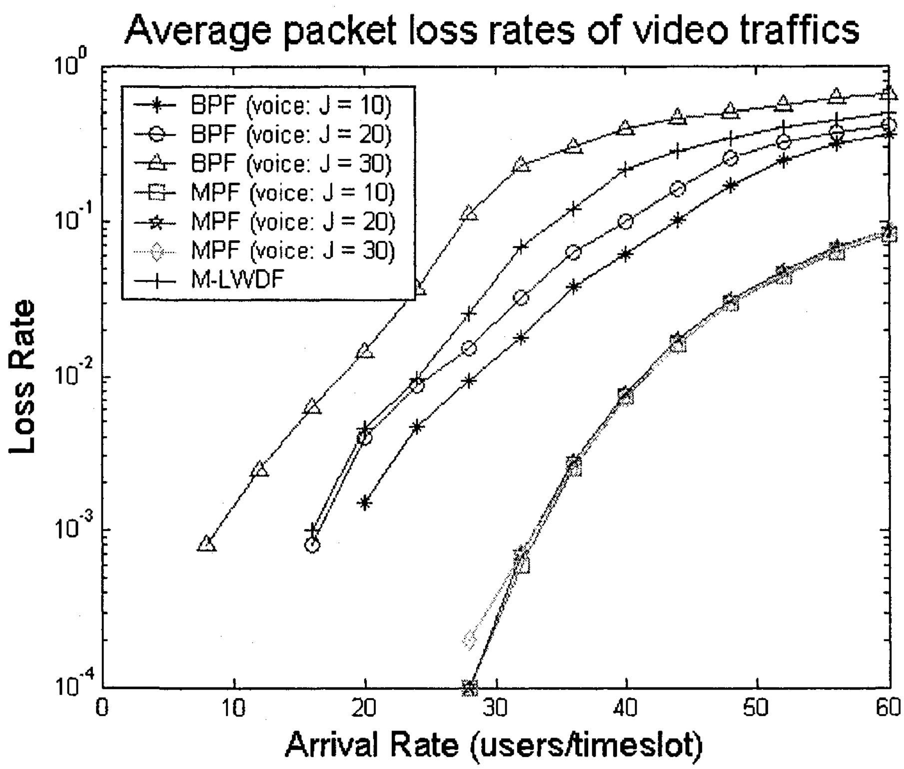



Figure 5.7 Average packet loss rate of video traffics ($J_{video} = 30ms$)

Figures 5.8 and 5.9 show the simulation results of the average packet delays of voice traffics. We set the length of jitter to 20 ms for voice traffic and 30 ms for video traffic in the MPF and the BPF algorithms. This means that the HOL packets of voice traffic will be transmitted after 20 ms and the HOL packets of video traffic will be transmitted after 120ms from their arrival instants.

When the number of users is less than 44, the M-LWDF algorithm provides lower average packet delays than the BPF and the MPF algorithms. When the number of users is greater than or equal to 48, the MPF algorithm provides higher average packet delays than the BPF algorithm, but provides lower average packet delays than the M-LWDF algorithm. However, for the MPF and the BPF algorithms, the average packet delays are always less than the maximum acceptable delay requirement, which is 40 ms, under all offered loads in our simulations.

For the average packet delays of video traffics, the MPF and the BPF algorithms provide higher average packet delays than the M-LWDF algorithm, but is always less than the maximum acceptable delay requirement, which is 150 ms, under all offered loads in our simulations.

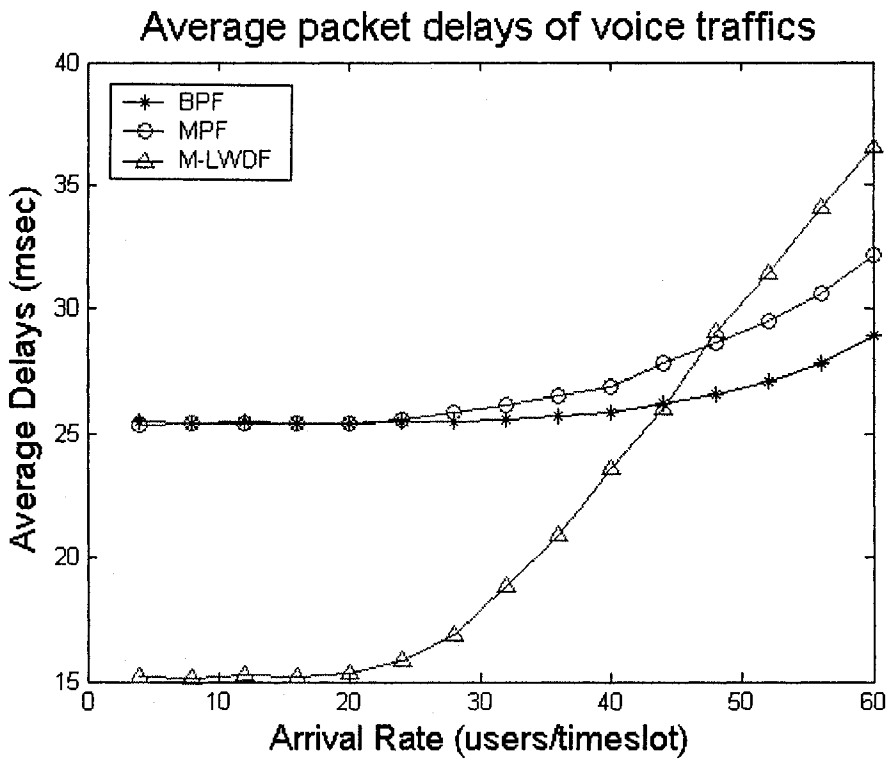



Figure 5.8 Average packet delays of voice traffics ( $J_{voice} = 20ms, \quad J_{video} = 30ms$ )

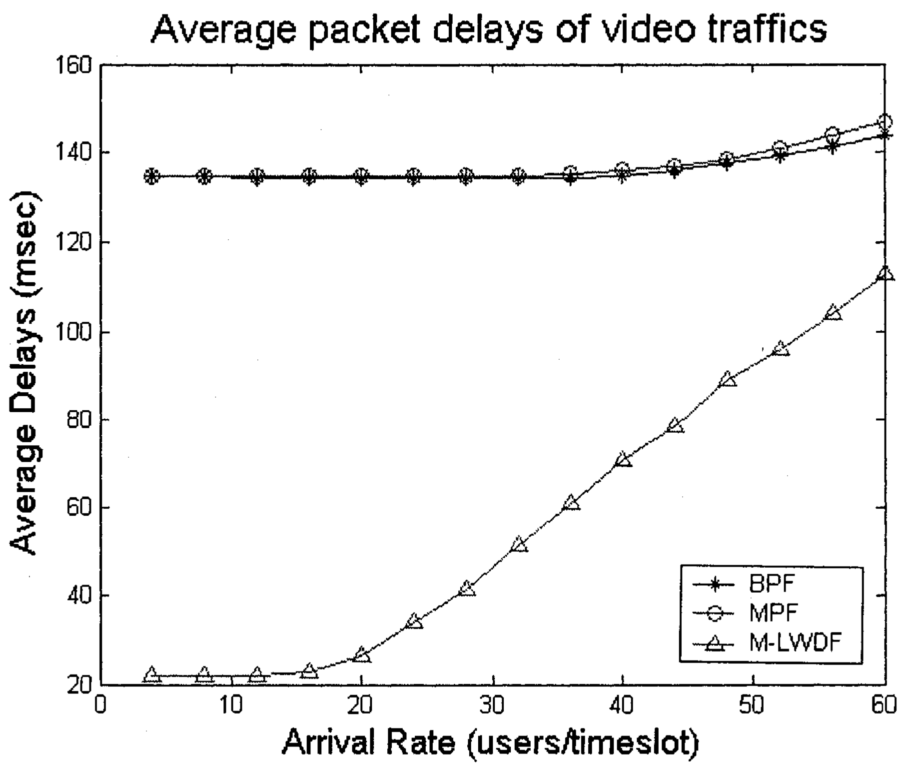



Figure 5.9 Average packet delays of video traffics ($J_{voice} = 20ms,\quad J_{video} = 30ms$)

**Simulation results of NRT traffics**: For NRT traffics, we need to maximize the throughput of NRT WWW and NRT Email traffics under different offered loads. The number of users of WWW traffics and Email traffics are varying from 4 to 60 as the offered load. When both RT and NRT traffics are supported, we set the length of jitter equal to 20 ms for voice traffic and 30 ms for video traffic in the BPF and MPF algorithms, and we assume that the number of NRT traffic users is the same as that of RT traffics.

Figures 5.10 and 5.11 show the throughputs of WWW and Email traffics of the MPF, BPF and M-LWDF algorithms respectively, when both RT and NRT traffics are supported. For WWW traffic, we notice that the throughput of the MPF algorithm is a little bit higher than that of the BPF and the M-LWDF algorithm under almost all offered loads. For Email traffic, we notice that the throughputs of the MPF, BPF and M-LWDF algorithms are almost the same under all offered loads.

Due to the existence of jitter of RT traffic, the channel capacity is wasted beyond the jitter period if only RT traffic is supported. In the BPF and MPF scheduling algorithms, the scheduler allocates the subchannels, which are left by RT traffic, to NRT traffic according to the priority and fairness factors. Hence, the throughput of NRT traffic is almost unchanged while the QoS requirements of RT traffic (such as packet delay and loss rate requirement) are satisfied. It indicates that the BPF and MPF packet scheduling algorithms can support NRT and RT traffics at the same time without degrading the performance.

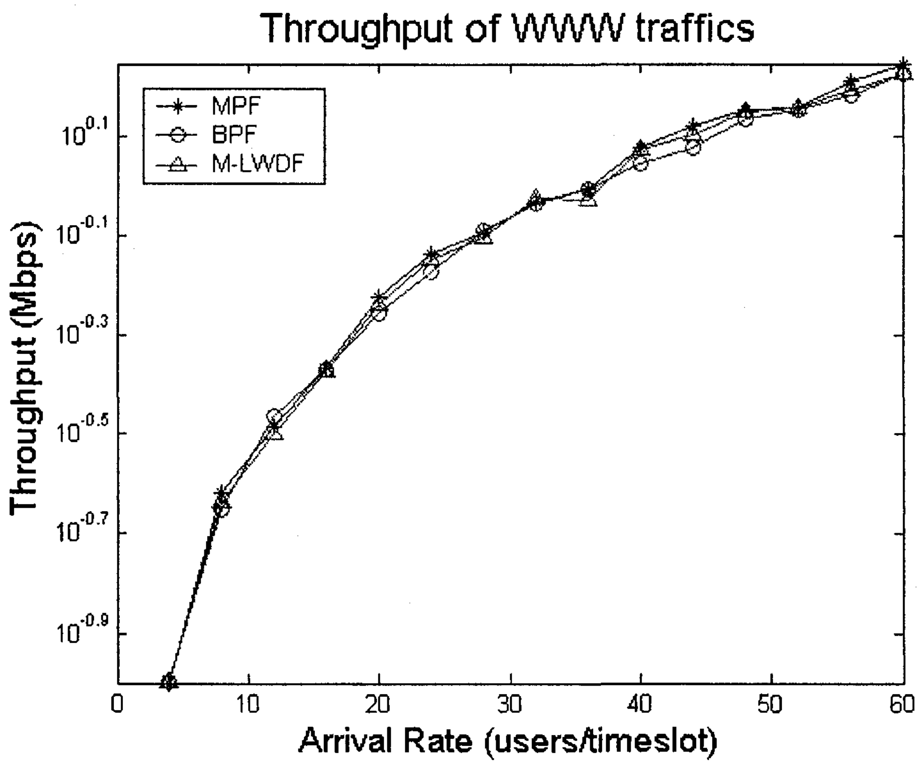



Figure 5.10 Throughput of WWW traffics

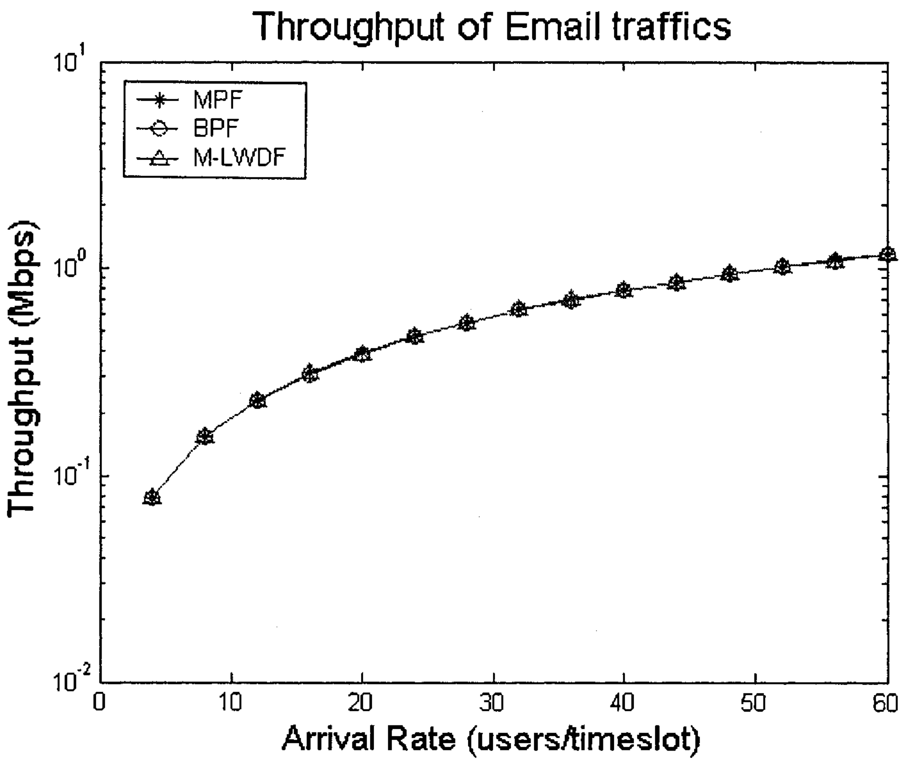



Figure 5.11 Throughput of Email traffics

Figures 5.12 and 5.13 show the average packet loss rate of voice and video traffics respectively under the situations of RT traffic only and RT and NRT traffics together.

From Figures 5.12 and 5.13, we notice that the average packet loss rate of voice and video traffics are almost the same respectively when the simulation condition is changed from RT traffic only to RT and NRT traffics together in the MPF and BPF algorithms. However, for the M-LWDF algorithm, the average packet loss rate of voice and video traffics increase apparently respectively when the simulation condition is changed from RT traffic only to RT and NRT traffics together.

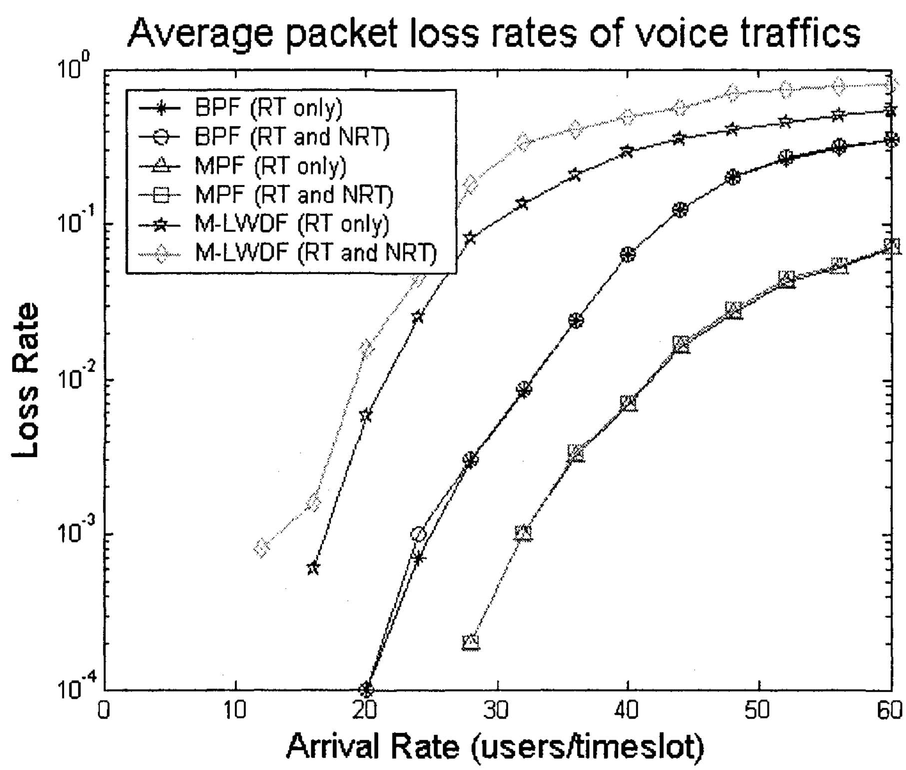



Figure 5.12 Average packet loss rates of voice traffics with RT traffic only and RT and NRT traffics together

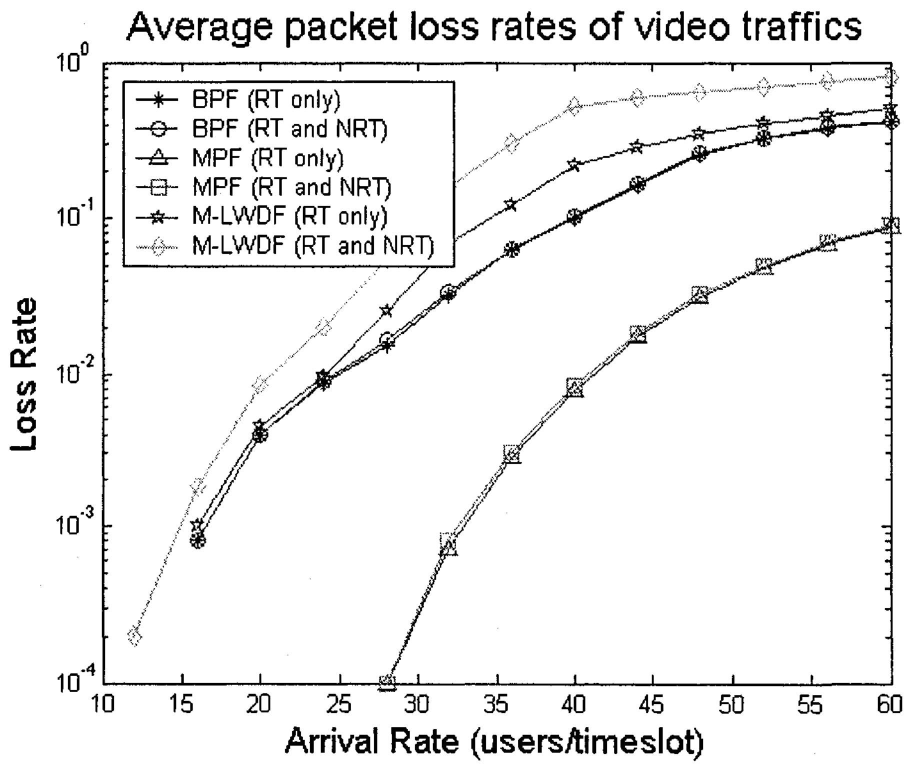



Figure 5.13 Average packet loss rates of video traffics with RT traffic only and RT and NRT traffics together

Figures 5.14 and 5.15 show the average packet delay of voice and video traffics respectively under the situation of RT traffic only and RT and NRT traffics together.

Similarly, the average packet delay of voice and video traffics are almost the same when the simulation condition is changed from RT traffic only to RT and NRT traffics together in the MPF and BPF algorithms. For the M-LWDF algorithm, the average packet delay of voice and video traffics increase apparently respectively when the simulation condition is changed from RT traffic only to RT and NRT traffics together, which are shown in Figures 5.14 and 5.15.

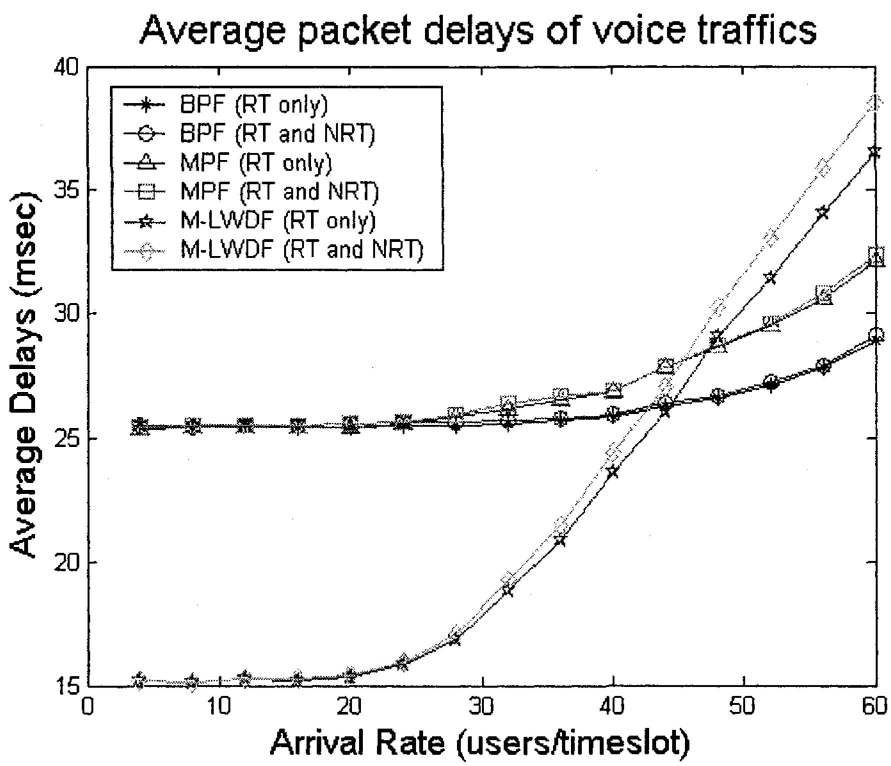



Figure 5.14 Average packet delays of voice traffics with RT traffic only and RT and NRT traffics together

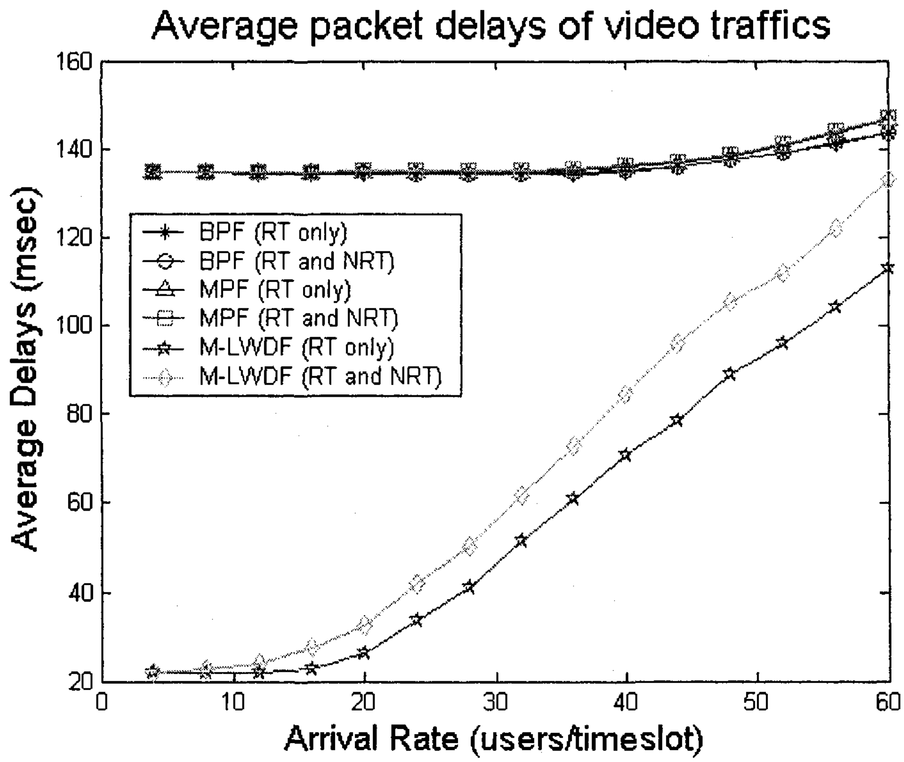



Figure 5.15 Average packet delays of video traffics with RT traffic only and RT and NRT traffics together

# Chapter 6

# Conclusion and Future Studies

In this thesis, the BPF and MPF packet scheduling algorithms are proposed to support both RT and NRT traffics. In the BPF and MPF packet scheduling algorithms, the priority of a user is determined based on the urgency factor of the incoming packets from the user and the efficiency factor of the wireless communication channel experienced by the user, while fairness factor is taken into account as well. The users are selected to transmit their packets according to the values of their priority and fairness factors. Under different simulation conditions, such as offered load and the length of jitter, the obtained results show that the proposed scheduling algorithms can satisfy the QoS requirements of different RT and NRT traffics, and also support NRT and RT traffics simultaneously without degrading the performance of RT traffic. Compared with the M-LWDF algorithm, the BPF and MPF algorithms provide better performance in terms of average packet loss rate in RT traffics and throughput in NRT traffics.

The M-LWDF algorithm is widely applied to the current OFDMA systems, thus we compare the proposed BPF and MPF algorithms with it to evaluate their performance in this thesis. However, there are also many other algorithms being proposed for OFDMA system. We will compare the proposed algorithms with more other algorithms to study their performance better and to find possible further improvements. For simplification, it is assumed that each user only generates one type of traffic in this thesis. In general, a user may generate multiple types of traffics at the same time. A large amount of information will be sent to BS from the subscriber station. The packet scheduler at BS needs to adjust adaptively under various wireless packet scheduling situations. In addition, in our studies, we only considered the performance of the proposed packet scheduling algorithms. In order to evaluate the efficiency of the proposed packet scheduling algorithms, the scheduling time overhead of the proposed BPF and MPF algorithms needs to be studied. These can be my future study topics.

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