Simulation of QoS Based Call Admission Schemes for DS-CDMA

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Abstract

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Code division multiple-access (CDMA) has been receiving considerable attention as the core multiple-access technology in the development of upcoming ubiquitous personal communication service (PCS) system with flexible quality of service (QoS) requirements at any time and anywhere. Moreover, future wireless networks are expected to integrate different types of multimedia traffic, such as voice, data, and compressed images and video.

This work aims to investigate the resource allocation problem for wireless direct sequence spread-spectrum CDMA cellular networks supporting heterogeneous multimedia applications where individual users have different QoS requirements. We propose optimal rate and power distribution approaches for source allocation to individual users, subject to a constraint on the total available bandwidth, to maximize the per-cell capacity and maintain the quality of service (QoS) for different multimedia services.

Connection oriented admission control schemes based on the required QoS are proposed in this thesis. Several policies are introduced for use in CDMA network. Simulation results show that the proposed Radio Resource Management scheme can achieve both effective QoS guarantee and efficient resource utilization.

Acknowledgements

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I am deeply grateful to my parents for their loving care and guidance throughout my entire life.

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List of Abbreviations

ABO	Average buffer overflow
AMPS	Advanced mobile phone services
AR	connection Arrival Rate
AWGN	additive white Gaussian noise
BER	Bit error rate
Bps	Bit per second
BPSK	Binary Phase Shift Keying
Bs	Buffer Size
BS	Base station
CAC	Connection admission control
CBR	Constant bit rate
CDMA	Code division multiple access.
DCA	Dynamic Channel Allocation
DS-CDMA	Direct sequence code division multiple access
ECSD	Enhanced Circuit-Switched Data
EDGE	Enhanced Data for Global Evolution
ETSI	European Telecommunications Standards Institute
FDD	frequency division duplexing
FDMA	Frequency division multiple access
GPRS	General Packet Radio Services
GSM	Global System for Mobile communications
IMT-2000	Standard for 3G
IMTS	Improved Mobile Telephone Service
ISR	Interference-to-Signal Ratio
ITU	International Télécommunications Union
MAC	Medium access control
MAS	Maximum SIR algorithme

MC-CDMA	Multi code code division multiple access
ME	Mobile equipment
MS	Mobile station
MSC	Mobile Switching Center
MTP	Minimum total power algorithm
NCBP	New connection blocking probability
PLMN	Public Land Mobile Network
PSD	Power spectrum density
QoS	Quality of Service
RLC/MAC	Radio Link Control/Multiple Access Control
RRM	Radio resource management
STA	Static algorithm
SIR	Signal-to-Interference Ratio
TDD	Time division duplexing
TDMA	Time Division Multiple Access
VBR	Variable bit rate
WCDMA	Wideband Code Division Multiple Access

Chapter 1

Introduction

1.1 Motivation

With the advance of radio and network technologies, the mobile system has evolved from preliminary communication devices to a ubiquitous personal communication service system that provides users new channel capacity and features, such as data, voice and multimedia applications and services with flexible quality of service (QoS) requirements at any time and anywhere. The wireless system development has come to an era of the convergence of information and telecommunication technologies.

To enable these services it is essential that next generation wireless networks support multiple classes of traffic with diverse quality of service (QoS) requirements in terms of data rate, bit error rate, delay, etc. For example, voice traffic generally requires a data rate of 8 kbps and bit error rate (BER) less than 10e-2 to 10e-3, while video traffic requires data rate of 64 kbps and up, with BER less than 10e-6. To provide users with improved QoS it is desirable for next generation wireless networks to have enhanced performance, in terms of higher data rate. It is equally important for next generation wireless networks to have higher system capacity in order to meet the rapidly escalating demands on accessing the backbone networks from anywhere at any time.

One of the viable technologies to meet these challenges is code division multiple access (CDMA). During the evolution of wireless systems in the past, CDMA has been receiving considerable attention as the core multiple-access technology, and the development of CDMA network has been struggling with different fundamental key problems.

Due to the unique feature that users are sharing the allocated bandwidth and competing for the bandwidth by adjusting their power level at the receiver in a CDMA system, the problem can also be formulated from a resource allocation point of view. To meet the large bandwidth requirement of multimedia traffic, it is important to utilize the system resource efficiently and provide preferential treatment according to mobile user's traffic. The Radio Resource Management (RRM) module [15] in the CDMA network system is responsible for the management of air interface resources. RRM is needed to offer efficient system utilization and guarantee a certain QoS level to different user. Meanwhile, The call admission control (CAC) mechanism is one of the most important components of RRM that affects the resource utilization efficiency and QoS guarantees provided to users.

In CDMA systems the spreading sequences of different users are usually not mutually orthogonal due to the asynchronous transmission, the multi-path fading channel, and the limitation on the number of available orthogonal codes. This causes multiple access interference (MAI) when many users access the allocated spectrum simultaneously in a CDMA system. A direct result of that is the capacity vs. quality tradeoff, i.e. as more and more users access the spectrum simultaneously the performance of each user's communication link deteriorates. In order to maintain the link quality above certain specified level, it is necessary to limit the number of users accessing the spectrum at the same time. Therefore capacity of CDMA system is said to be limited by multiple access interference, or interference limited.

Current CDMA systems (e.g. IS-95) [2] use a conventional matched filter receiver structure that treats other user's signal as noise and simply correlates the received signal with desired user's spreading codes. While easy to implement, this receiver suffers from performance degradation caused by severe multiple access interference when there are a large number of users accessing the spectrum. To improve the performance and capacity it is desirable to apply so called "performance enhancing" techniques in the next generation wireless networks. Two performance-enhancing techniques in European and Japan's third generation (3G) systems standard use advanced receiver structures and smart antenna arrays. Advanced receiver structures like multi-user receiver and interference cancellation exploit structure in other users' signaling to mitigate or cancel the multiple access interference. Smart antenna arrays jointly process signals received from multiple antennas to enhance the desired user's signal and reduce interference from other users. While there exist many works that propose different multi-user receiver and antenna array structures, most of them focus on demonstrating the performance improvement in the physical layer in terms of lower bit error rate. They generally assume either no power control or some simple fixed power control scheme in which all the users' received power at the receiver is kept the same.

Furthermore, in recent research, several radio resource management algorithms for CDMA were proposed for the purpose of improving the capacity of cellular system, such as [6], [7], [8] and [11]. However, most of them are proposed for circuit switched system or single class traffic with same data rate and target bit error rate. The diversified QoS requirements of users in joint rate and power allocation were not taken into consideration. Issues like power and bandwidth resource allocation and system capacity for multi-class traffic are still unclear.

1.2 Problem to Solve

The objective of resource allocation in wireless networks is to decide how to allocate resources such as power, bandwidth and channel such that quality of service requirements of all the users' can be satisfied.

In this thesis we focus on solving a kind of resource allocation problem in multi-class traffic system. We also provide a solution to the call admission control problem in CDMA cellular systems. By doing so, the objective of this research is to propose and to evaluate the performance of effective RRM (Radio Resource Management) schemes for CDMA networks that can:

1. Provide services to heterogeneous traffic with a wide range of parameters, including power allocation and transmission rate;
2. Guarantee a wide range of QoS levels, including delay requirement and BER requirements, while maintaining a minimum resource cost.

3. Achieve high resource utilization at the network layer, i.e. support more connections in the system.
4. Improve the transmission efficiency of the system.
5. Controls the new connection admission in adaptation to power distribution in cellular networks.
6. Allocate power and bandwidth resource to multiple classes of traffic.
7. Be implemented with reasonable complexity in practice.

This problem is challenging because of the difficulties associated with supporting multimedia applications in wireless communications environment:

1. The wireless channel is time varying due to the multi-path fading effect, the resource allocation scheme needs to be adaptive.
2. The radio bandwidth is a scarce resource; the resource allocation scheme needs to be efficient.
3. To support multiple classes of traffic with different data rates and SIR requirements, the resource allocation scheme needs to be flexible.
4. Due to the limited power available in the handset, the resource allocation scheme should minimize transmit power.
5. Due to the significant multiple access interference in CDMA systems, it is desirable that the resource allocation scheme incorporates performance-enhancing techniques.

The RRM scheme jointly takes the factor in the physical layer, link layer and network layer into consideration to achieve effective and efficient resource allocation. The basic approach is summarized as follows.

The power distribution is a way to allocate power to all participating connections to maximize the number of simultaneously transmitting users subject to satisfaction of SIR specifications. The power from each mobile should be large enough to achieve the required minimize SIR for its type of connections in the system, but should be kept low enough in order to reduce MAI to other connections in the system, and to allow more active users in the system.

The power distribution is based on a given transmission rate from each participating connection. We then study allocating the transmission rate for each connection. The allocated transmission rate to each connection is determined by the SIR requirement. A buffer with limited size should be allocated to each real-time connection to temporarily store the delayed packets. The required packet level performance for each connection can be guaranteed, as long as the total load is within the system capacity. The packet transmission is based on the obtained power distribution and rate allocation, and is to guarantee the required QoS performance while improving the system efficiency.

Based on our algorithm, the connection admission control (CAC) scheme is developed to achieve an effective and admission control. The CAC scheme make an decision for each connection based on the power distribution and the rate allocation, so that once admitted, the connection can be served with guaranteed QoS performance through packet scheduling. We proposed three policies and the performance of three policies are compared.

The CAC scheme also incorporates the effect of users' rate by changing power so that once admitted, the connection can be serviced with guaranteed QoS throughout its lifetime.

1.3 Summary of Contributions

In this research, a novel Joint power distribution and rate schemes is proposed for cellular CDMA systems supporting multi-class services to achieve required SIR. These schemes can improve quality and capacity of the system and guarantee users' QoS requirements. First, we propose two optimization algorithms, Based on these architecture, three schemes are proposed enabling connection admission control in DS-CDMA cellular systems. An optimized allocation scheme aims at minimizing the total available power while meeting the minimum QoS requirement of users; another is to maximizing QoS for admitted users while meeting limitation of power for each user. The third scheme we propose is a simple fixed allocation scheme that acts a reference for the former two schemes. Computer simulations assuming a CDMA system using C programing indicate that the proposed QoS based allocation algorithms exhibit good performance with respect to blocking probability, system capacity, delay and overflow, which indicate higher satisfaction of users and better performance. Last, we find QoS based idea can also be used for relative billing policy. The concept is to charge more to users who consume more of the total system user capacity. Unlike fixed QoS systems, the effect of admitting a new user depends heavily on the QoS requirements of the new user. The schemes we proposed achieve effective and efficient radio resource utilization, which can benefit both mobile users and network service providers. It is more reasonable than traditional billing method. Therefore, we also include in the simulation the effect of billing policy.

The main contributions of this thesis are listed as follows.

- Optimized power and rate distribution schemes have been derived for CDMA systems providing services to multi-class traffic.

- A wide range of QoS levels is guaranteed, including delay requirement and BER requirements, while maintains a minimum resource cost.
- Performance for CDMA systems supporting multi-class services is derived, including New Connection Blocking Probability (NCBP), traffic intensity, queuing delay, etc.
- By comparing the three schemes we propose in the thesis, A solution to the connection admission control problem in CDMA cellular systems has been proposed by taking joint power and rate distribution into consideration. The scheme ensures that once a connection is admitted into the system, its required QoS is satisfied throughout its lifetime.
- In Minimum Total Power algorithm (MTP), by distributing only the necessary amount of power to each connection, maximum number of simultaneously transmitting users can be supported. This achieves high resource utilization at the network layer, i.e. support more connections in the system.
- The schemes can achieve guaranteed QoS performance at the packet level, and achieve better GOS performance at the connection level.
- A multi-class traffic is proposed.

Since some of the objectives are paradoxical, it is impossible to simultaneously optimize all of them. Our objective is to improve the system utilization subject to satisfying a constraint on the others.

1.4 Overview of the Thesis

The remainder of this thesis is organized as follows. Chapter 2 introduces the evolution of cellular communication and Code Division Multiple Access principle. The technologies about CDMA are briefly discussed. As we mainly focus on resource problem in CDMA, Multimedia CDMA Resource Allocations research is described in chapter 3. Chapter 4 studies the optimization theory we use and an optimized power distribution simulation is proposed. We define our system model and propose the CAC performance measurements with resource allocation in chapter 5. Simulation model and numerical results are demonstrated in chapter 6. The conclusion remarks, and future research work are listed in Chapter 7.

Chapter 2

Code Division Multiple Access

CDMA (Code Division Multiple Access) is an advanced digital wireless transmission technology. It is one of the driving forces behind the rapidly advancing personal communications industry. In this chapter, we presents an overview of concepts related to communication system and CDMA technology, describes the principal attributes and techniques. We also give a brief introduction to some of applications in the CDMA system.

2.1 Basic Concept in Mobile Communication System

The ability to communicate with people on the move has evolved remarkable since Guglielmo Marconi first demonstrated radio's ability to provide continuous contact with ship sailing the English Channel. Today, Wireless communications have shown a profound effect on our daily life. In less than ten years, cellular telephones have attracted more than a hundred million subscribers in the United States, Europe, and Asia. As larger capacity is needed due to the high growth rate of cellular mobile subscribers, a transition from analog to digital system is expected. Digital communication has been a popular solution. In this section, we introduce some basic knowledge about mobile communication.

2.1.1 Multiple Access Method

Wireless systems transmit and receive signals over a common resource: the air. This may lead to conflicts if several users want to transmit at the same time. Multiple access means that multiple, simultaneous users can be supported. In other words, a large number of users share a common pool of radio channels and any user can gain access to any channel (each user is not always assigned to the same channel). A channel can be thought of as merely a portion of the limited radio resource, which is temporarily allocated for a specific purpose, such as someone's phone call. A multiple access method is a definition of how the radio spectrum is divided into channels and how channels are allocated to the many users of the system.

The original system was called the Advanced Mobile Phone System, or AMPS. It is the system we use throughout North America. Similar systems, with slight variations, are Nordic Mobile Telephone (NMT) in Scandinavia, and Total Access Communications System (TACS) used in the United Kingdom, China, and other countries. Spectral allocations are in the 800-900 MHz region.

Several hundred channels are available within the spectrum allocation. One channel of one base station is used for each conversation. Upon handoff, the subscriber station is directed via messaging to discontinue use of the old channel and tune to the new one, on which it will find the new cell.

2.1.2 FDMA, TDMA and CDMA

In order to separate each user so that they do not interfere with one another, service providers may use one of the three primary multiple access systems: Frequency Division Multiple Access (FDMA), Time Division Multiple Access (TDMA), and Code Division Multiple Access (CDMA). Figure 2.1 below shows these three access methods.

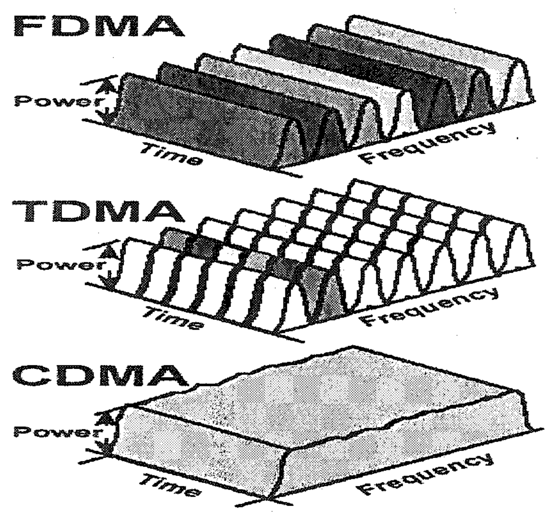


Figure 2.1: Multiple Access Methods

FDMA: Frequency Division Multiple Access

FDMA divides radio channels into a range of radio frequencies and is used in the traditional analog cellular system. With FDMA, only one subscriber is assigned to a channel at a time. Other conversations can access this channel only after the subscriber's call has terminated or after the original call is handed off to a different channel by the system. FDMA cellular standards include AMPS (Advanced Mobile Phone Service) and TACS (Total Access Communications System).

TDMA: Time Division Multiple Access

TDMA is a common multiple access technique employed in digital cellular systems. It divides conventional radio channels into time slots to obtain higher capacity. Hence, channelization of users in the same band is achieved through separation in time. Its standards include North American Digital Cellular, Global System for GSM (Mobile Communications), and PDC (Personal Digital Cellular).

CDMA: Code Division Multiple Access

CDMA assigns each subscriber a unique "code" to put multiple users on the same wideband channel at the same time. The codes, called "pseudo-random code sequences", are used by both the mobile station and the base station to distinguish conversations. The signals are separated at the receiver by using a correlator that accepts only signal energy from the desired channel. Undesired signals contribute only to the noise.

The IS-95 CDMA standard was adopted by the TIA (Telecommunications Industry Association) and became a digital cellular standard in 1992. The J-STD-008 standard for personal communications services was also accepted by ANSI. Depending on the level of mobility of the system, it provides 10 to 20 times the capacity of AMPS, and 4 to 7 times the capacity of TDMA. CDMA is the only one of the three technologies that can efficiently utilize spectrum allocation and offer service to many subscribers without requiring extensive frequency planning.

2.1.3 The Cellular Concept

For certain types of services the aim is to achieve full spatial coverage. In conventional wireless systems a mobile entity is linked to a base station (BS). BS is connected to a radio network controller, which uses additional interfaces that cater for the access to the public switched telephone network (PSTN). The principle structure of a cellular wireless system is shown in Figure 2.2. The signals on the air–interface experience a distance dependent attenuation.

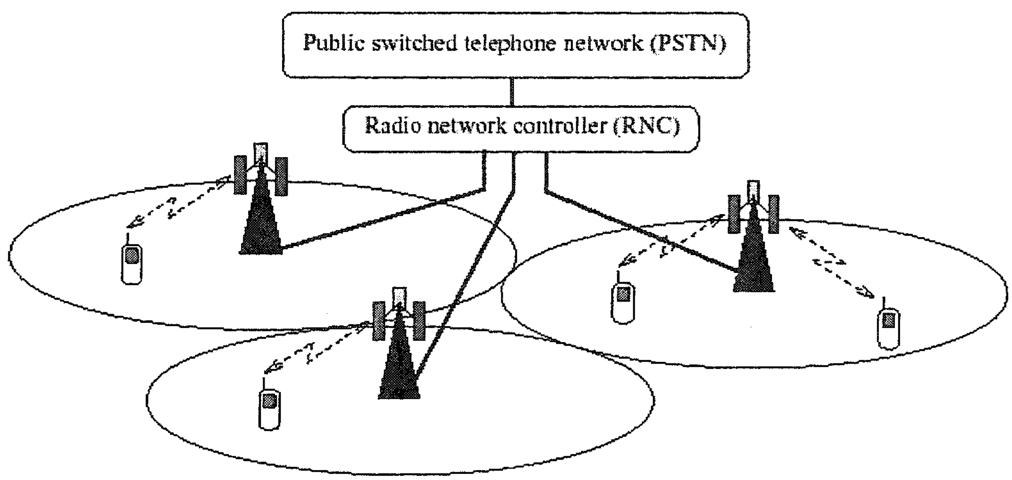


Figure 2.2: A cellular wireless system

Since the transmission powers are limited, the coverage area of a BS is limited, as well. Due to the radial signal propagation, in theory, a single BS covers a circular area. The area that is covered by a BS is also referred to as a cell. When modeling cellular systems, cells are approximated by hexagons as they can be used to cover a plane without overlap (tessellation) and represent a good approximation of circles. Since the total available radio resource is limited, the spatial dimension is used to allow wide area coverage. This is achieved by splitting the radio resource into groups. These groups are then assigned to different contiguous cells. This pattern is repeated as often as necessary until the entire area is covered. A single pattern is equivalent to a cluster. Therefore, a radio resource which is split into i groups directly corresponds to a cell cluster of size i. In this way it is ensured that the same radio resource is only used in cells that are separated by a defined minimum distance. This mechanism is depicted in Figure 2.3 (A group of radio resource units is indicated by a certain color). As a consequence the separation distance grows if the cluster size increases. Hence, increasing the cluster size acts in favor of low interference.

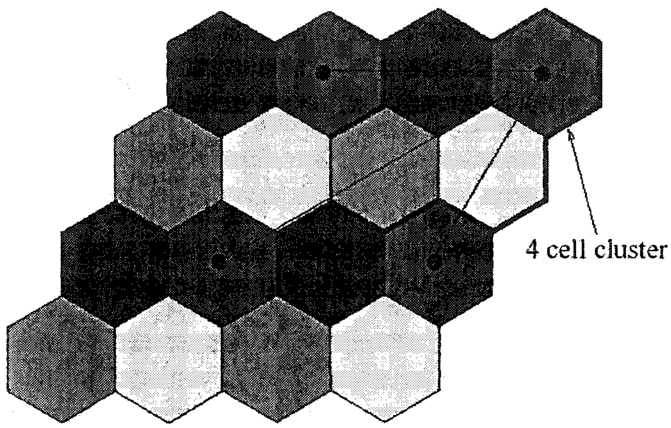


Figure 2.3: The cellular concept

However, an increased cluster size means that the same radio resource is used less often within a given area. As a result, fewer users per unit area can be served. Therefore, there is a trade–off between cluster size and capacity. In an ideal scenario the total available radio resource would be used in every cell while the interference was kept at a tolerable level. Herein lies a particular advantage of CDMA over all other multiple access modes since the same frequency carrier can be re–used in every cell. It is clear that this results in increased co–channel interference (CCI) which gradually reduces cell capacity, but the magnitude of the resulting reduction of spectral efficiency is usually less than would be obtained if a fixed frequency re–use distance was applied. The cell capacity, finally, is dependent on many system functions such as power control, handover, *etc.* that is why capacity in a CDMA system is described as soft–capacity. However, the fact that in a CDMA system frequency planning can be avoided may not only result in capacity gains, but it eventually makes CDMA a more flexible air interface.

2.1.4 Structure of a Typical Wireless System

The figure below is the structure of a typical wireless system. The MSC has wire connections to all its base stations; link the wireless communication network to fixed PSTN. The base station, working as an intermediate system, has radio link connections to mobile equipment. The main purpose of the base station is to process the received signal and to transfer information and control message between the mobiles and the MSC.

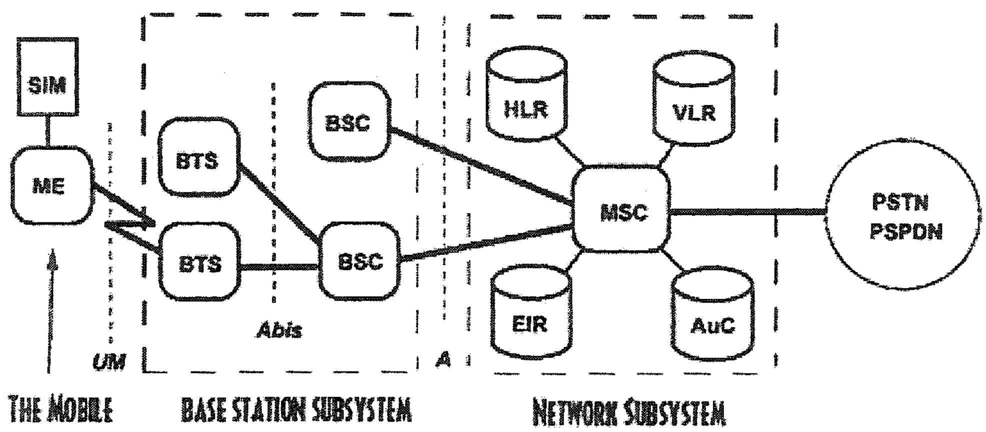


Figure 2.4: The structure of a typical wireless system

2.1.5 Modes of Channel Operation

There are three basic modes for operating a communication channel, namely: simplex, half duplex and full duplex. The basic mechanisms are depicted in Figure 2.5.

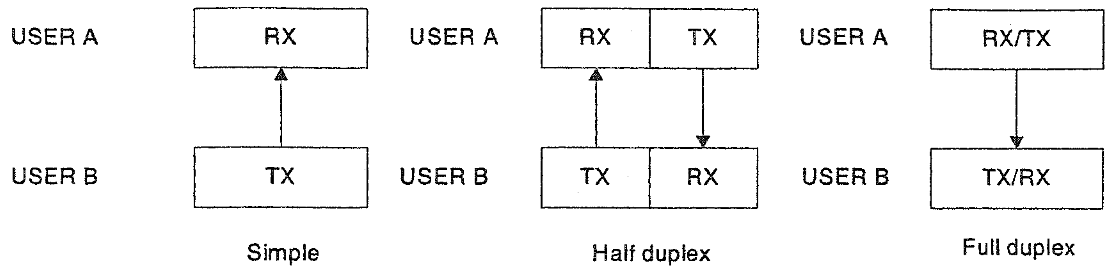


Figure 2.5: Modes of channel operation

In the case of a simplex communication the information is passed from one entity to another without permitting any acknowledgement (one way communication). Notable examples are television and radio broadcasting.

A half duplex channel can send and receive, but not at the same time. This means one entity transmits at a time while the other entity listens, and vice versa. A user A indicates when he wishes to terminate transmission giving the counterpart, user B in this case, the chance to talk. This leads to a 'ping–pong' type of communication. This technique is used in talk–back radio and CB (Citizen Band) radio where only one person can talk at a time. Note that access to the Internet merely requires a half duplex channel: consider user A sending a download request in principle, no further information needs to be transmitted and, thus, user A can go into the receive mode until all the required information is downloaded.

Information that travels in both directions simultaneously is referred to as a full duplex channel. Two entities can receive and transmit at the same time. Telephony is an eminent example from this category.

In wireless communication systems two methods are used to achieve a full duplex channel —time division duplex (TDD) and frequency division duplex (FDD). If the receive and transmit slots of a half duplex channel are repeated periodically in short intervals, a full duplex channel can be emulated by a half duplex channel. This is exactly the mechanism used in TDD. In contrast, an FDD system separates both directions in the frequency domain so as to eliminate cross–talk. This means the full duplex channel is accomplished by two independent simplex channels. The basic mechanism of TDD and FDD are shown in Figure 2.6.

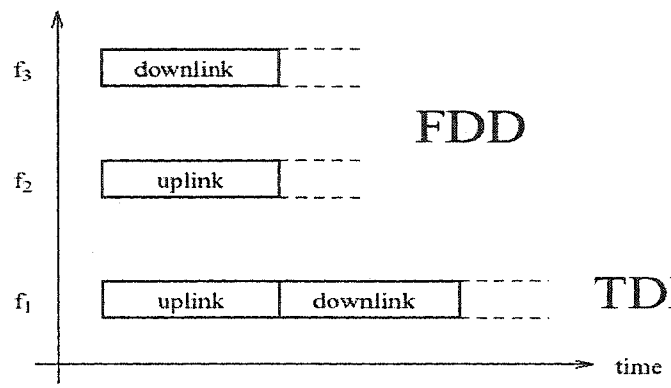


Figure 2.6: The principles of TDD as compared with FDD

In cellular communication, the direction from the BS to the MS is referred to as the downlink. Similarly, the direction from the MS to the BS is the uplink.

The advantage of FDD is that it represents a true full duplex channel that does not need any coordination between uplink and downlink transmission. The disadvantage is that two separated channels have to be maintained. Given that many new services do not require a full duplex channel (predominately data applications as illustrated by an Internet session), FDD offers more performance than would be required. In the case of a file download, for instance, the uplink channel is underused or even unused which results in the waste of expensive radio resources. In comparison, the TDD technique does not represent a true full duplex channel. It requires co–ordination (synchronization), but due to its nature, it ideally supports services that basically only require an asymmetric half duplex channel. Given that future wireless communication is evolving towards the wireless Internet, the significance of TDD will grow.

2.2 Evolution of Cellular Communication

Wireless communications have experienced an enormous amount of growth during the last two decades. In this section, we will review the evolution of cellular systems and highlight improvements from one generation to the other. Figure 2.7 shows a history of cellular communication evolution.

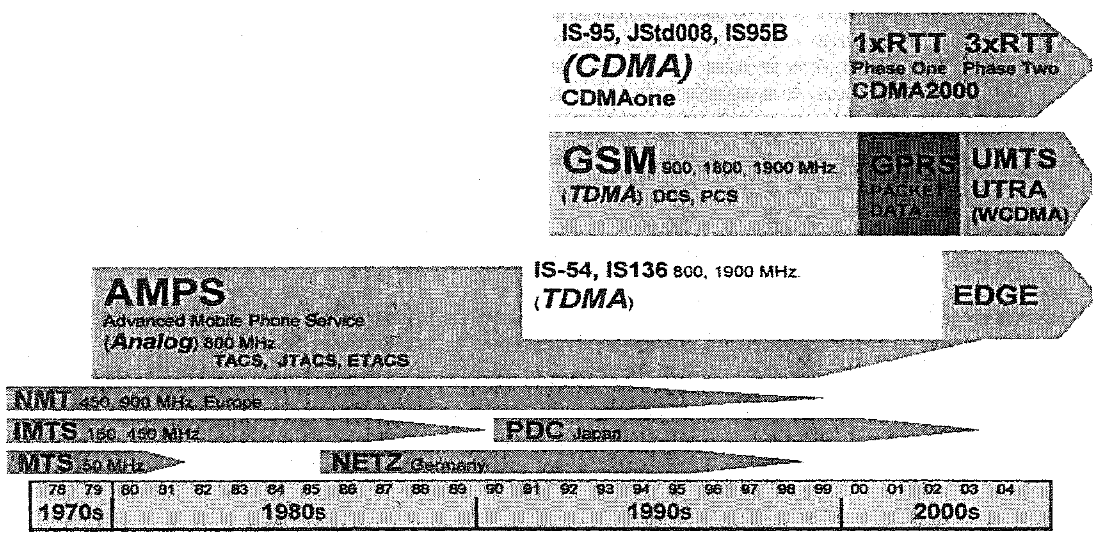


Figure 2.7: A history of cellular communication evolution

2.2.1 The First Generation (1G) Systems

The first-generation (1G) wireless communication systems that used analog transmission for speech services were introduced in early 1980s. At the beginning, cellular service providers were given 50 MHz of channel spectrum, divided by two systems (A&B), further divided by forward and reverse communication paths, effectively leaving just 12.5MHz each way per system. Since this spectrum is not expected to increase in foreseeable future, the cellular industry must look at different technologies to increase capacity without sacrificing quality of service.

The most popular one is 30kHz analog channel, which is known as FDMA. There were several types of analog cellular systems including the Nordic Mobile Telephone (NMT) [Nordic countries: Sweden, Norway, Finland and Denmark, 1981], the Advanced Mobile Phone Service (AMPS) [U.S., 1983], the Total Access Communications System (TACS) [UK, 1984], etc. All 1G cellular systems used FDMA method to achieve spectrum sharing among multiple users. To allow simultaneous transmission and reception, the BS transmits along one set of radio channels, called forward channels or downlink, and receives along another set of channels, which are reverse channels or uplink from the MT. Since the number of channels is limited by the allocated frequency spectrum of a system, a cellular system adopts the frequency reuse strategy to tackle this problem to increase the number of radio channels. As cellular systems get advanced, directional antennas are applied to sector a cell, so the interference can be minimized.

2.2.2 The Second Generation (2G) Systems

To meet the need of the increasing capacity of the cellular system and to establish compatibility with the evolution of wired networks towards digital systems, the second-generation (2G) wireless cellular systems based on digital transmission techniques were introduced in early 1990s. Digital cellular systems fall into three basic types of cellular technologies: frequency division multiple access (FDMA), time division multiple access (TDMA) and code division multiple access (CDMA).

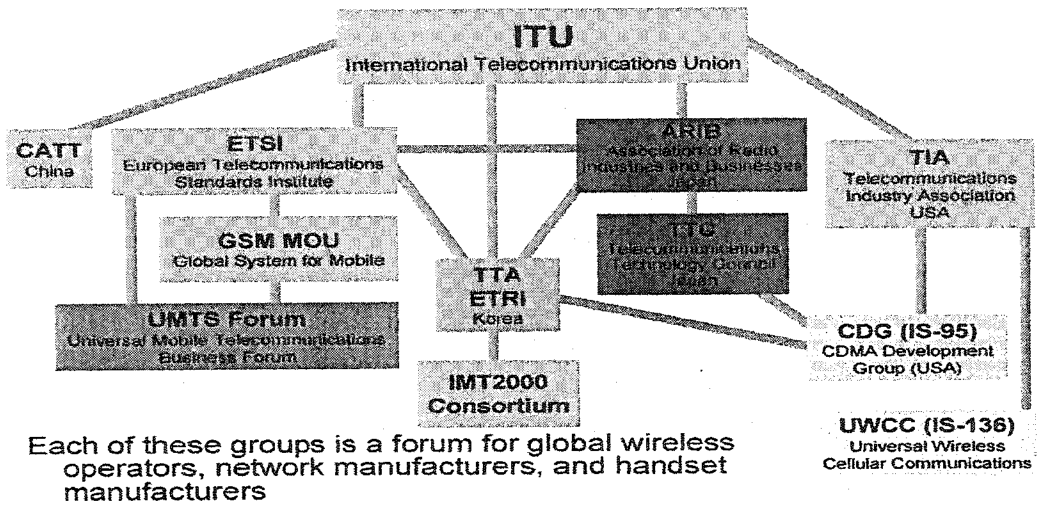


Figure 2.8: A simple structure of different research groups

In the TDMA system, the cell sequentially scans through the mobiles, each of which uses the same 30 kHz frequency band, but at different time slot. In this way, the number of cells does not increase. But since there are now more than one user per 30kHz channel, the total number of users per cell increases. The IS-136 system is sometimes referred to as Digital AMPS (DAMPS), it is the new generation of the TDMA system beyond the IS-54 dual-mode analog-digital system used in north American. IS-136 was led by the Universal Wireless Communications Consortium (UWCC) and Committee T1 (T1) sponsored by the Alliance for Telecommunications Industry Solutions and accredited by the American National Standards Institute. A simple structure of different research groups is shown in Fig.2.9. The Global System for Mobile Communications (GSM) developed in Europe is a mixed type of TDMA/FDMA system. The development of GSM was led by the GSM association and the European Telecommunications Standards Institute (ETSI). CDMA appeared in this period. It differs from FDMA and

TDMA systems through the use of coded radio channels. IS-95 (cdma One) is an example of the CDMA technology, which was originated by the CDMA Development Group (CDG) and Telecommunications Industry Association (TIA)[14]. The 2G digital system has many advantages over the 1G system in terms of capacity, quality, flexibility, security, and system complexity. With the demand of new innovative services in general, and wide-band multimedia services in particular, the currently deployed 2G wireless systems have further evolved towards the third generation (3G) systems to offer more advanced service features.

Some systems that extend the existing 2G systems are called 2.5G systems. The main feature of 2.5G systems is the data packet service enhancement. They are developed to bridge the 2G and the 3G systems. One example of the 2.5G systems is the General Packet Radio Services (GPRS) that can provide higher data-rate packet-switching services up to 115Kbps. This effort leads to Enhanced Data Rates for GSM Evolution (EDGE), which can support services with a data rate up to 384 Kbps. The other example is the step-by-step synchronous approach of CDMA2000 evolving from current IS-95 networks [2], led by the 3GPP2 effort in North America.

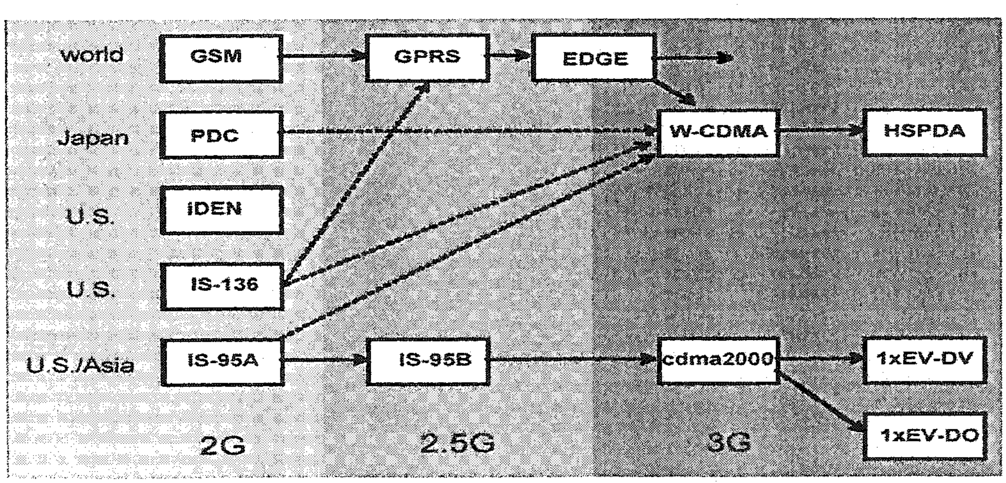


Figure 2.9: 2G to 3G Migration Paths

The shift to 3G goes through CDMA2000 1X and CDMA2000 3X. With this approach, the maximum transmission speed of 2G is 64Kbps in IS-95A/B specification while the maximum transmission speed of 2.5G is up to 384 Kbps and the maximum Transmission speed of 3G is up to 2Mbps.

2.2.3 The Third Generation (3G) Systems

Third-generation (3G) systems are critical to the wireless Internet services often touted as the future of mobile communications. It began work on a project called *IMT-2000* (International Mobile Telecommunication 2000) by ITU (the International Telecommunications Union). Under the IMT-2000 framework, the third generation (3G) air interfaces include CDMA or TDMA technologies will be developed [31]. IMT-2000 [15] will provide packet and circuit switched services with on-air data rates between 384 Kbps (for the wide-area coverage), and 2 Mbps (for indoor coverage). At first, they will offer permanent access to the Web, interactive video, and voice quality that sound more like a CD player than a cell phone. Many of their future applications are as yet unknown, with industry pundits saying that we will discover them as we go along.

There are two kinds of systems in *IMT-2000* technology, WCDMA that comes from Europe's GSM method and CDMA2000 that comes from North America's IS-95 method. Some manufactures of two systems are listed in Table 2.1. It is expected that CDMA2000 and WCDMA will compete for the 3G markets.

Key features of IMT-2000 [15] include the following:

- Bit rates up to 2 Mbps (high bit rate transmission);

- A variable bit rate to offer bandwidth on demand (rate adaptively);
- High quality requirements from 10% frame error rate to 10e-6 bit error rate (a wide range of QoS requirement);
- High spectrum efficiency (via radio resource management and other mechanisms);
- Multiplexing of services with different quality requirements (multimedia support);
- Support packet-based transmission.

IMT-2000 provides a framework for worldwide wireless access by integrating a diverse system consisting of both terrestrial and satellite networks. It also exploits the potential synergy between digital mobile telecommunication technologies and fixed wireless access (FWA) Systems. The detail introduction of CDMA standard is given in [2] and [29].

2.2.4 The Fourth Generation (4G) Systems...

Research on fourth generation (4G) mobile communication systems is already underway. Internet protocol (IP) can potentially become the universal network-layer protocol over all wireless systems as it already is for wire line packet networks. Currently, the 4G researches are mainly on three areas:

1. To provide higher data rate.
2. To achieve global roaming and horizontal communications between different access technologies.
3. To provide a common platform that can provide more advanced types of services.

In general, the rapid evolution of wireless mobile communications has been driven by i) the high demand of wireless communications and mobility, ii) the requirements of high quality applications, iii) the need of new applications, and iv) the technical development. In the near future, mobile users expect to enjoy the same set of services and high QoS that are seamless in both fixed and mobile environments. Future wireless systems are required to provide greater mobility and adequate QoS support as a user moves from place to place. In addition, future applications will demand more resource than ever. Therefore, resource allocation will play a vital role in future wireless communication system in delivering a target QoS and optimizing network resource utilization.

Regional Standards Body	Technology	Group
TIA (TR45.5)	Wideband cdmaOne	Lucent, Motorola, Nortel, Qualcomm
		HNS
		Nokia
		Samsung
		Hitachi
ETSI (SMG2)	W-CDMA W-TDMA/CDMA	Nokia, Ericsson, NEC, Panasonic, Fujitsu Siemens

Table 2.1: Some manufactures of 3G systems

2.3 CDMA

2.3.1 Introduction to Spread Spectrum Communications

Spread spectrum (SS) multiple access transmits the entire signal over a bandwidth that is much greater than that required for standard narrow band transmissions in order to gain signal-to-noise (S/N) performance. In channels with narrowband noise, increasing the transmitted signal bandwidth results in an increased probability that the received information will be correct. Because each signal is a compilation of many smaller signals at the fundamental frequency and its harmonics, increasing the frequency results in a more accurate reconstruction of the original signal. The effective drawback of narrowband data communications is the limitation of bandwidth; thus signals must be transmitted with enough power so the corruption by Gaussian noise isn't as effective and the probability that the data received is correct will remain low. This means that the effective SNR must be high enough so that the receiver can recover the transmitted code without error.

From a system viewpoint, the performance increase for very wideband systems is referred to as "process gain". This term is used to describe the received signal fidelity gained at the cost of bandwidth. Errors introduced by a noisy channel can be reduced to any desired level without sacrificing the rate of information transfer using Claude Shannon's equation describing channel capacity:

$$C = W \log_2 (1 + S/N) \qquad (2.1)$$

where **C** = Channel capacity in bits per second, **W** = Bandwidth, **S/N** = Energy per bit/Noise power.

The benefits of increasing bandwidth become clearer. The S/N ratio may be decreased without decreasing the bit error rate. This means that the signal may be spread over a large bandwidth with smaller spectral power levels and still achieve the required data rate. If the total signal power is interpreted as the area under the spectral density curve, then signals with equivalent total power may have either a large signal power concentrated in a small bandwidth or a small signal power spread over a large bandwidth.

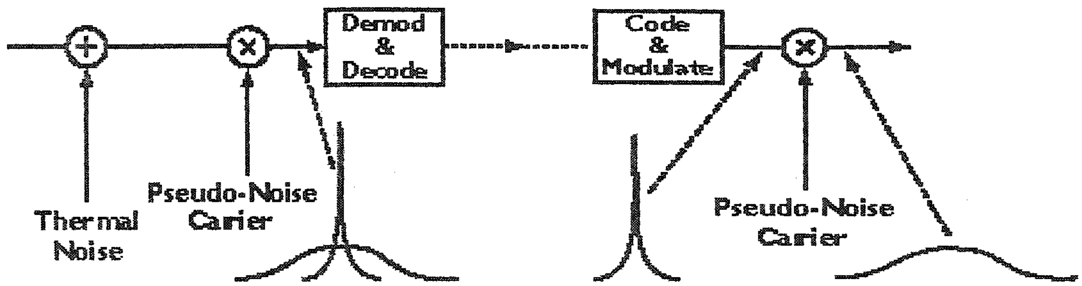


Figure 2.10: Modulation of a CDMA spread spectrum signal

A CDMA spread spectrum signal is created by modulating the radio frequency signal with a spreading sequence (a code consisting of a series of binary pulses) known as a pseudo-noise (PN) digital signal, because they make the signal appear wide band and "noise like". The PN code runs at a higher rate than the RF signal and determines the actual transmission bandwidth. Messages can also be cryptographically encoded to any level of secrecy desired with direct sequencing.

A SS receiver uses a locally generated replica pseudo noise code and a receiver correlator to separate only the desired coded information from all possible signals. A SS correlator can be thought of as a specially matched filter -- it responds only to signals that are encoded with a pseudo noise code that matches its own code. Thus a SS correlator (SS signal demodulator) can be "tuned" to different codes simply by changing its local code. This correlator does not respond to man made, natural or artificial noise or interference. It responds only to SS signals with identical matched signal characteristics and encoded with the identical pseudo noise code.

Many spread spectrum radios can share the same frequency band, provided that each system uses a unique spreading code to reduce interference between the different radios. Because only the receiver with the identical code can despread the signal to recover the signal, SS radios can tolerate a high level of interference unlike conventional radios, providing much greater capacity increase in frequency reuse. Since many users can share the same spread spectrum bandwidth without interfering with one another, SS systems become bandwidth efficient in multiple user environments. This reason makes SS communication an ideal choice for metropolitan areas with large blocking rates. Frequency reuse is universal, that is, multiple users utilize each CDMA carrier frequency.

The spread of energy over a wide band, or lower spectral power density, makes SS signals less likely to interfere with narrow band communications, because the spread signal power is near that of Gaussian noise levels.

In general, Spread Spectrum communications is distinguished by three key elements:

1. The signal occupies a bandwidth much greater than that, which is necessary to send the information. This results in many benefits, such as immunity to interference and jamming and multi-user access, which we'll discuss later on.

2. The bandwidth is spread by means of a code, which is independent of the data. The independence of the code distinguishes this from standard modulation schemes in which the data modulation will always spread the spectrum somewhat.

3. The receiver synchronizes to the code to recover the data. The use of an independent code and synchronous reception allows multiple users to access the same frequency band at the same time.

In order to protect the signal, the code used is pseudo-random. It appears random, but is actually deterministic, so that the receiver can reconstruct the code for synchronous detection. This pseudo-random code is also called pseudo-noise (PN).

2.3.2 Three Types of Spread Spectrum Communications

There are three ways to spread the bandwidth of the signal:

- Frequency hopping. The signal is rapidly switched between different frequencies within the hopping bandwidth pseudo-randomly, and the receiver knows before hand where to find the signal at any given time.
- Time hopping. The signal is transmitted in short bursts pseudo-randomly, and the receiver knows beforehand when to expect the burst.
- Direct sequence. The digital data is directly coded at a much higher frequency. The code is generated pseudo-randomly, the receiver knows how to generate the same code, and correlates the received signal with that code to extract the data.

CDMA is a form of Direct Sequence Spread Spectrum communications.

2.3.3 Direct Sequence Spread Spectrum

CDMA is a Direct Sequence Spread Spectrum system. The CDMA system works directly on 64 kbit/sec digital signals. These signals can be digitized voice, ISDN channels, modem data, etc.

Figure 2.11 shows a simplified Direct Sequence Spread Spectrum system. For clarity, the figure shows one channel operating in one direction only.

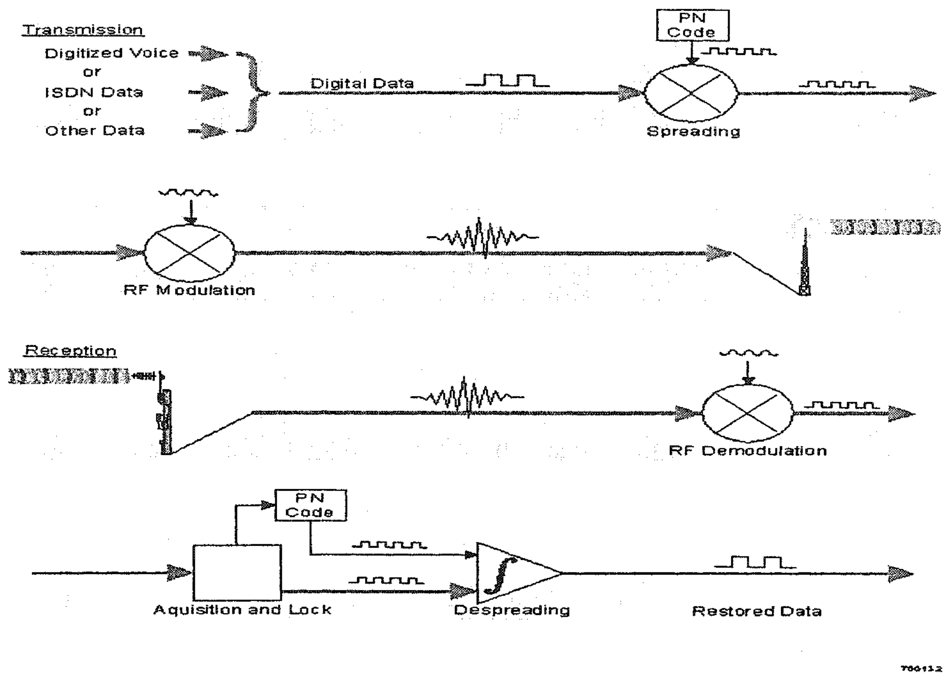


Figure 2.11: A simplified Direct Sequence Spread Spectrum system

Signal transmission consists of the following steps:

1. A pseudo-random code is generated, different for each channel and each successive connection.
2. The Information data modulates the pseudo-random code (the Information data is "spread").
3. The resulting signal modulates a carrier.
4. The modulated carrier is amplified and broadcast.

Signal reception consists of the following steps:

1. The carrier is received and amplified.
2. The received signal is mixed with a local carrier to recover the spread digital signal.
3. A pseudo-random code is generated, matching the anticipated signal.

4. The receiver acquires the received code and phase locks its own code to it.
5. The received signal is correlated with the generated code, extracting the Information data.

2.3.4 CDMA Concepts

CDMA stands for "Code Division Multiple Access." It is a radically new concept in wireless communications. It is a form of spread-spectrum, an advanced digital wireless transmission technique. The core principle of spread spectrum is the use of noise-like carrier waves, and, as the name implies, bandwidths much wider than that required for simple point-to-point communication at the same data rate.

Instead of using frequencies or time slots, as do traditional technologies, CDMA uses mathematical codes to transmit and distinguish between multiple wireless conversations. Its bandwidth is much wider than that required for simple point-to-point communications at the same data rate because it uses noise-like carrier waves to spread the information contained in a signal of interest over a much greater bandwidth. However, because the conversations taking place are distinguished by digital codes, many users can share the same bandwidth simultaneously.

Old-fashioned radio receivers separate stations or channels by filtering in the frequency domain. CDMA receivers do not eliminate analog processing entirely, but they separate communication channels by means of a pseudo-random modulation that is applied and removed in the digital domain, not on the basis of frequency. This universal frequency reuse is not fortuitous. On the contrary, it is crucial to the very high spectral efficiency that is the hallmark of CDMA.

Since all cells in CDMA use the same frequency, it is possible to make the connection to the new cell before leaving the current cell. As the mobile nears the boundary of a neighboring cell, it receives transmissions from both cells. The mobile will receive some message from one cell, and some from the other until it has moved into one or the other cells. This is known as a 'make-before-break" or "soft handoff" because the user never experiences any glitch and certainly never a dropped call. Soft handoffs require less power, which reduces interference and increases capacity.

When implemented in a cellular telephone system, CDMA systems provide operators and subscribers with significant advantages over analog and conventional TDMA-based systems. The main advantages of CDMA are as follows:

- Increased capacity
- Improved voice quality, eliminating the audible effects of multipath fading
- Enhanced privacy and security
- Improved coverage characteristics which reduce the number of cell sites
- Simplified system planning reduces deployment and operating costs
- Reduced average transmitted power, thus increasing talk time for portable devices
- Reduced interference to other electronic devices
- Reduction in the number of calls dropped due to handoff failures
- Development of a reliable transport mechanism for wireless data communications
- Coexistence with previous technologies, due to CDMA and analog operating in two spectrums with no interference

CDMA changes the nature of the subscriber station from a predominately analog device to a predominately digital device. Commercial applications became possible because of two evolutionary developments. One was the availability of very low cost, high density digital integrated circuits, which reduce the size, weight, and cost of the subscriber stations to an acceptably low level. The other was the realization that optimal multiple access communication requires that all user stations regulate their transmitter powers to the lowest that will achieve adequate signal quality. The advanced methods used in commercial CDMA technology improve capacity, coverage and voice quality, leading to a new generation of wireless networks.

The world's first CDMA network began commercial operation in Hong Kong in September 1995. This was followed in 1996 by the launch of two public CDMA systems in Korea. These three networks were closely monitored by the cellular industry to see if the promises of the new technology could really be delivered. By the middle of 1997, with one and a half million users being supported on these systems, it was clear that most of the questions had been satisfactorily answered. CDMA is now fully accepted as a mainstream cellular standard and the number of systems is rapidly rising.

Qualcomm has been at the forefront of CDMA with its IS-95 technology forming the basis for most current networks. However, other variations of CDMA are being proposed and trialed for future services. Some of these are aimed at wireless local loop applications for developing countries; others will provide support for wireless connected broadband data applications. CDMA has emerged as the dominant technology for the new PCS networks in the USA. Japan has adopted CDMA as the answer to its long term cellular capacity. In the UK, Vodafone has begun trials that will integrate a CDMA air interface into a GSM network structure. Overall, forecasts for future CDMA subscriber numbers and market shares are being constantly revised upwards.

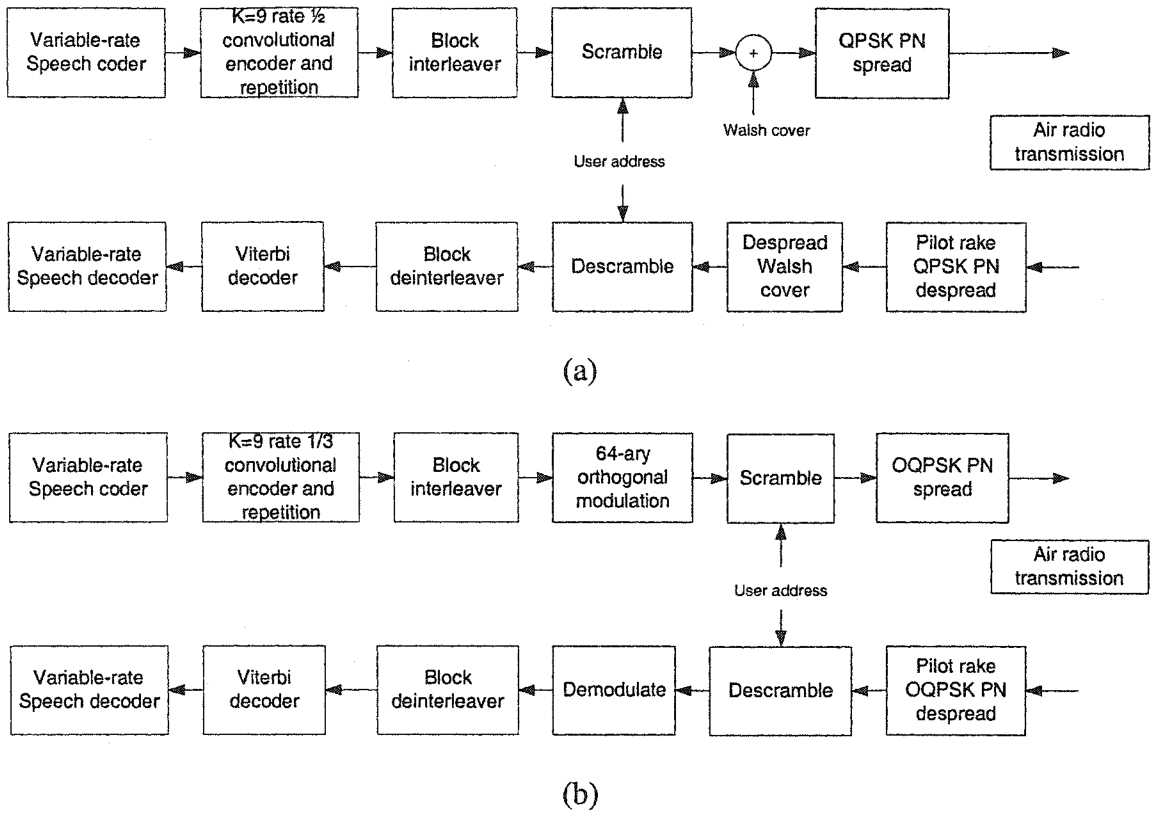


Figure 2.12: Conceptual block diagram of IS-95 (a) forward link and (b) reverse

2.3.5 North American DS-CDMA Digital Cellular System (IS-95)

As noted earilier, in the United States and in many other parts of the world, an ever-increaseing demand has been placed on the resources of the existing analog cellular systems [11]. In June 1989, Qualcomm Inc. proposed the use of DS-CDMA technology as a means of overcoming the expected capacity limits of analog systems. In the following years, Qualcomm has further developed the idea of a DS-CDMA digital cellular system into a actual implementation that has now been accepted as an Interim Standards body (IS-95) by the TIA, an ANSI-approved standards body, as a standard accepted for deployment in North America at 800MHz. The IS-95 DS-CDMA personal communication systems (PCS) frquency band of 1850 to 1970 MHz which the Federal

Communication Commission released for PCS applications in september 1993. at the present time these include Qualcomm, OKI, Motorola, Nokia, and Sony for subscriber equipment and Motorola and AT&T for base equipment. Conceptual block diagrams of the differences between the forward link and reverse link are shown in Figure 2.12 a and b, respectively.

Chapter 3

Multimedia CDMA Resource Allocations

With the increasing demand for wireless communication services and more stringent service quality requirements, new transmission technologies and improved radio resource management are required for wireless communication systems. In this chapter, we will explain the basic ideas of resource management in cellular radio systems. This introduction will lead to the resource management problems that we will investigate in this thesis.

3.1 Challenges of Radio Resource Management

The objective of a cellular communication system is to provide communication services anywhere and anytime and also to maintain connection quality. The mobility of the users in cellular systems causes new problems that do not occur in wired telephone systems.

When the number of users increases, a more careful resource allocation should be considered. In general, in the uplink, system assigns a channel and required transmitted power to a user. However, restricted radio bandwidth limits the number of users that can have acceptable connections.

A user's access to a cellular communication system can be divided to two stages, the call set up stage and the maintenance stage. As show in figure 3.1, the call set up stage occurs when there a user tries to obtain the resources necessary for a radio link connection. The maintenance stage occurs when there is no new user coming but due to the user motion each user will change his share of resources.

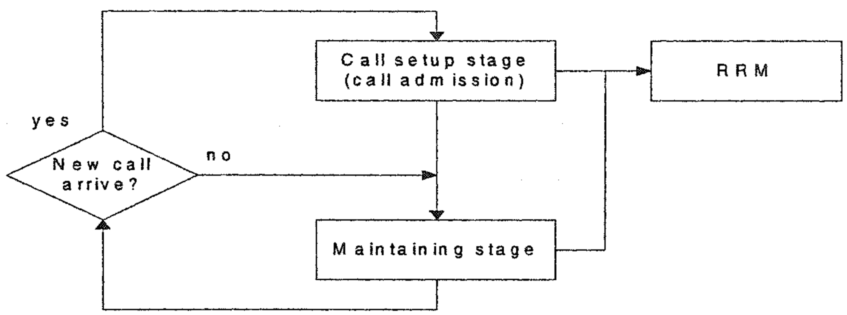


Figure 3.1: Two stages RRM

In recent research, several radio resource management algorithms for CDMA were proposed for the purpose of improving the capacity of cellular system. However, most of them are proposed for circuit switched system or single class traffic with same data rate and target bit error rate. The diversified QoS requirements of users in joint rate and power allocation were not taken into consideration. Issues like power and bandwidth resource allocation and system capacity for multi-class traffic are still unclear.

3.2 Radio Resource Management in CDMA

With more and more QoS requires and multimedia support in the next generation cellular system, such as packet dalay/ jitter requirements, bit error rate (BER), and signal-to-interference rate (SIR), the radio resource management (RRM) becomes a crucial task to efficiently utilize the limited radio resources while guaranteeing the required QoS performance for mobile users. RRM is the process of developing decisions and taking actions to optimize the system utilization. Due to the unique feature that in CDMA system users are sharing the allocation bandwidth and competing for the bandwidth by adjusting their power level at the receiver, the problem can also be formulated from a resource allocation point of view. The objective of resource allocation in wireless network is to decide how to allocate resource such as power, bandwidth and channel such that quality of service requirements of all the users' can be satisfied.

The resource allocation issues in CDMA networks include those related to some problems. In a wide sense, Technical challenges in RRM include two aspects:

1. How to design a network infrastructure such that the system capacity is maximized? This includes waveform design, BS and antenna deployment, modulation and coding schemes, etc.
2. Given a certain infrastructure design, how should the wireless resource be allocated to meet the QoS requirements of the mobile users while accommodating more mobile users in the network?

The first aspect has been studied extensively. The RRM research in this thesis is based on a given network infrastructure and therefore emphasizes on the second aspect. The aim of the RRM is to provide guaranteed QoS for heterogeneous services, and to efficiently utilize the precious radio resource. While the cellular CDMA offers a lot of advantages, such as high capacity, high spectrum utilization, it also brings a lot of challenging issues when supporting multimedia services. We will discuss these in the following.

3.2.1 Multimedia Services

Future wireless cellular communication networks are expected to support heterogeneous traffic with a wide range of transmission rates, BER requirements, and delay/jitter requirement. For example, voice and video are real-time traffic and cannot tolerate long delay; while most data traffic can tolerate large delay, they are very sensitive to transmission errors. Different amounts of resource should be allocated to different types of traffic in order to satisfy their unique QoS requirement. Services requiring higher QoS require more system resources. Since QoS corresponds to the system capacity, heterogeneous services make the system capacity study a complex issue. In cdma, the resource sharing property opens the opportunity to flexibly multiplex traffic and the available resource, and the potential of efficient resource utilization. How to coordinate the different QoS requirements for different types of traffic coexisting in the same system is a very challenging problem.

3.2.2 Limited Radio Resource

The available radio frequency bands are limited due to technical limitation. The usage of the available radio frequency bands is also regulated and coordinated by special organizations. Unlike wire line system, which can be added simple by adding more devices, the available radio frequency in wireless network is very limited and precious. Radio transmission experience path loss and is range-limited. Since signals are transmitted in open space, transmission from different users at the same time interface with each other if they are at the same frequency bands. Higher transmission power from one user increase the interference to other users. Too much interference reduces the SIR and increases the BER. Therefore, wireless systems are also power limited or interference limited, and the radio resource should be utilized efficiently. For a given amount of bandwidth, an efficient RRM should admit into the system as many as connections as possible, and achieve high packet throughput for the admitted connections. The efficiency is achieved subject to the effectiveness. In practice, however, the system resources can never be fully utilized due to intolerably high connection blocking probability.

3.2.3 Capacity Analyses

Third generation mobile communication standards such as CDMA2000 are aiming to provide multimedia services and support a number of different classes of users with different data rates and QoS requirements. These characteristics affect multiple access interference and hence capacity.

In a CDMA system the quality of a link (BER) and the total system capacity depend largely on the total interference in the system [18]. Thus the number and the type (QoS) of the interference are crucial system parameters. High data rate users require greater BER and thus greater SIR. They achieve this by transmitting at a higher power. This in turn implies higher level of interference and consequently a decrease in capacity in the network.

For a single cell CDMA system, suppose all users have the same traffic to transmit, and each user's transmitted power is perfectly controlled so that all signals are received at the BS at equal power levels. If the received signal power of each user is P_i, and the background noise is additive white noise, the following inequality holds for the uplink:

$$SIR_i = \frac{W}{R_i} \frac{d_i^{-\gamma_i} P_i \alpha_i}{\sum_{j=1, j\neq i}^{n} P_j \alpha_j + N} \tag{3.1}$$

Where the factors α_j are introduced to accommodate normalization constants and other factors, such as the effects of beam forming in multi antenna systems. N is power of the background noise, W is the spread spectrum bandwidth, R_i is transmission rate. n is the number of total supportable users. The maximum value of n is the uplink capacity.

3.2.4 Multi-rate Schemes

Different from the first two generations of wireless networks that were mainly designed for providing voice and low rate data services, the third generation wireless network, such as CDMA2000 is to provide high rate integrated services. Radio resources are scarce and the demand for wireless services keeps increasing, hence the efficient management of the radio resources in wireless networks is important in achieving a high level of utilization. Application such as Internet access and video can thus be supported over wireless. It is expected that services will have significantly differing characteristics from the current voice-dominated wireless networks [10]. Already, the demand for various services with different QoS requirements such as video and data is increasing.

Since CDMA is the most important candidate for 3G networks, the focus of this thesis is on how to provide Quality of Service for multi-rate services in CDMA network.

Multi-rate design means that different services with different quality of service requirements are multiplexed together in a flexible and spectrum-efficient way. The provision of variable data rates with different quality of service requirements can be divided into three sub-problems:

- How to map different bit rates into the allocated bandwidth;
- How to provide the desired quality of service;
- How to inform the receiver about the characteristics of the received signal.

The first problem concerns issues like multi-code transmission and variable spreading. The second problem concerns coding schemes, and the third problem concerns control channel multiplexing and coding.

In addition to the above basic requirements, the design of a multi-rate solution needs to consider a number of other requirements and constraints. The uplink and power amplifier requirements should not be too stringent, in order to facilitate use of power-efficient power amplifiers. In the downlink, the multi-rate solution should allow the mobile station receiver to save processing power.

A dynamic rate allocation scheme is proposed in [26] to maximize the system transmission throughput while guaranteeing the required QoS performance. SIR and rate allocation for CDMA data users on the forward link is studied in [20].

3.2.5 Accommodation of Higher Data Rates

In a DS-CDMA system, Multi-Code and OVSF schemes have been proposed to achieve variable data rates. In Multiple-SF scheme, the spreading ratio is reduced as the data rate increases. In a multi-code scheme, additional parallel codes are allocated as the data rate increased. It is also possible to have combinations of these. Multi-Code CDMA achieves higher data rates by assigning multiple orthogonal codes of constant spreading factor. In present generation DS-CDMA systems such as IS-95 each user is assigned a single orthogonal spreading code. In order to support higher and variable data rates 3G proposes using Orthogonal Variable Spreading Factor (OVSF) codes. Shorter codes with lower spreading length are assigned as the data rate increases. In this way variable data rate is achieved while keeping the chip rate constant.

In the selection of a multi-rate transmission scheme, the following criteria need to be considered:

- Performance in the presence of multi-path;
- Multiple access interference characteristics;
- Complexity of the receiver and transmitter;
- Forward link orthogonality;
- Power control requirements.

3.2.5.1 Multi-rate CDMA and Multi-code

Multi-code CDMA is adopted to support multi-rate transmissions because it has some advantages. The multi-modulation CDMA degrades the performance for the users with high data rates. When multiprocessing gain CDMA is used, users with very high source rate have very small processing gain, because the chip rate is constant for all the users. Therefore, both multi-modulation CDMA and multiprocessing gain CDMA are biased against high data rate users, and the higher is the transmission rate, the lower is the communication quality. MC-CDMA is expected to work well with multimedia traffic. Because of its unified architecture, when it integrates multimedia traffic, traffic streams with significantly different transmission rates can be easily integrated with all the transmission channels having the same bandwidth and spectrum processing gain.

In an MC-CDMA system, all the data signals over the radio channel are transmitted at a basic rate, R. Any connection can only transmit at rate Mr, referred to as m-rate, where m is a positive integer. Fig 3.2 shows the block diagram of an MC-CDMA transceiver. When an MS need to transmit at m-rate, it converts its data stream, serial-to-parallel, into m basic-rate streams. It first spreads each basic rate stream with a different code and superimposes them before spread spectrum modulation. The multiple simultaneously transmitted packets from the same connection is first spread by a set of orthogonal codes generated by the so called "sub-code concatenation", and then spread by a long code B, which an MS uses to indicate the BS that it is communicating with. The sub-code concatenation generated codes are unique to each mobile, while the long code is unique to each radio cell.

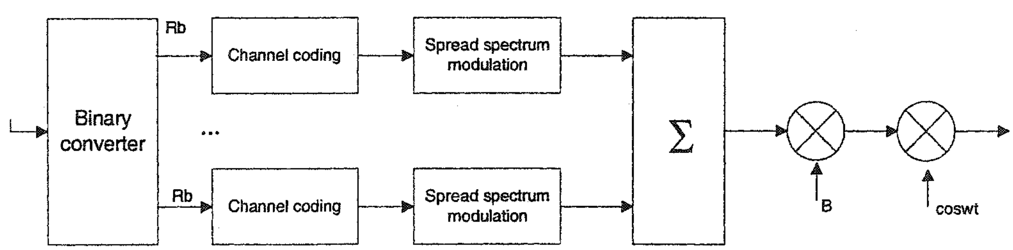


Figure 3.2(a): MC-CDMA transmitter

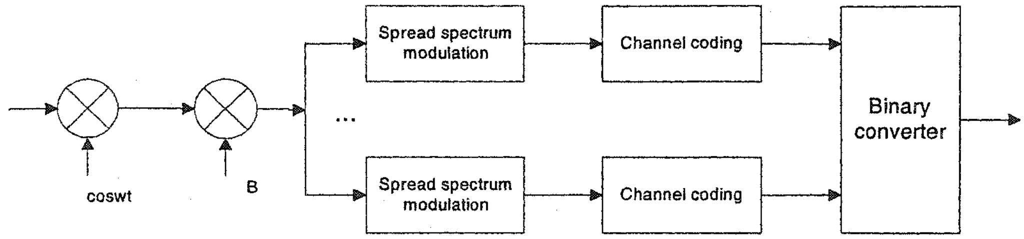


Figure 3.2(b): MC-CDMA receiver

3.2.5.2 Multi-Rate CDMA of OVSF Codes

The third generation (3G) wireless standards UMTS/IMT-2000 use the wideband CDMA (WCDMA) to support high data rate and variable bit rate services with different quality of service (QoS) requirements. In WCDMA, all users share the same carrier under the direct sequence CDMA (DS/CDMA) principle. In the 3GPP specifications [29], orthogonal variable spreading factor (OVSF) codes are used as channelized codes for data spreading on both downlink and uplink. OVSF codes also determine the data rates allocated to calls. Because OVSF codes require a single RAKE combiner at the receiver, they are preferable to multiples of orthogonal constant spreading factor codes, which need multiple RAKE combiners at the receiver.

The migration towards the 3G wireless systems is driven by the need to provide multimedia services in a mobile environment. Such services tend to generate variable rate traffic, which is drastically different from the conventional voice traffic in today's second-generation systems. OVSF codes preserve orthogonality between channels of different rates and spreading factors. The spreading Factor SF (also called Spreading gain) is the ratio of the signal bandwidth, or equivalently, the ratio of the time duration of a data bit to the time duration of a chip (chips per bit). The spreading factor is controlled by increasing or decreasing the time duration of the data bit, while maintaining a constant chip rate. This is achieved by using variable length spreading codes. The supported data rate of each user varies inversely with the length of the spreading code. OVSF codes are generated by a modified Hadamard transformation [33], which is represented by the tree structure as shown in Figure 3.3. Each level in the code tree corresponds to particular spreading factor. In Wideband CDMA standard the spreading factors range from 4 to 256. Codes at the same level are orthogonal however the codes on the same path or the in the sub-tree below a specific code assigned are not orthogonal. For example in Fig.3.3, C4.1 is orthogonal to every code except C2.1, C8.1 or C8.2.

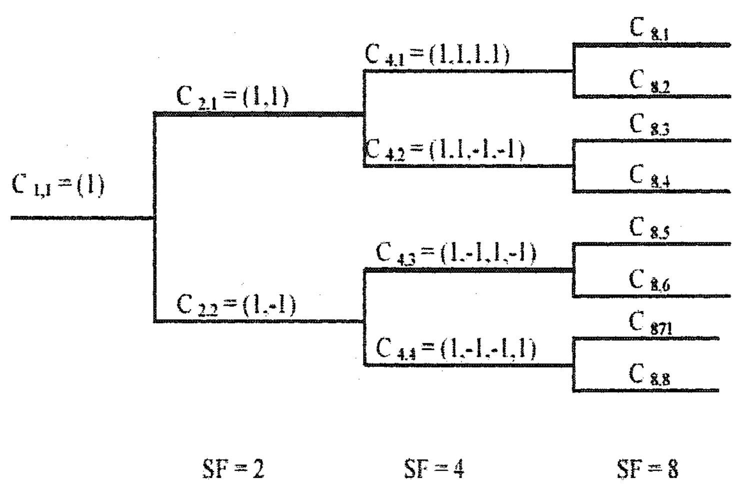


Figure 3.3: Generator Tree for Orthogonal Hadamard Codes

The cross correlation and auto correlation properties of the codes to be used in a CDMA system are important to be considered. The OVSF codes have the property that at zero lag, there is zero cross correlation between the codes while in case of non-alignment, OVSF codes have a significant cross correlation. Orthogonal codes are known to have poor correlation properties and this is shown in Fig 3.4(a) and Fig 3.4(b) below.

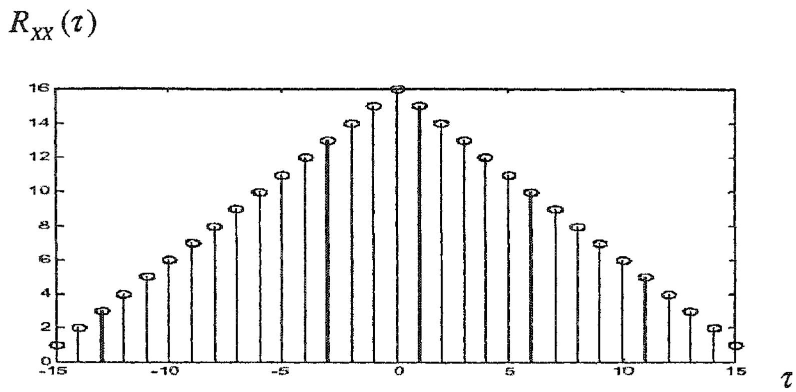


Figure 3.4(a): Auto-Correlation of Orthogonal codes with SF = 16

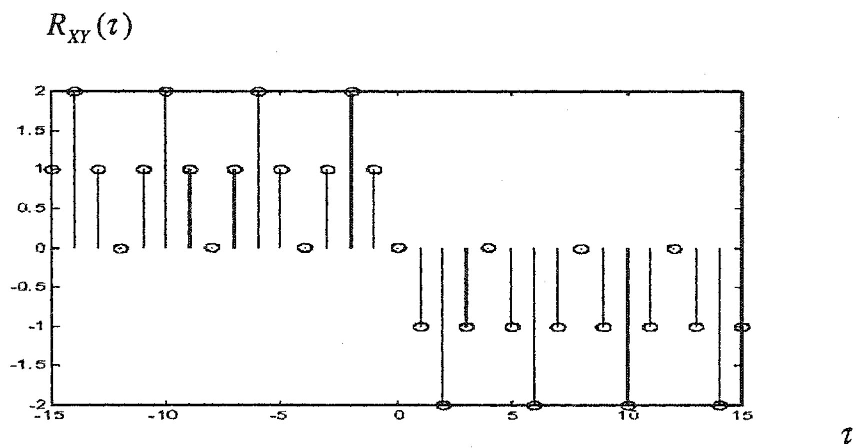


Figure 3.4(b): Cross-Correlation of Orthogonal codes with SF = 16

3.2.6 Power Control

Without very good power control, in direct-sequence and certain types of frequency hopping CDMA cellular system, multiple access communication is impossible. Assume, for example, that a cellular system has two mobiles. Each try to communicate with the BS with the same transmit power. One mobile is located near the base station, the other near the cell boundary. Considering the path-loss equations, a 60dB or more signal power difference at the base station between the two mobiles is quite possible. At the base station, the near-in mobile appears as a wideband jammer with 60dB more power than the far-out jammer. Unless very wideband spread is used, the base station will never receive the far-out mobile. This is the familiar near-far problem of spread-spectrum multiple-access communication systems. The close users have stronger signals and would cause unacceptable interference. The solution to overcome the near-far problem is the use of stringent power control, so that the received power for all the connections are the same in order to achieve the same QoS.

In CDMA systems with heterogeneous traffic, power control could also be used to achieve a target power for each connection in order to limit the MAI interference to other users and to achieve its required SIR.

In 3G WCDMA and CDMA2000 systems, the reverse link power control includes the open loop power control and the closed loop power control. The open power control is based on the principle that a mobile closer to the Bs needs to transmit less power as compared with a mobile that is far away from the BS. The mobile adjusts its transmit power based on the received power level from the BS. If the received power is high, the mobile reduces its transmit power. The BS is not involved in the process. The reverse link closed loop power control consists of two parts: the reverse inner loop power control and reverse outer loop power control. The inner loop power control keeps the mobile as close to its target signal-to-interference ratio (SIR) as possible (noise is neglected compared with interference), and the outer loop power control adjusts the target SIR for a given mobile. The target SIR may vary with vehicle speed and RF environment. The relatively fast variations associated with Rayleigh fading may at times be too rapid to be tracked by the closed-loop power control, but variations in relative path losses and shadowing effects is generally slow enough to be controlled. Also, while Rayleigh fading may not be the same for forward and reverse links. Lognormal shadowing normally will exhibit reciprocity.

Ideally, if perfect power control can be achieved, the required SIR for each user can be guaranteed by accurately distributing the received power for all the users, as long as the traffic load is within the system capacity. In reality, however, due to channel estimation errors, power control delay, power control command transmission errors, etc. imperfection in power control always exists. Due to the randomness of power control errors, it is unlikely to achieve the target SIR exactly at the receiver and communication outage occurs occasionally. The outage probability is defined as the probability that the actual received SIR is below the minimum required SIR of a connection. Capacity of CDMA system under imperfect power control has been analyzed in [13] for homogeneous voice traffic.

3.2.7 Power Distribution

Power as the common resource makes CDMA very flexible in handling mixed services. By allocation a different amount of power to each user according to its unique traffic parameters and QoS requirements, multiplexing of services with very different characteristics can be achieved in an efficient way [25]. Much work has been performed in the area of designing QoS-based power distribution algorithm for finding the minimum power or maximizing the system capacity for given SIR required by users. The goal of power distribution is to guarantee their QoS requirements in terms of data rate and bit error rate.

Power distribution for data traffic only and for intergrated voice and data is studied in [35], where different transmission rates are supported by variable processing gain.

3.2.8 Admission Control

The connection admission control (CAC) is to make a decision about whether a new or handoff connection should be admitted into a system according to the current system traffic load and the QoS requirements of the new/handoff connection. The admission decision is critical to both the system and the users, since admitting a connection that should not be admitted results in overload of the system, and the QoS of all the current connections would be deteriorated. The call admission control that incorporates rate assignment and power distribution is required for the call set up management. The decision of whether to admit a call should be based on the joint consideration of rate and power assignment.

On the other hand, rejecting a connection decision-making should have the following at its disposal:

1. A mechanism to compute the resource usage of ongoing connections and the amount of unallocated resource for allocation.
2. A mechanism to allocate the available resource to admit new connections.
3. A mechanism to enforce control subject to satisfaction of QoS specification.
4. Provision to admit handoff connections with a higher probability than new connections.

Many connection admission control techniques have been extensively investigated for application in cellular communication systems. An extensive survey of most of these techniques can be found in [36]-[40].

Chapter 4

SIR Optimization

In several application areas, there are obvious advantages in having computational tools that help the engineer to evaluate the properties of proposed designs, or partially or even totally automate the design process. For these tasks, when applicable, optimization based tools are uniquely suited: they provide global solutions, with the corresponding certificates, and perform very efficiently both in theory and practice. Hence, we choose one kind of optimization algorithm as our tool to solve the power and rate distribution problem. Our approach has been to try to formulate those into optimizations [22], in particular geometric programs, and then use the efficient interior point algorithms to solve them. In this chapter, we introduce some basic knowledge about the optimization theory and our idea about SIR optimization.

4.1 SIR Optimization Problem

We need efficient algorithms to find the optimal solution to our nonlinear problems of admission control. Fortunately, these problems can be turned into optimization [22] formulations, which refer to minimizing a convex objective function over constraint sets. The particular type of optimization we use is in the form of geometric program. Geometric programming focuses on monomial and posynomial functions.

Definition 1: A *monomial* is a function *f:* $R_n \to R$, where the domain contains all real vectors with non-negative components:

$$h(x) = cx_1^{a_1} x_2^{a_2} \dots x_n^{a_n}, \quad c \geq 0 \; and \; a_i \in R \tag{4.1}$$

Definition 2: A *posynomial* is a sum of monomials

$$f(x) = \sum_{k=1}^{K} c_k x_1^{a_{1k}} x_2^{a_{2k}} \dots x_n^{a_{nk}} \tag{4.2}$$

Geometric program is an optimization problem with the following form:

$$\text{Minimize} \quad f_0(x) \tag{4.3}$$

$$\text{Subject to} \; f_i(x) \leq 1$$

$$h_j(x) = 1$$

Where f_0 and f_i are posynomials and h_j are monomials. Geometric programming in the above form is not a convex optimization problem. However, with a change of variables: $y_i = \log x_i$ and $b_{ik} = \log \; c_{ik}$, we can put it into convex form:

$$\text{Minimize} \;\; p_0(y) = \log \sum\nolimits_k \exp(a_{0k}^T y + b_{0k}) \tag{4.4}$$

$$\text{Subject to} \;\; p_i(y) = \log \sum\nolimits_k \exp(a_{ik}^T y + b_{ik}) \leq 0$$

$$q_j(y) = a_j^T y + b_j = 0$$

Therefore, we have an optimization problem. Apart from computational efficiency, optimization also offers stability analysis and accommodates a variety of constraints. Solution algorithms also unambiguously and efficiently determine feasibility. We find optimization problems can solve many versions of QoS provisioning and resource allocation problems in wireless cellular system.

4.2 An Example of Optimization Scheme

To clearly understand the application of theory of optimization to the resource allocation problem, we give an example here. We first consider power distribution in a wireless cellular network in this section. For simplicity, this section considers a single base station. Extensions to multiple base stations and the associated links are straightforward. Each link is a unidirectional path from the transmitter to the receiver. The propagation model [1] used in this section is as follows:

$$P_r = P_0 \left(\frac{d_0}{d}\right)^{\gamma} \tag{4.5}$$

or

$$P_r(dBm) = P_0(dBm) + 10\gamma \log\left(\frac{d_0}{d}\right) \tag{4.6}$$

Where P_r is the received power, P_0 is the transmitted power, d is the propagation path length, and d_0 is a reference distance for the antenna far field. The path loss exponent γ is usually between 2 and 6 for most indoor and outdoor environments. The interfering users' powers are decreased by the inverse of G_p, which can be the spreading gain for a CDMA system. Accordingly, SIR_i [41] [1] for the *ith* user is defined as

$$SIR_i = \frac{W}{R_i} \frac{d_i^{-\gamma_i} P_i \alpha_i}{\sum_{j=1, j \neq i}^{N} P_j \alpha_j + N} \tag{4.7}$$

Where the factors α_j are introduced to accommodate normalization constants and other factors, such as the effects of beam forming in multi antenna systems. N is the power of the background noise. *SIR* is well justified to be used as a *QoS* parameter [24]. For example, channel capacity scales with log *SIR* for high SIR, and the probability of symbol decoding error depend on the modulation type and the Q function what is the complementary Gaussian CDF.

The problem of *SIR* maximization can be formulated as an optimization program [22]. In the following basic formulation, the SIR is maximized for a particular mobile *i*. At the same time QoS for the other mobiles should also satisfy certain requirements or constraints, such as interference due to users has to be smaller than the received signal power so as to achieve a required SIR= β_k.

With the objective and constraints thus formulated and upper bounds *P* on all transmitted powers, we obtain the following non-linear optimization formulation:

Formulation 1: (SIR constrained optimization of power distribution) The following nonlinear problem of optimizing node powers to maximize *SIR* (MAS) or a particular user under QoS constraints for other users in a cellular network is an optimization problem.

$$\text{Maximize } SIR_i = \frac{W}{R_i} \frac{d_i^{-\gamma_i} P_i \alpha_i}{\sum_{j=1, j \neq i}^{N} P_j \alpha_j + N} \tag{4.8}$$

Subject to

$$\beta_k \ \left(\sum_{j=1, j\neq i}^{N} P_j \alpha_j + N \ \right) < \frac{W}{R_i} d_i^{-\gamma_i} P_i \alpha_i$$

$$P_j >= 0$$

$$P_j <= \text{Max}\, P_j$$

The equality constraints are monomials in the same parameters. The variables are the transmitted powers P_i. Therefore, this is indeed a geometric program, and therefore an optimization problem with efficient algorithms that obtain the global optimality.

This general formulation can be applied to different power distribution situations. For example, if there is no objective function, the above formulation reduces to a *SIR* requirement feasibility problem. Also, the objective function can be replaced by "minimize $\sum_i P_i$" as in the following formulation, and then the minimum power vector under the QoS constraints can be determined.

Formulation 2: (SIR constrained optimization for minimum total power MTP)

The following nonlinear problem of minimum power allocation in a cellular network is a optimization problem [22].

$$\text{Minimize} \sum_i P_i \tag{4.9}$$

$$\text{Subject to } \beta_k \left(\sum_{j=1, j\neq i}^{N} P_j \alpha_j + N_0 \right) < \frac{W}{R_i} d_i^{-\gamma_i} P_i \alpha_i$$

$$P_j >= 0$$

$$P_j <= \text{Max}\, P_j$$

Additionally, a weighted sum or powers, or the maximum user power can be minimized. The d_i can also be treated as optimization variables for optimization of antenna sectoring, which is a popular technique for interference mitigation.

4.3 Computer Simulation

Assume there is a simple system that comprised by four different rate categories. They are voice, video, email and FTP. Their rate arranges are listed in Table 4.1. Special number of users in each category is used for a simulation of Formulation 1.

Table 4.1: Rate arrange

	RATE ARRANGE Min (Kbps)	RATE ARRANGE Max (Kbps)
VOICE	1	8
VIDEO	32	120
FTP	8	40
EMAIL	16	24

We ignore the distance here, and just put d=10. The power drop-off factor $\gamma = 4$, and $\alpha = 1$. The users in each category are in rate arrange of their limitation.

Each user has a maximum power constraint of $P_{\max} = 0.5W$. The noise power is $N=0.5\mu W$ for all users. Bandwidth is 1MHz. The spreading gain *W/R* here is calculated according to the user categories. The SIR of all users, other than the user we are optimizing for, must be greater than a common threshold SIR level β. β is varied to observe the effect on the Optimized user's SIR. This is done independently for the voice users, the video users, the email users and the FTP users. The results are plotted in figure 4.1.

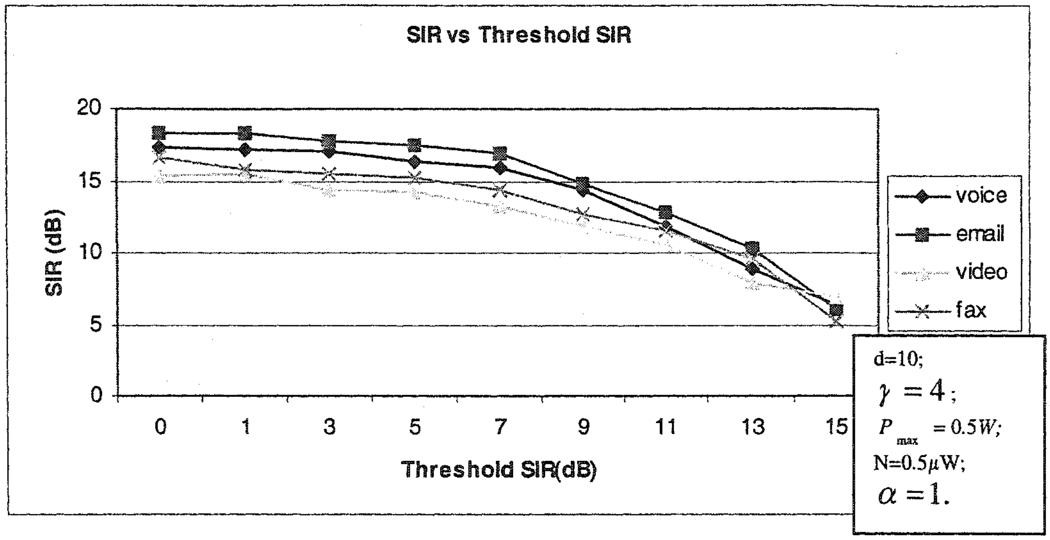


Figure 4.1: SIR optimization simulation

This simulation illustrates several interesting effects.

First, when the required threshold SIR for the non-optimized users is high there are no feasible power control solutions. At moderate threshold SIR, as threshold SIR β is decreased, the optimized SIR initially increases rapidly. This is because intended user power is allowed to increase relative to the sum of the other users' power, which then decreases in this category, and the noise is relatively insignificant.

At low threshold SIR, the noise becomes more significant and the power trade-off from the other users less significant. Eventually, the optimized user reaches its upper bound on power and cannot utilize the excess power allowed by the lower threshold SIR. Therefore, during that stage, the only gain in the optimized SIR is the lower power transmitted by the other users. This is exhibited by the sharp bend in the curve to a much shallower sloped curve.

Chapter 5

Connection Admission Control of Multi-class CDMA System

From the theoretical knowledge about CDMA, capacity, power, multi-rate system, optimized power distribution scheme was derived. In this chapter, we will first introduce our CAC schemes, and then describe the system models used in the connection admission control in our simulation. Then, we discuss quality measures for radio link connections used in simulation.

5.1 Multi-class Connection Admission Control

Due to the statistical multiplexing inherent in CDMA systems there is a tradeoff between the system capacity and the level of quality of service. As more and more users access the system at the same time, the quality of communication link for individual user deteriorates due to excessive multiple access interference. In order to guarantee the quality of service for all the traffic in a cellular system, call admission control (CAC) must be applied to control the number of users in each cell such that an appropriate level of communication quality can be maintained [21]. When a new call request is received, admission control algorithm should decide whether it should be admitted into the system or not. If it is expected that after admitting the new call into the system the QoS of all the users can still be guaranteed, then the new call should be admitted. If it is expected that after admitting the new call into the system users' QoS requirements are likely to be violated then the new call request should be blocked. CAC schemes that are too conservative tend to admit less number of users than that can be accommodated by the system, therefore under-utilize the precious bandwidth resource. CAC schemes that are too optimistic tend to admit more number of users than that can be accommodated by the system, therefore violate users' QoS requirements. A good admission control scheme should achieve high capacity utilization without violating QoS guarantees [27]. Existing works on call admission control in CDMA systems are interference-based or SIR-based and apply only to systems with conventional matched-filter receiver, fixed power control and single-class traffic.

Our goal is to develop an admission control scheme that supports multiple classes of traffic with different QoS requirements and incorporates power and rate distribution.

5.2 Problem Formulation

Consider the uplink operation in a cellular network. Before transmitting any packets, a user makes a connection request to the system. Upon receiving the connection request, the CAC at the MSC makes the decision about whether its connection can be admitted or not. The admission decision is made based on the declared traffic parameters and the required QoS performance in the connection request for the user, and the available resources in the system. The admission decision is then sent back from the MSC to the MS's currently associated BS, which notifies the MS about whether or not its connection request can be accepted. If the admission decision is to accept, a connection can be made between the user and the BS before packet transmissions. Fig.5.1 shows the state diagram of a connection upon its arrival. The entire traffic is divided into different traffic classes, and each user can be transmitted with different rates. I.e. each user is a group of all the connections with the same traffic parameters and QoS requirements. The new connection arrival process for each class is assumed to follow a uniform process. Each new connection request is subjected to a connection admission test at the BS. Failure of the test results in connection blocking, and the probability of this defined as new connection blocking probability (NCBP). After successfully passing the admission test, a connection is established between the user and the BS for packet transmission.

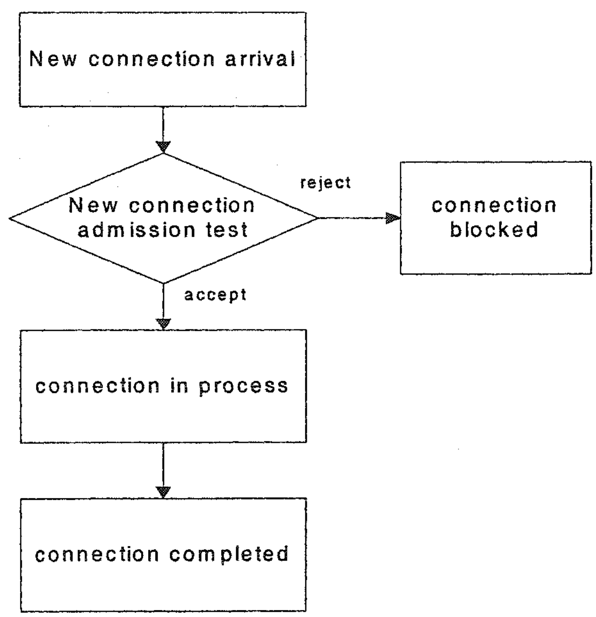


Figure 5.1: The state diagram of connection establishment

During the lifetime of its connection, real-time traffic is delay sensitive, and their packets must be transmitted more urgently than the packets from non-real-time traffic, i.e., the transmission rate for each real-time connection should be high enough to vacate the backlogged packets. On the other hand, some data traffic requires very low BER, although it can tolerate long delay and overflow. In this case, higher transmission powers are needed to ensure successful transmission with high probability. We will evaluate the transmission performance in terms of queuing delay and number of instance of buffer overflow. Therefore, transmission power and rate allocation are the basis for effective and efficient resource management, and should be performed based on the traffic parameters and required QoS performance for each connection.

The variable SIR is considered in our simulation. Users rates change every window. Accordingly, users are allocated powers dynamically. The user will leave after the lifetime of its connection.

5.3 System Model

5.3.1 Network Configuration

Wireless cellular communication is the public communication approach to provide high system capacity and user mobility for mobile user. Future wireless communications are expected to support multimedia services. Fig. 5.2 shows a network architecture delivering end-to-end transportation in a hybrid wireless and wire line network. A mobile switching center (MSC) serves as the access point for the wireless network to the wire line backbone network. Sometimes, the wireless segment is called radio access network (RAN). The service area of each RAN consists of several radio cells, each of which is the coverage area of a BS. The MSC is responsible for RRM, CAC, mobility management, and etc. the transmission between an MSC and the wire line backbone network is a wire line channel, while that between an MS and a BS is a wireless channel. There are two types of wireless channels: uplink, the link from MS to a BS; and downlink, the link from a BS to MSs. The uplink channel is a multiple access channel, where all the MSs in the cell share the radio resource and compete for service. In the downlink channel, the BS has the knowledge of the transmitted traffic. Communications between two or more users use the end-to-end transport through access networks and the wire line backbone network. The challenging issued behind supporting multimedia in wireless networks is the limited system resource. Which usually make the wireless access networks as the bottleneck in end-to-end transportation.

This thesis is confined to one BS under one MSC. We assume that the backbone network always has enough resources to serve all admitted connections in the system. This thesis focuses on the uplink, since the uplink support multi-user transmissions that impact on the capacity of the system.

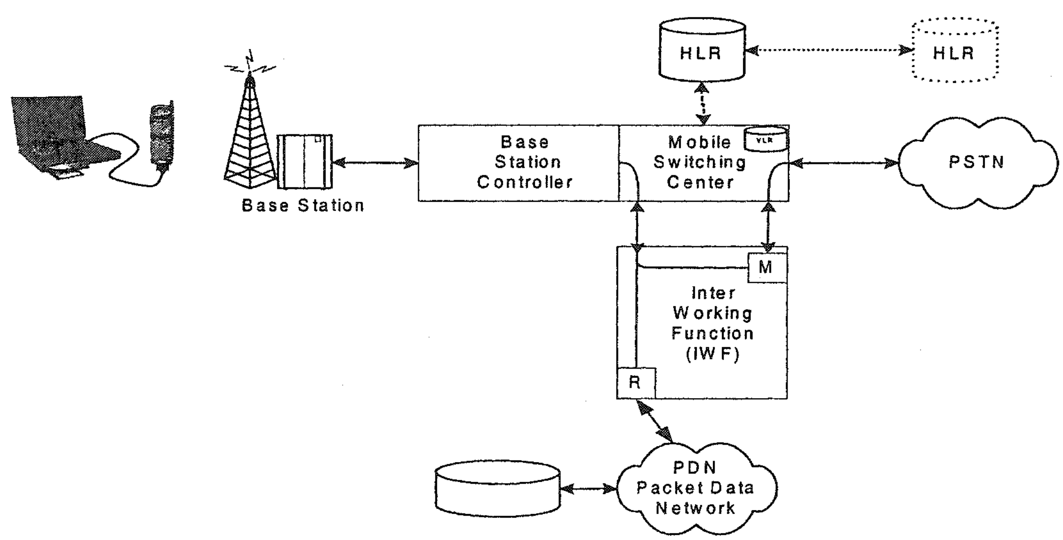


Fig. 5.2: Network architecture transportation in a wireless and wire line network

5.3.2 Multi-class Traffic CDMA System With Multi-rate

We consider a direct-sequence CDMA system that supports multiple classes of traffic with different data rates and SIR requirements, as illustrated in Figure 5.3.

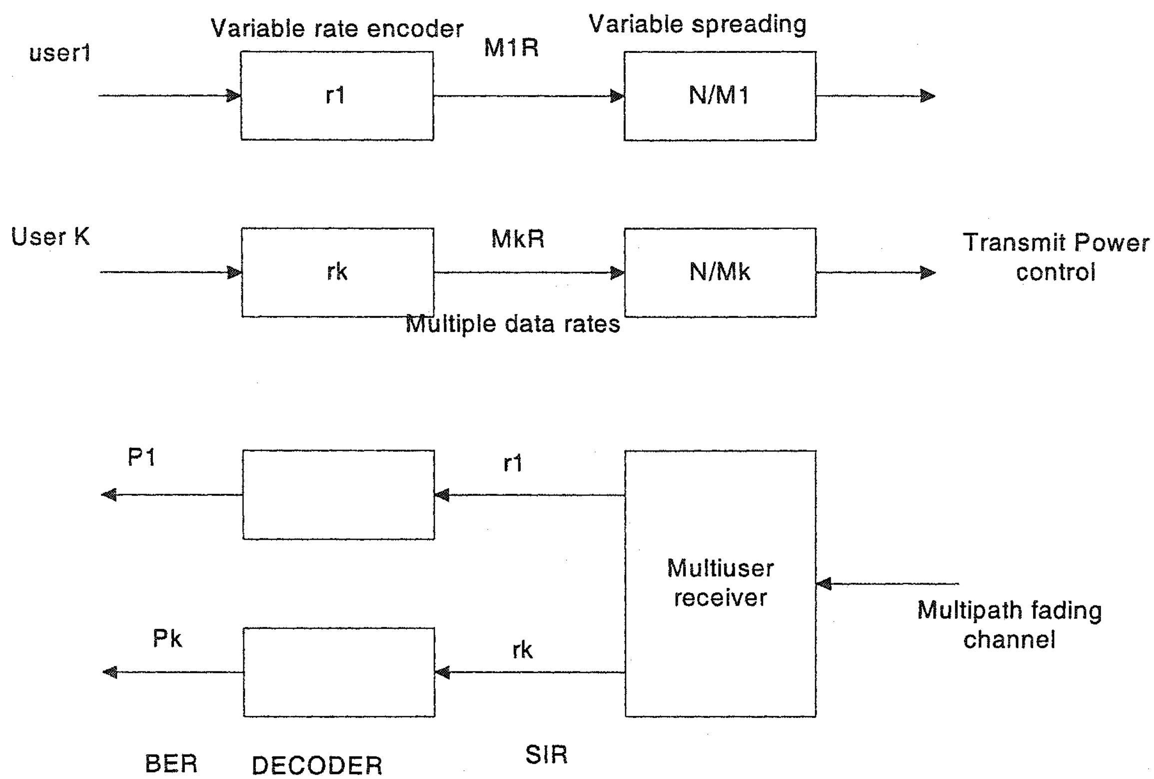


Figure 5.3: Multi-class Traffic CDMA system

There are K users in the CDMA system, each transmitting information bits with source data rate of R_k and bit error rate requirement of P_k. The transmitter consists of variable rate error correction coding, variable spreading and transmission power distribution. Multiple symbol rates are supported by varying the spreading factor while maintaining fixed chip rate and symbol alphabet, as shown in Figure 5.3. The lowest symbol rate in the system is said to be the basic rate, denoted as R, and the symbol period for the basic rate is termed as basic symbol period. User k transmits M_k symbols in a basic symbol period, where M_k is an integer. This corresponds to an Mk-fold increase in symbol rate of user k at $M_k R$. The spreading factor for user k is N/M_k, where N is the spreading factor corresponding to the basic rate. User k can choose a coding rate r_k, and source data rate and symbol rate are related by $R_k/r_k = M_k R$. The modulated signal of user k is transmitted at power p_k.

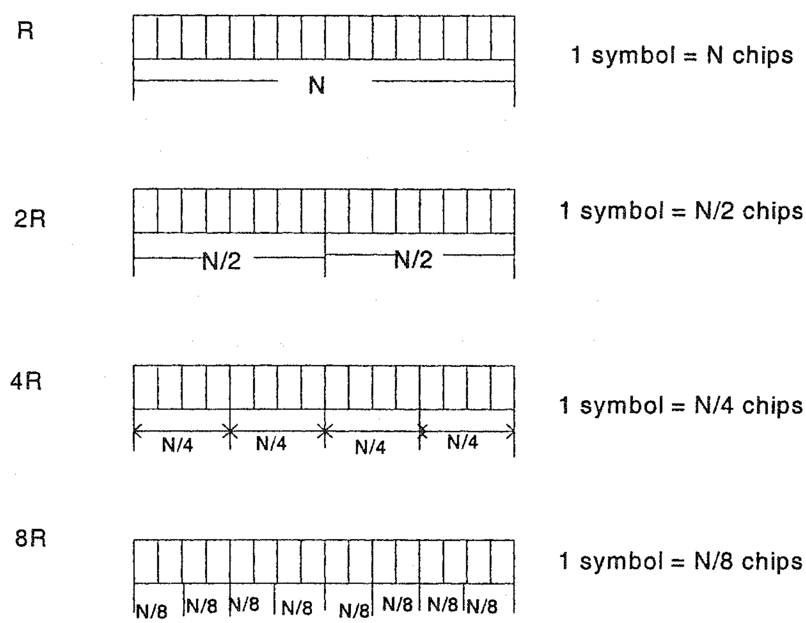


Figure 5.4: Multi-rate scheme

The received signals are jointly demodulated by a single antenna or antenna array multi-user receiver, followed by separate decoding for each user. The signal-to-interference ratio of user k's demodulated signal is β_k, while the bit error rate of user k's decoded information bits is P_k. To maintain bit error rate of user k below a required level of P_k, it is necessary to keep SIR of user k higher that certain value β. The required SIR β is decided by the target bit error rate P_k, the coding rate r_k, and distribution of β. The adaptive power distribution algorithm adjusts the transmit power p_k such that the SIR at the receiver exceeds or equal to the required SIR for the entire user, $\beta_k >= \beta$.

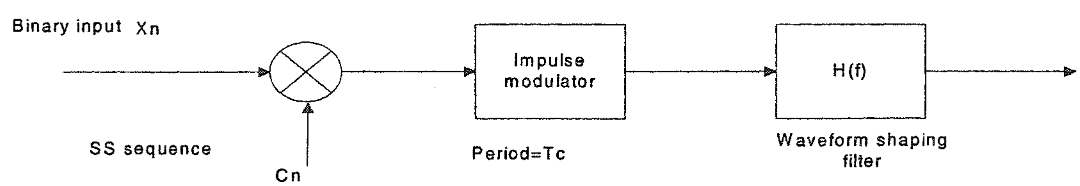
We consider the case where the spread spectrum bandwidth, *W*, which is also the total system operation band, is shared by all the users in the system. The isolation between uplink and downlink uses FDD, so there is no interference from the downlink to uplink transmission. Transmission in each connection encounters background AWGN. The effect of soft handoff on the uplink system is not explicitly considered in this research. To keep our simulation model simple and tractable we have neglected inter-cell interference and any sort of error correction scheme.

5.3.3 Modulation Method

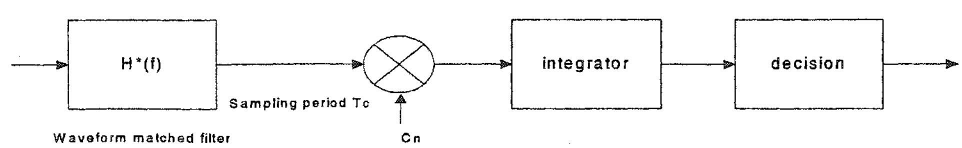
The BPSK modulation is assumed to use in our system. The block diagrams for modulation and demodulation are shown in Fig.5.5.

The relation between bit error probability and SIR is

$$P_b = Q(\sqrt{2SIR}); \qquad (5.3)$$



Modulator



Demodulator

Figure 5.5: Modulation and demodulation of BPSK

5.3.4 Power Distribution

In a CDMA system, the available system resource is shared by all active users. An appropriate amount of power resource should be distributed to each connection to ensure its required SIR and at the same time to allow more active users in the system.

Power distribution is necessary to maintain all users' signal powers at the desired target values. We focus on how much the target power is required for each connection with given SIR requirements. The detail of how to implement the power control is beyond the scope of this research. There are methods for power control reported in the literature. Ideally, if perfect power distribution can be achieved, the required SIR for each user can be guaranteed by accurately distributing the received power for all the users, as long as the traffic load is within the system capacity. In reality, however, due to channel estimation errors, power control delay, power distribution command transmission errors, etc. imperfection in power distribution always exists.

The base station measures the average received power from an MS every period, e.g. 0.625 ms for W-CDMA and 1.25 ms for CDMA2000. We here assume user power be checked very 2 packets, i.e. 2 second. After comparing the average value with the target power, a power distribution command is sent to the MS to raise or lower transmission power. With this process, power distribution can overcome the effect of the path loss and most of the effect caused by slow varying lognormal shadowing. We assume that with well-designed receiver structure, e.g. a RAKE receiver, the fast fading can be ideally mitigated.

5.3.5 Call Admission Control

The call admission control makes admission decisions for new connections. It is performed at the MSC instead of at each BS, because making an admission decision for a new connection requires information about the MSs and traffic load in the entire procedure [35]. An admission decision is based on the traffic parameters and required QoS from both the new and handoff request and the existing connections. The CAC admits a new connection as long as its required QoS and the QoS of ongoing connections can be guaranteed. The reserved resource for data connections can still be efficiently utilized at the packet level to increase the packet throughput of non-real-time traffic. We will not consider handoff in our simulation.

5.4 Three Allocation Policies

Based on the SIR optimization simulation we proposed in chapter 4, we will use some resource allocation policies in our call connection admission simulations. Three types of power and rate distribution policies experiments are described in this section, Minimum Total Power algorithm (MTP), Maximum SIR algorithme (MAS), and Static algorithm (STA). A performance comparison of these schemes is introduced in chapter 6 in processing of connection admission control. Their basic formulations are following.

5.4.1 MTP Policy

In this section, we consider the MTP scheme. The goal of this algorithm is to obtain the power allocation for each mobile that maximizes the total system capacity while meeting all the users' minimum SIR requirements. Power from each mobile should be large enough to achieve the required minimum SIR for its class of connections in the system, while it should be kept low enough in order to reduce MAI to other connections in the system, and to allow more active users in the system. The basic formulation of this problem is presented by the following optimization problem.

$$\text{Minimize} \sum_{i=1}^{N} \sum_{j=1}^{4} P_{ij} Y_{ij} \tag{5.1}$$

$$\text{Subject to} \quad P_i < P_{\max}$$

$$SIR_i = \sum_{j=0}^{3} \frac{8 * 2^j}{120} SIR_{ij} > \beta_i$$

$$\beta_{ij} R_{ij} \Big(\sum_{k=1,k\neq i}^{N} \sum_{j=1}^{4} Y_{ij} P_{kj} + N_0 \Big)/W \leq Y_{ij} P_{ij}$$

$$R_{ij} \geq \mathrm{R}_{insti}$$

Where β_i is the minimum SIR for each class. According to rate allocation diagram in Figure 5.6, We denote the power consumption for user i in rate j as P_{ij}, the transmission rate for user i in channel j as R_{ij}, Y_{ij} is the number of rate for user i in rate j, *SIR* is defined as the total signal to interference ratio for the whole system, and SIR_i is defined as the signal to interference ratio for user *i*, and SIR_{ij} is defined as the signal to interference ratio for user *i* in rate j. SIR_i of user *i* depends on the SIR_{ij} of constant rate. We assume the four rate channels (8K, 16K, 32K and 64K) are transmitted in our system.

Therefore, we assume that $SIRi = \sum_{j=0}^{3} \frac{8*2^j}{120} SIR_{ij}$, where denominator 120 is the sum of the four rates. A weight is assumed to each SIR_{ij} to yield SIR_i. High rate user are multiplied by large number so as to get SIR of high rate, same as SIR of low rate.

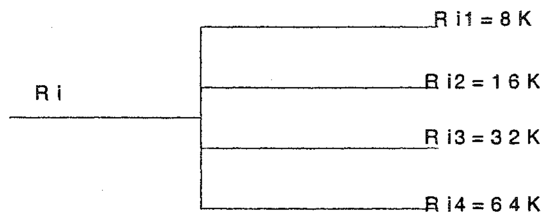


Figure 5.6: Rate allocation diagram

The SIR level for each user should be greater than a minimum SIR level to achieve certain level of QoS, i.e. $SIR_i \geq SIR_{\min}$.

Given these constraints does there exist any P and Y such that these constraints are met, if not, new users will have to be rejected. If there exists more than one feasible solution, it is natural to pose the question as which one to choose.

5.4.2 MAS Policy

The goal of this algorithm is to obtain the power and rate distributions for each mobile that maximizes the SIR quality, named MAS, which try to let every user get a perfect SIR quality, but ignore the number of user to admit. It satisfies all the users' SIR quality requirements but does not consider the system capacity.

Maximize SIR_i (5.2)

Subject to $P_i < P_{\max}$

$$SIRi = \sum_{j=0}^{3} \frac{8*2^j}{120} SIR_{ij} > \beta_i$$

$$\beta_{ij} R_{ij} (\sum_{k=1,k\neq i}^{N} \sum_{j=1}^{4} Y_{ij} P_{kj} + N_0)/W \leq Y_{ij} P_{ij}$$

$$R_{ij} \geq \mathrm{R}_{insti}$$

Where β_i is the minimum SIR for each class. We denote the power consumption for user i in rate j as P_{ij}, the transmission rate for user i in channel j as R_{ij}, Y_{ij} is the number of rate for user i in rate j, SIR is defined as the total signal to interference ratio for the whole system, and SIR_i is defined as the signal to interference ratio for user i, and SIR_{ij} is defined as the signal to interference ratio for user i in rate j.

5.4.3 A Constant Static Policy STA

The effects of the two algorithms above on the system performance are investigated with comparing another CAC scheme named STA. STA stand for Constant Static policy. The policy we propose is a fixed allocation scheme that acts a compromise of the former two schemes, MTP and MAS. By this conventional algorithm, fixed expected number of users and the constant received rate and powers are allocated without considering user's maximum and minimum QoS requirements and the changing resource availability. We try to define it as classic policy. Some parameters of STA are listed in chapter 6.

5.5 Performance Measures

Typically, data transmission schemes tend to measure performance in terms of delay and overflow; connection admission schemes often use probability of blocking against offered traffic load. We will introduce the performance measures used in our thesis below.

5.5.1 Quality of Service in CDMA

The Quality of Service described in this section is based on information produced by the standardization body 3GPP. Today's communication systems try to provide best effort services. Traffic is processed as quickly as possible, but there are no guarantees as to timeliness or actual delivery. However, different applications generate traffic with very different characteristics. Moreover, they have different demands on network performance in terms of bandwidth, delay and reliability to name a few. A way to overcome the varying demands is to provide a single level of quality that meets the requirements of all applications. That is, the abundant capacity of the wired network could be resembled to a single level of quality, meeting the demands of all applications.

In the case of wireless networks the situation is in fact quite different. In second-generation systems, such as GSM, a single level of quality has been provided, enabled through the use of resource reservation. This level of quality has fixed guarantees, however not negotiable for the users. Since there is a tradeoff between quality and capacity, the approach of providing a single level of quality meeting all demands would result in an over-dimensioned system. That is, if all types of traffic would be treated the same, delay insensitive traffic such as email would consume as much resources per transmitted bit as would delay sensitive voice traffic. In other words, the former would be given a better service quality than needed, resulting in an over-dimensioned and hence poorly efficient system. In addition, over provisioning of the radio network would not be possible for a number of reasons. The very nature of the radio spectrum is that it is inherently scarce. Further, an excessive deployment of base stations would be an extremely costly adventure, beyond the rationales of common sense investment estimates.

Hence, applying a differentiated service approach to wireless systems facilitates a more efficient use of the resources. A way to measure different levels of quality is by the use of parameters. In summary, the need for a differentiated service approach, e.g. QoS, comes into light. The growing demand for bandwidth in mobile environments places tight restraints on the wireless resources. One is presented with the options to further increase the capacity of the network, e.g. deploy more efficiently allocate the existing resources through the use of QoS.

QoS implies differentiation in services available to users according to, for instance, the requirements of a particular type of traffic. The traffic type can be real time voice, video conferencing, file transfer, real time video, etc. The mobile user and the network will agree on a particular QoS profile during the initial service negotiation stage and the network will attempt to deliver this QoS.

Therefore, the quality of service supported by the network depends on the type of service. Each has different a Quality of Service requirement.

In our CDMA systems, the quality of service measure is based on *SIR*.

$$SIR_i = \frac{(W / R_i)P_i}{I_i} \tag{5.4}$$

Where, W is the spread-spectrum bandwidth, R_i is the transmission rate for user i, P_i is received power for user i, I_i is the interference power from all other users.

Since, processing gain G_p is

$$G_P = \frac{W}{R_i} \tag{5.5}$$

Therefore,

$$SIR_i = \frac{G_p P_i}{I_i} \tag{5.6}$$

In CDMA systems, the required *SIR* is usually less than 10 dB. Each cell can support more than one user sharing the same frequency spectrum.

5.5.2 Traffic Intensity, iteration and window

Traffic intensity [1]: measure of channel time utilization, which is the average channel occupancy measured in Erlangs. This is a dimensionless quantity and may be used to measure the time utilization of multiple channels.

The traffic intensity offered by each user is equal to the call request rate multiplied by the holding time. That is, each user generates a traffic intensity of A_u Erlangs given by

$$A_u = \lambda H \tag{5.7}$$

where H is the average duration of a call and λ is the average number of call requests per unit time. For a system containing U users and an unspecified number of channels, the total offered traffic intensity A, is given as

$$A = UA_u \tag{5.8}$$

Traffic can be measured in several different ways. One of them is *offered traffic*. Offered traffic is an average measure of traffic on a channel or system, i.e. if a channel is used on average 50% of the time then the offered traffic is 0.5 Erl. We use the bandwidth 3MHz in our simulation system, and from the statistic simulation, we assume the average user rate is 60Kbps. Therefore the average system capacity is C=50 users per second.

From 5.7 and 5.8, we define normalized traffic intensity ρ in our simulation as

$$\rho = A / C = UA_u / C \tag{5.9}$$

In this study, we assume that the new user arrival is generated by uniform distribution. Each new call is equally likely to occur anywhere on the ring and the duration of each call is exponential distributed with time H. And the users normalized accepted traffic is a value less than user arrival traffic. i.e.

$$\rho' = \rho \times \textit{(1-blocking probability)} \tag{5.10}$$

We define "**iteration**" and "**measurement window**" as time concept in our programming. Iteration indicates the numbers of we call uniform connection access routine, i.e., each run of uniform access routine is one iteration. For the users who have been in the network, one packet was transmitted per iteration. Measurement window acts as the period measurement for the delay, overflow and change of power or rate. Each window may be equal to 2, 4, 6, 8 iterations or packets here. In our study, we assume that one iteration is equal to one packet time. Each packet is t=*8Kbit*, and R_b=*8Kbps*. each packet transmission time:

$$\mathrm{T}_p = \frac{t}{R_k} \tag{5.11}$$

Therefore, one iteration in our simulation corresponds to 1 second of real time. The data packets are transmitted according to unit "packet" per iteration.

5.5.3 CDMA Cell Load Measurement

In this section, we discuss cell load measurement in CDMA systems. Since the capacity at each base station is shared by all users, a meaningful cell load measure should also include the interference. In CDMA systems, we can view the system as one isolated system with finite resources. The capacity of each base station is limited by the total interference received at each BS [29]. For CDMA digital cellular systems that (1) employ power control such that all uplink (mobile-to-base) signals are received at the same power level. And (2) all users are spread (via either direct-sequence or frequency-hop spread spectrum) over the total available bandwidth *W*; a simple equation for capacity may be developed as follows. Assuming that a cell has a total of *N* mobile users, the composite uplink received signal at the base site will consist of the desired signal with power S and *N-1* interference users, each also having power *S*, by the power control constraint. The signal-to-interference ratio at the base is thus $S/I=S/(N-1)S=1/(N-1)$. The energy/bit to interference power spectral density $E_b\ /I_0$ may be derived similarly. The energy/bit is just the signal power *S* divided by the information bit rate *R* (i.e., E_b=*S/R*). The interference power spectral density is the interference power divided by the spreading bandwidth W [i.e., $I_0 = S(N-1)/W$].

We assume that each user I has an identical SIR requirement $\beta_i = \beta$. All users assigned to base station k will have equal received powers P_k. We now develop a characterization of the cell capacity. When each user assigned to base station has the same received power P_k and rate *R*, we can write the SIR requirement in terms of

$$SIR = \frac{W}{R} \frac{P_k}{(N_k - 1)P_k + N} \geq \beta \qquad (5.12)$$

In CDMA systems, the maximum capacity for each cell is achieved when there is no other inference except the users assigned to the same base station. That is N =0. From the above equation, the number of users assigned to base station k, N_k must satisfy the following condition.

$$N_k \leq 1 + \frac{W}{\beta R} \tag{5.13}$$

We have the maximum capacity $N_{\max}$ for base station k is

$$N_{\max} = 1 + \frac{W}{\beta R} \tag{5.14}$$

5.5.4 Call Blocking Probability

In general, connection-blocking probability has been used to measure the probability that a call will fail during the set-up phase. In CDMA systems, upon arrival of a fresh call, the respective base station assigns a channel and power to it according to a specific strategy. Different allocation algorithms in our simulation have been described before. If the SIR is smaller than the minimum SIR of that type, then the call has to be blocked.

In our measurement, a call that is accessed but not admitted to the system is regarded as blocked. That is, for every user who requires service, it is assumed there is no setup time and the user is given immediate access to a channel if one is available. If no channels are available, the requesting user is blocked without access and is free to try again later. This type is called *blocking calls cleared.* Call Blocking Probability P_{block} is defined in our simulations as:

$$P_{block} = \frac{Number\ of\ blocked\ calls}{Total\ Number\ of\ Access\ Calls} \times 100\% \qquad (5.15)$$

Call blocking probability at a particular level can be used to determine the capacity of a scheme, i.e. how many active or potentially active users the system can support.

5.5.5 Queuing Delay

Packet delay in a packed-switched network normally consists of three components: transmission delay, retransmission delay as well as queuing delay. In our simulation, only queuing delay is calculated. Each user has its own buffer. The packets that have been generated by users but cannot get access to the network will be left in the buffer. The unit of buffer overflow is indicated as Kbps or 1/8 packet. .

Normalized Queuing Delay:

The queuing delay is defined in our simulations as the time elapsed from the instant of the packets input into user's buffer to the instant of successful output of these packets, we can write it as

$$\mathrm{D}_i = \frac{\sum_{w=1}^{M} B_{i,w}}{M} \qquad (5.16)$$

Where $B_{i,w}$ is buffer content of user i at window w, M denotes the number of window in whole run.

Average Normalized Queuing Delay:

Average Normalized Delay $\overline{D}$ denotes, for each window of data transmitted, how much delay is incurred. It is defined as:

$$\overline{D} = \frac{\sum_{i=1}^{N}\sum_{w=1}^{M} B_{i,w}}{MN} \text{ (In units of measurement window)} \quad (5.17)$$

Where $B_{i,w}$ is the number of packets in buffer of user i at window w. M is the number of window during user *i* is accepted. N is the total number of online users.

Variance of Average Queuing Delay:

The Variance of average queuing delay is a measure for instant user delay from average user delay, which is defined as

$$\sigma_D^2 = \frac{\sum_{i=1}^{N}(D_i - \overline{D})^2}{N-1} \quad (5.18)$$

Where N is the total number of online users.

5.5.6 Buffer Overflow

If the packets in the number buffer are larger than the buffer size, the buffer will overflow, and extra packets generated above the buffer size will be lost. We define the buffer overflow as the times of packets dropped when a given threshold buffer size is exceeded. The unit of buffer overflow is indicated as Kbps or 1/8 packet. The buffer overflow of user i is defined as the total number of times of exceeding user's buffer size to be divided by the number of users N and by multiplying the number of window M, i.e.

$$\mathrm{OV}_i = \frac{\sum_{w=1}^{M_j} O_{(B_{i,w}-B_L)}}{M_j} < 1 \quad (5.19)$$

Where $O=1$ is defined as exceeding user's buffer size for user i in window w. M_j is the number of iteration in each jth user call. $B_{i,w}$ is the number of packets in buffer for user i at window w. B_L is the buffer size.

Average Buffer Overflow

The Average Buffer Overflow Rate $\overline{OV}$ is defined as:

$$\overline{OV} = \frac{\sum_{j=1}^{N} \left[\frac{\sum_{i=1}^{Mj} O(j,i)}{M_j} \right]}{N} \times 100\% \tag{5.20}$$

Where $O(j,i)=1$ whenever user j has buffer overflow at window i, N means total number of online users; M_j stands for total number of window during which user j is calling.

Variance of Average Buffer Overflow

$$\sigma_{OV}^2 = \frac{\sum_{i=1}^{N} (OV_i - \overline{OV})^2}{N-1} \tag{5.21}$$

Where N is the total number of online users.

5.5.7 Average Packet Loss Rate

Due to buffer overflow, some data packets are lost before being served by the network. Average Packet Loss Rate P_l is used to denote what percentage of packets is lost due to buffer overflow. It is defined as:

$$P_l = \frac{\sum_{j=1}^{N} \{ [\sum_{i=1}^{Mj} Pg(j,i) - \sum_{i=1}^{Mj} Ps(j,i)] \Big/ \sum_{i=1}^{Mj} Pg(j,i) \}}{N} \times 100\% \qquad (5.22)$$

Where *Ps (j,i)* is number of packets that have been served successfully in measurement window, whereas *Pg (j, i)* is number if data packets generated by *j* th calls in the window *i*. M_j is the call duration for user *j*.

5.5.8 Average Bit Error Rate

The bit stream transmission with different rate in our simulation is shown in Figure 5.7.

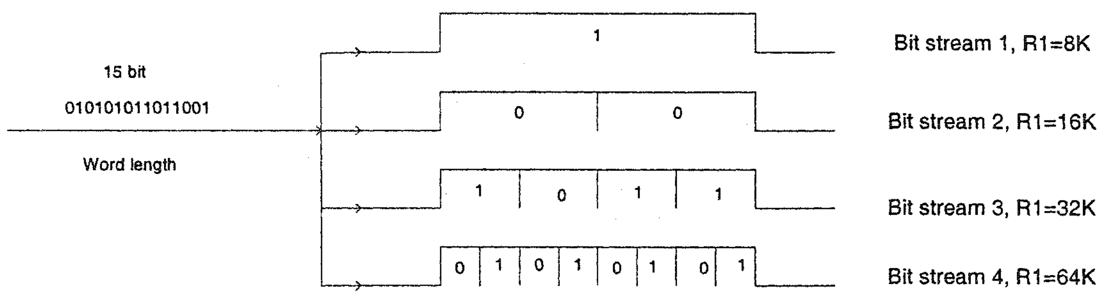


Figure 5.7: bit stream transmission with different rate

Based on the parallel to serial bit conversion in figure 5.7, the average bit error rate for a connection is presented as follow,

$$P_{bi} = \frac{word \text{ error}}{15} \qquad (5.23)$$

i.e.

$$P_{bi} = \frac{1-(1-Q\sqrt{2\mathrm{SIR}_{i1}})^1(1-Q\sqrt{2\mathrm{SIR}_{i2}})^2(1-Q\sqrt{2\mathrm{SIR}_{i3}})^4(1-Q\sqrt{2\mathrm{SIR}_{i4}})^8}{15} \qquad (5.24)$$

Where, SIR_{ij} is the signal-to-interference of *i*th connection and *j*th rate channel. $Q\sqrt{2SIR_{i1}}$ is the probability of one symbol error of the lowest rate data branch. $Q\sqrt{2SIR_{i4}}$ is the probability of one symbol error of the highest rate data branch. Q is Q-function [31], which is cumulative normal distribution function giving the probability that a various assumes a value in the range [0,x],

$$Q(x) = \frac{1}{\sqrt{2\pi}} \int_{x}^{\infty} e^{-x^2/2} \cdot dx \tag{5.25}$$

Recall *i*th user bit stream is multiplied (by serial to parallel carrier) into 4 data channels, the first has rate of 8Kbps, 2nd rate is 16K, 3rd rate is 32K, and so on, 4th branch rate is 64K, so we get one symbol of the least rate, one would have 2 symbols of second rate and 4 symbols of 3rd rate and 8symbols of highest rate stream giving user to correct symbol error probability as in 5.24, and divided by 15 stems from the fact that 15 bits of original user i bit steam are demultiplied as above.

Chapter 6

Simulation and Numerical Results

In this chapter we provide simulation details for three admission control schemes for CDMA cellular system using joint power and rate distribution. Then admission control is implemented in a cell based on the simulation. Performance of the CAC schemes is evaluated by comparing the system parameters.

6.1 Simulation Description

6.1.1 Input Parameters

The proposed CAC schemes are demonstrated via simulation of a CDMA system. The system parameters used in this numerical result are listed in Table 6.1.

Table 6.1: System parameters

Spread spectrum Bandwidth: W=3Mhz
Basic rate: R_b =8kbps
The white noise PSD: N_0 = 10 e–6 W/Hz

For simplification, our research is limited to one cell, and there are four types of traffics—voice, video, email and Ftp. The traffic parameters are listed in table 6.2. Each user has only one application (one kind of aforementioned four traffics). The chip rate for all the users is fixed so is the total bandwidth W used by all users. Also, OVSF codes are used as orthogonal codes that allow a constant spread bandwidth for various information rates. We assume there are four kinds of basic rates that are orthogonal each other. The transmission rates are 8kps, 16kps, 32kps, and 64kps respectively, and each user's actual transmission rate is a linear combination of those four kinds of basic rate as in Figure 5.7. Bandwidth is a limitation of network. Our goal is to achieve the best system QoS under the condition of satisfying each user's power consumption requirements and limited total system power consumption at a certain level in the mean time.

Table 6.2: Traffic parameters

	RATE ARRANGE MIN (Kbps)	RATE ARRANGE MAX (Kbps)	POWER ARRANGE MIN (Kbps)	POWER ARRANGE MAX (Kbps)	MIN SIR
VOICE	1	8	0	0.2	6
VIDEO	32	120	0	0.1	7
FTP	8	40	0	0.16	5.5
EMAIL	16	24	0	0.21	5

For STA user SIR is equal to 5.

6.1.2 Formulations

For each user, we can calculate the instantaneous transmission rate by:

$$R_{ins,j} = R_{\min j} + (R_{\max j} - R_{\min j}) * 6 \qquad (6.1)$$

Where, 6 is random coefficient value from 0 to1. j is one of four classes of traffic, from 1 to 4; The rate assigned to user i so as to satisfy his required rate $R_{ins,j}$ is a linar combination of 4 basis rates i.e.,

$$R_{assi} = 8Y_{i1} + 16Y_{i2} + 32Y_{i3} + 64Y_{i4} \geq R_{ins,j} \tag{6.2}$$

Where Y_{ij} is the number of composite rate assigned user i in rate j, R_{assi} is the actual assigned transmission rate for user i. E.g. if a user requires $R_{ins,j}$=40Kbps, this could mean $Y_{i1} = 0$, $Y_{i2} = 1$, $Y_{i3} = 1$, $Y_{i4} = 0$; or $Y_{i1} = 0$, $Y_{i2} = 0$, $Y_{i3} = 0$, $Y_{i4} = 1$; so many possibilities for rate splitting are possible, which gives more degrees of freedom for optimization process.

Combined 6.1 and 6.2, we have:

$$8Y_{i1} + 16Y_{i2} + 32Y_{i3} + 64Y_{i4} \geq R_{\min j} + (R_{\max j} - R_{\min j}) * 6 \tag{6.3}$$

We further assume that

$$Y_{i1} + Y_{i2} + Y_{i3} + Y_{i4} <= 4; \tag{6.4}$$

and Y_{ij} is equal to either 1 or 0. Each user that carries different information rates and QoS requirements can occupy a maximum of four channels.

We denote the power consumption for user i in rate channel j as P_{ij}, the transmission rate for user i in channel j as R_{ij}, SIR is defined as the total signal to interference ratio for the whole system, and SIR_i is defined as the signal to interference ratio for user i, and SIR_{ij} is defined as the signal to interference ratio for user i in rate j. therefore,

$$SIR_{ij} = \frac{(W / R_{ij}) P_{ij}}{I_{ij}} \tag{6.5}$$

And the background AWGN N is assumed. Thus the total interference is the sum of interference from other users in the cell plus background interference. I_{ij} is defined as,

$$I_{ij} = P_{\Sigma} - P_{ij} + N \tag{6.6}$$

Where,

$$P_{\Sigma} = \sum_{i=1}^{N} \sum_{j}^{4} P_{ij} Y_{ij} \tag{6.7}$$

Therefore,

$$SIR_{ij} = \frac{W}{R_{ij}} \frac{Y_{ij} P_{ij}}{\sum_{k=1, k \neq i}^{N} \sum_{j=1}^{4} Y_{ij} P_{kj} + N}, \text{ for user } i \text{ in channel } j, \tag{6.8}$$

Where, W is bandwidth of CDMA wireless system. R_{ij}, Y_{ij} and P_{ij} are defined before.

The following assumptions are made in the simulations:

1. All packets transmission has equal priority in the network. Yet we are trying to represent different packet data users in terms of different basic rates.
2. There is perfect signaling control between BS and MS, i.e., no channel estimation error, command transmission error, etc.
3. All users have the same distance to BS. Distance of user to BS is not considered in this thesis.

6.1.3 Simulation Procedure

In every program iteration,

1. A new call arrives to the network and initiate connection assignment request.

2. System process resource allocation to new connection request and optimization algorithm is invoked. At same time, admission system decides if the new connection is admitted according to whether there exists feasible assignment with which users' SIR requirements can be met simultaneously.
3. Calculate system performance in each iteration.
4. Rate of each accepted users changes from $R_{\min}$ to $R_{\max}$ of their classes every other window.
5. Output the performance statistic stream to files.
6. Draw performance curves with Matlab and Excel.

The flow diagram of new connection admission process is as shown in Figure 6.1 to 6.3.

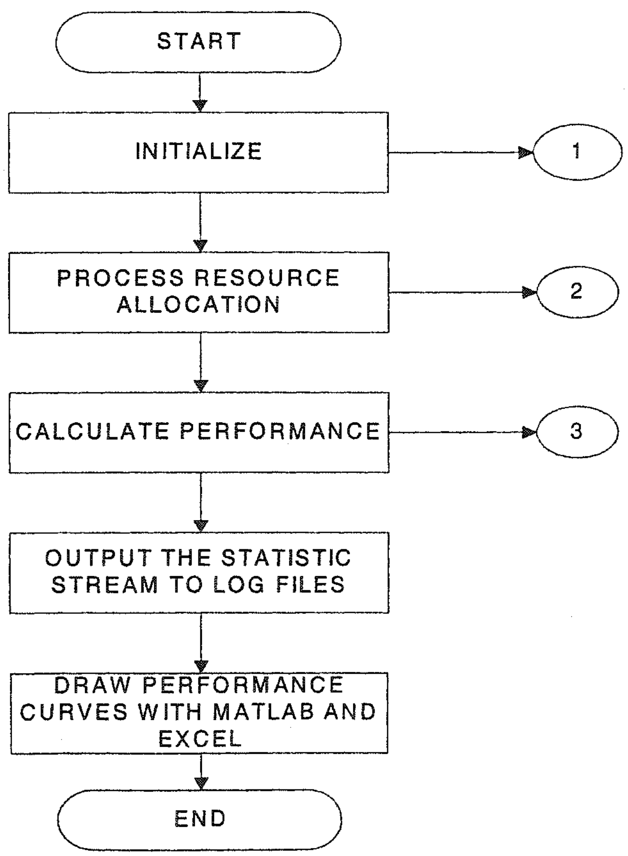


Figure 6.1: The main flow diagram of new connection admission processing

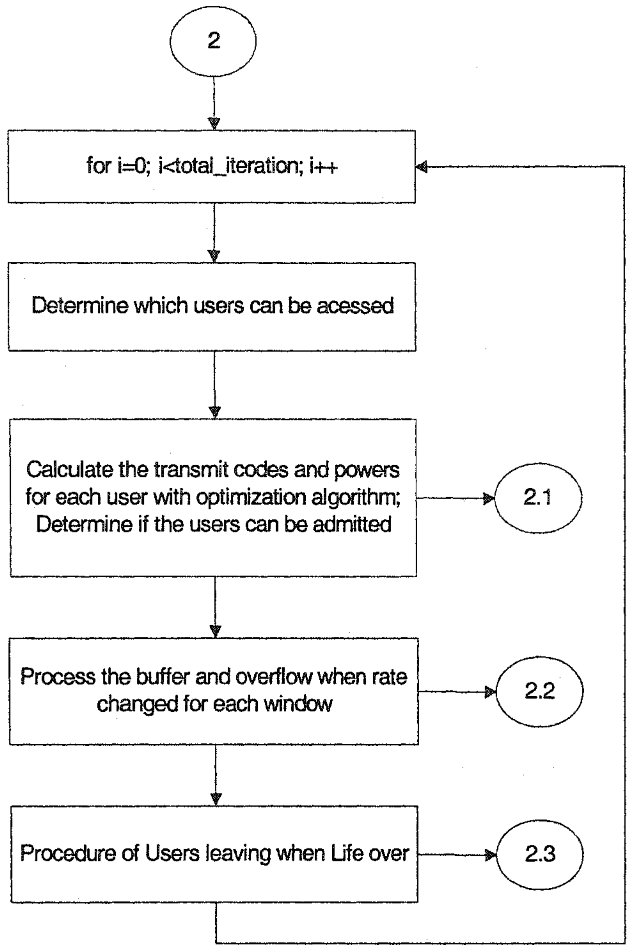


Figure 6.2: The flow diagram of resource allocation processing

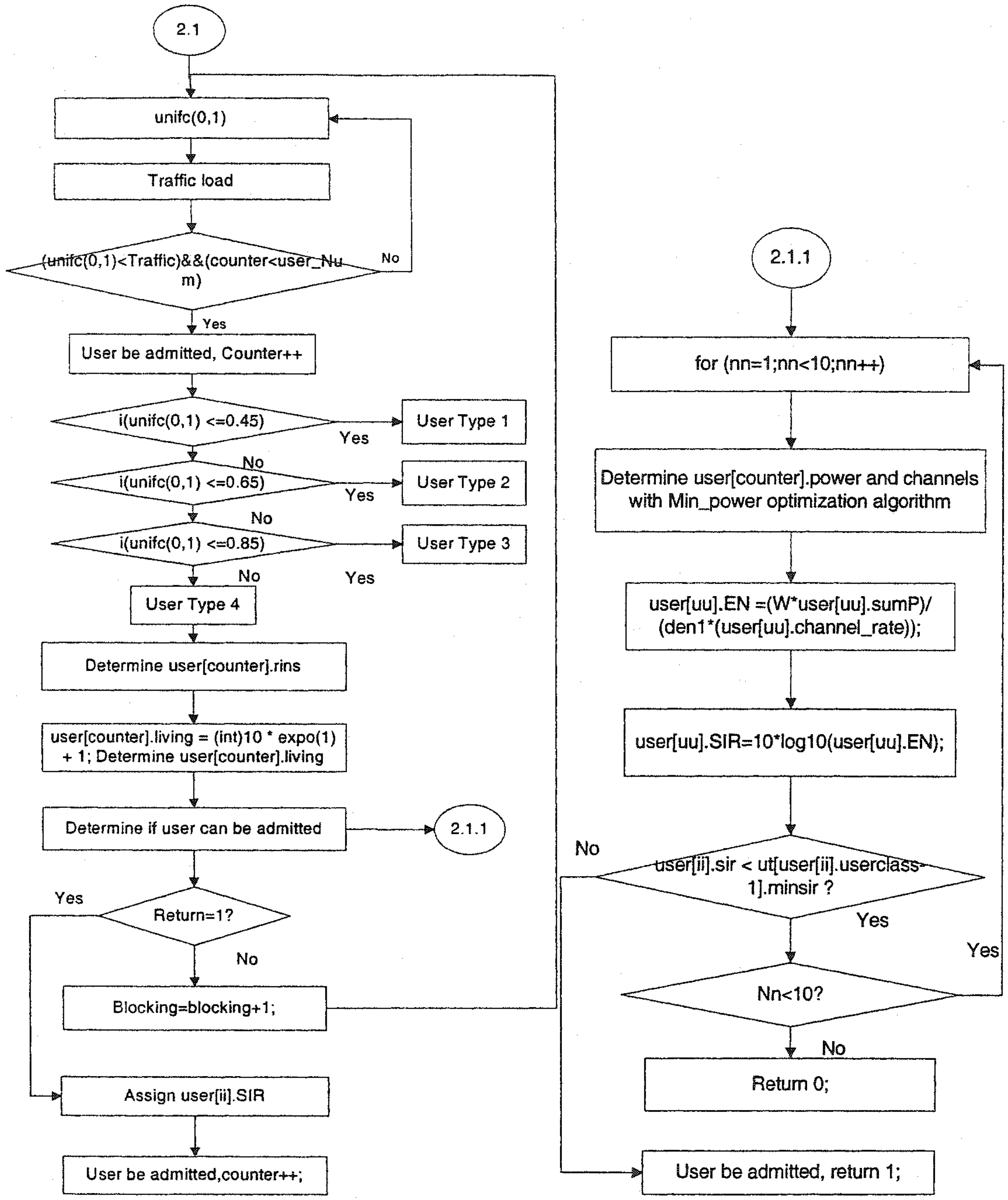
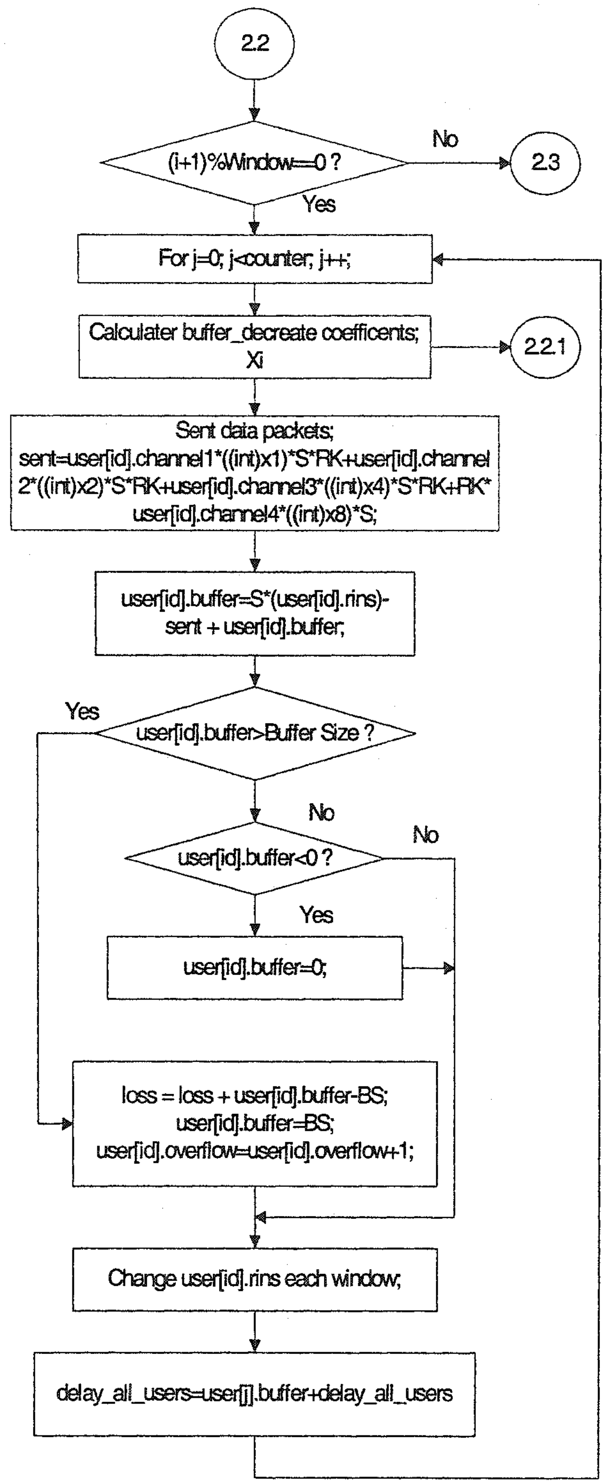


Figure 6.2(a): The detail flow diagram of resource allocation processing



2.2
(i+1)%Window==0 ?
No
2.3
Yes
For j=0; j<counter; j++;
Calculater buffer_decreate coefficents; Xi
2.2.1
Sent data packets;
sent=user[id].channel1*((int)x1)*S*RK+user[id].channel2*((int)x2)*S*RK+user[id].channel3*((int)x4)*S*RK+RK*user[id].channel4*((int)x8)*S;
user[id].buffer=S*(user[id].rins)-sent + user[id].buffer;
Yes
user[id].buffer>Buffer Size ?
No
No
user[id].buffer<0 ?
Yes
user[id].buffer=0;
loss = loss + user[id].buffer-BS;
user[id].buffer=BS;
user[id].overflow=user[id].overflow+1;
Change user[id].rins each window;
delay_all_users=user[j].buffer+delay_all_users

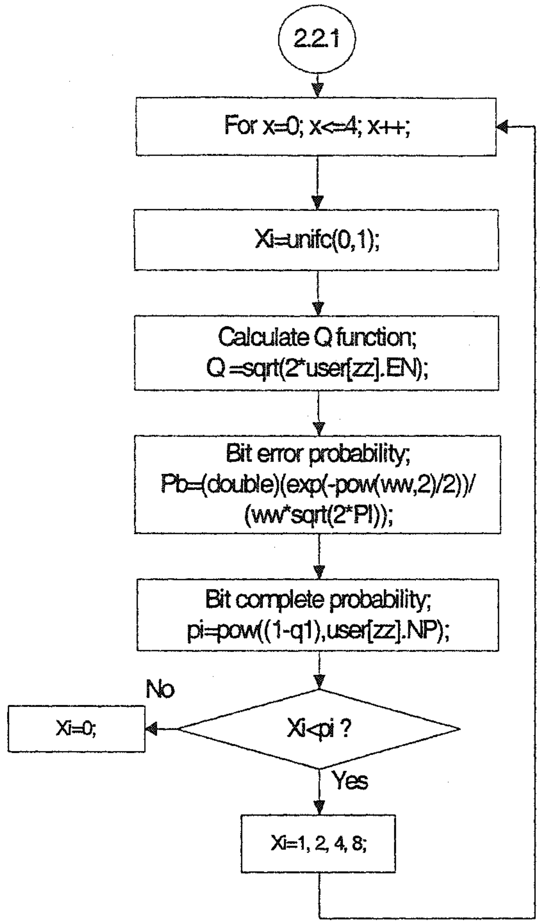


Figure 6.2(b): The detail flow diagram of resource allocation processing

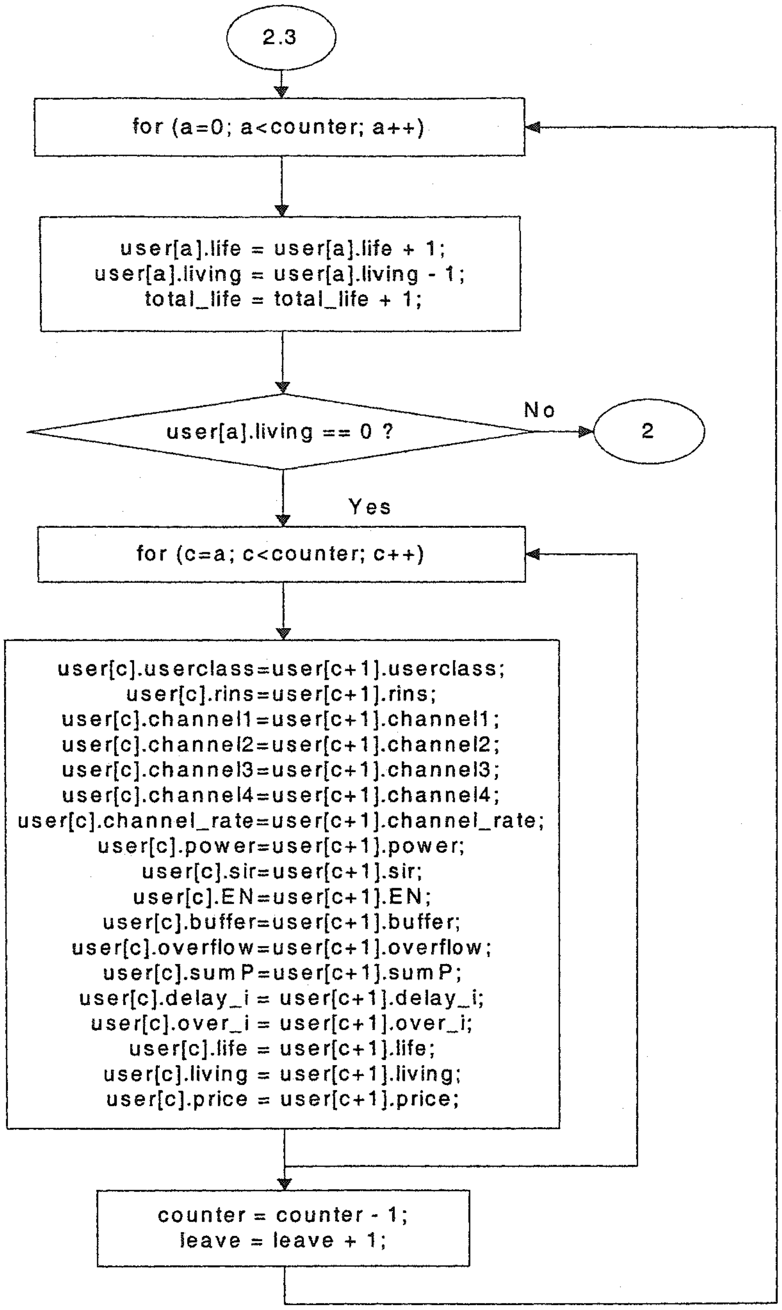


Figure 6.2(c): The detail flow diagram of resource allocation processing

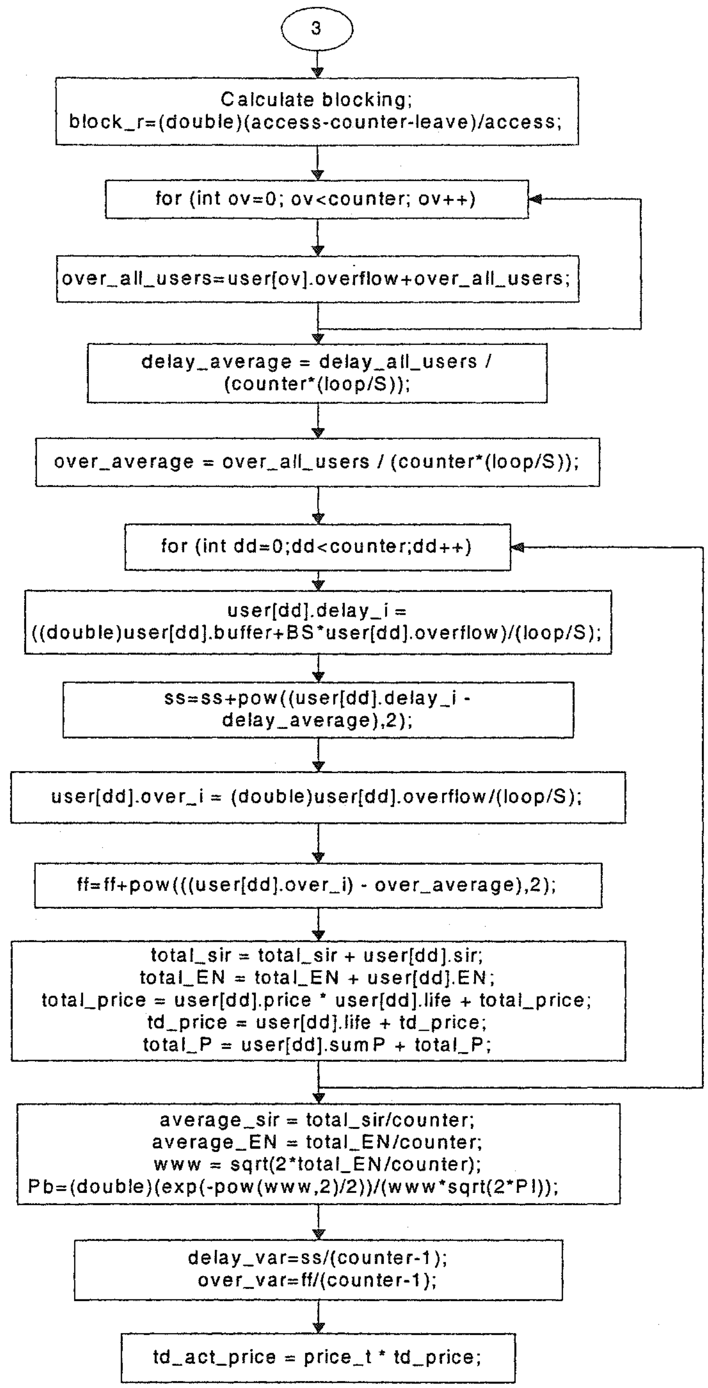


Figure 6.3: The detail flow diagram of performance calculation

6.2 Simulation Case1: Multi-class User CAC

6.2.1 MTP Policies Result Analysis

In this section, we consider CAC performance with the MTP policy.

6.2.1.1 Performance of New Connections

The simulation results for mixed new connections are shown below. The performance of SIR&power, buffer overflow and queuing delay are illustrated in figure 6.4 to figure 6.12. The unit of buffer overflow and queuing delay are described as Kbps. The descriptions of the abbreviations in the legends of Fig 6.4 to Fig 6.27 are following:

Policy: the algorithm that used

Window: the number of iterations of Measurement Window

Bs: buffer size, unit of it is Kbps

NI: Number of iteration

N: number of user

1. Performance of SIR and Power of user

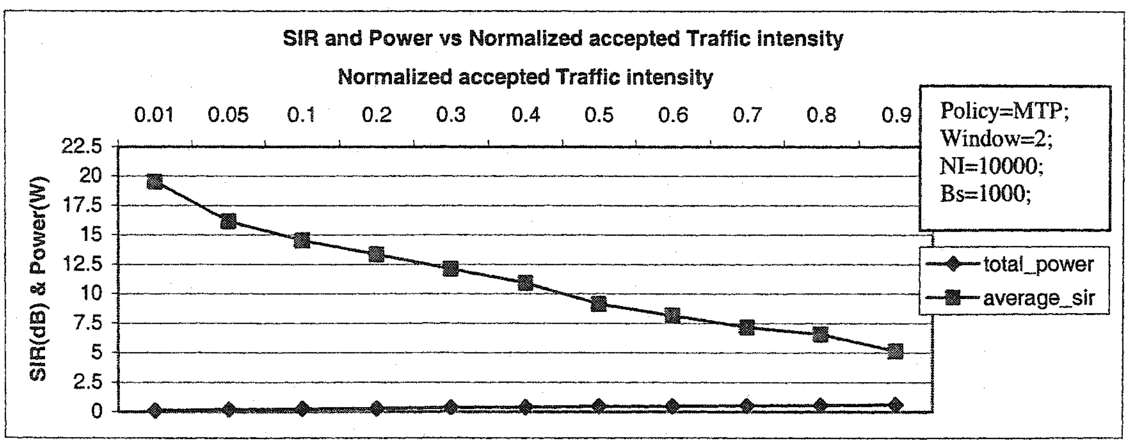


Figure 6.4: SIR and Power vs. Normalized accepted traffic intensity

We access the voice users in this simulation. The meaning of Normalized accepted traffic intensity is explained in section 5.5.2. From Figure 6.4, we can see that the performance changes of a user. With the increase of normalized accepted traffic intensity, i.e. increase of number of user, the SIR decreases rapidly and the system total power increases tardy. It is because CDMA is an interference limitation system.

2. Buffer overflow of different measurement windows

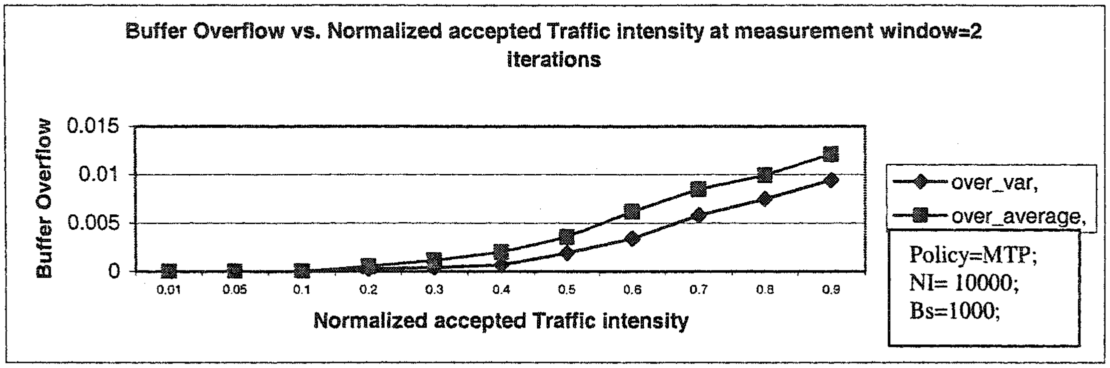


Figure 6.5: Buffer overflows vs. normalized accepted traffic intensity at window=2

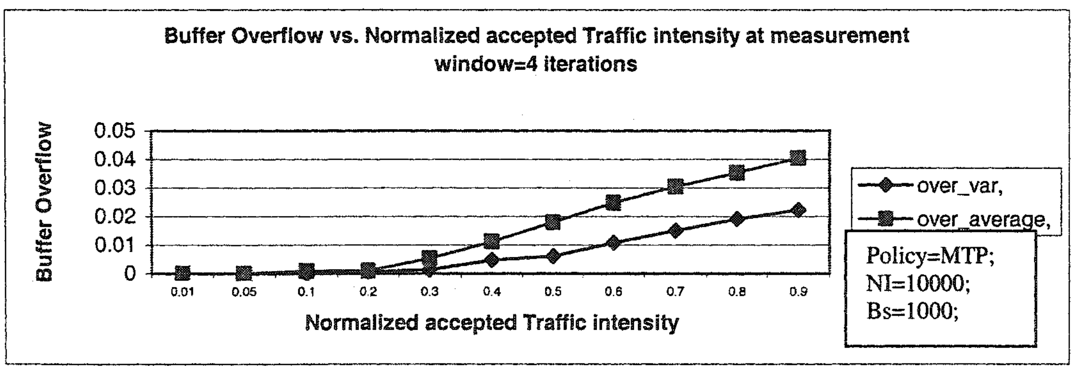


Figure 6.6: Buffer overflows vs. normalized accepted traffic intensity at window=4

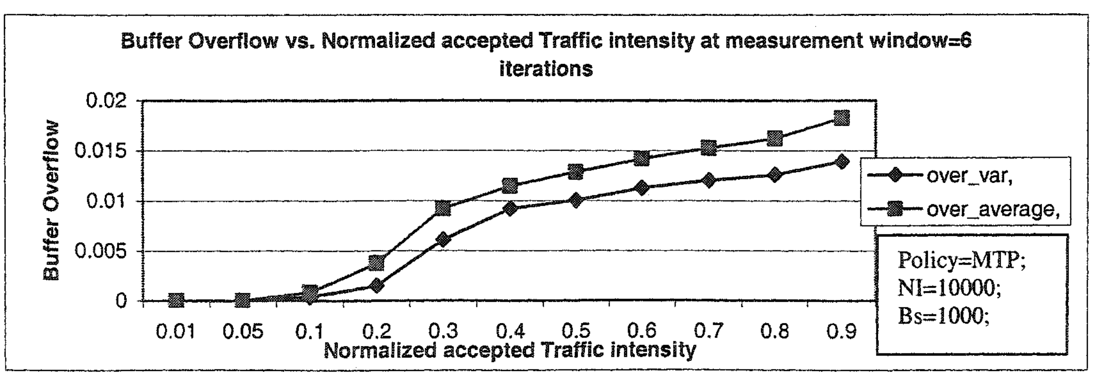


Figure 6.7: Buffer overflows vs. normalized accepted traffic intensity at window=6

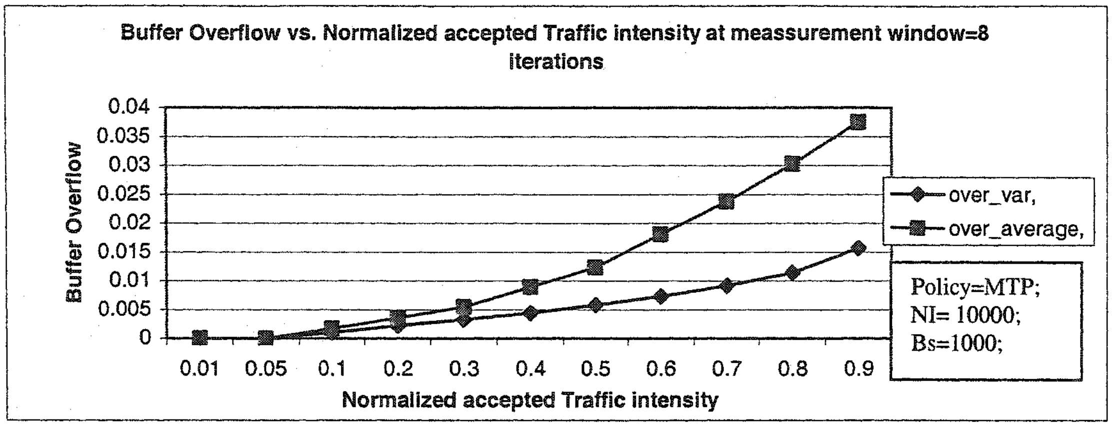


Figure 6.8: Buffer overflows vs. normalized accepted traffic intensity at window=8

The performances of Buffer overflows are shown with four different measurement windows in Figure 6.5 to Figure 6.8 above. The definitions of buffer overflow and measurement window are mentioned at section 5.5.6 and 5.5.2. Accepted traffic intensity means the performance we measured is focused on admitted users. Also, we access randomly the mix four classes' users in these simulations. There are two curves in each figure. Over_average indicates average buffer overflow, while over_var indicate the variance of average buffer overflow. The tendency of curves in these four figures show buffer overflows increase with the increase of normalized accepted traffic intensity, and the data of queuing delay are also reasonable. We also find variance of average buffer overflow is a little bit lower than average buffer overflow in each traffic intensity.

Note that for the different window, the users overflow and queuing delay vary irregularly in same buffer size. This is due to the date rate of connections may change in its range every specific window. The change of buffer overflow would not be shown relationship with window size.

3. Queuing delay of different measurement windows

Queuing delay vs. normalized accepted traffic intensity are shown from Figure 6.9 to Figure 6.12 below.

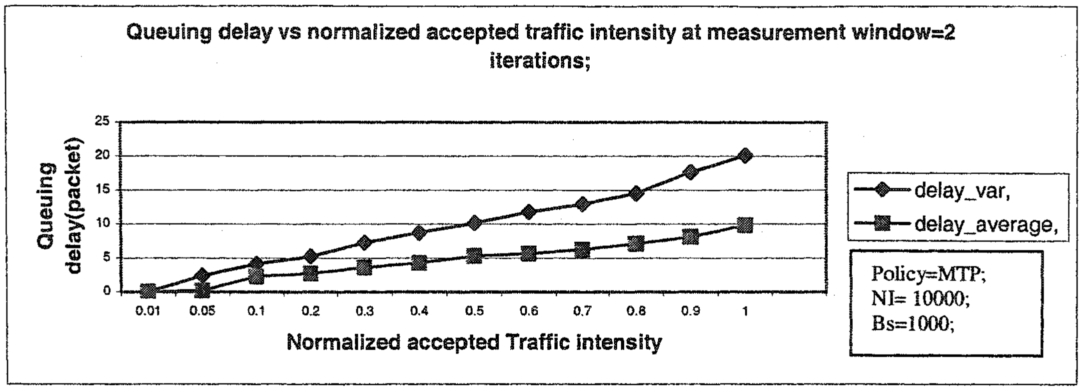


Figure 6.9: Queuing delay vs. normalized accepted traffic intensity at window=2

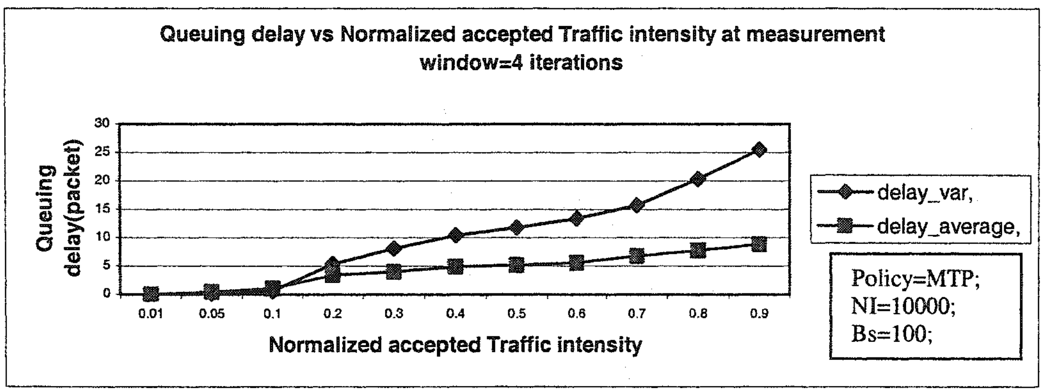


Figure 6.10: Queuing delay vs. normalized accepted traffic intensity at window=4

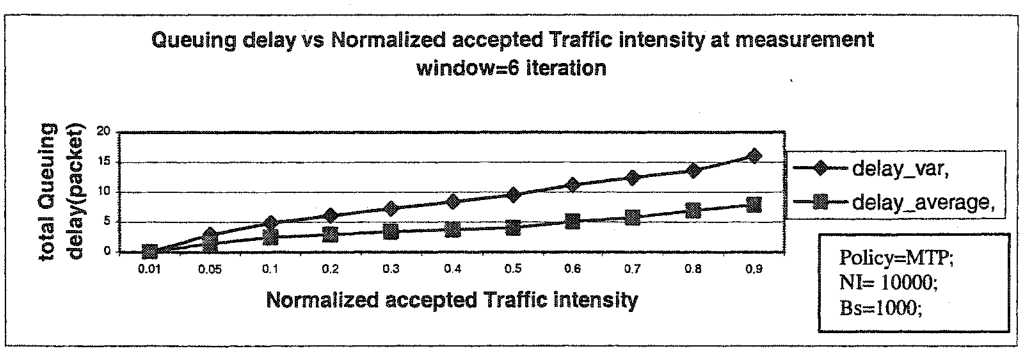


Figure 6.11: Queuing delay vs. normalized accepted traffic intensity at window=6

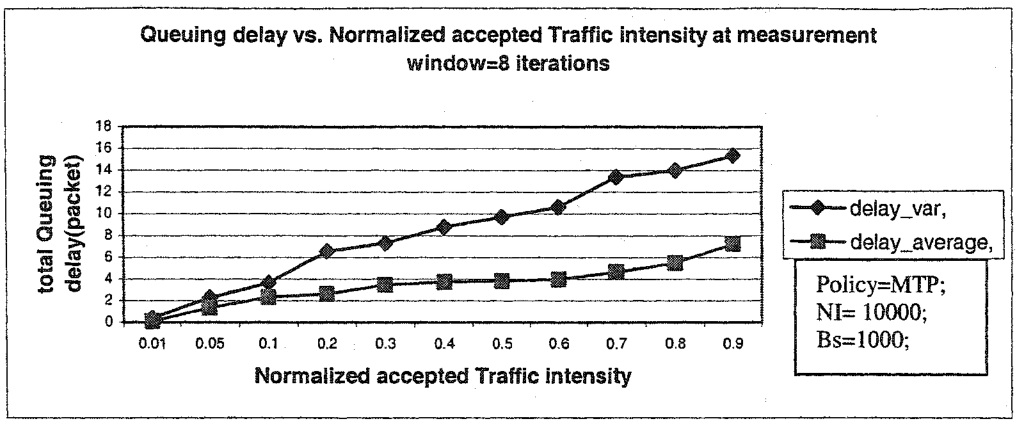


Figure 6.12: Queuing delay vs. normalized accepted traffic intensity at window=8

The mix users of four different classes are accessed randomly in these four simulations. The definitions of buffer overflow and measurement window are mentioned at section 5.5.5 and 5.5.2. Delay_average indicates average queuing delay, while delay_var indicate the variance of average queuing delay. We can see from figures that, the performance varies with different traffic intensity. The tendency of curves in this four figures show queuing delay increase with the increase of normalized accepted traffic intensity. We also find variance of average delay is a little bit higher than average delay in each traffic intensity of each figure.

Some seemingly unreasonable fluctuations may be explained by the random nature of users' data. This phenomenon may also happen in the following simulations.

6.2.1.2 Performance Comparison of Different Traffic Classes

1. **Bit Error Probability**

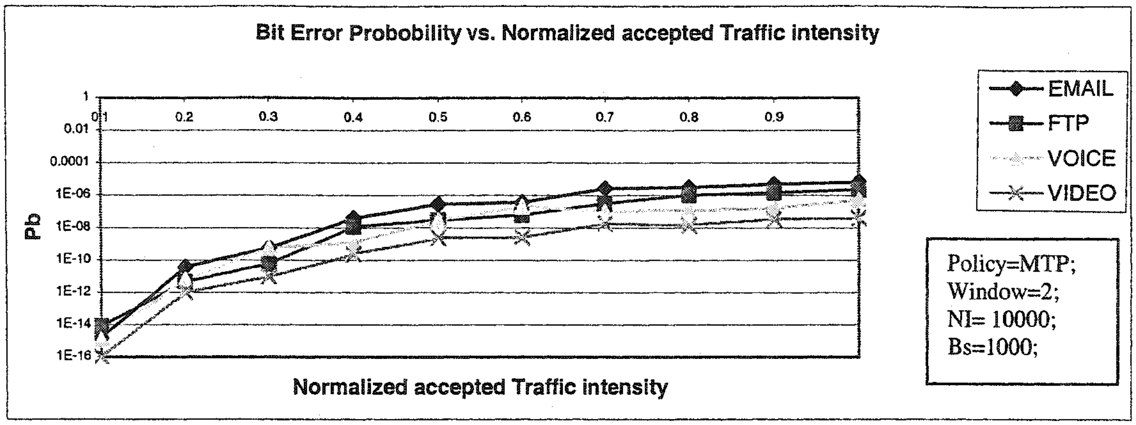


Figure 6.13: Bit error probability vs. normalized accepted traffic intensity

Figure 6.13 shows that bit error probability change with traffic for different user classes. Bit error probability increase as the traffic load increase.

2.packet loss rate

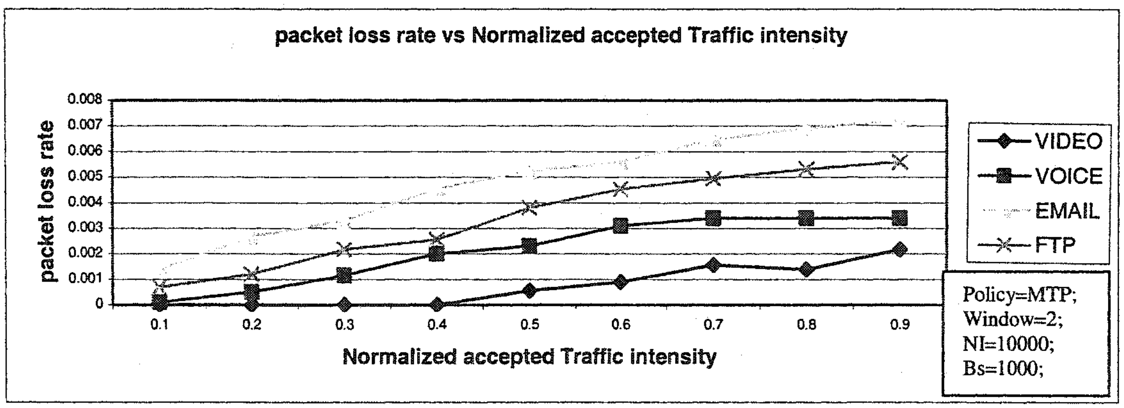


Figure 6.14: Data loss vs. normalized accepted traffic intensity

Figure 6.14 shows data loss for four classes traffic admission. Where, we can see that packets loss increases when the number of user increases. Meanwhile, the data classes like FTP and Email increase quickly than the voice and video. This is because the data classes less sensitive to loss than the voice and video.

3. NCBP of Different Traffic Classes

Comparison of NCBP between different traffic classes when they arrive uniformly is shown in Figure 6.15. Normalized traffic intensity includes the traffic of blocking and admitted later. We also give two specific cases to bright the NCBP performance in Figure 6.16 and Figure 6.17.

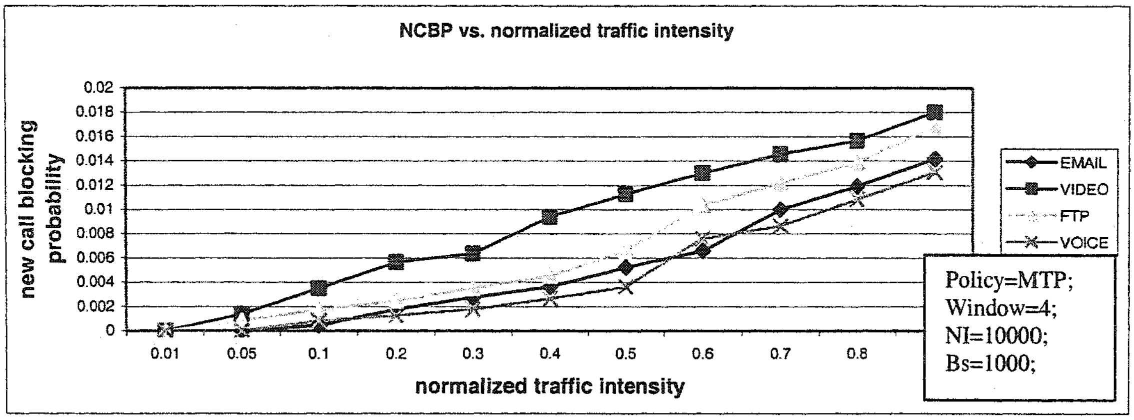


Figure 6.15: NCBP of different traffic classes when they arrive uniformly

Figure 6.15 shows that as connection traffic intensity increases, NCBP for voice and email traffic class increase slightly, while NCBP for video increase rapidly. The difference is duo to the fact that a video connection requires much more resource than a voice connection.

Figure 6.16 shows NCBP vs. normalized traffic intensity of voice, while the traffic intensity for other three traffic classes are 0.3.

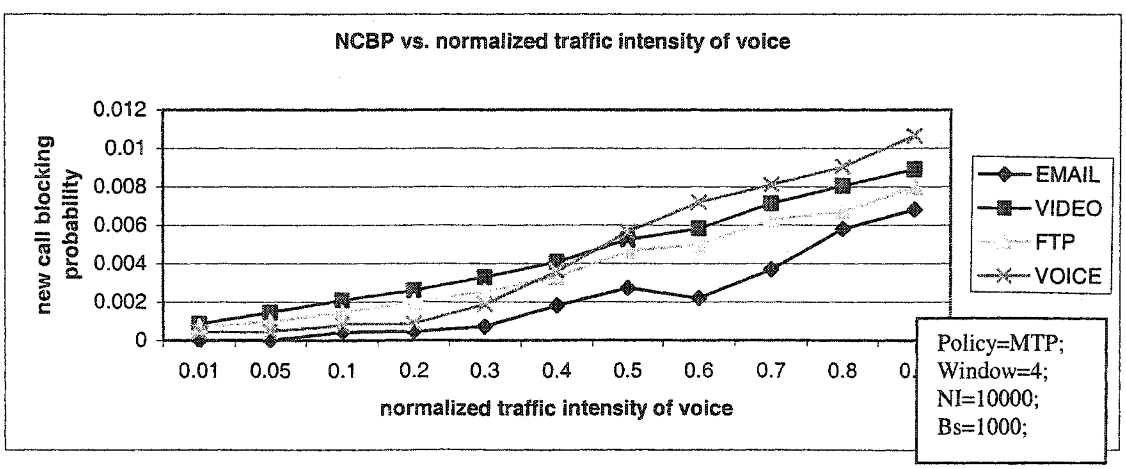


Figure 6.16: NCBP vs. normalized traffic intensity of voice

Figure 6.17 shows NCBP vs. normalized traffic intensity of video, while the traffic intensity for other three traffic classes are 0.25.

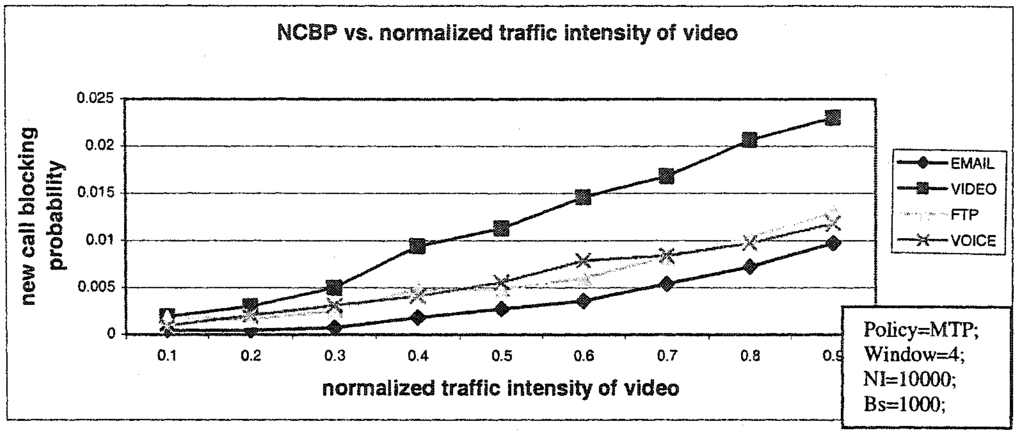


Figure 6.17: NCBP vs. normalized traffic intensity of video

Figure 6.16 and Figure 6.17 show that as voice connection traffic intensity increases, NCBP for video, ftp and email traffic classes increase slightly. On the other hand, as video connection traffic intensity increases, NCBP for voice, ftp and email traffic classes increase rapidly.

All the above figures also show that, when the connection traffic intensity increase, NCBP of each traffic class increases. It is because the available resource is shared by all the connection in the system. Besides, voice connections always receive better NCBP performance than video or data connections, because a voice connection requires much less resource than a video, ftp or email connection does.

6.2.2 Performance Comparison with Different Policies

1. **Delay and Overflow**

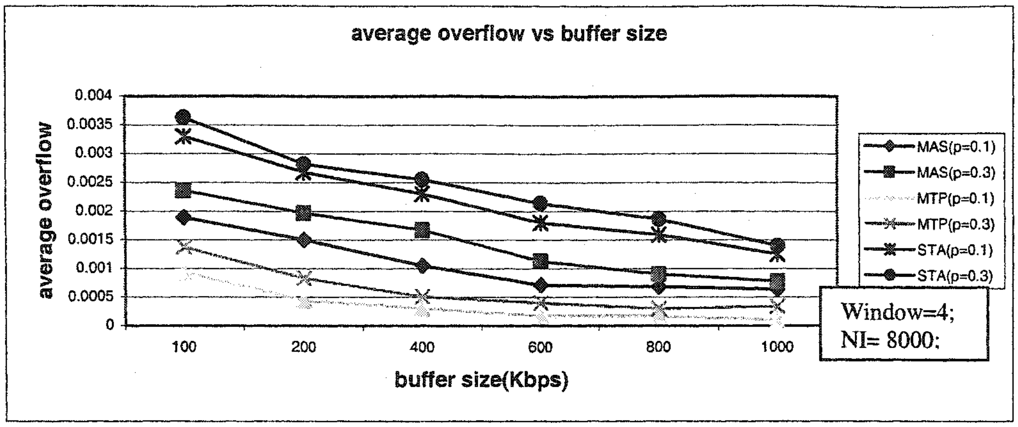


Figure 6.18: Average buffer overflow vs. buffer size

Figure 6.18 shows that average buffer overflow (ABO) in three types of policies relates to different user buffer size and traffic intensity. We access the mixed traffic here. The ρ in the legend indicates the normalized accessed traffic intensity.

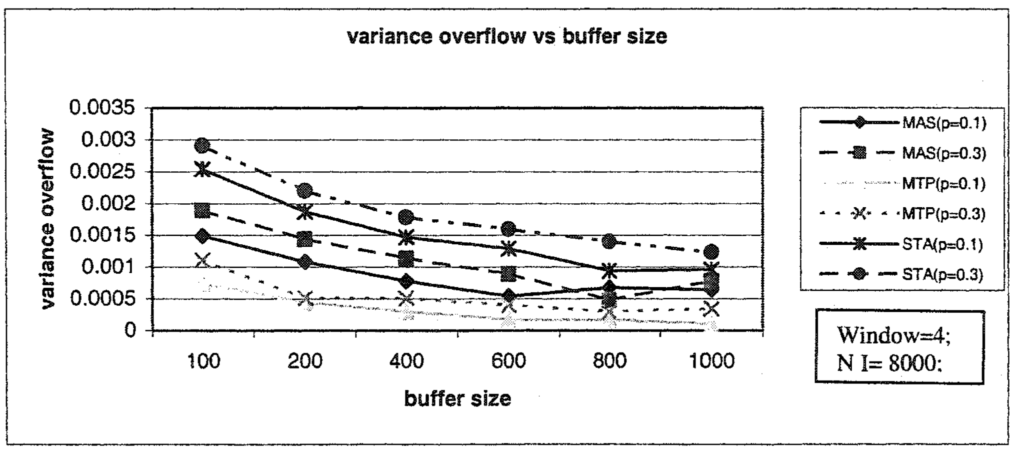


Figure 6.19: Variance buffer overflow vs. buffer size

Figure 6.19 shows the variance of buffer overflow in three types of policies relates to different user buffer size and traffic intensity. The ρ in the legend indicates the normalized accessed traffic intensity here.

We also notice that the bigger of buffer size that are assigned, the less is the buffer overflow. Moreover, in comparison with fixed assignment schemes, the proposed two schemes achieves better overall performance, with very little variance of buffer overflow, and average overflow. Moreover, MPT seems better than MAS.

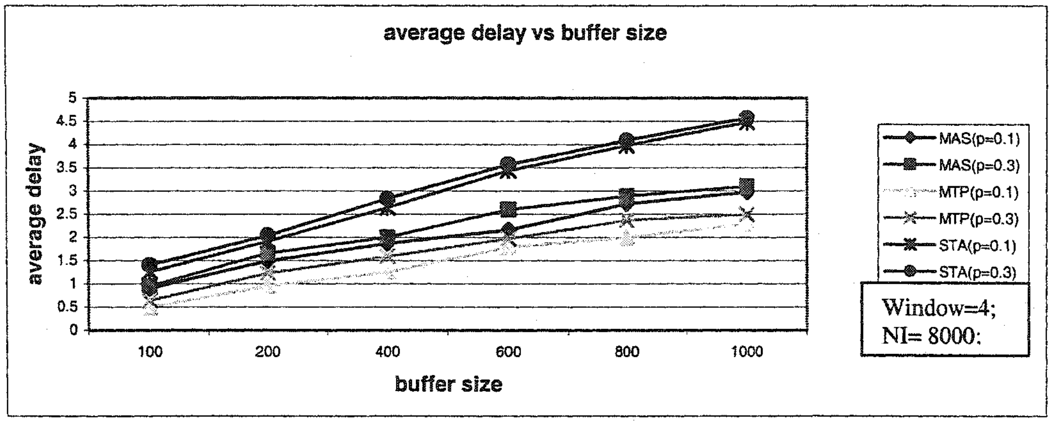


Figure 6.20: Average queuing delay vs. buffer size

Figure 6.20 shows the average queuing delay for the three types of policies relates to different user buffer size and traffic intensity.

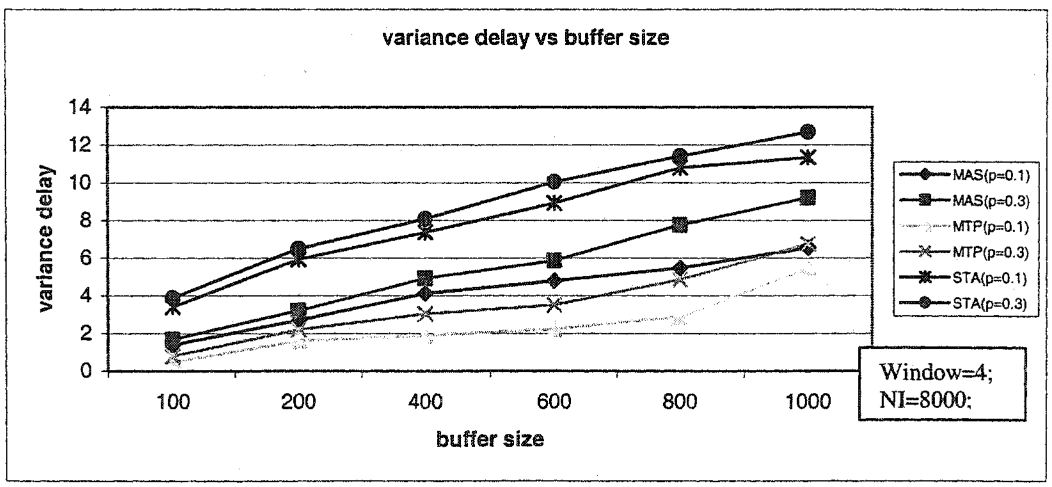


Figure 6.21: Variance queuing delay vs. buffer size

Figure 6.21 shows the variance of queuing delay in the three types of policies relates to different user buffer size and traffic intensity. The average queuing delays increase as buffer size increases.

2.NCBP

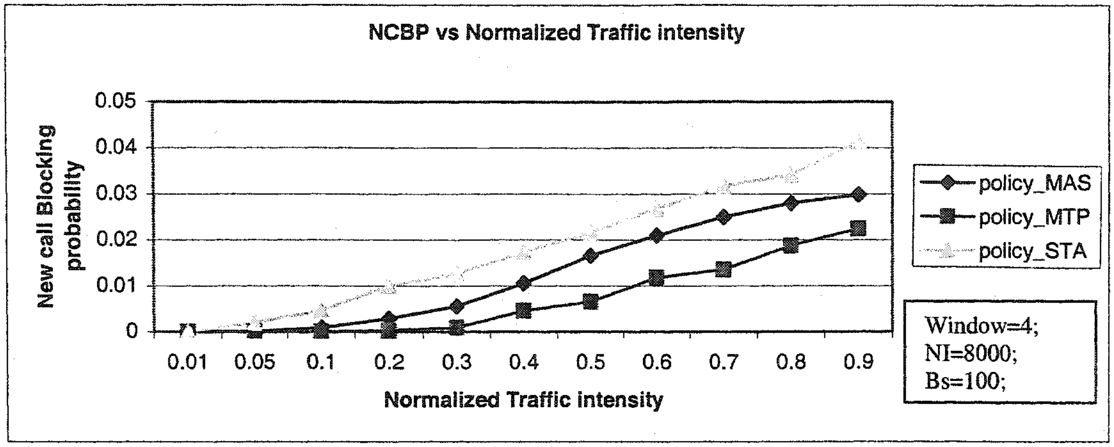


Figure 6.22: Call blocking probability vs. normalized traffic intensity

Figure 6.22 indicates the call blocking probability of the three policies when all users arrive according to uniform distribution. The performance curves are plotted vs. the new connection traffic intensity. The MTP curve indicates the performance of the Minimum Total Power policy. The MAS curve indicates the performance of the maximum SIR algorithm. From figure 6.22, we can see that the proposed policy 1 & 2 exhibit significant improvements on call blocking probability. This is because by using proposed MTP, minimizing total power means more new user can be admitted into system while meeting the QoS requirement. When the traffic intensity is heavy, the blocking probability increases. Moreover, by using the proposed MAS, the QoS of users are granted relatively higher, as they have to meet the best quality of their class, consequently the call blocking probability is further higher.

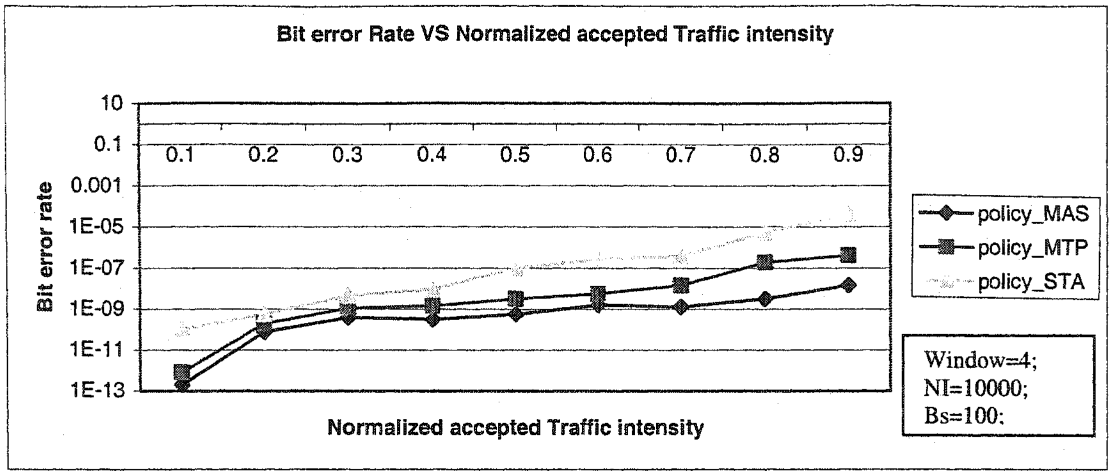


Figure 6.23: Bit error rates vs. normalized accepted traffic intensity

Figure 6.23 indicates the bit error rate of three policies when all users arrive according to uniform distribution. The performance curves are plotted against traffic intensity. From the figure, we can see that the proposed MTP and MAS algorithms exhibit significant improvements on bit error rate than STA. When traffic intensity is heavy, BER increases.

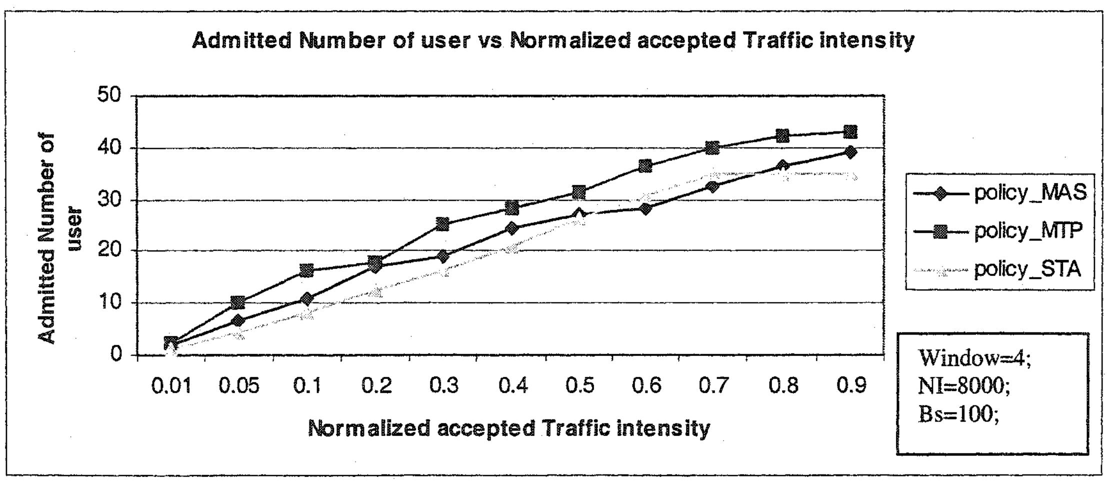


Figure 6.24: Admitted Number of User of three policies vs. traffic intensity

Figure 6.24 shows the comparison of admitted Number of User of three policies when all four classes users arrive according to uniform distribution. The definition of traffic intensity is mentioned at section 5.5.2. The user capacity is a main performance in mobile system. The tendency of curves in this figure shows that, for the curve of using STA policy, the number of user increase stably when traffic raise. But some bad QoS are yielded; for the curves of using MTP and MAS policies, the number of user increase around the curve of STA, but the users got good QoS and resource like power and rate are allocated more reasonable and flexible.

6.3 Simulation Case 2- QoS Based Billing Scheme

6.3.1 QoS Based Billing Scheme Introduction

The effect of adding a new user to the system could be used to establish billing for that user. The concept is to charge more to users who consume more of the total system resource. Unlike fixed QoS systems, the effect of admitting a new user depends heavily on the QoS requirements of the new user. A user with a high QoS requirement will reduce system user capacity more than a user with easily supported QoS requirements and should be commensurately charged more. For example, a user seeking high data rates close to a group of other users will more adversely affect user capacity, through interference and power limitations, than a user seeking a lower data rate in a low interference region. The effect on capacity can be modeled using geometric programming by determining the number of standardized users that could be added to the system both before and after the new user is admitted [22]. The difference between these two numbers can be taken as the reduction in the system (standard) user capacity that results from admitting the new user to the system. The price could then be set as a linear function of this difference. Under this approach, a user could experience different "spot" billing at different times depending on the existing load on the system when the user sought to access the network. Spot billing is different from the current rate based billing approaches, and more realistically models the effect a user has on the network from a revenue potential point of view. Moreover, from a user's point of view, this method is more sound and fair. Better is the quality, the more is the charge.

6.3.2 Simulation Processing and Result Analysis

We use the same system model and simulation parameters of call admission before at our billing simulation. All assumption in call admission simulation is valid in this section. The policy MTP is used when the user is accessed. We compare the performances for the traditional billing method and QoS_based billing method. The main programming flow diagram of billing processing is shown at Figure 6.25. We use C as our computer language too. The minute rate of different classes users that we assumed in this simulation is listed in Table 6.3.

Table 6.3: Minute rates of different classes users ($)

	QoS_based rate	Traditional rate
VOICE	0.20	0.25
VIDEO	0.50	0.25
FTP	0.30	0.25
EMAIL	0.10	0.25

The curve based Table 6.3 is shown at Figure 6.26. The rates of classes are depended on both QoS and real time transmission or not. Mixed four classes admission users were admitted uniformly. For each user, the price is calculated according to the duration of user admission. i.e.

$$\sum_{i=1}^{n} \text{Price}_j = \sum_{i=1}^{n} (\text{duration of admission} * \text{the rate of traffic rate})_j \quad (6.9)$$

Where n is the number of user in traffic class j. For traditional rate calculation model, we adjusted traffic intensity to as same as the other four traffic intensity. The billing data according to QoS based price and traditional price calculation are collected in ten minute (600 iterations). Their performance is shown at Figure 6.27.

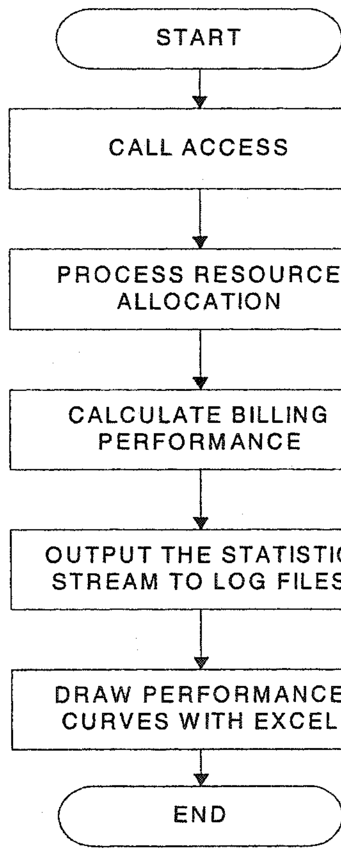


Figure 6.25: The main flow diagram of billing processing

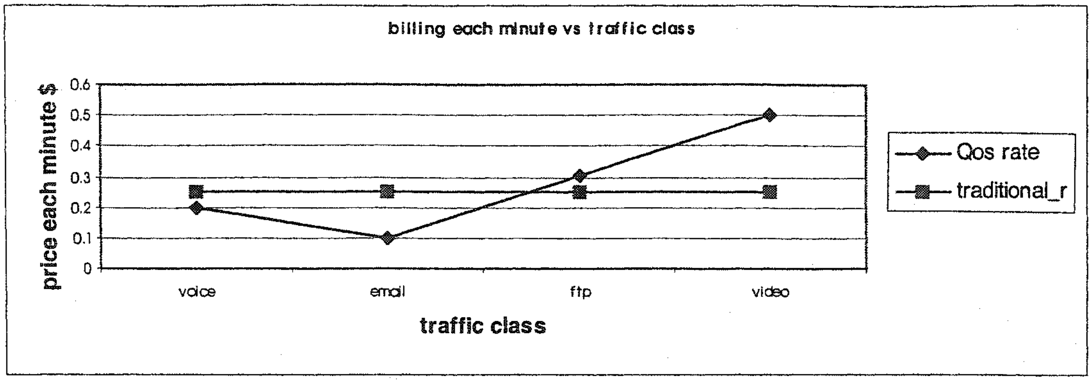


Figure 6.26: Price per minute vs. traffic classes

Figure 6.26 is a comparison of the QoS based rate and traditional rate in a unit time. The user of traditional rate and the user of QoS based rate are assumed work in same time. "traditional_r" in the legend here means the billing is calculated by traditional rate.

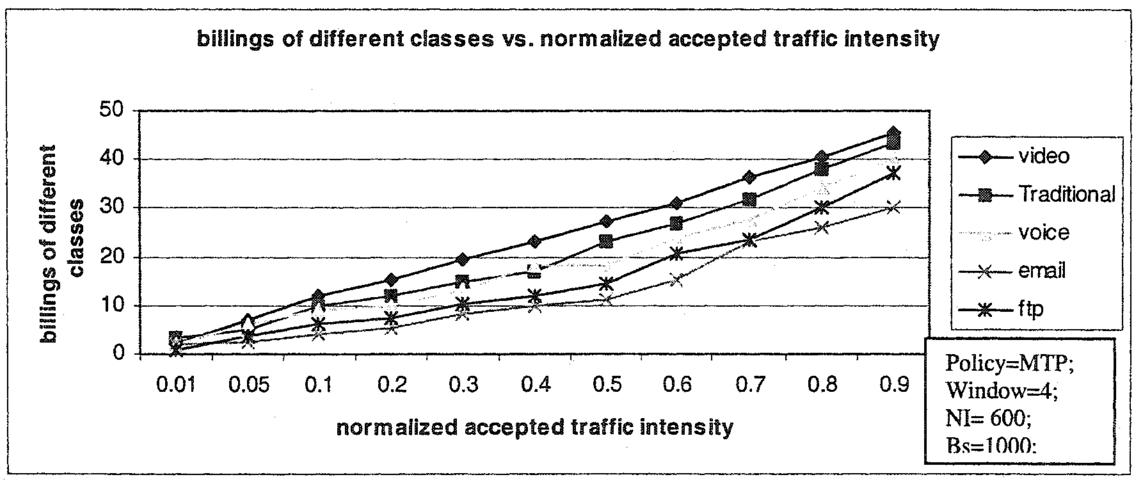


Figure 6.27: Billing of CAC vs. normalized accepted traffic intensity

The simulation result is shown in Figure 6.27. Five curves are drawn with the increase of normalized traffic intensity. The policy MTP is used when the user is accessed. Measurement window is 4 iterations. Buffer size is 1000. We can see from the curve that different traffic users are charged differently. Compared with the traditional method, the QoS based billing method is more reasonable and easy to accept by customers.

6.4 Summary

Numerical results with our three policies have been presented to demonstrate the performance of connection level QoS performance of a single cell CDMA system. The results show that:

1. Compared with other non-joint RRM schemes, the proposed joint RRM scheme can achieve low NCBP while keeping relatively high system resource utilization and QoS;
2. Compared with the MTP (minimum total power algorithm), the MAS (maximum SIR algorithm) result in reduction in system capacity while higher user satisfaction;
3. Due to the resource sharing effect, traffic change in one traffic class affects the performance of all the traffic classes in the system;
4. Voice traffic change has much less effect on the system performance than other traffic classes because a voice connection requires much less system resource than a video or data connection.
5. The billing scheme according to QoS is more reasonable than the traditional billing method, which could be accepted by more users.

Chapter 7

Conclusions and Future Work

7.1 Conclusions

A resource allocation scheme supporting multi-class traffic CDMA system has been proposed in this thesis. The policies achieve effective and efficient radio resource utilization, which can benefit both mobile users and network service providers. From all the work in this thesis, we can conclude that:

- Proposed resource allocation policies are verified effectively by simulation results.
- In multi-class traffic CDMA system, traffic change in each class affects the performance of all the classes in the system. Therefore, improving the efficiency in resource allocation in one traffic class can improve the performance of all the traffic classes in the systems.
- Due to the differences among resource sharing policies, connection policy requiring less system resource always achieves better system capacity than those requiring more system resources.
- Joint consideration of different network characteristics in the RRM in cellular CDMA systems can achieve better resource utilization and effective QoS guarantee.

- Compared with other non-joint RRM schemes, the proposed MTP scheme can achieve low NCBP while keeping relatively high system resource utilization and QoS.
- Compared with the minimum power algorithm, the maximum SIR algorithm results in reducing system capacity but yields higher user QoS.
- Since some objectives are paradoxical, it is impossible to simultaneously optimize all of them. Our research is to improve the system utilization subject to satisfying a constraint on other parameters.

7.2 Future Work

- In the thesis it is assumed that power control is perfect, i.e. it always adapts to variation of the channel and keeps the SIR at a constant required level. While this is realistic when fading is slow, it is interesting to study the resource allocation problem in fast fading environment. There the SIR at the receiver is not constant but randomly distributed due to the error in power control. It is possible to extend the resource allocation scheme in the thesis to this case using adaptive power control based on the mean value of the SIR.
- The optimization algorithms we studied occasionally yield no feasible solution. In this case, we have to rerun the optimization program. Added, these are centralized algorithms. It will more effort to find corresponding distributed algorithms.
- The cellular system demonstrated in the thesis is in a single cell environment. An interesting question is resource optimization of multi-cell CAC scheme based on the estimation of other-cell interference.

- How this system scales is another question that should be considered in future, i.e. what is the capacity of cellular system in a cell environment with cell dimension of several hundred meters to tens of meters.
- This thesis proposes a call admission control scheme in multi-class traffic for CDMA cellular networks. It does not model user mobility. It would be more practical to incorporate these effects and extend the admission control scheme to a more realistic scenario.
- It is also interesting to develop a joint admission and handoff control scheme to better adapt to the dynamic and non-uniformly distributed traffic in CDMA cellular networks.
- In this thesis we focus on solving resource allocation problems in the uplink (from mobile to base station) transmission. Traditionally this has been a bottleneck in the performance of wireless networks due to the difficulties in keeping users synchronized at the base station, the limited handset power and battery life, and the large overhead in packet access control and scheduling. In next generation wireless networks traffic in the downlink (from base station to mobile) is likely to increase significantly due to the need to access and download information from the Internet. Meanwhile performance of uplink transmission will be greatly improved using performance-enhancing techniques such as the antenna array multi-user receiver. Therefore downlink could be a bottleneck and it is desirable to enhance its performance using various technologies. A resource allocation scheme using joint transmit power control and beamforming using antenna array transmitter at the base station could be one of the solutions.

References

[1] T. S. Rappaport, *Wireless Communications - Principles and Practice*, New Jersey, NJ, Prentice Hall, 1996

[2]3GPP2, "The third generation partnership project 2 (3gpp2)," *http://www.3gpp2.org*, 2000.

[3] M. Andersin, Z. Rosberg, and J. Zander, "Gradual removals in cellular PCS with constrained power control and noise." *Wireless Networks*, 2(1):27-43, 1996.

[4] F. Adachi, M. Sawahashi and H. Suda, "Wideband DS-CDMA for next generation mobile communication systems", *IEEE Communications* 36(9) , 1998.

[5] J. T. H. Wu and E. Geraniotis, "Power control in multimedia CDMA networks", *Proc. of IEEE VTC'94*, pp. 789-793.

[6] A. Yener, R. Yates and S. Ulukus, "Joint power control, multiuser detection and beamforming for CDMA systems", in: *Proc. of IEEE VTC'99*, 1999.

[7] E. H. Dinan and B. Jabbari, Spreading codes for direct sequence CDMA and wideband CDMA cellular networks, *IEEE Communications* 36(9) 1998.

[8] A. Sampath, P. S. Kumar and P. Holtzman, "Power control and resource management for a multimedia CDMA wireless system", *Proc. of IEEE International Symposium on Personal, Indoor and Mobile Radio Communications*, Vol. 1, 1995, pp. 21-25.

[9] S. Yao and E. Geraniotis, Optimal power control law for multimedia multirate CDMA systems, *IEEE VTC'95* , pp. 392-396.

[10] T.H. Hu and M. M. K. Liu, "Power control for wireless multimedia CDMA systems", *Electronic Letters* 33(8), 1997, 660-662.

[11] Roger L. Peterson, Rodger E. Ziemer, David E. Borth, *Introduction to Spread Spectrum Communications*, Englewood Cliffs, NJ, Prentice Hall, 1995

[12] W. S. Wong and S. E. Sousa, "Power assignment strategies and traffic control for wireless multimedia DS-CDMA systems", *Proc. of IEEE ICC'95*.

[13] M.G.Jansen and R.Prasad, "Capacity, throughput, and delay analysis of a cellular DS CDMA system with imperfect power control and imperfect sectorization", *IEEE Trans. Veh. Techno,* vol. 44, no. 1, pp. 67-75, 1995.

[14] TIA TR-45.5, "The cdma2000 RTT Candidate Submission," Tech.Rep., U.S. *Telecommunication Industry Association (TIA),* 1998.

[15] R.D. Carsello, "IMT-2000 Standards: Radio Aspects," *IEEE Personal Communications*, August 1997.

[16] F. Leite, "Regulatory Considerations Relating to IMT-2000," *IEEE Personal Communications*, August 1997.

[17] TIA/EIA/IS-95-A, "Mobile Station - Base Station Compatibility Standard for Dual-Mode Wideband Spread Spectrum Cellular System," Tech. Rep., *Telecommunications Industry Association*, 1995.

[18] K.S. Gilhousen, "On the Capacity of a Cellular CDMA System," *IEEE Trans. Veh. Techno.,* vol. 40, no. 2, pp. 303, 1991.

[19] R. Padovani, "Reverse Link Performance of IS-95 Based Cellular Systems," *IEEE Personal Communications*, no. 3, 1994.

[20] L.Song and N.B.Mandayam, "Hierarchical SIR and rate control on the forward link for CDMA data users under delay and error constraints", *IEEE J.Select. Areas. Commun.*, vol. 19, no. 10, pp. 1871-1882, 2001.

[21] N. Bambos, S. Chen, and G. Pottie, "Radio link admission algorithms for wireless networks with power control and active link quality protections." *Proc. of the IEEE Infocom*, 1995.

[22] S. Boyd and L. Vandenberghe, *Convex Optimization and Its Engineering Applications*, Stanford University EE 364 Course Reader 2001.

[23] K. Chawla and X. Qiu, "Throughput performance of adaptive modulation in cellular systems." *IEEE Proc. on Universal Personal Communications*, 2:945-950, 1998.

[24] M. Chiang, D. ONeill, D. Julian, and S. Boyd, "Resource allocation for QoS provisioning in wireless ad hoc networks." *Proc. IEEE Globecom*, San Antonio, TX, Nov. 2001.

[25] G. Foschini and Z. Miljanic, "A simple distributed autonomour power control algorithm and its convergence." *IEEE Trans. Vehicular Technology*, 42(4), 1993.

[26] J.Yang,I.Koo, Y.Chio and K.Kim, "a dynamic resource allocation scheme to maximize throughput in a multimedia CDMA system", *Proc. IEEE VTC 2000*, Fall, pp.348-351.

[27] D. Julian, M. Chiang, and D. ONeill, "Robust and QoS constrained optimization of power control in wireless cellular networks." *Proc. IEEE VTC Fall*, Atlantic City, NJ, Oct. 2001.

[28] S. Kandukuri and S. Boyd, "Optimal power control in interference limited fading wireless channels with outage probability specifications." *To appear IEEE JSAC*, 2002.

[29] 3GPP, "The third generation partnership project (3gpp)," *http://www.3gpp.org.*

[30] Yu. Nesterov and A. Nemirovsky, *Interior Point Polynomial Methods in Convex Programming,* SIAM 1994.

[31] Tero Ojanpera, Ramjee Prasad, "*Wideband CDMA For Third Generation Mobile Communications*", Artech House Publishers.

[32] R. Yates, "A framework for uplink power control in cellular radio systems." *IEEE J. Sel. Areas in Comm.*, 13(7): 1341-1347, 1995.

[33] Bernard Sklar, "*Digital communications, fundamentals and applications*", Prentice Hall PTR, Communication Engineering Service, Tarzana, California. University of California, Los Angeles.

[34] O. Yu and V. Leung, "Adaptive Resource Allocation for Prioritized Call Admission over an ATM-Based Wireless PCN," *IEEE J. Select. Areas. Commun.*, vol. 15, no. 7, pp. 1208, 1997.

[35] J.B.Kim and B.Jabbarri, "performance analysis for a CDMA network with mixed traffic using uniquely power control", *IEEE Trans. Veh Technology*, 49(2), Mar 2000.

[36] Zhao Liu and Magda El Zarki, "SIR-Based Call Admission Control for DS-CDMA Cellular Systems," *IEEE J. Select. Areas. Commun.,* vol. 12, no. 4, pp.638, 1994.

[37] Yoshihiro Ishikawa and Narumi Umeda, "Capacity Design and Performance of Call Admission Control in Cellular CDMA Systems," *IEEE J. Select. Areas. Commun.,* vol. 15, no. 8, pp. 1627, 1997.

[38] W. Yang and E. Geraniotis, "Admission Policies for Integrated Voice and Data Traffic in CDMA Packet Radio Networks," *IEEE J. Select. Areas. Commun.*, vol. 12, no. 4, pp. 654, 1994.

[39] C. Chao and W. Chen, “Connection Admission Control for Mobile Multiple-Class Personal Communications Networks," *IEEE J. Select. Areas. Commun.,* vol. 15, no. 8, pp. 1618, 1997.

[40] D. Levine and I. Akyildiz, “A Resource Estimation and Call Admission Algorithm for Wireless Multimedia Networks Using the Shadow Cluster Concept," *IEEE/ACM Trans. Networking,* vol. 5, no. 1, pp. 1, 1997.

[41] A.K.Elhakeem, “Notes of ELEC6141”, 2001.