Adaptive Radio Resource Management for Mobile Satellite Systems

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A Thesis Submitted to the

Faculty of Engineering and the Built Environment

Department of Electrical Engineering

For the Degree of Master of Science in Engineering

University of Cape Town

South Africa

December 1, 2007

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Abstract

The next generation wireless networks will realised the true broadband communication. The existing research to optimise the bit rate had explored numerous techniques, which include adaptive modulation and coding (AMC) scheme. With this technique, a channel is estimated and based on target BER, certain parameters of the transmitted signals are dynamically changed for efficient utilisation of scarcely available radio resources.

In this thesis, a set of unique strategies and enhanced schemes for adaptive CDMA modulation are devised. A graded resource system is proposed for better radio resource management. Subsequently, a successful adaptive CDMA algorithm is designed and a prioritised processing gain for adaptive CDMA algorithm in satellite system is introduced. The idea of the critical section in the downlink system when a user controller scheme has to be activated to improve the performance is initiated. The diversity technique and rate compatible punctured turbo-code (RCPT), which has been found to give improved throughput performance in a direct sequence (DS) CDMA, are exploited. An analytical process and simulations were carried out. The algorithm presented has the potential to improve reliability, availability, performance and robustness. These improvements arise from radio resource management algorithms. The threshold based algorithm method adopted in this work have the capability to optimise both the throughput and performance of wireless systems .

The proposed adaptive CDMA algorithm was simulated and the results compared with existing systems. The simulation results show that the proposed adaptive CDMA algorithm provides better throughput performance. Also a remarkable gain can be obtained when the processing gain is prioritised over modulation and coding scheme (MCS). The throughput

performance of the algorithms is dependent on the channel situation. The influence of channel parameters on these relative gains is evaluated in this work as well. Furthermore, it is concluded that the spectral efficiency depends on the number of available resources for adaptation at the transmitter. Finally, the benefit of using adaptive modulation system in terms of spectral efficiency and probability of bit error for different applications is examined on the proposed scheme.

List of Publications

[1] Adeyemi A. Ajibesin and Emmanuel O. Bejide, "A Priority-based Adaptive CDMA Algorithm for Multimedia Wireless Systems," Australasian Telecommunications Networks and Applications Conference, ATNAC2007/IEEE, Christchurch, New Zealand, 2 - 5 December 2007.

[2] Adeyemi A. Ajibesin and Emmanuel O. Bejide, "An Agent Characterised BPSK/QPSK Modulation for Adaptive CDMA-based Rice-lognormal Channel," IEEE Africon, Windhoek, Namibia, 26 - 28 September 2007.

[3] Adeyemi A. Ajibesin and Emmanuel O. Bejide, "An Agent-based Adaptive Modulation for next Generation Mobile Satellite System," South African Telecommunications Networks and Applications Conference, SATNAC2006, Spier Wine Estate, Cape Town, 3 - 6 September 2006.

List of Abbreviations and Acronyms

3G *Third Generation mobile phone system*

4G *Fourth Generation mobile phone system*

ACI *adjacent cell interference*

AMC *adaptive modulation and coding*

ARRM *adaptive radio resource management*

AWGN *additive white Gaussian noise*

BER *bit error rate*

BPSK *Binary Phase Shift Keying Modulation*

DS-CDMA *direct sequence-code division multiple access*

DVB-S2 *Digital Video Broadcasting via Satellite*

ECC *error control code*

ETSI *European Telecommunication Standards Institute*

FDMA *frequency-division multiple access*

FES *fixed Earth stations*

FSL *free space loss*

G/T *receiving antenna gain over system noise temperature*

IMT *International Mobile Telecommunications*

LOS *line of sight*

ISI *inter symbol interference*

ITU *International Telecommunication Union*

K *Rice-factor*

LEO *low Earth orbit*

LLR *log likelihood ratio*

MA *multiple access*

MAI *multiple access interference*

MAP *maximum a posterior*

MCS *modulation and coding scheme*

MEO *medium Earth orbit*

MQAM *M-ary quadrature amplitude modulation*

NCC *network control centre*

PCE *power control error*

PDF *probability density function*

PLMN *public land mobile network*

PN *pseudo random noise*

PSTN *public switched telephone network*

QAM *quadrature amplitude modulation*

QoS *quality of service*

QPSK *quadrature phase shift keying modulation*

RCPT *rate compatible punctured turbo-code*

RLM *Rice-lognormal model*

RSC *recursive systematic convolutional*

SCC *satellite control centre*

SINR *signal to interference plus noise ratio*

SIR *signal to interference ratio*

SISO *soft-input soft-output*

SNR *signal to noise ratio*

SOVA *soft-output Viterbi algorithm*

S-PCN *satellite personal communication networks*

TDMA *time-division multiple access*

UMTS *Universal Mobile Telecommunications Services*

Chapter 1

Introduction

Radio resource management (RRM) is the system level control of radio transmission characteristics in wireless communication systems [1, 2]. Parameters such as transmit power, transmit rate and modulation scheme are controlled. The objective is to utilize the limited radio spectrum resources and radio network infrastructure as efficiently as possible. In static radio resource management fixed resource allocation are performed. Adaptive radio resource management scheme dynamically adjust the radio network parameters to the traffic load, user positions, quality of service requirements, etc. Adaptive RRM schemes have been considered in the design of wireless systems [3, 4], in view to maximize the system spectral efficiency and to achieve performance in order to satisfy the QoS required for fading channels [5]. Changing from relatively static radio resource management techniques generally in use today to dynamic methods like those discussed in this thesis helps to increase capacities and improve performance of wireless systems.

Examples of adaptive RRM schemes are power control algorithms, admission control, adaptive filtering, and adaptive modulation and coding. In this work, adaptive modulation and coding is considered. AMC is one of the techniques, which is defined in the third generation wireless system to improve the throughput on fading channels [6]. AMC is defined in wireless communications to denote the matching of the modulation, coding and other signal parameters to the conditions on the radio link [7]. These conditions involve pathloss, the interference due to signals coming from other transmitters, the sensitivity of the receiver and the available transmitter power margin [7].

1.1 Motivation

The driving force and main design objective of third-generation systems, as defined by the Universal Mobile Telecommunications Services/International Mobile Telecommunications in the year 2000 (UMTS/IMT-2000), has been high bit rate services [8]-[10]. Third generation systems should be able to offer at least 144 kb/s (preferably 384 kb/s) for high mobility users with wide area coverage and 2 Mb/s for low mobility users with local coverage [11, 12]. The need for high bit rate services, together with the scarce spectrum, motivate the development of more spectrum efficient radio technologies. The goal of UMTS/IMT-2000 is to support a large variety of services with different quality of service requirements, that is, multimedia services with bandwidth on demand [13]-[15]. The technical challenge is to achieve the required flexibility without overwhelming complexity in the network and terminal [13]-[15]. Another challenge is the fourth generation wireless systems (4G) requirements to achieve true broadband access and to provide the spectral efficiency needed for some applications that 3G is incapable of supporting at low cost. AMC is a solution to these problems. RRM algorithms that guarantee the required quality for all users in a fair manner is required [16].

1.2 Objective of Thesis

The objective of this thesis is to propose a set of unique strategies and prioritised AMC schemes, which are efficient for adaptation in broadband direct sequence code division multiple access (DS-CDMA) satellite networks. This could lead to an optimised utilisation of scarcely available radio resources for the realisation of next wireless generation in satellite systems.

To achieve the above objective two processes have to be followed

- Develop an efficient adaptive CDMA scheme
- Evaluate the schemes via simulation

The purpose of adaptive CDMA algorithms is to change some parameters in satellite transmission for the optimisation of available radio resources. The adaptive CDMA system has the advantage of flexibility, and it can provide high transmission quality and throughput.

The scope of the thesis focuses on adaptive CDMA algorithms for mobile satellite systems, but some references will be made to adaptive modulation and coding in CDMA and adaptive selection of multiple access (MA) scheme. The work is divided into three parts: satellite channel and system modelling, modulation and coding scheme (MCS), and an adaptive CDMA algorithm. Instant and perfect channel estimation is assumed to be available for systems in this thesis.

1.3 Problem Description

One of the important properties of a wireless channel is its non-deterministic characteristic. This results in a stochastic performance and throughput. In other words, system performance is not optimised for a fixed wireless system because the mobile communication channel is varied in time. Consequently, the channel bandwidth cannot be used efficiently. However, in wireless communications, spectrum is the most costly parameter that determines the rate at which information can be transmitted. For many years, researchers have been finding ways of either conserving or exploiting spectrum for its most efficient use. Also, it was reported that to achieve broadband multimedia services, which was the next phase in the development of mobile satellite networks, the L-/S-bands (390-1550Hz/1550-3900MHz) already set aside for Satellite-UMTS/IMT-2000 had a limited bandwidth and would not be suitable for high-speed applications [17, 18]. It will be necessary to move up in the frequency band to an allocation where sufficient bandwidth is available [19]. The next suitable frequency band is the Ka-band (17250-36000MHz) [20]. However, an inherent problem of this band is the channel characteristic, which is discussed in the next chapter. The most notable characteristic of land mobile channel at this frequency is *fading* due to shadowing [20].

1.4 Related Work and Approach

Several techniques have been identified to counteract the effect of fading [19]. One of these techniques is adaptive modulation and coding (AMC). The use of AMC techniques is particularly recommended to address the fading issue and to improve the availability of the mobile link [19, 20]. Moreover, an effective technique is needed to improve the throughput and to meet the quality of service (QoS) required for next generation services as defined in the UMTS standard by the ITU for satellite mobile systems [16]. Some of the techniques in the literature which attempted to solve this problem include *variable-rate* techniques, in which the symbol rate is fixed while changing the constellation size or modulation type [21, 22]; *variable-power* techniques, in which the power adaptation inverts the channel fading so that the channel appears as an AWGN channel to the modulator and demodulator [22, 23]; and *Variable-coding* techniques, whereby different channel codes are used to provide different amounts of coding gain to the transmitted bits [24, 25]. Variable-coding is implemented by multiplexing different error correction capabilities. This method uses a strong error correction code when the strength of the received signal is small, and a weak code or no coding when the strength of the received signal is large [26, 27]. However, because of complexity, modulation must remain fixed [27]. An alternative technique to this is the use of *rate-compatible punctured turbo (RCPT) codes*, which is an effective adaptive coding technique [28]. In our work, we considered adaptive modulation and coding (AMC) and exploited the advantages of RCPT codes to solve the complexity problem imposed by adaptive modulation.

Many papers present different approaches to the implementation of adaptive techniques. Some of these approaches is described in [29].

In this work, an improved and robust AMC scheme that is based on the threshold method is examined for throughput maximization in broadband CDMA satellite networks. The processing gain is prioritised over MCS. Satellite path diversity using RAKE receiver technique is exploited, as well as RCPT. Simulations were carried out. The simulation result shows an improved average throughput performance over traditional modulation and coding scheme.

1.5 Contributions

The contributions of the thesis are the following;

- Proposed a prioritised processing gain adaptive CDMA algorithm in Ka-band LEO broadband CDMA satellite networks.
- Evaluate the BER performance and throughput of the proposed adaptive CDMA algorithm and compare the results with the existing system of modulations.
- Combine different adaptive CDMA schemes, which composed of different grades of resources set and examine their throughput maximization.
- Investigate the proposed adaptive CDMA algorithm for different applications in Ka-band LEO broadband CDMA satellite networks.

1.6 Outline of the Thesis

Figure 1.1 presents the organization of the thesis. This thesis starts by providing the motivation, which includes the description of some requirements and current challenging technical problems associated with wireless transmissions. It further lays out the objectives to be achieved by the end of the research work. One of the problems affecting the wireless transmission, fading, is described. Related works, which have been reported in the literature including the techniques, the methods employed and our approach are briefly discussed.

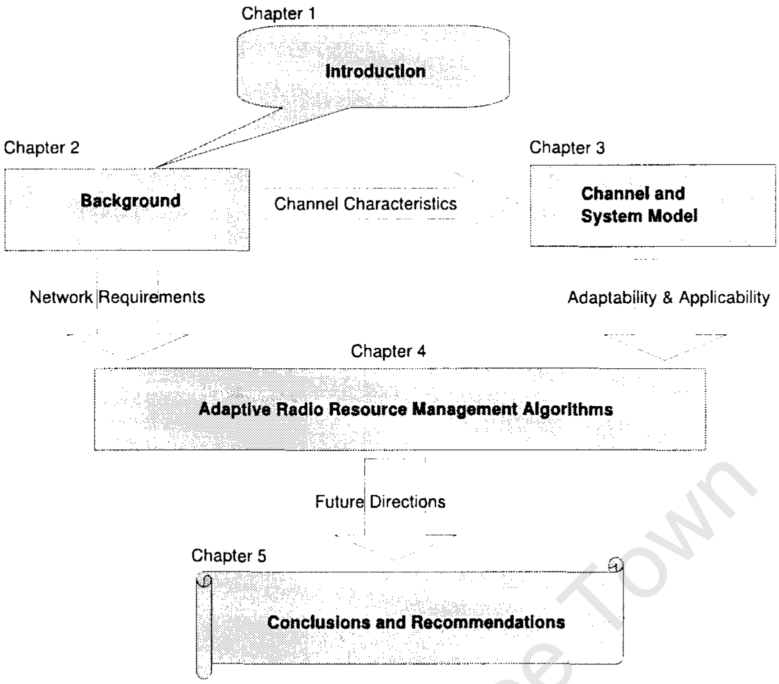

Figure 1.1: Thesis Outline Flowchart

The expected contributions to knowledge are highlighted and the thesis outlines are briefly discussed.

In Chapter 2, the background of the thesis is discussed. The mobile satellite network requirements are identified. The existing satellite personal communication networks (S-PCN) for mobile satellite systems are compared with the proposed satellite-UMTS, which have been accepted for the next generation satellite network system. One of the existing satellite channel models is examined for consideration in our system model.

In Chapter 3, the channel and the system model is presented, and their performances are evaluated via simulation.

In Chapter 4, the concept, architecture, design objective, and problem definition are described. Formula derivations, development process of the AMC, strategy, and work procedure of adaptive CDMA modulation are discussed and analysed. The adaptivity is based on the current channel condition described in Chapter 2 and 3. The proposed adaptive CDMA modulation algorithms for satellite channel with prioritised processing gain is presented. The algorithm is evaluated for throughput performance and compared with the existing AMC system. The effect of varying the BER is examined for system flexibility. Finally, the performance of the proposed adaptive system is compared with the fixed system.

In Chapter 5, conclusions and recommendations are summarised and some future directions related to the work presented in this thesis are discussed.

Chapter 2

Background

2.1 Introduction

The recent developments in satellites technology, transmission techniques, antennas, and launch capabilities have enabled a new generation of services to be made available to users [30]. Communications satellites have been widely considered to complement the terrestrial systems. [31, 32]. There are four orbit types for satellite systems: geostationary orbit, highly elliptical orbit, low Earth orbit (LEO), and medium Earth orbit (MEO) [33, 34, 35]. Traditionally, geostationary satellites were used as the sole basis for the provision of communication satellite services. Recently, LEO and MEO orbits are being used to provide voice and low data rate services in hand-held phones. This system is implemented by combining QPSK modulation with any of the multiple access techniques such as frequency-division multiple access (FDMA), time-division multiple access (TDMA), and code-division multiple access (CDMA)[36, 37, 38, 39].

In this thesis, a direct sequence CDMA access technique is considered and implemented for satellite systems. CDMA technology is a method of providing multiple access in a wireless system where each user is assigned a unique code that identifies the user to the system [40]. CDMA has several advantages over other more traditional multiple access methods. One of the most important is its inherent noise-rejection capability. It is a good option to be considered in terms of spectral efficiency. Also, satellite systems have utilised lower constellations such as BPSK, QPSK, offset QPSK, pi/4 shift QPSK, and 8PSK for modulation. This is because satellite transmission requires low signal-to-noise ratio (SNR) due to the noise-limited channel. Higher constellations require high signal-to-noise ratio (SNR) and they may not be desirable for satellite transmission [33, 34]. In this work, both BPSK and QPSK techniques are considered for adaptive system. The combination of this with CDMA allows the satellite transmission system to overcome fading and interference.

Satellite personal communication networks (S-PCN) using the above technology were developed to provide voice and low data rate services. These services are similar to those available via terrestrial cellular networks, using hand-held phones via satellite in either the LEO or MEO orbits. The allocated frequency bands for S-PCN ranges between 140 and 400 MHz for data systems, and 1.610-1.665GHz (uplink) and between 2.4835 and 2.500GHz (downlink) for voice systems [39]. The major operators of S-PCN are Globastar, Iridium, and Intermediate Circular Orbit (ICO) [41]. Different environments require different values of *Rice-factor*, which is the power ratio of the direct wave to the diffuse component. Typical values of measured Rice-factors are within the range of 5-18dB for the satellite elevation angle of 10-50 degrees with a fade rate of less than 200Hz for a mobile device travelling at less than 100km/h. The Rice-factor is in the range of 10-18dB for land mobile channels in rural or suburban areas and 5-10dB for mobile channels in urban areas [37, 42].

The common frequency bands that are used in satellite communications are the L, S, C, and Ka band, which are in the ranges of 390-1550Hz, 1550-3900MHz, 3900MHz and 17250-36000MHz, respectively [43]. S-PCN operate in the L- and S-bands between 1 and 3 GHz. However, the S-PCN has a limited bandwidth and this makes it less suitable for high-speed applications [43]. Ka bands are widely considered these days for satellite transmission because of their wider bandwidth, which allows the use of high data rate for broadband multimedia applications. Also, Ka band is not overcrowded [44, 45]. Furthermore, a number of different satellite roles have been defined in the framework of the third generation of mobile systems, which is referred to as Satellite Universal Mobile Telecommunications System (S-

UMTS) by the European Telecommunication Standards Institute (ETSI) and International Telecommunication Union (ITU). Part of the S-UMTS objectives are to enable global roaming of UMTS users and to provide QoS commensurate with that of terrestrial QoS at an affordable cost [9, 33, 46].

The remainder of this Chapter is as follows. Section 2.2 describes the mobile satellite requirements, which are further discussed in Subsections 2.2.1, 2.2.2 and 2.2.3. The basic architecture of the mobile satellite networks is discussed in Section 2.3 and the two major satellite networks are described in Subsections 2.3.1 and 2.3.2. In Section 2.4, satellite radio propagation is discussed. Subsections 2.4.1, 2.4.2, 2.4.3, and 2.4.4 describe the channel characteristics, examine the land mobile signal components, investigate the channel probability density functions and present joint probability distribution modelling respectively. The channel coding using turbo codes are considered in Section 2.5. The Chapter summary is drawn in Section 2.6.

2.2 Mobile Satellite Requirements

General requirements for third-generation services are defined in terms of bit rate, bit error rate, and delay. In order to offer multimedia applications, the S-UMTS system should at least be able to support user bit rates of up to 144kb/s in a rural outdoor environment at a maximum speed of 500km/h, 384kb/s with limited mobility in a macro and micro-cellular suburban outdoor environments at a maximum speed of 120km/h, and 2Mbit/s with low mobility in home and pico-cellular indoor and low-range outdoor environments at maximum speed of 10km/h [9]. The CDMA proposal has been submitted to ITU for UMTS/IMT-2000 satellite component. The proposal is aimed at providing services up to 144kb/s. In terms of spectral efficiency, the bit rate is expected to be maximised as much as possible using an adaptive system. In term of QoS and end-to-end delay requirements, a target BER of 10^{-3} and a maximum delay of 400ms is defined for voice services. A BER of 10^{-6} and different delay requirements have been considered for each class of data service (e.g., a few seconds for Internet access) [38, 41].

2.2.1 Digital satellite network Requirements

Satellite and terrestrial systems are considered as complimentary networks to achieve full coverage needed for communication systems. Mobile satellite services have been given attention for internet access and telephones services. The standard for satellite system need to be adopted for better performance and utilisation. This will include frame length, allocation frequency, transmission scheme, throughput and QoS requirements [47].

2.2.2 Broadband Satellite Requirements

The next wireless generation will achieve the true broadband with broadband satellite service to provides affordable multimedia services at a lower cost. This could be achieved through the throughput maximisation and efficient AMC system. Better performance could also be achieved by considering the higher frequency bands such as Ka-band [47, 48, 49].

2.2.3 Physical Layer Requirements

The QoS at the physical level is expressed in terms of the BER. The physical link design is expected to put into consideration problems such as the signal fade. AMC have been considered at the physical layer to counteract these effects. The implementation of AMC schemes in satellite network can result in significant saving in the transmission power and bandwidth [47, 50, 51].

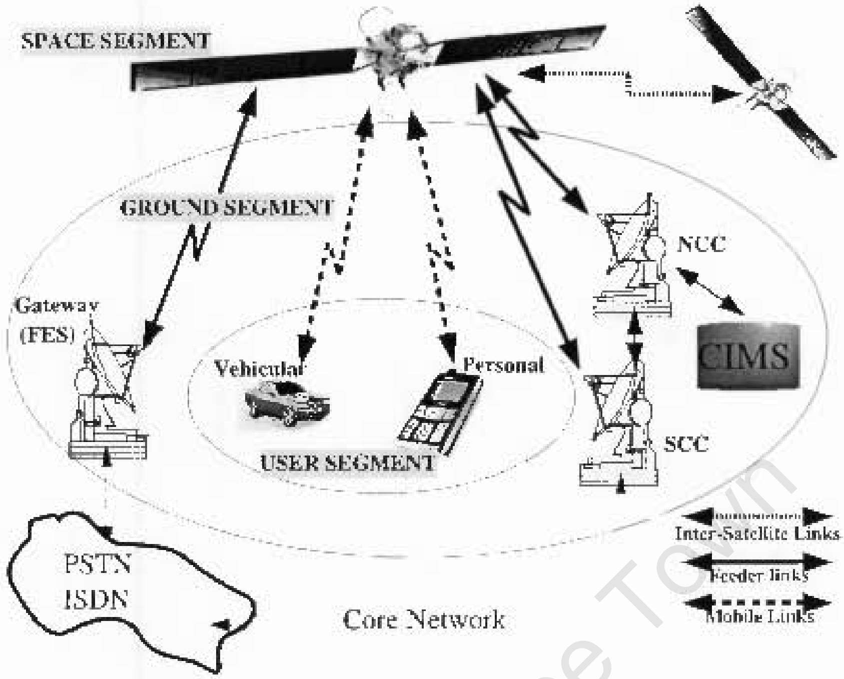

Figure 2.1: Basic Satellite Network Architecture

2.3 Mobile Satellite Networks

The basic architecture of a mobile-satellite access network is shown in Figure 2.1. The network architecture is divided into three segments: user segment, ground segment, and space segment. The roles of each segment are discussed in the following.

The User Segment : The user segment is of two types:

- *Mobile terminals* - Mobile terminals are those that support full mobility during operation. They can be further divided into two categories: mobile personal terminals and mobile group terminals. Mobile personal terminals often refer to hand-held and palm-top devices. Other mobile personal terminal categories include those situated on board a mobile platform such as a car. Mobile group terminals are designed for group usage and for installation on board for a collective transport system such as a ship, cruise liner, train, bus, or aircraft.

- *Portable terminals* - Portable terminals are the type with dimension similar to a brief-case or lap-top computer. As the name implies, these terminals can be transported from one site to another, but operation while mobile is normally not supported.

The Ground Segment : The ground segment consists of three main network elements: gateways, sometimes called fixed Earth stations (FES), the network control centre (NCC), and the satellite control centre (SCC). Gateways provide fixed entry points to the satellite access network by furnishing a connection to the existing core networks (CN), such as the public switched telephone network (PSTN) and public land mobile network (PLMN), through local exchanges. A number of gateways can be located within a spot-beam, or a gateway could provide access to more than one spot-beam, depending on the satellite coverage [52].

The Space Segment : The space segment provides the connection between the users of the network and the gateways. A direct connection between users via the space segment is also achievable using the latest generation of satellites. The space segment consists of one or more constellations of satellites, each with an associated set of orbital and individual satellite parameters. Satellite constellations are usually formed by a particular orbital type; hybrid satellite constellations also exist in the space segment. One such example is the ELLIPSO network [53].

2.3.1 Satellite Personal Communication Network

Geostationary satellites were conventionally used as the only technology for the provision of satellite mobile in one form or another [36]. Later, other satellite systems such as low Earth orbit (LEO) and Medium Earth orbit (MEO) were launched to enable satellite personal communication networks (S-PCN). LEO satellites are placed between 750 and 2000 km above

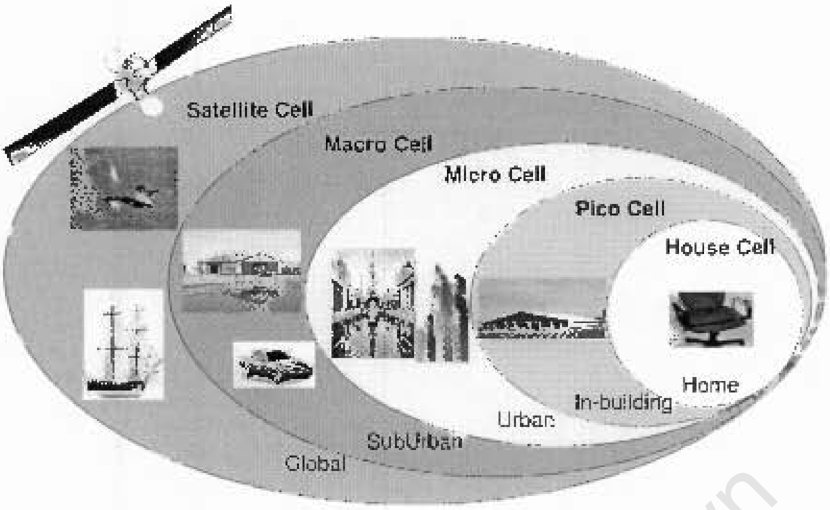

Figure 2.2: Cell Types

the Earth and MEO satellites are placed between 10 000 and 20 000 km above the Earth. GLOBALSTAR is a system that uses the LEO, while NEW ICO exploits the MEO system. Satellite Personal Communication Network (S-PCN) aimed primarily at the provision of voice and low data rate services, similar to those available via terrestrial cellular networks. S-PCN networks operate in the L- and S-bands [33, 34, 41].

2.3.2 Satellite-UMTS Networks

As mentioned before, S-PCN has a limited bandwidth and this makes it less suitable for high-speed applications. Satellite-UMTS networks provide higher bandwidth and are being considered in the third generation of mobile systems by Universal Mobile Telecommunications System (UMTS) to enable global roaming of UMTS users and to provide QoS. This will allow a user to operate in a variety of transmission environments, which include office, home, urban-vehicular and -pedestrian, rural-vehicular and -pedestrian, satellite-fixed, satellite-rural, aeronautical, and maritime environments. Five basic cells (satellite, macro, micro, pico, and home cells) that form an hierarchical cell-structure, as illustrated in Figure 2.2, are identified to cover a user environment [54].

2.4 Satellite Radio Propagation

The performance of a satellite system depends on the propagation mode of the radio waves [55]. The nature of the propagation channel is the biggest obstacle facing the design of communications systems. This makes satellite channels characterised by fading and interference [56]. In a mobile satellite network, two types of channels exist: the *mobile channel*, between the mobile terminal and the satellite; and the *fixed channel*, between the fixed Earth station or gateway and the satellite. These two channels have very different characteristics, which must be taken into account during the system design phase. The more critical of the two links is the mobile channel, since transmitter power, receiver gain and satellite visibility are restricted more in comparison to the fixed link [57, 58]. Thus, this thesis focuses on the mobile channel.

The mobile terminal, unlike a fixed system, operates in a dynamic environment where propagation conditions are constantly changing. As a result, the local operational environment has a significant impact on the achievable quality of service (QoS) and throughput of the system. The different categories of mobile terminals used in different environments such as land, aeronautical, or maritime, also have different impacts on channel characteristics, and these should be considered in channel modelling [59, 60].

2.4.1 Mobile Channel Characteristics

The first step toward modeling the mobile satellite channel is to identify and categorise typical transmission environments [61]. This is usually achieved by dividing the environment into three broad categories as shown in Figure 2.3.

- **Urban areas**, environment characterised by almost complete obstruction of the direct wave.
- **Rural areas**, open place with no obstruction of the direct wave.

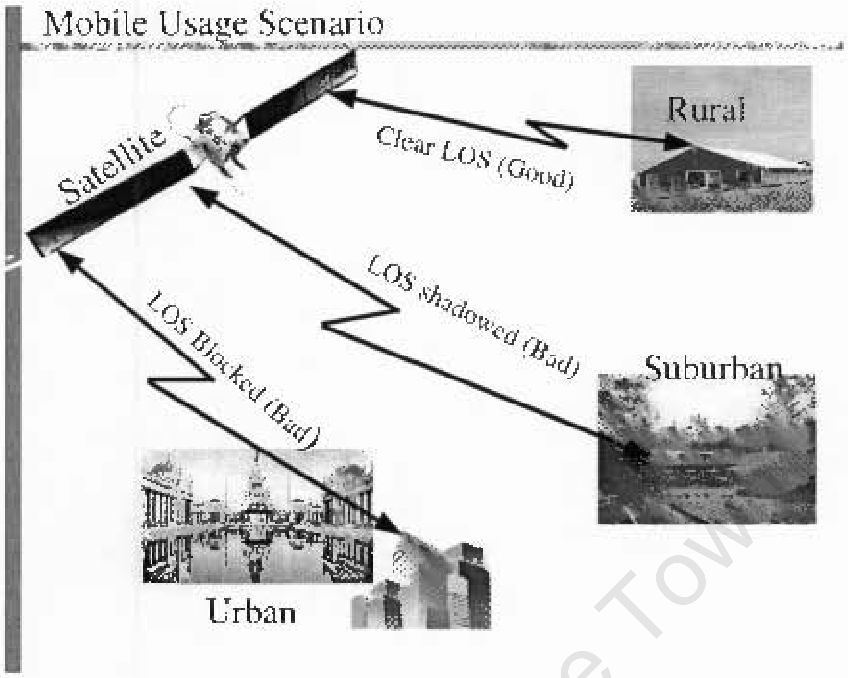

Figure 2.3: Land Mobile Satellite Environment

- **Suburban areas**, a tree shadowed environment, where intermittent partial obstruction of the direct wave occurs.

In urban areas, visibility to the satellite is difficult to guarantee, and this results in the multipath component that dominates reception of the signal. Thus, a receiver receives a random amplitude and phase signal. In order to reduce this effect, satellite constellations with a high guaranteed minimum elevation angle are required. Also, satellite diversity is needed to allow optimum reception of one or more satellite signals so as to counteract the effect of shadowing. These factors cause the urban propagation environments to place severe constraints on the mobile satellite network. For example, in order to achieve a fade margin in the region of 6-10 dB in urban environments, a continuous guaranteed minimum user-to-satellite elevation angle of at least 50 degree and a constellation of 100 satellites is required [62, 63, 64]. Because of these constraints satellite networks have been integrated with a terrestrial system in urban environments. GLOBALSTAR is a good example of an operator using this scenario.

In rural areas, where direct line-of-sight (LOS) to the satellite can be achieved with a fairly high degree of certainty, the multipath effect could cause link impairment. This effect can be either constructive or destructive to signal components. A constructive multipath component enhances the transmitting signal, while the destructive multipath component causes fade to the direct wave component and result in signal power fluctuations [56, 65]. In suburban areas, the major signal degradation is due to buildings and other man made obstacles. These obstacles cause shadowing to the direct LOS signal and result in attenuation of the received signal. The presence of trees is also another obstacle. The depth of the fade is dependent on a number of parameters: tree type, height, and the leaf density of the trees [66]. Also, the motion of the mobile through suburban areas results in the continuous variation in the received signal strength and phase. The effect of moving up in frequency to the Ka-bands imposes further constraints on the design of the link [57].

2.4.2 Land Mobile Signal Components

The received mobile satellite signal consists of a combination of three components: the direct line-of-sight (LOS) wave, the diffuse wave, and the specular ground reflection, as shown in Figure 2.4. The direct LOS wave arrives at the receiver without reflection from the surrounding environment. However, direct components are affected by free space loss (FSL). FSL is related to the operating frequency and transmission distance. Another loss experienced by the direct component is due to the ionospheric and tropospheric effects. Tropospheric effects can be considered negligible at frequencies below 10 GHz, but systems operating at above 10 GHz need to take this effect into account. The impairments introduced by the ionosphere is always counteracted by the selective use of transmission polarisation [67, 68, 69].

The diffuse component comprises of multipath reflected signals from the surrounding environment, such as buildings, trees, and telephone poles. Multipath has an effect on mobile satellite links in most practical operating environments [61]. The specular ground component results from the reception of the reflected signal coming from the ground near to the

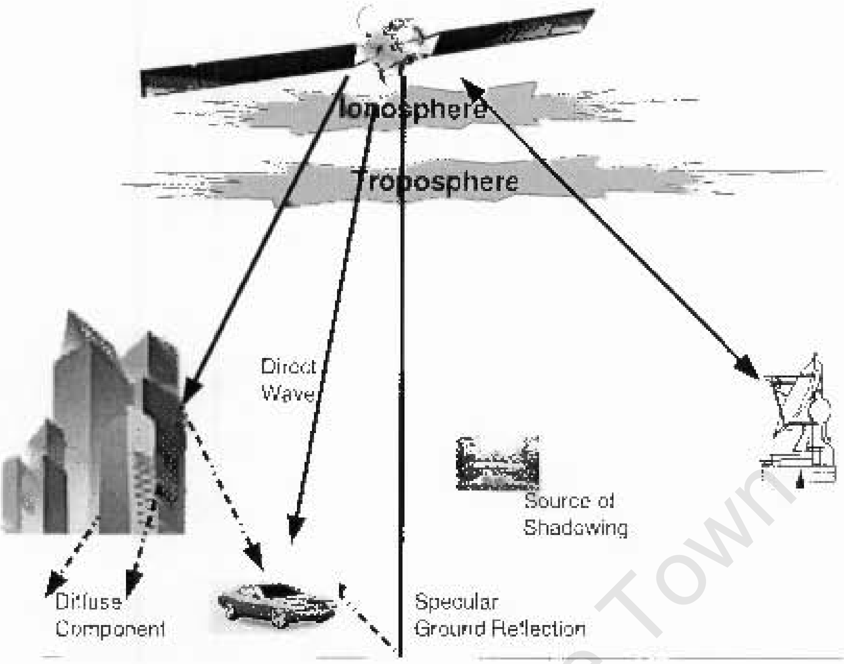

Figure 2.4: Satellite Channel Scenario

mobile. Antennas of low gain, wide beam width operating via satellites with low elevation angle are particularly susceptible to this form of impairment. Such a scenario could include hand-held cellular like terminals operating via a non-geostationary satellite.

2.4.3 Channel Modeling

The statistical model is an accurate model to characterise the multipath and shadowing phenomena of fading channels. This is because it- allows the dynamic nature of the channel to be modelled and enables the performance of the system to be evaluated for different environments. The combination of three probability density functions (PDF) is used to characterise the channel. Rician pdf is considered when a direct wave dominates over the multipath reception, Rayleigh pdf is used when multipath reception dominates over the direct wave, and lognormal pdf is always used to represent shadowing effects associated with the received wave.

Rayleigh PDF

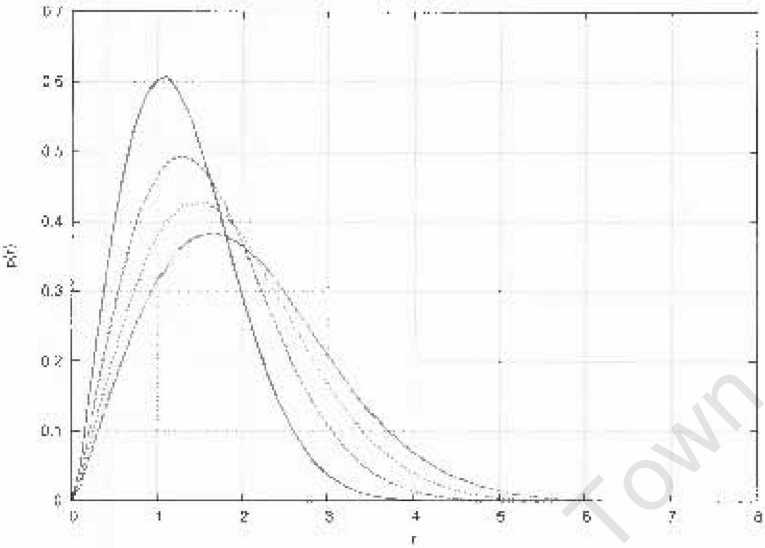

Figure 2.5: Rayleigh Channel Model

The Rayleigh distribution shown in Figure 2.5 is used to represent the propagation environment where the mobile antenna receives a large number of reflected and scattered waves. The Rayleigh distribution assumes that all the components that make up the resultant received signal are reflected or scattered and there is no direct path (i.e. LOS) from the transmitter to the receiver. This PDF is suitable for urban areas where there is complete obstruction of the direct wave.

The pdf of the Rayleigh model is given as [70]

$$p(r) = \frac{r}{\sigma^2} exp\left(\frac{-r^2}{2\sigma^2}\right), \tag{2.1}$$

where r is the received signal envelope and σ^2 is the mean of the received scattered power because of the multipath effects.

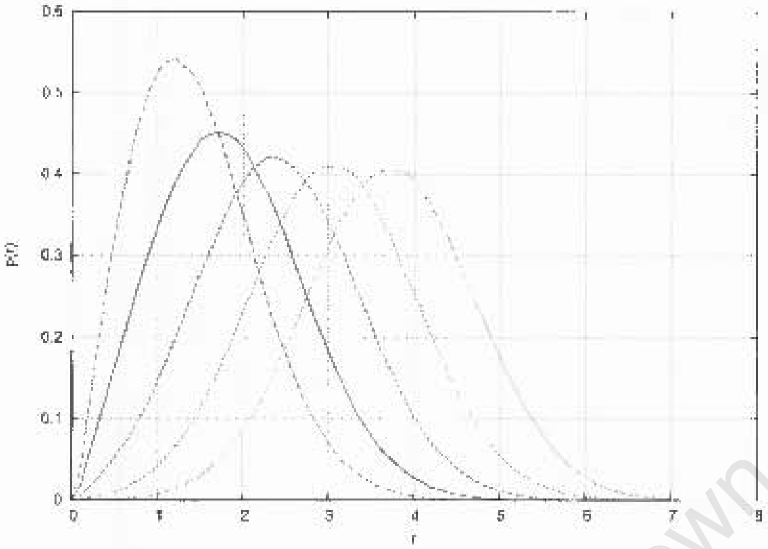

Figure 2.6: Rician Channel Model

Rician PDF

The Rician distribution shown in Figure 2.6, is used to represent the propagation environment in which LOS propagation is dominant. In such case the resultant signal amplitude follows the Rician distribution.

The model, in terms of probability distribution of the received signal envelope, r, is given as [71]

$$p(r) = \frac{r}{\sigma^2} exp\left(-\frac{r^2 + A^2}{2\sigma^2}\right) I_o\left(\frac{rA}{\sigma^2}\right), \tag{2.2}$$

where A represents a signal with clear LOS and I_o is the modified Bessel function of order zero. The phase distribution is no longer uniform like a Rayleigh distribution. When A = 0, it becomes the PDF of a Rayleigh fading process. It could be observed that Rayleigh distribution is a special case of the Rician distribution and arises when there is no LOS component available.

Lognormal PDF

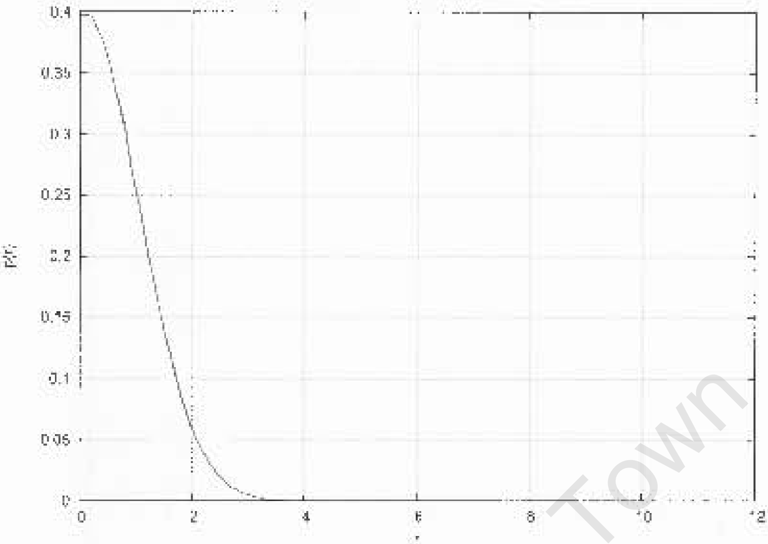

Figure 2.7: Lognormal Channel Model

Lognormal distribution shown in Figure 2.7, is used to represent the propagation environment in which the direct wave is shadowed. The model, in terms of probability distribution of the received signal envelope, r, is given as

$$p(r) = \frac{1}{\sigma_s r \sqrt{2\pi}} exp\left(\frac{-\ln r - {\mu_s}^2}{2\sigma_s^2}\right), \tag{2.3}$$

where σ_s is the standard deviation of the shadowed component ($\ln r$) and μ_s is the mean of the shadowed component ($\ln r$).

This distribution combined with any of those mentioned is suitable for a suburban area in which random shadowing of the direct wave occurs due to the presence of trees, buildings, and so on. The lognormal distribution is used to represent the effect of this environment on the direct wave component.

2.4.4 Joint Probability Distribution Modelling

The statistical distributions described in the last Section are combined to characterise a complete transmission environment. Two of the most widely referenced statistical models are those developed by Loo [71] and Lutz [72]. The Loo model is an example of how the constituents of the channel are combined into a single probability distribution with associated parameters. The Lutz approach, on the other hand, employs state orientated statistical modelling, whereby each particular state of the channel is separately characterised by a probability distribution, with a specified probability of occurrence. A third model, which is an alternative to Loo's model is the Corazza and Vatalaro model. These models are explained in more detail.

Loo's Channel Model: Loo's Model [71] is based upon measurements for mobile transmission link in rural environments. The model assumed:

- The received voltage due to diffusely scattered components is Rayleigh distributed;
- The voltage variations due to attenuation of the direct path signal are log-normally distributed.

Loo's PDF for a signal envelope r is given by [71]

$$p_{Loo}(r) = \frac{r}{\sigma^2\sqrt{2\pi\sigma_s^2}} \int_0^\infty \frac{1}{Q} exp\left(-\frac{(\ln Q - \mu_s)^2}{2\sigma_s^2} - \frac{r^2+Q^2}{2\sigma^2}\right) I_o\left(\frac{rQ}{\sigma^2}\right) dQ, \qquad (2.4)$$

where σ^2 is the mean of the received scattered power due to multipath propagation, σ_S^2 is the standard deviation of the shadowed component $(\ln A)$ and μ_s is the mean of the shadowed component $(\ln A)$.

Lutz's Channel Model: Lutz's Model [72] employs a two-state Markov model and assumes that the propagation link has two distinct states: shadowed and un-shadowed. The un-shadowed is called a good fading state, or clear LOS, where the received signal is comprised of the direct component and multipath reflections, and is assumed to be Rician distributed.

The shadowed is called a bad fading state (LOS shadowed or blocked, or both), where the received signal is characterised by a Rayleigh distribution, with a short time varying mean received power, for which a lognormal distribution is assumed. The resultant probability density function of this model is given by

$$P_{Lutz}(s) = (1 - \Omega)P_{Rice}(s) + \Omega \int_0^\infty P_{Rayleigh}(s|s_0)P_{Lognormal}(s_0)ds_0, \tag{2.5}$$

where Ω is the proportion of the time spent in each fading state.

Corazza and Vatalaro Channel Model: The Corazza and Vatalaro model [65] is an alternative model to Loo's approach for non-geostationary satellite constellations. In this model the direct and scattered components are both considered to be affected by shadowing. This is termed the Rice-lognormal model (RLM). This is the basis for our channel model. The Corazza and Vatalaro model is preferred to other existing channel models because it is well suited for the application area that we consider here, and the model provides a deep insight into the nature of the satellite channel characteristics, which is useful when designing satellite radio systems [65]. Corazza and Vatalaro model is explained in details in Chapter 3.

2.5 Channel Coding

Satellite communication systems are generally limited by available power and bandwidth. It is therefore of interest if the signal power can be reduced while maintaining the same grade of service. This can be achieved by adding redundant bits to the information content, using a channel encoder. The two main classes of channel encoder that are most widely used for satellite communications are: block encoders and convolutional encoders. At the receiver, the additional bits are used to detect any errors introduced by the channel. One of the techniques employed in satellite communications to achieve this is forward error correction

(FEC), where errors are detected and corrected at the receiver. Turbo codes are one of the powerful types of error control codes currently available. They are used as the building blocks for bandwidth efficient code schemes and have recently been included in the 3G standard.

2.5.1 Overview of Turbo Codes

Turbo codes were first presented by Berrou, Glavieux, and Thitimajshima in 1993 [73]. Turbo codes comprises of encoder and decoder. The encoder consists of two recursive systematic convolutional codes in parallel separated by an interleaver. The decoder consists of two Maximum A Posterior (MAP) decoders connected via interleavers and pass soft decisions from the outer of one decoder to the input of the other. This process is iterated several times to produce better decisions. The information, which is available to one of the component decoders before it starts its own decoding process, is called intrinsic information. Extrinsic information is derived by each decoder itself.

2.5.2 Convolutional Codes

Convolutional codes generates a digit sequence from the information digits in which no finite group of information digits can be ascribed to one information codeword. A convolutional code adds redundancy to a continuous stream of inputs data by using shift register. For instance, a convolutional encoder rate $r = k/n$ will have each set of n output symbol as a linear combination of the current set of k input bits and m bits stored in the shift register. The total number of bits that each output depends on the constraint length, and is denoted by k [74].

2.5.3 RSC Component Codes

The recursive systematic convolutional (RSC) encoder is derived from the nonrecursive non-systematic convolutional encoder by feeding back of its encoded output. A simple RSC

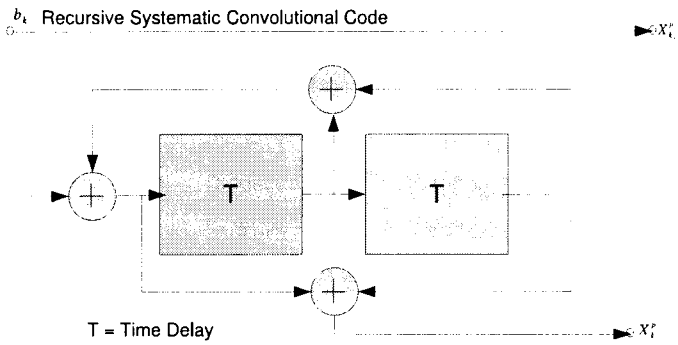

Figure 2.8: Recursive Systematic Convolutional Code

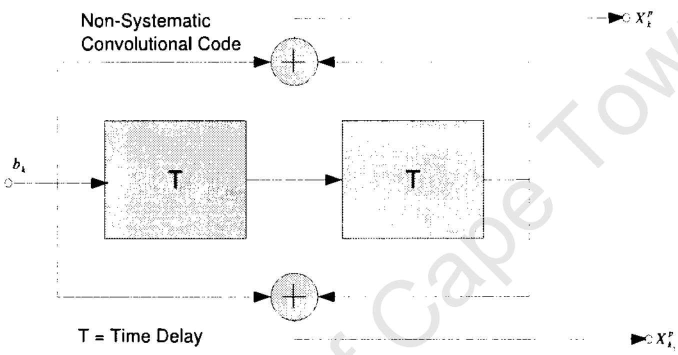

Figure 2.9: Non-Systematic Convolutional Code

encoder is shown in Figure 2.8 along with a non-systematic (NSC) encoder in Figure 2.9, for comparison. A convolutional encoder, which has its input block unchanged is called systematic while convolutional encoder, which is implemented by including feedback is caled a recursive convolutional encoder [75].

2.5.4 Encoder for Turbo Codes

Turbo encoder is a parallel concatenated convolutional code. Figure 2.10 shows a block diagram of the turbo encoder consists of two convolutional encoders, separated by pseudo-random interleaver. The encoder structure is a rate 1/3, mapping N data bits to 3N code bits and is formed by parallel concatenation of two identical rate $\frac{1}{2}$ and generator polynomial

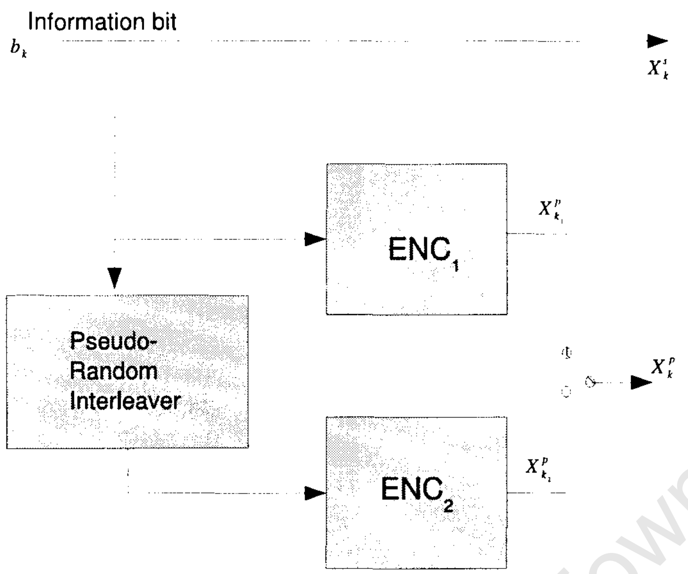

Figure 2.10: The Encoder for Turbo Codes

$G = \{g_1, g_2\} = \{7, 5\}$ where g_1 is the feedback connectivity and g_2 is the output connectivity, in octal notation [75].

2.5.5 Interleaving

The interleaver is used in turbo codes to take the incoming block of bits and rearrange them in a pseudo-random fashion prior to encoding by the second encoder. This is necessary to spread bursts of errors evenly and over as large a distance as possible. An interleaver is a random mapping between input and output positions, generated by means of some form of random. The Figure 2.11 shows the original data sequence represented by the sequence of white squares, and the interleaved data sequence represented by the black squares. Turbo code BER performance improves with interleaver length [75].

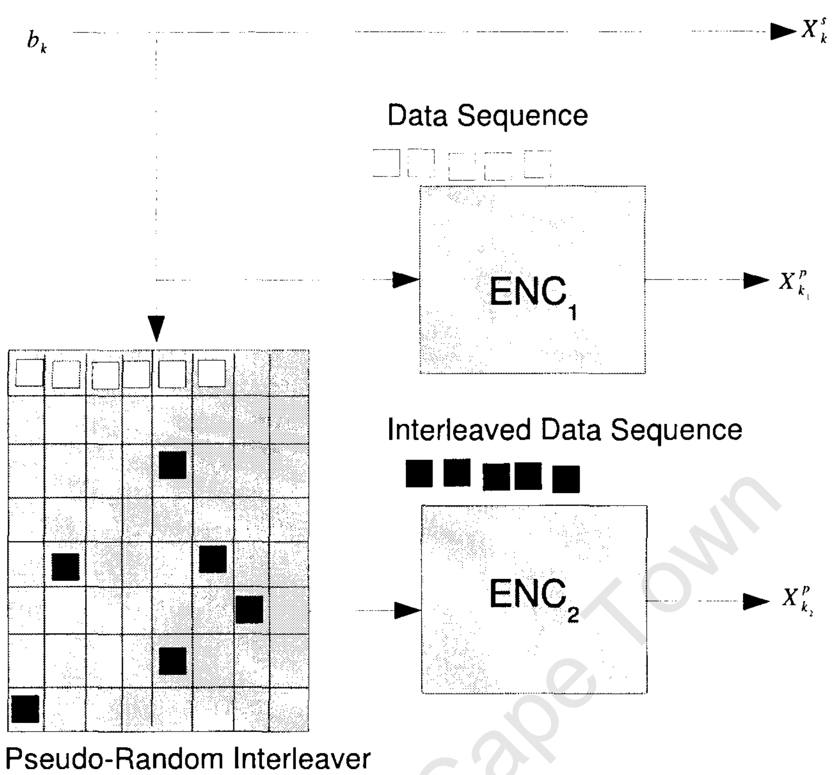

Figure 2.11: Interleaving

2.5.6 Turbo Decoding Algorithm

The decoding algorithm, which has been derived and appear in many literatures [79, 80, 75] is reviewed. The MAP algorithm is considered in this thesis and is described in Appendix.

2.6 Chapter Summary

In this Chapter, the background of the thesis has been presented. First, an overview of satellite systems was briefly discussed. After, mobile satellite requirements, satellite networks, satellite radio propagation, and channel coding were reviewed. The major types of mobile satellite networks in existence such as S-PCN and S-UMTS, were discussed. According to the review, for high-speed transmission requirements and roaming capability, S-UMTS networks have been considered because they are suitable for broadband satellite communications.

Radio propagation, which is one of the important components in a satellite network, has been studied. Two types of satellite propagation were identified and discussed, namely mobile and fixed channel. The mobile channel, unlike a fixed channel, operates in a dynamic environment where propagation conditions are constantly changing. In a mobile channel environment, three types of mobile usage scenarios were discussed. In urban areas, the user environment is characterised by almost complete obstruction of the direct wave. In rural areas, the user experiences a clear line of sight of the direct wave, while in suburban areas the user environment is characterised by shadowing from the tree and terrain features. Other factors effecting the satellite propagation are ionospheric and tropospheric conditions.

Three types of fading channels probability density function (PDF) corresponding to the identified mobile satellite environment were discussed. These distributions are Rician, Rayleigh, and lognormal PDF. The Rician PDF is used to model rural area where there is no obstruction of the direct wave. The Rayleigh PDF is used to model urban areas where the environment is characterised by almost complete obstruction of the direct wave. Lognormal PDF is used when the performance Rician and Rayleigh channel is affected by shadowing and this could lead to a joint distribution to form a typical channel model such as in Rice-lognormal and Rayleigh-lognormal. In essence, Loo's model is an instance of Rice-lognormal that model suburban areas defined by a tree shadowed environment, where intermittent partial obstruction of the direct wave occurs. The Rician-lognormal model, which is the alternative model to Loo's approach proposed by Corazza and Vatalaro was considered in our system model, and will be discussed in the next chapter. This model was considered because it provides a deep insight into the nature of the satellite channel characteristics, which is useful when designing satellite radio systems.

The performance of the satellite channels have been determined and compared. However, the performance obtained are not optimal, due to the noisy channel. An error control code (ECC) has become a vital part of modern digital wireless systems, enabling reliable transmission to be achieved over noisy and fading channels. Turbo codes, which have been widely considered was extensively discussed. Various constituents of turbo codes were described and examined for wireless channels.

Chapter 3

Satellite Channel and System Model

3.1 Introduction

In a wireless communications system, the channel characteristics are of fundamental importance because they limit the transmission quality and throughput. In traditional radio systems (i.e., non-adaptive radio systems), the long-term statistical properties of the channel are measured and evaluated before system design. In adaptive modulation systems, the situation is different. To guarantee that the adaptive system function works effectively, information about the short-term statistical or even instantaneous properties of the channel in different domains are continuously required. The main limiting factors of a mobile communications system originate from the radio medium. These were discussed in the previous chapter. In this Chapter, channel and system model are discussed.

The organization of this Chapter is as follows: Section 3.2 describes the system model, and is divided into five Subsections. The transmitter model is described in Section 3.2.1. In this Section the CDMA parameters are exploited. Section 3.2.2 describes the channel model and satellite path diversity system. The influence of factor K and PCE on the channel model are evaluated in Section 3.2.3. This is necessary to understand the importance of the main parameters on the system model, and how they benefit the fading channel with adaptive parameters. In Section 3.2.4, the receiver model is presented, while signal to interference plus noise ratio analysis for CDMA based system is discussed in Section 3.2.5. Some simulation results are presented in Section 3.3. These results involved the performance evaluation of turbo coded channel parameters, the performance evaluation of turbo coded CDMA parameters and the performance evaluation of turbo code parameters. Conclusion is drawn in Section 3.4.

3.2 The Satellite System Model

3.2.1 Transmitter model

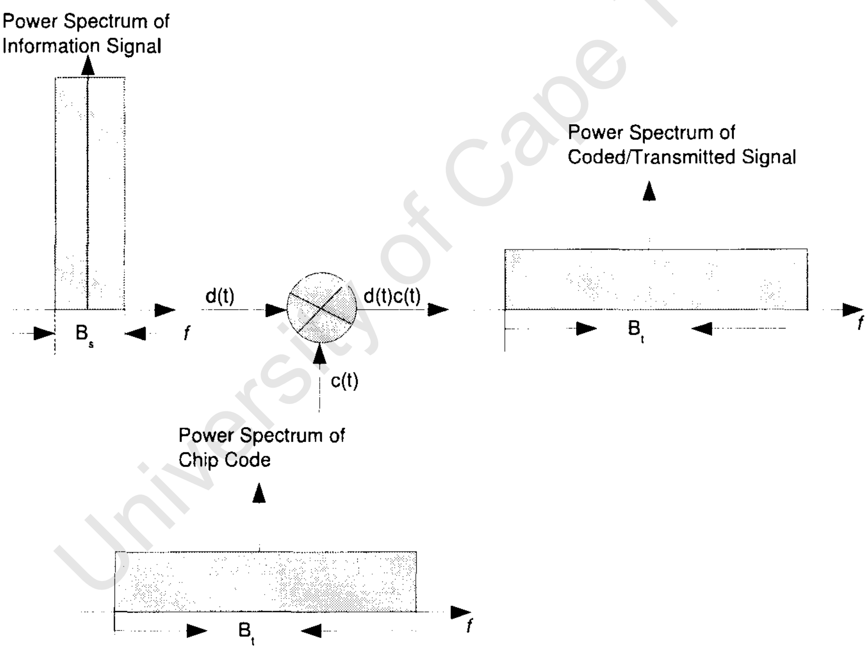

Figure 3.1: Block Diagram of Spread Spectrum

In a code division multiple access (CDMA) system, each user is assigned a unique code, as shown in Figure 3.1, in which only the intended receiver with the same code can recover the user-specific transmitted signal. The effect of this coding assignment is the expansion of the transmitted signal bandwidth. This method has the property that the unwanted signals appear like noise to the unintended receiver. All users exploit the whole frequency bandwidth all the time, and they are differentiated by their unique codes [83, 84]. In Figure 3.2, the

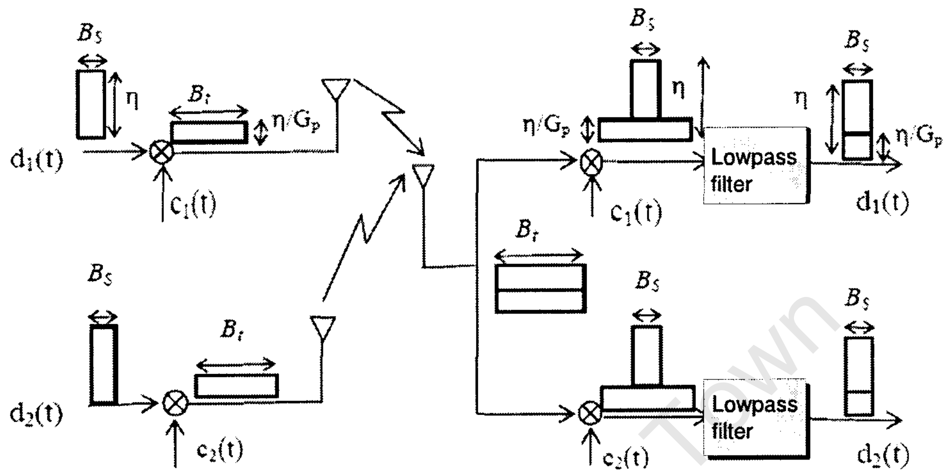

Figure 3.2: Block Diagram of Spread Spectrum

information signal $d(t)$ of bandwidth B_S multiplies with a chip code $c(t)$ of bandwidth B_t to become a coded signal with a bandwidth of B_t, as observed in the Figure. This form of CDMA is called direct sequence CDMA (DS-CDMA). For k-th user, $d_k(t)$, $c_k(t)$, $S_k(t)$ denote the information signal before spreading, the chip code, and the transmitted signal, respectively. The spreading process in time domain can be expressed as

$$S_k(t) = d_k(t)c_k(t) \tag{3.1}$$

Although Figure 3.1 describes a single-user spread spectrum system, it can be easily generalized to multiple users. Figure 3.2 shows a multi-user spread spectrum system. Once again, it is observed that signals from other users appear like noise after dispreading. In the Figure, η is the power spectrum density of the system; and G_p is the processing gain, which equals the total transmit bandwidth divided by the signal bandwidth. DS-CDMA is the spread spectrum technique, where each user is assigned a unique code sequence (i.e., a spreading code) which is being used to encode the user-specific information. One particular code of interest is the pseudo random noise code (PN code). A PN code is a sequence of chips valued -1 and 1 (polar) or 0 and 1 (non-polar), and has noise-like properties, namely low cross-correlation values among the codes. The receiver, which has the code sequences of the user, decodes the received signal after reception and recovers the original data. In direct sequence systems, the length of the code is the same as the spreading factor [85]. Hence, the length of the direct sequence code is assumed to be equal $\frac{T_s}{T_c}$. In time domain, the CDMA system is modeled as

$$c_k(t) = \sum_{n=0}^{N_c-1} c_{k,n} \Pi(t - nT_c) , \tag{3.2}$$

where $c_k(t)$ is the spreading waveform for the k-th user, $c_{k,n} \in \{+1, -1\}$ and the chip pulse waveform $\Pi(t)$ is a rectangular pulse of duration T_c. The spreading or code waveform is composed of N_c chips, and we assume BPSK for the spreading modulation [83]. If R_c stands for the rate of chip code, we have

$$B_t = R_c = \frac{1}{T_c} . \tag{3.3}$$

This equation relates the time domain parameter, i.e. the chip period, to the frequency domain parameter, i.e. the transmission bandwidth. In other words, the chip period limits the transmission bandwidth. With regard to the information signal, if R_s is the symbol rate and T_s the symbol period, we have a similar relationship, as

$$R_s = \frac{1}{T_s} . \tag{3.4}$$

This equation relates the time domain parameter of an information signal, i.e. its symbol period, to its frequency domain parameter, the symbol rate. It is straightforward to extend this relation to the bit rate R_b and bit period T_b for a certain modulation with modulation level M ($M = 2$ for BPSK, 4 for QPSK, 16 for 16QAM, for example). We then get the following equations:

$$B_s = R_s = \frac{1}{T_s} , \tag{3.5}$$

$$T_s = T_b \log_2 M , \tag{3.6}$$

$$R_b = R_s \log_2 M , \tag{3.7}$$

where B_s stands for the bandwidth of the information signal. It is easy to see that in one symbol the number of bits is $\log_2 M$ for M level modulation.

3.2.2 Channel Model and Satellite Path Diversity

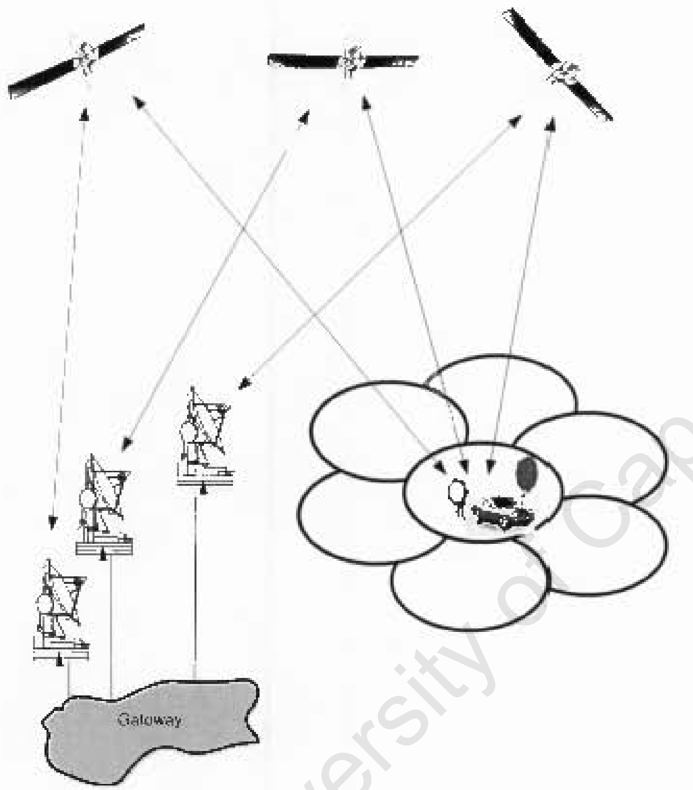

Figure 3.3: Satellite Path Diversity

The channel model under consideration is the Corazza and Vatataro model [65], which has been mentioned in Section ?? of Chapter 2. Here, we consider low Earth orbiting (LEO) satellite communication network systems and CDMA technology. The DS-CDMA based mobile satellite system for binary phase-shift keying (BPSK) and quadrature-shift keying (QPSK) are assumed to occupy the Ka-band of land mobile satellite systems. As shown in Figure 3.3, in addition, we considered the system to exhibit L-order path diversity, where each satellite contained S spot beams and is transmitting via a transponders from gateways with directional antennae. Each antenna is directed to the satellite in view and satellite tracking is performed. It is assumed that each spot beam has U simultaneous active users.

The per user fading signal envelope is described by the path gain $\alpha_l; l = 1, 2, \ldots, L$, where α_l is a Rice-lognormal random process and is the product of two independent processes. It can be written as $\alpha_l = S_l R_l$, where S_l is a lognormal random variable used to model the long-term shadowing effects, while R_l is a Rice random variable to model the short-term diffuse multipath fading over the direct signal component. The probability density function of the instantaneous received signal power is given by [86]

$$\zeta_{\alpha_l^2}(\alpha) = \int_0^\infty \zeta_{\alpha_l^2|\beta_l}(\alpha|\beta_l)\zeta(\beta_l)d\beta_l \,, \tag{3.8}$$

where $\zeta_{\alpha_l^2|\beta_l}(\alpha|\beta_l)$ is the probability density function of the instantaneous received signal power condition on the mean square value of the signal. This is modelled as the non-central chi-square distribution and is given by

$$\zeta_{\alpha_l^2|\beta_l}(\alpha|\beta_l) = \frac{K_l+1}{\beta_l} exp\left[-\frac{(K_l+1)\alpha}{\beta_l} - K_l\right] I_o\left(2\sqrt{\frac{K_l(K_l+1)\alpha}{\beta_l}}\right) . \tag{3.9}$$

The Rice factor $K = K_l$ is the ratio of the direct signal power to the diffuse multipath power. The mean power and variance are resulted from $\beta_l = E[R_l^2]S_l^2 = (K_l+1)2\sigma^2 S_l^2$, and can be determined in the evaluation of $\frac{E[R_l^2]S_l^2}{2(K_l+1)}$. In the presence of shadowing, S_l^2 is a random variable with log-normal probability density function of

$$\zeta_{S_l}(s) = \frac{1}{\sqrt{2\pi}h\sigma_{slS}} exp\left[-\frac{(\ln S - \mu_{sl})^2}{\sqrt{2}h\sigma_s l}\right] , \tag{3.10}$$

where $h = (\ln 10)/20$, μ_{sl} and σ_{slS} are the mean and standard deviation (in dB) of the associated normal variate, respectively.

3.2.3 Influence of Rice K Factor and Power Control Error (PCE) on the Channel Model

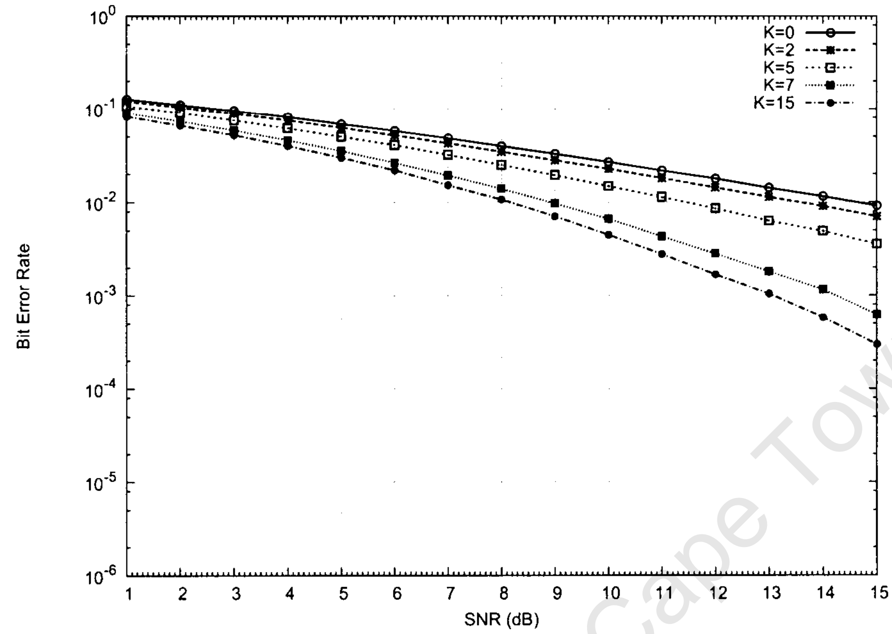

Figure 3.4: Uncoded BER of BPSK versus Eb/No(dB) on Rice-lognormal Channel: Rice factor, K = [0, 2, 5, 10, 15], PCE = constant and mean = 0

The channel model is validated by varying the two channel parameters, namely K and PCE, and the outputs are compared using simulation results. Also, in order to optimize modulation, coding and CDMA parameters under a certain channel behaviour, we needed to know how these parameters influence the channel condition and performance. Regarding the PCE, we adopted a closed-loop power control scheme as proposed in [87, 88], where it is assumed that the mean and standard deviation of the power envelope $p = s$ of the received signal are μ_{pl} and σ_{pl}, respectively. In the power control algorithm, the received power is varied around the target power and, hence, generally μ_{pl} = 0dB and the PCE is measured

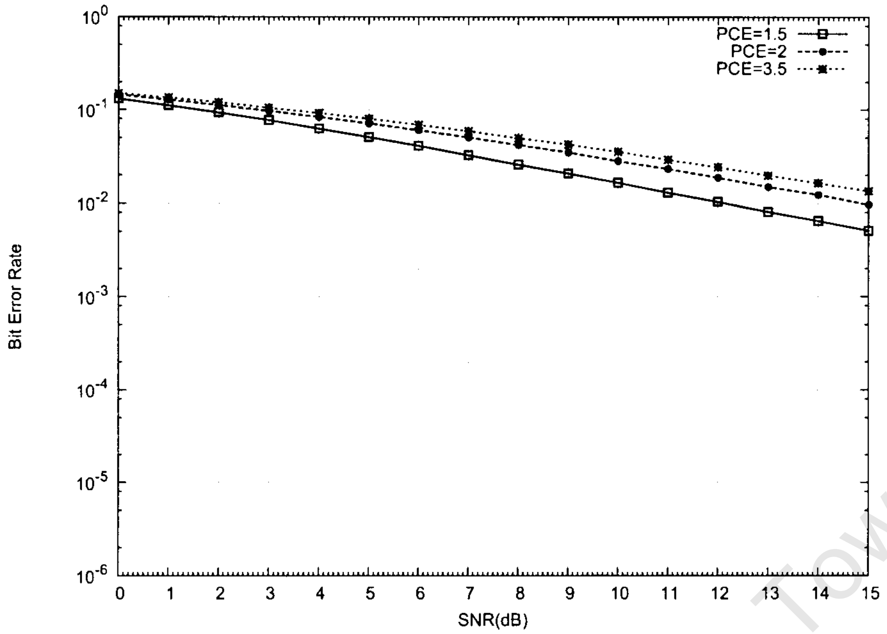

Figure 3.5: Uncoded BER of BPSK versus Eb/No(dB) on Rice-Log-normal Channel: PCE = [1.5, 2, 3.5], Rice factor, K = constant and mean = 0

by σ_{pl}. The typical values of PCE ranges between $6 - 9$dB, but for a perfect control system, the values are shown to range between $2.8 - 4.5$dB. In our adaptive modulation and coding algorithm, which is presented in Chapter 4, 4dB of PCE is assumed. The Rice K factor is the power ratio of the direct wave to the diffuse component. Typical values of the Rice-factor are between 10 and 20dB [88].

From simulation, when we decrease the Rice factor K and holding PCE constant, performance degradation is observed. Likewise, when we increase PCE while holding K constant, the performance degrades. These are shown in Figures 3.4 and 3.5, respectively.

3.2.4 Receiver Model

We consider M simultaneously active users of a particular spot beam transmitting via satellite transponders and a L-order path diversity. The received signal is given as

$$r(t) = \sqrt{2P_t} \sum_{k=1}^{U} \sum_{l=1}^{L} \alpha_l^k d^k(t - \tau_l^k) c^k(t - \tau_l^k) \cos(\omega_c t + \phi_l^k) + n(t)\,, \tag{3.11}$$

where P_t is the transmitted power, $c^k(t)$ is the spreading sequence of the k-th user, $d^k(t)$ is the message generated at rate $1/T$, τ_l^k, and ϕ_l^k represent independent time delays and carrier phases, respectively. The fading envelope $\alpha_l(t)$ describes the Rice-lognormal statistics, generated at rate $1/T_c$. $n(t)$ is the AWGN with two-sided power spectral density of $N_o/2$. Here, our processing gain is $G_p = \frac{T}{T_c} = 128$.

In a DS-CDMA system with RAKE receiver techniques, the output of the n-th transmitted bit of user 1, assuming perfect synchronization is given by

$$Z_l^1(n) = \int_{nT+\tau_l^1}^{(n+1)T+\tau_l^1} r(t) \alpha_l^1 c^1(t - \tau_l^1) \cos(\omega_c t + \phi_l^1) dt\,. \tag{3.12}$$

Assuming perfect channel gain estimates, the above equation becomes

$$Z_l^1(n) = \sqrt{P_t} T b_n^1 \left\{\alpha_l^1\right\}^2 + \sqrt{2P_t} \alpha_l^1 \sum_{k=2}^{M} \sum_{i=1}^{L} \alpha_i^k I_{il}^{k1}(T) cos\phi_{il}^{k1} + \alpha_l^1 N(T), \tag{3.13}$$

where $I_{il}^{k1}(T)$ is the cross correlation of i-th path of the k-th user with the local spreading code at the l-th branch of the correlation receiver and $N(T)$ is zero mean Gaussian process with variance $n_o T\{\alpha_l^1\}^2$.

3.2.5 Signal to Interference Plus Noise Ratio (SINR) Analysis for CDMA Systems

The system capacity of a traditional CDMA system is interference limited. The traditional CDMA systems are all self-interference systems. The three predominant types of interference are inter-symbol interference (ISI), which is created by signal dispersion due to the time-varying channel, multiple access interference (MAI), which is the interference from other users in the same cell, and adjacent cell interference (ACI), which is the influence of all the interfering signals from adjacent cells onto the useful signal. ISI can be mitigated by sequential detection and adding the delayed versions to the signal energy. The effect of ACI is not consider in this work [84].

A traditional CDMA system has a normalized signal to interference ratio (SIR) of [84]

$$SIR = \frac{E_s}{N_J} = \frac{G_p}{N_u - 1}, \tag{3.14}$$

where E_S is the received symbol energy in Joules, N_J is the received interference power spectrum density in W/Hz, G_p is the processing gain or spreading factor, and N_u is the number of users per sector. Therefore, in a CDMA system the number of users per sector limits the channel signal-to-interference ratio according to this formula, and can be changed according to the channel condition. In CDMA, the channel condition can be represented by the normalized SINR, which is

$$SINR = \frac{E_s}{N_J + N_0}, \tag{3.15}$$

where N_0 is the received noise power spectrum density in W/Hz. If we only consider MAI and ignore other interference, we can obtain

$$N_0 = \frac{E_s}{SINR} - N_J = \frac{E_s}{SINR} - \frac{E_s}{G_p/N_u - 1} = E_s(SINR^{-1} - \frac{N_u - 1}{G_p}). \tag{3.16}$$

In a traditional CDMA system, the signal power and inference power are much greater than the noise. Hence, it is assumed that $SINR = SIR$ [84]. Further, if only the interference from the other users in the same cell is considered in the error probability analysis of a Rice lognormal channel DS-CDMA system, the variance of the interference from multiple users in the same cell and from multipaths can be written as [84].

$$\sigma_{int}^2 = \frac{T_s^2}{3G_p}[1 + \frac{N_u(L+K)}{1+K}]. \tag{3.17}$$

Table 3.1: Simulation Parameters for turbo coded CDMA based Rice lognormal Channel

Parameters	**Value Considered**
Rice K Factor	6, 10, 14, and 18dB
Power Control Error (PCE)	2.5, 3.0, 3.5, and 4.5dB
Number of Users N_u	5, 7, and 9
Number of Iterations	4, 6, and 8 Iterations
Frame Length	100, 500, and 1000 bits
Component Encoder	(13,15)RSC
Component Decoder	MAP
Number of Satellites	8
Rate of Transmission	1/3
Processing Gain G_p	32, 64, 128
Channel	Rice-lognormal

3.3 Simulation Results

This Section discusses the performance evaluation of turbo coded channel CDMA based parameters. Examination of our system model with turbo coded transmission and varying system parameters is performed via the Monte Carlo method of simulation. The parameters of interest for channel are K, PCE, G_p and N_u for the CDMA system. Turbo code parameters of interest include the frame length and the number of iterations. The purpose of the simulations is to determine the average bit error rate (BER) with differing system parameters. We implement the MAP algorithm using C++ and apply the method to Rice-lognormal channel. The parameters used for simulation are summarised in Table 3.1.

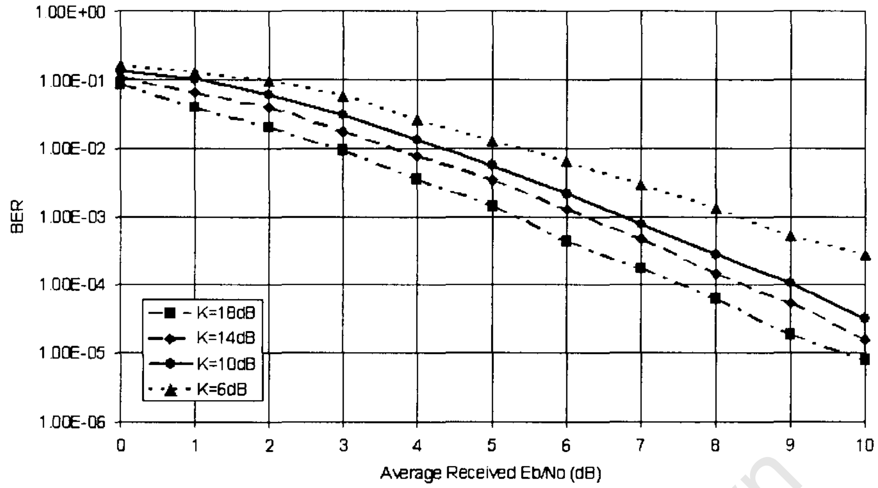

Figure 3.6: Simulation results of the turbo code performance with Rice Factor, K = {6, 10 , 14, 18}dB, Rate 1/3, 6 iterations and MAP Turbo code over Rice-lognormal channel

3.3.1 Performance Evaluation of Channel Parameters with Turbo Codes

The result of a varying K in an uncoded system was presented in the previous section. For comparison purposes, we once again simulate the same K, namely 6, 10, 14, and 18dB with a turbo coded system. As expected, and as shown in Figure 3.6, the performance improves with the increase of the K when PCE is constant. Furthermore, an increase of K means that the channel fading is reducing and the channel is approaching the performance of the AWGN channel. This confirms how significant the parameter K is in our system model. Additionally, we simulated the coded system with a fixed K, μ_{pl} = 0dB, and a varying PCE. These parameters are necessary when considering the mitigation of the lognormal fading and path loss. Unfortunately, the delay in executing power control commands does not allow shadowing to be completely removed, thus causing signal power fluctuations as the mobiles move in the cell. It has been shown that signal fluctuations after imperfect power control can be fitted accurately by a lognormal distribution with the variation ranges from 2.8 – 4.8dB. We choose the variance σ_{pl} of our system to be within this range, namely

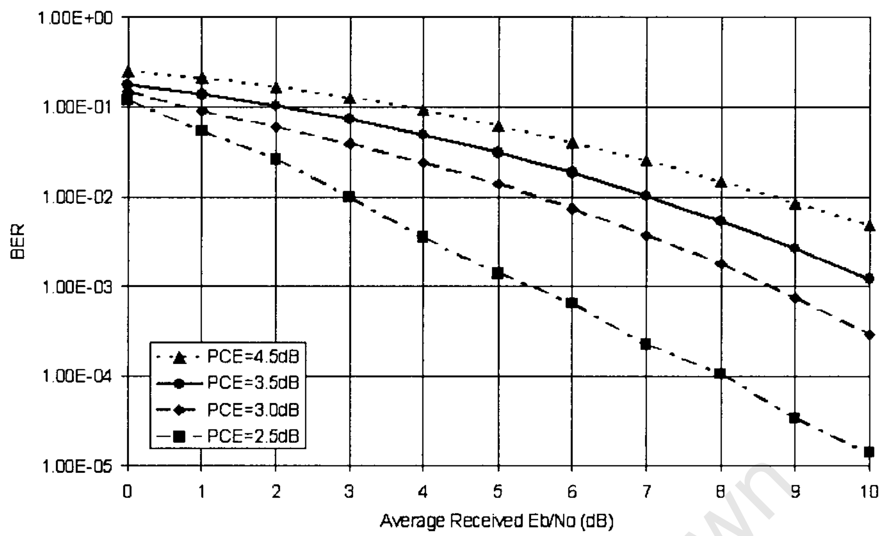

Figure 3.7: Simulation results of the turbo code performance with power control error, PCE = {2.5, 3.0 , 3.5, 4.0}dB , Rate 1/3, 6 iterations and MAP Turbo code over Rice-lognormal channel

2.5, 3.0, 3.5, and 4.5dB. The results from the simulation is shown in Figure 3.7. Here, we observe a proportional increase in the turbo coded system performance as the PCE value decreases with a constant K. It is clear from the results obtained that the K and PCE are inversely related.

3.3.2 Performance Evaluation of CDMA Parameters with Turbo Codes

The performance of a varying N_u = 5, 7, and 9 and a fixed G_p, and the performance of varying G_p = 32, 64, and 128 and a fixed N_u are evaluated by simulation for the Rice-lognormal channel, the results of which are shown in Figures 3.8 and 3.9, respectively. The BER performance of varying N_u is substantially degraded as the number of users increases. In contrast, there is improvement with the increased G_p. In an asynchronous CDMA system that is dominated by the multiple access interference, the performance is expected to degrade as the number of users increases. The degradation can be reversed by using a large processing

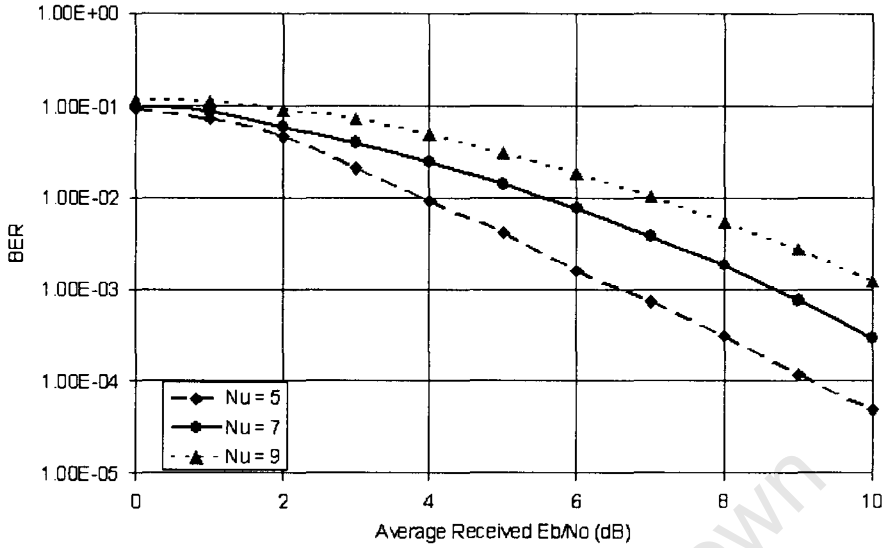

Figure 3.8: Simulation results showing the turbo code CDMA based system performance as user N_u = 5, 7 and 9, Rate 1/3, 6 iterations and MAP Turbo code over Rice-lognormal channel

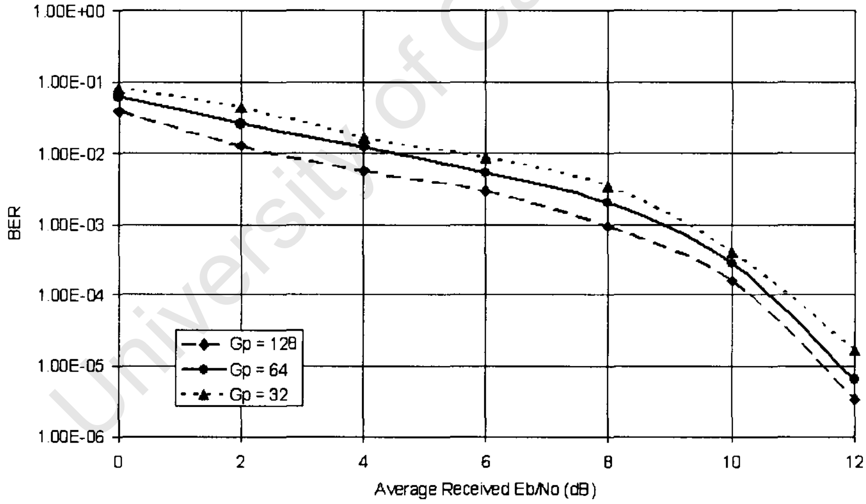

Figure 3.9: Simulation results showing the turbo code CDMA based system performance as processing gain Gp = 32, 64 and 128, Rate 1/3, 6 iterations and MAP Turbo code over Rice-lognormal channel

gain.

3.3.3 Performance Evaluation of Turbo Codes Parameters

The performance of turbo codes at a varying frame length of 100, 500, and 1000 bits, and a different number of turbo iterations are also simulated over a Rice-lognormal fading channel and compared in terms of the BER. This performance is shown in Figure 3.10. The results of a varying number of iteration of 4, 6, and 8 are displayed in Figure 3.11. In both cases the coding gain is obtained as the respective parameter values are increased.

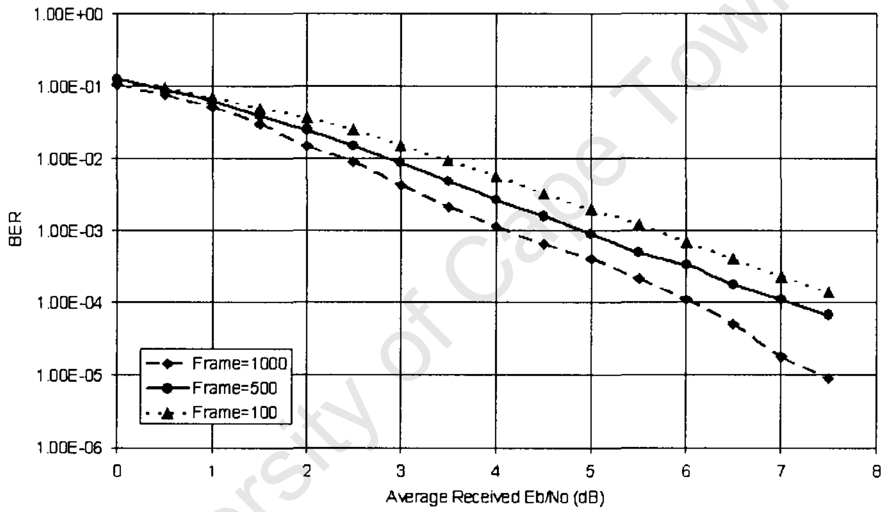

Figure 3.10: Simulation results showing the turbo code performance with frame length 100, 500, and 1000 bits, Rate 1/3, 6 iterations and MAP Turbo code over Rice-lognormal channel

3.4 Chapter Summary

In this Chapter, the channel and system model have been presented. The channel model for the work in this thesis have been briefly described. This model is based on the Rice-lognormal distribution. The influence of K and PCE on channel models was discussed. It

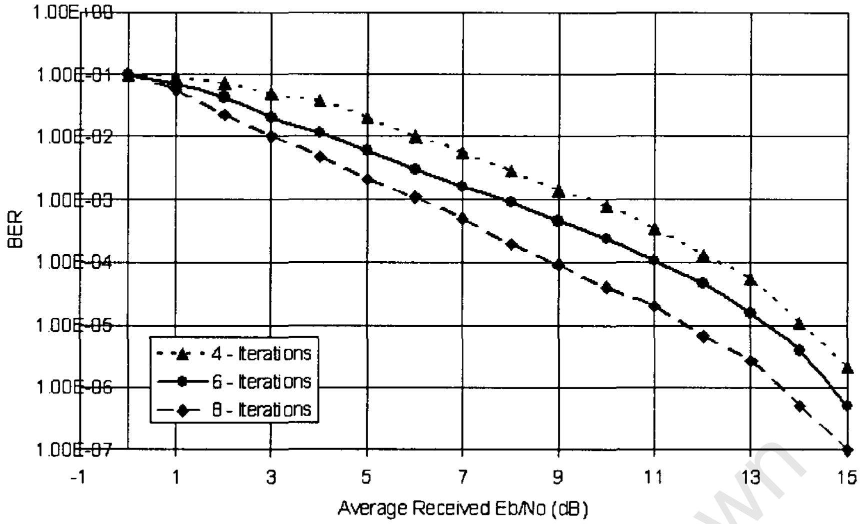

Figure 3.11: Performance Comparison by Simulation of the number of iteration 4,6 and 8 for 1000 bits, Rate 1/3, MAP Turbo code over Rice-lognormal channel

is found by simulation that as the K factor decreases, there is a degradation in the system performance when the PCE is held constant. Similarly, performance degradation occurs as the PCE is increased when the K factor is held constant. This behaviour has been validated in literature. The effect of parameters, such as the processing gain and the number of users, were also simulated. Based on the continuously monitored channel condition, our proposed adaptive algorithm will adjust the modulation, coding, and CDMA parameters to yield an overall system performance. Henceforth, when the channel condition is changed, new values of these parameters will be assigned to improve the system.

Chapter 4

Adaptive Radio Resource Management Algorithms

4.1 Introduction

In this Chapter, the adaptive CDMA algorithm is designed, evaluated and simulation results are presented. The AMC is first introduced and schemes for its development in a three-stage process and grading is proposed in Section 4.2. In Section 4.3, the existing AMC transmission architecture is described. The block diagram of the proposed AMC transmission system is presented in Section 4.4. Objectives of AMC are stated in Section 4.5. Following the requirements for AMC algorithms, in Section 4.6 the parameters influencing the throughput of a CDMA system are presented. The BER formulas in terms of modulation and coding parameters for CDMA systems are described. With these formulations, the BER can be obtained at a certain point of time given the values of the modulation and coding parameters. In Section 4.6.2, the parameters influencing the transmission are explained. All parameters in the throughput formula of CDMA systems are explained and the influence of CDMA parameters on the transmission QoS is briefly described.

Adaptive modulation is introduced in in Section 4.7. In Section 4.8 adaptive CDMA algorithm for radio resource management is proposed. The actual adaptive CDMA algorithm for satellite systems is built and presented in Section 4.8.1. Section 4.8.2 outlines the simulators and assumptions that are exploited in the simulations. Sections 4.9 to 4.9.7 present simulation results of all these algorithms. Conclusions are given in Section 4.10.

4.2 Development of Adaptive Modulation and Coding

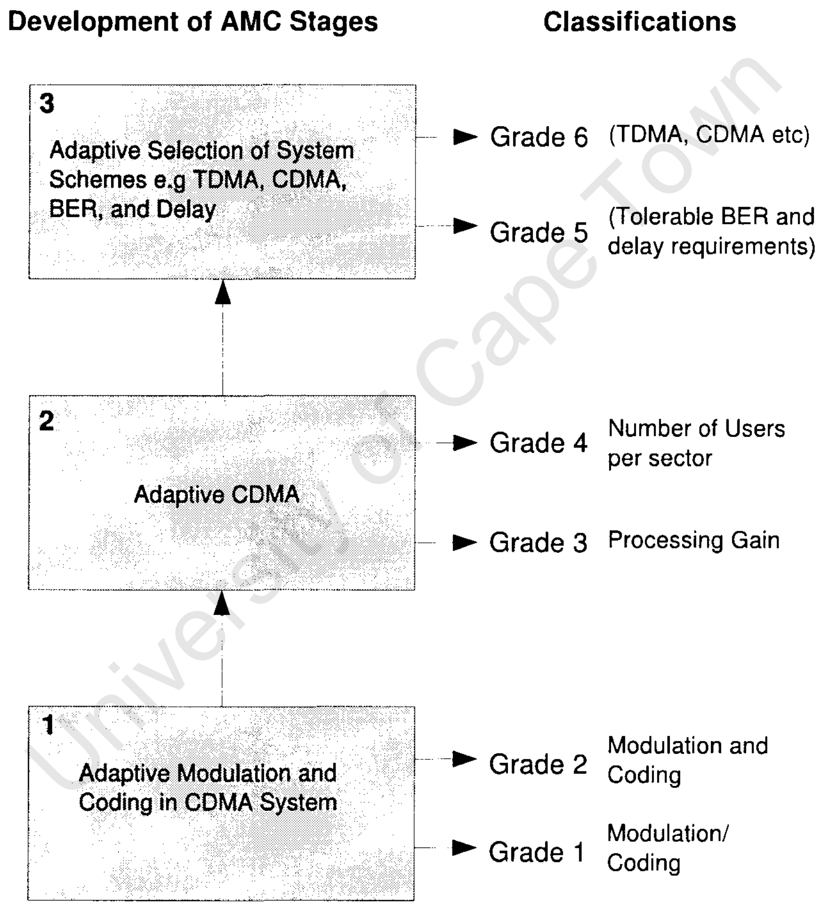

Figure 4.1: The AMC Developmental stages

The AMC technique can be divided into three stages, as depicted in Figure 4.1. The first stage is known as adaptive modulation and coding in the CDMA system. It makes use of available resources and modulation parameters such as modulation level and coding rate. Related work has been reported in the literature [89, 90]. The second stage is called adaptive CDMA. This stage includes CDMA and its parameters. Here, the multiple access (MA) parameters such as the processing gain are adaptive [91]. The third stage of the adaptive modulation and coding technique involve changing the multiple/multiplexing access (MA) schemes (e.g. TDMA, CDMA) by a certain criterion. Changing the MA scheme on the requested users QoS, which might include tolerable BER, delay, to maximize system capacity under current channel conditions and frequency bandwidth available[92]. Further, grading is done at each stage based on the number and type of resources available. For instance, the grade 2 has two resources available for adaptation while grade 3 has three resources for adaptation. This initiative enables us to examine the performance of adaptive modulation and coding and to effectively manage radio resources. Many work reported in literature are base on grade 1 and grade 2 [89, 90]. Our focus is adaptive CDMA, which is the second stage of the AMC development, and grade 3 and grade 4 falls into this stage. However, references will be made to adaptive modulation and coding in CDMA and adaptive selection of a multiple access scheme, which are stage one and stage three respectively.

4.3 Architecture of the Existing AMC Systems

Figure 4.2 presents the architecture of AMC system. It shows that adaptive modulation is used to change the modulation parameters with the channel situation. In order to do this, the channel situation should be known by the transmitter before transmission, for which reason the acquisition of the channel information is necessary. This is called channel estimation. There are two different methods for channel estimation. One is that the transmitter gets the feedback of the channel information from the receiver; the other is that the transmitter itself estimates the channel [92, 93]. Channel estimation is an essential prerequisite for an adaptive modulation system. Channel estimation is out of the scope of this thesis, so we do not discuss it in depth within this thesis. To perform the function of AMC, an efficient scheme has to be built, by means of which modulation parameters can be configured based on

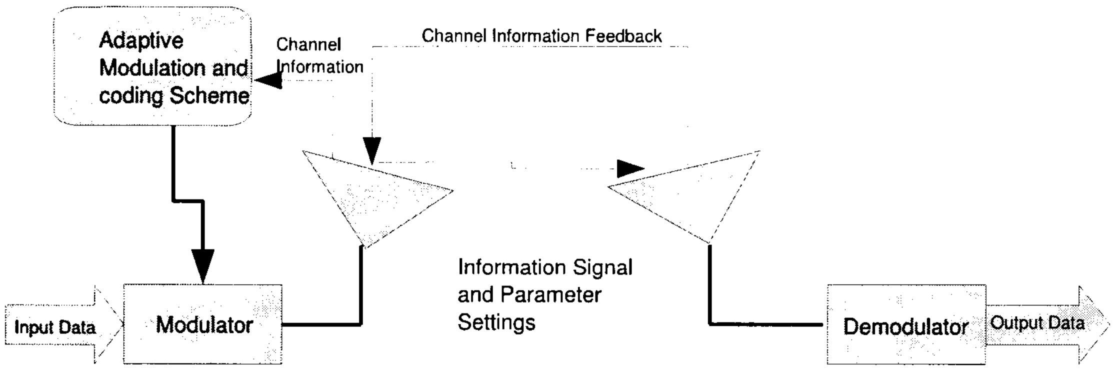

Figure 4.2: The Architecture of Adaptive Modulation

the instantaneous channel condition. The channel information is the input for the adaptive modulation and coding scheme. The resulting flow chart of adaptive modulation systems looks as follows:

- Measure and predict the channel conditions accurately, i.e. estimate the channel ;
- Build up an algorithm to adapt the modulation and coding parameters in terms of the channel conditions ;
- Obtain the settings of the modulation and coding parameters at the receiver.

This thesis concentrates on the second step, that is, on building the AMC scheme. In the simulations, perfect channel estimation, and that the receiver knows the exact settings of the modulation and coding parameters are assumed.

4.4 The Proposed AMC Transmission System

The block diagram of a DS-CDMA transmission system model is shown in Figure 4.3. The transmitter is composed of RCPT encoder, interleaver and data modulator. The information bits are initially encoded by a turbo encoder, interleaved and the encoded bits are modulated by data modulator. Rice-lognormal fading channel is assumed and AWGN is added. The

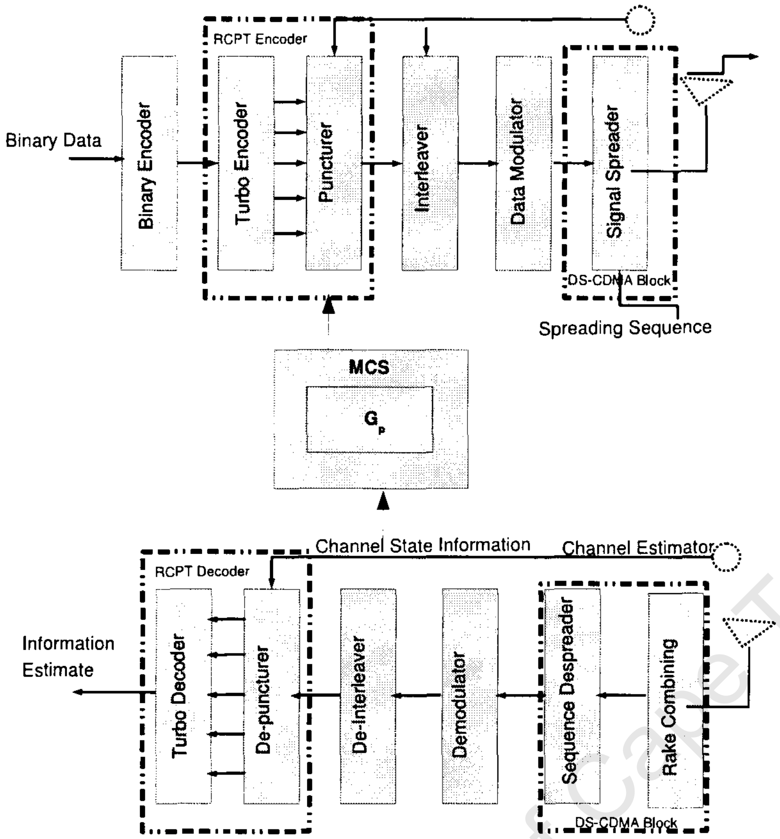

Figure 4.3: Block Diagram of Transmission System

receiver is composed of data demodulator, turbo decoder, de-interleaver, RCPT decoder, and channel estimator. A turbo code of length n corresponding to k information bits is employed such that the rate R is given by k/n. The key code is a rate 1/3 mothercode and punctured accordingly. The punctured sequences which are of different lengths for different puncturing periods are block-interleaved before mapping to a binary phase-shift keying (BPSK) or quadrature-shift keying (QPSK) signal constellation.

In DS-CDMA, the transmission channel is multi-user shared. The orthogonality of individual user is achieved by spreading the information data with a long scrambling code of a spreading factor (SF) length through a user-specific pseudo random sequence (PN). In the system, channel code gain and spreading gain are simultaneously obtained. At the receiver, the SINR is estimated at the channel estimator. Modulation and Coding Scheme (MCS) selector and other transmission parameters are modified based on the received SINR at the mobile receiver.

4.5 Objective of Adaptive Modulation and Coding

The task in wireless communications consists mainly of two aspects: the transmission QoS and the throughput. AMC is designed to achieve this purpose. The advantage of AMC is that it can realise higher transmission QoS and higher throughput by efficient usage of the channel situation. This work on adaptive CDMA is based on these objectives:

- Achieving higher transmission QoS (bit error rate);
- Achieving higher throughput (bit rate).

4.6 Parameters Influencing Throughput and QoS

One of the objectives of AMC is to improve throughput. In order to do that, the parameters influencing throughput must be known.

In this Section, the parameters influencing throughput are investigated. In terms of the SNR of the channel, there is a well-known formula for channel capacity [93]:

$$C = B\log_2(1 + SNR)[bps], \tag{4.1}$$

where C is channel capacity, and B is the channel bandwidth. Adaptive modulation is used to change modulation and coding parameters to channel situations in order to guarantee the required transmission quality and to reach a certain throughput. This means that the systems need to know the throughput as a feedback. Meanwhile the required transmission quality can be guaranteed. Hence the need for a formula to calculate the throughput with modulation, coding, and CDMA parameters.

4.6.1 Throughput Formulas for CDMA Systems

As established in Chapter 3, in CDMA systems the throughput can be given as

$$R_t = (R_c \cdot log_2 M) \cdot B = (R_c \cdot log_2 M) \cdot \frac{B_t}{G_p}. \tag{4.2}$$

This is the throughput formula for CDMA systems in terms of bit rate, where B_t is the total data frequency bandwidth, R_c, coding rate, M, modulation level, and G_p processing gain. All the values of parameters on the right side of the formula are determined by the channel situation. It could be observed that the bit rate of CDMA systems is linearly proportional to the coding rate and transmission bandwidth, logarithmically proportional to modulation level, and inversely proportional to processing gain. We can conclude that in order to increase the bit rate, we can either increase the modulation level (M), coding rate (R_c), or $1/G_p$ inverse. Two methods can thus be used to increase the bit rate: the first is to increase the modulation level or coding rate and the second is to increase $1/G_p$, that is, to decrease the processing gain. These methods can only be used when the channel situation permits a change in the values of these parameters. In an adaptive CDMA system we can vary these parameters according to the channel situation to achieve good QoS and to reach a certain bit rate.

4.6.2 Parameters Influencing QoS

This Section examined how some parameters (i.e. the coding rate, modulation level, and processing gain) affect the transmission QoS. To evaluate the transmission QoS, the quantity BER versus SINR is employed. Given that the other parameters remain constant, in CDMA systems, BER will increase with coding rate and modulation levels; it will not change with the transmission bandwidth. For the processing gain, the BER decrease with the increase of processing gain.

4.7 The Existing Adaptive Schemes - Adaptive Modulation

This Section begins with the illustration of adaptive modulation, which is a simple case of adaptive process. Adaptive modulation adapts one parameter that is modulation level. The method here is based on the idea of partitioning the estimated channel SINR into regions using a set of threshold values. Each region is associated with a particular modulation scheme m_i while the threshold values are optimised to maximise the overall throughput [94, 95]. The system has a set of modulation schemes $m_i, i = 1, \ldots, M$ corresponding to throughput versus average SINR. The average SINR values can be graphically represented, where the curves intersect with each other. This is shown in Figure 4.4. The average SINR values corresponding to the intersection points are chosen as the range of SINR(γ) into n regions, denoted by $[\gamma_i, \gamma_{i+1})$ for $i = 0, \ldots, n-1$. A particular modulation scheme m_i is assigned to the region $[\gamma_i, \gamma_{i+1})$. The adaptive boundaries that determine the switching condition for different target BER is shown in Table 4.1.

Table 4.1: Condition for switching

SINR	Modulation	Target BER
SINR $\leq 2dB$	BPSK	10%
2dB $< SINR \leq 8dB$	QPSK	10%
8dB $< SINR \leq 12dB$	16QAM	10%
12dB $< SINR \leq 40dB$	64QAM	10%
SINR $\leq 8dB$	BPSK	1%
8dB $< SINR \leq 14dB$	QPSK	1%
14dB $< SINR \leq 20dB$	16QAM	1%
20dB $< SINR \leq 40dB$	64QAM	1%
SINR $\leq 11dB$	BPSK	0.1%
11dB $< SINR \leq 17dB$	QPSK	0.1%
17dB $< SINR \leq 25dB$	16QAM	0.1%
25dB $< SINR \leq 40dB$	64QAM	0.1%

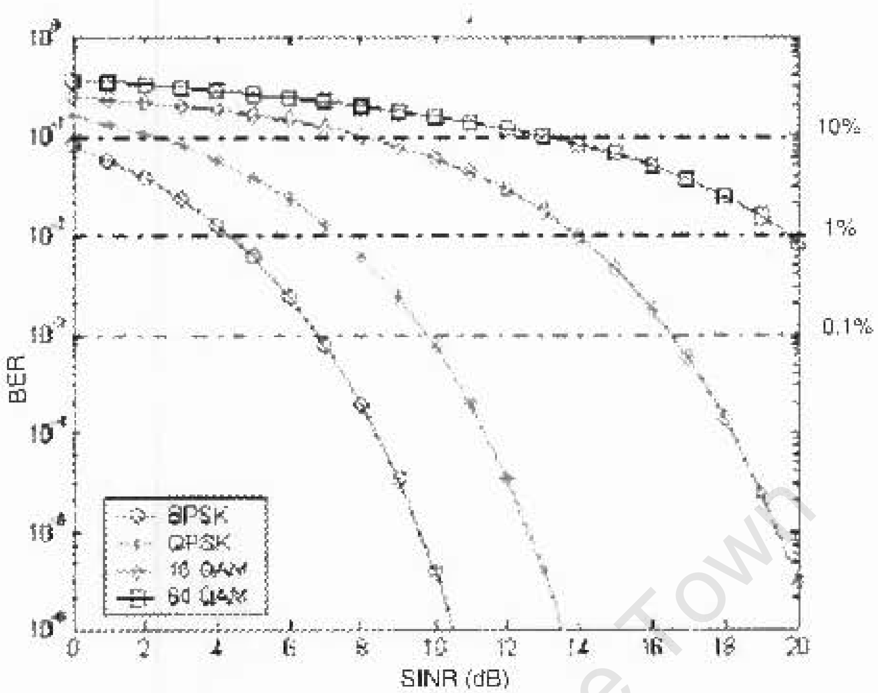

Figure 4.4: Adaptive Modulation Boundaries

4.7.1 Adaptive Modulation Algorithm

This section present an adaptive modulation algorithm for user k given the modulation scheme, $m = 1, \ldots, M$. The transmit power is the same for all users denoted by SNR_m^k and is uniquely dependent on the instantaneous channel coefficient $\alpha_{k,m}$. An adaptive modulation algorithm for QAM modulation is described as follows:

Initialisation process

Reordering $|\alpha_1^k| < |\alpha_2^k| < \cdots < |\alpha_M^K|$ in ascending order and allocate bits (modulation mode) for users equally

$Q_m^k = TP \qquad m = 1, \ldots, M$

where $Q_m^k = TP$ is the initial modulation mode allocation on m for user k

$TP = 1, 2 \cdots, M$ correspond to the throughput of QAM

Allocation process

loop1: $m = 1$ to $M - 1$, step 1

loop2: $n = M$ to $m + 1$, step-1

if $Q_m^k == 0$,

break loop2

end if

switch $Q_{k,n}$

case1: if $|\alpha_n^k|^2 > 2|\alpha_m^k|^2$

$Q_n^k = Q_n^k + 1$;

$Q_m^k = Q_m^k - 1$;

end if

case2: if $|\alpha_n^k|^2 > 3|\alpha_m^k|^2$

$Q_n^k = Q_n^k + 1$;

$Q_m^k = Q_m^k - 1$;

end if

end switch

end loop2

end loop1

Then $Q_m^k, m = 1, \ldots, M$ is the final allocation modulation modes for user k. The above algorithm can be performed for all the users in the system separately, since the channels from the transmitter to different user are independent and they do not interfere with each other.

4.7.2 Simulation Results

In this Section simulation results for adaptive modulation are provided. Ideal conditions including no feedback delay or error and perfect channel estimation are assumed. We investigated the adaptive CDMA algorithm for Rice-lognormal channel. The simulation was set up with $N_u = 9$, $G_p = 64$, coding rate 1/3, (13,15)RSC, 6 iteration, MAP turbo code and frame length of 1000 bits. Throughput using adaptive CDMA algorithm is evaluated. The performance of adaptive modulation for BER targets 10% is simulated and shown in Figure 4.5. It was observed that the performance of adaptive modulation begins by overlapping the BPSK curve. However, as the SNR is increased to 5 dB, the performance of adaptive modulation begins to decrease.

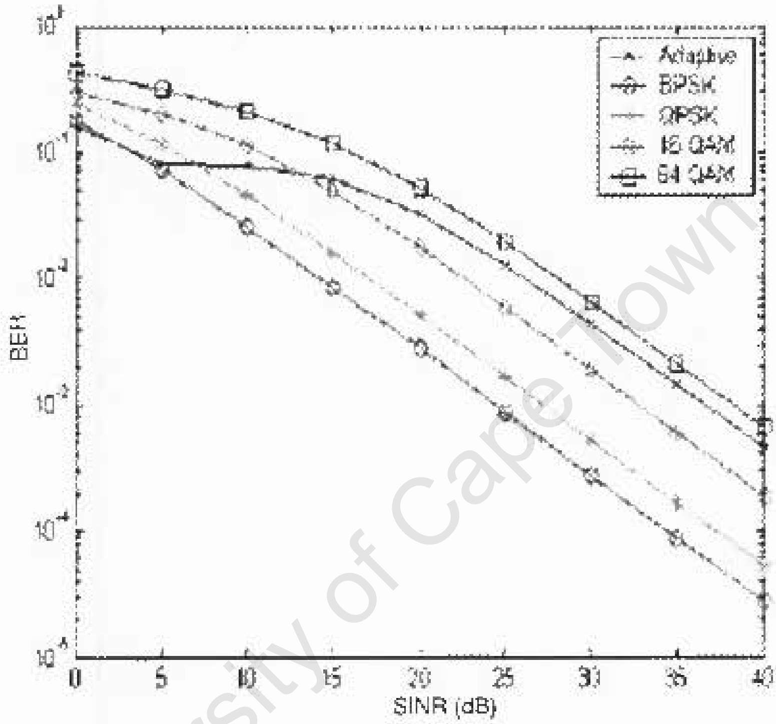

Figure 4.5: BER Performance of Adaptive Modulation

4.7.3 Adaptive Modulation and Coding

In this Section the parameters to be adapted are increased to two and these are modulation levels and coding rates. In AMC system, a specific combination of modulation scheme and coding rate is treated as one called MCS. In this thesis, 6 MCSs are considered in the AMC system. Table 4.2 shows the MCS set. One of the key technologies for AMC is how to switch (or select) MCS. SIR is a common criterion to switch MCS. The switching boundaries for

Table 4.2: Modulation and Coding Schemes (MCS): Mode dependent parameters for AMC

Mode (MCS)	Modulation (M)	Coding Rate (R)
MCS1	BPSK	1/3
MCS2	BPSK	1/2
MCS3	BPSK	2/3
MCS4	QPSK	1/3
MCS5	QPSK	1/2
MCS6	QPSK	2/3

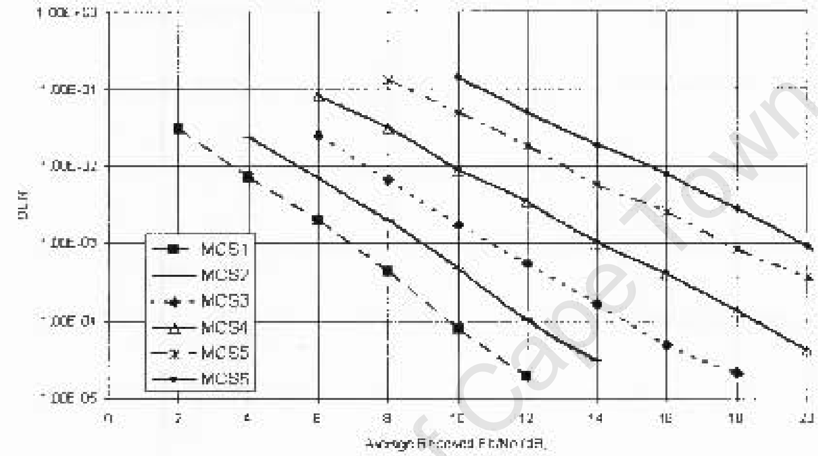

Figure 4.6: Simulation results showing the performance of MCS for the determination of adaptive boundaries using rate 1/3 and 6 iterations

MCS are shown in Figure 4.6. For example if the measured SIR is above the SIR threshold between MCS1 and MCS2 but below the SIR threshold between MCS2 and MCS3, MCS is switched to MCS2. If the measured SIR is above the threshold between MCS2 and MCS3, MCS3 will be selected. The threshold values are decided in order to achieve maximum throughput. The threshold values are a function of channel conditions and depend on which particular MCS set is used [96].

Radio link control select among the MCS options, in response to the channel condition measured. The operation of selecting the rate is made by sending the frame with different puncturing patterns from the same mothercode and allows transmission to begin with low redundancy. It increases redundancy only when errors occur, thus adaptively changing the effective data rates [97]. Adaptive Coding required the coding rates to be selected from predetermined values bounded by [1, 1/3]. The code construction allows for a family of code of rates $R_l = P/(P+l)$, where $l = 0, 1, \cdots, (M-1)P$. For each value of l, a binary puncturing matrix $a(l)$ is defined. An RCPT code with $M = 3$ and $P = 2$ is constructed and then the RCPT is made to select any of code rate in the set {1, 2/3, 1/2, 2/5, 1/3 }. In implementation, all possible codes rates are not required to be utilised [28, 98]. In this work, a special case where $l = 2^n$ and $n = 0, 1, 2$ are considered. Hence, rates 1/3, 1/2, and 2/3 are used in our design.

In the case of adaptive modulation for instance, if given an adaptive modulation system that uses BPSK and QPSK for a target BER_0 of approximately 10^{-3}, the adaptive modulation uses QPSK modulation for good channel and BPSK modulation for bad channel. This type of modulation technique is called variable-rate techniques, where symbol rate is fixed while changing the constellation size or modulation type. In other words, if considering a transmission that is encountering a deep fade, the option is to use one of the modulation schemes, which differ in spectral efficiency and robustness. If fading is extremely deep, perhaps half of all bits will be in error, it is advantageous to send fewer bits because the total number of errors will be decreased. This influences BER much more than total number of bits sent. When the channel is not in a fade, then many bits could be sent. In this situation, BER could be lower by increasing the number of bits sent because errors become less frequent [93, 97].

Code rate and modulation level are use to design modulation and coding schemes (MCS). Both the set of coding rates and modulation levels are made available so that the system is selected from values bounded [1, 1/3] and [2, 2M] of the code rates and signal constellation respectively. In this work, the set of coding rates {1,2/3,1/2,2/5,1/3} of $l = 2n$ and $n = 0, 1, 2$ were made available. The modulations level considered are BPSK and QPSK for satellite transmissions. It is the combination of those two principles that allows the BER performance of adaptive systems to be more robust than static modulation systems while simultaneously providing better spectral efficiency at most ranges of SIR.

Table 4.2 summarises the different operating modes available for adapting the system. They range from BPSK 1/3 rate (Mode 1) to QPSK 2/3 rate (Mode 6). The BPSK 1/3 rate mode provides a more reliable transmission link than the QPSK 2/3 rate mode for a given receive power. Figure 4.6 shows the frame performance of the 6 main modes of the system.

4.7.4 AMC Algorithm

The algorithm for the implementation of AMC is presented below.

$$\text{MCS} = \begin{cases} No\ Transmission\ if\ SINR < \Gamma_{min} \\ MCS1 \text{ if } \Gamma_{min} \leq \text{SINR} < \Gamma_1, \\ MCS2 \text{ if} \Gamma_1 \leq \text{SINR} < \Gamma_2, \\ \cdots \\ \cdots \\ \cdots \\ MCSN \text{ if} \Gamma_{N-1} \leq \text{SINR} < \infty, \end{cases}$$

where $MCSk = 1, 2, \cdots, MCSN$ are modulation and coding schemes and are listed from the lowest data bits per symbol. $\{\Gamma_1, \Gamma_2, \cdots, \Gamma_{N-1}\}$ are the corresponding scheme switching thresholds. The flowchart of the algorithm is shown in Figure 4.7

4.8 The Proposed Adaptive Scheme - Adaptive CDMA

In this Section, an adaptive CDMA algorithm is proposed for efficient radio resource management (RRM) for satellite systems. This algorithm combines CDMA modulation adaptation with MCS parameter adaptation. It is well known that higher order modulations may give better spectral efficiency at the expense of worse bit error performance [99, 100]. A lower channel coding rate has a better error correction capability than the same type of coding

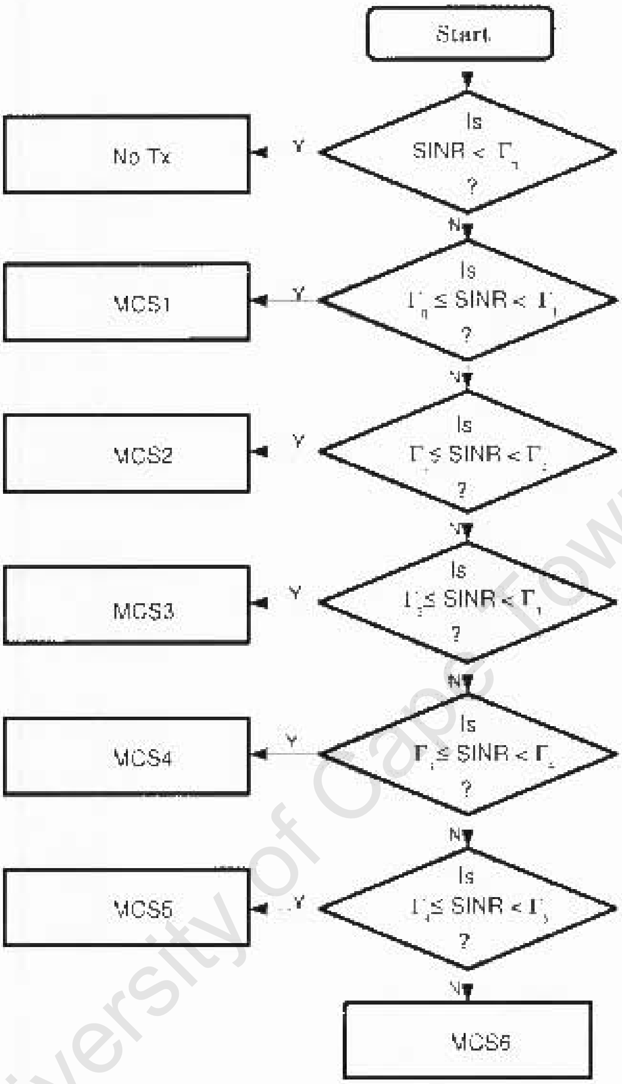

Figure 4.7: Flowchart of adaptive modulation and coding

with a higher coding rate. Thus, with a proper combination of the modulation order and channel coding rate, it is possible to design a definite set of modulation and coding schemes such that an increased spectral efficiency can be achieved in good channel conditions as illustrated in the previous sections of this Chapter. It is also possible to increase the bit rate further by the use of the processing gain for a given MCS, when the channel conditions allow. The algorithm presented in this work selects the number of processing gain and the

MCS per transmission such that the increase in the number of processing gain is prioritised over the increase in the higher order modulation and coding scheme.

Together with the AMC, processing gain transmission provides an additional dimension and thereby increased granularity for the link adaptation. While AMC provides a coarse adaptation to the channel, the use of processing gain brings the 'fine turning' to the selected MCS. Obviously, an algorithm is needed to select the MCS and the number of processing gain. The objective is to maximize the bit rate by selecting the right combination for the number of processing gain j and the MCS i, given a channel condition ρ. Let $f(\rho_{i,j}) = f_{i,j}$ be the frame error rate associated with the state (i, j) as a function of ρ , and $g(\rho)$ be the probability density function of ρ . Also, let $\varepsilon_{threshold}$ be the frame error threshold which defines the maximum tolerable error. The minimum channel condition which is required for state (i, j) such that $\varepsilon_{threshold}$ is not exceeded is generalised and shown in Figure 4.8, and is given by

$$\rho_{i,j} = f_{i,j}^{-1}(\varepsilon_{threshold}), \tag{4.3}$$

and the error curves are typically given by

$$f(\rho_{i,j}) = f(\rho_{i,1} - 10\log_{10}(j)), \tag{4.4}$$

where ρ is in dB. The bit rate associated with the state (i, j) is given by $r_{i,j} = jr_i$, where r_i is the bit rate associated with the MCS i.

4.8.1 Adaptive CDMA Algorithm for Rice-lognormal Channel

This Section presents an adaptive CDMA algorithm for different users given the set of processing gains and MCS. In the traditional adaptive CDMA algorithm the parameters such as Rician K factor, SINR, N_u are input into the system. The Rician K factor are used to set the channel parameters. After acquisition of the required parameters, the algorithm asks whether or not the CDMA modulation parameters have been set. If so, the algorithm compares the current channel parameters with the ones of the previous moment. The algo-

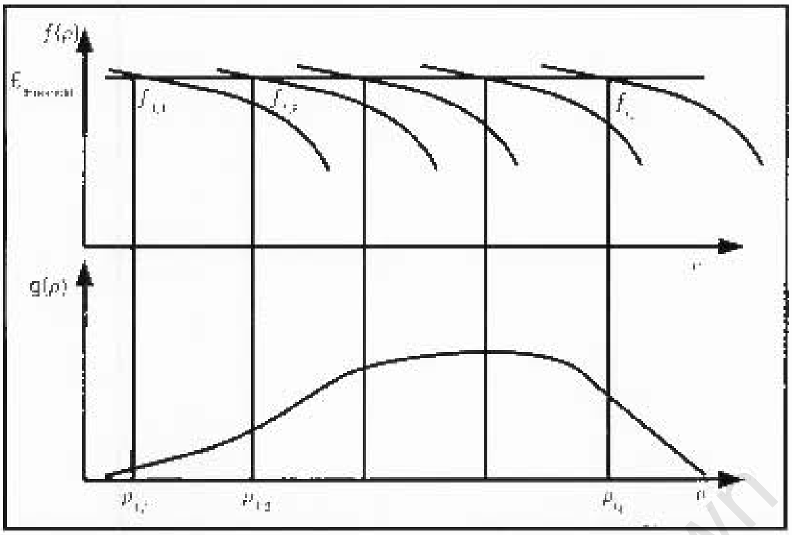

Figure 4.8: Generalised link performance in changing channel conditions

rithm only changes the settings if the channel situation changes. If not, parameter setting begins. The channel parameters are set first. Next, CDMA modulation parameters (modulation level M and processing gain G_p) are assigned. The proposed algorithm consider RCTP rate puncturing method and a prioritised processing gain over the modulation and coding scheme. The operations of the algorithm is explained below and the flowchart is shown in Figure 4.9. The algorithm highlights how the processing gain and the MCS are selected in order to increase the transmission bit rate.

- Step 1: Initialize the state (i, j) to $(1, 1)$.
- Step 2: Check the channel conditions whether it is good with respect to the threshold corresponding to the current state (i, j). If yes, go to step 3, else go to step 7.
- Step 3: Check whether the maximum number of processing gain has been reached. If not, go to step 4. Otherwise, go to step 5.
- Step 4: Increase the number of processing gain. Go to step 2.
- Step 5: Check whether the maximum MCS is reached. If not, go to step 6. Otherwise, go to step 7.

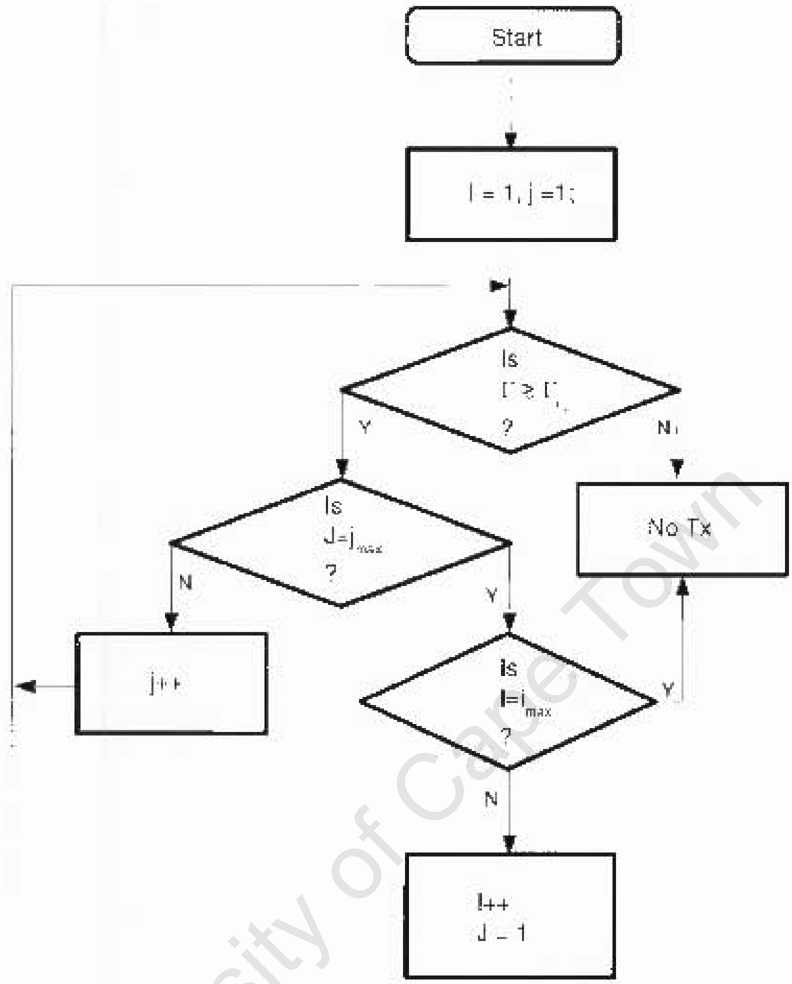

Figure 4.9: Flowchart of adaptive CDMA algorithm with a prioritised processing gain

- Step 6: Change the state to $(m_2 + 1, 1)$. Go to step 2.
- Step 7: Compare the bit rates that correspond to states (m_1, n_1) and (m_2, n_2) , and select the state which gives the higher bit rate.

In step 2, the algorithm checks the channel conditions, and increases the number of processing gain in order to increase the bit rate as shown in steps 3 - 4. The next higher MCS is examined only when the maximum number of processing gain is reached as in steps 5 - 6. In fact, given a channel condition, higher MCS with a given number of processing gain does not necessarily give a higher bit rate than the previous MCS with a maximum number of processing gains. Thus, the last step is to check and make sure that the state with the highest bit rate is selected. In this algorithm, a higher priority is given to increase the number of processing gains instead of the MCS.

In this work, i denotes joint modulation and coding scheme (MCS), j denotes processing gain. The processing gain is prioritised over MCS. Three processing gains are considered. They are 32, 64 and 128. Let $f_{i,j}(\rho) = f_{i,j}$ denote the frame error rate (FER) associated with the state (i,j) then $\rho_{i,j} = f_{i,j}^{-1}(\varepsilon_{threshold})$ and error curve is given as: $f_{i,j}(\rho) = f_{i,1}(\rho - 10\log_{10}(j))$

The adaptive CDMA algorithm for satellite channels has been designed to get as much throughput as possible while guaranteeing a BER of 10^{-3}. If translating this into prioritised adaptive CDMA method, it means that the system will first exploit the processing gain starting with the lowest. The lower the processing gain the higher the spectral efficiency (throughput). The algorithm considered the MCS after all the processing gains have been exploited. The algorithm sets its parameters based on this rule.

4.8.2 Simulator and Assumptions of the Algorithms

In the last Section, adaptive CDMA algorithms for Rice-lognormal channel have been proposed. The algorithm, which is for satellite channel combines Rice-lognormal parameters with CDMA modulation adaptation parameters. In order to evaluate the algorithms by simulations, a simulator is built. Figure 4.10 presents the block diagram of the simulator that is used to evaluate the algorithms designed. First, random channel parameters are generated. Based on the channel parameters, the adaptive CDMA algorithms assign all CDMA modulation parameters. At the receiver, the signals come to the CDMA demodulator through the channel. The BER and throughput for each transmission are calculated.

In this simulation for all AMC algorithms the following assumptions are used:

- The Rician-lognormal fading channel is assumed;

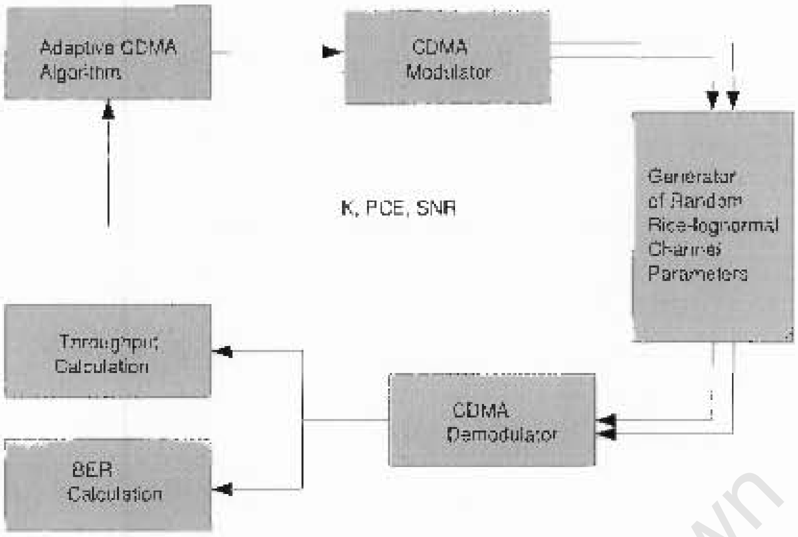

Figure 4.10: Simulator of Adaptive CDMA Algorithms

- Perfect channel estimation is assumed, and the algorithm obtains the channel situation without any delay;
- The time duration between two consecutive transmissions is assumed to be enough for algorithms to finish assigning parameters. In other words, the algorithms can always obtain the current channel situation in time and assign parameters successfully;
- Signals at the transmit antennas are completely uncorrelated, as well as at the receive antennas;

4.9 Simulation Results

MAP turbo decoder and 1/3 turbo encoders with the generator polynomials of $G = (7, 5)_8$ were considered for the simulation. The information sequence length, k, which represents the encoded sequence length, is assumed to be 1,024 bits. The turbo encoded sequence is interleaved with a size $2^a \cdot 2^b$ block-interleaver, where 'a' and 'b' are the maximum allowable integers for a given sequence size. BPSK/QPSK modulation is assumed for data demodulation at the receiver. $G_p = 32, 64, 128$ is assumed for CDMA system. Unless otherwise stated, we assume the uplink channel to be composed of $L = 16$ paths and uncorrelated, Rice-lognormal faded paths.

4.9.1 Throughput Performance of the Proposed Adaptive CDMA Algorithm

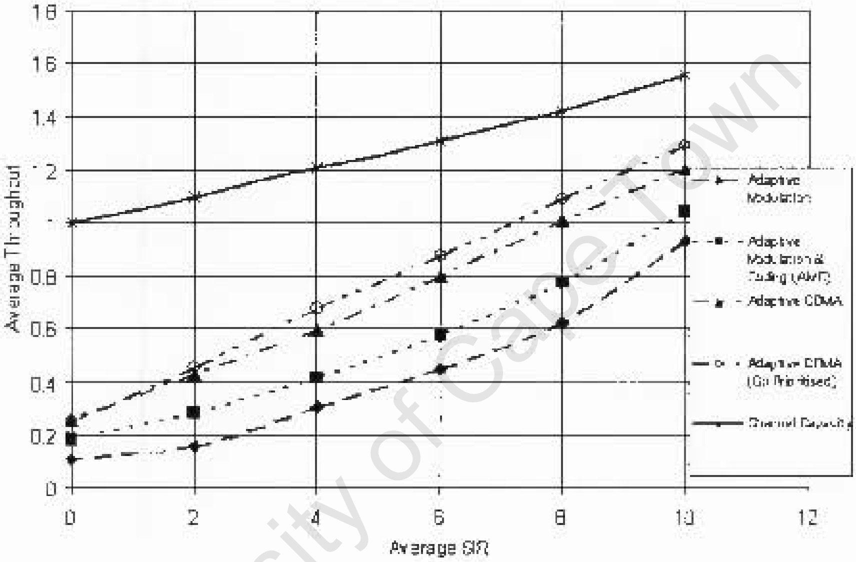

Figure 4.11: Simulation results showing the throughput of the existing adaptive algorithms and the proposed adaptive algorithm

The throughput performance of the existing adaptive algorithms is compared with the proposed adaptive algorithm. Adaptive modulation, AMC, adaptive CDMA and the proposed algorithm are examined. The results are presented in Figure 4.11. Given the same condition, it is observed that the proposed adaptive scheme shows an improved throughput over other existing algorithm. Essentially, the traditional adaptive CDMA algorithm is compared with the prioritised adaptive CDMA system. As shown, the proposed algorithm is an enhanced version of the existing adaptive CDMA system for satellite channel.

4.9.2 Effect of Throughput on Radio Resource Management Algorithms

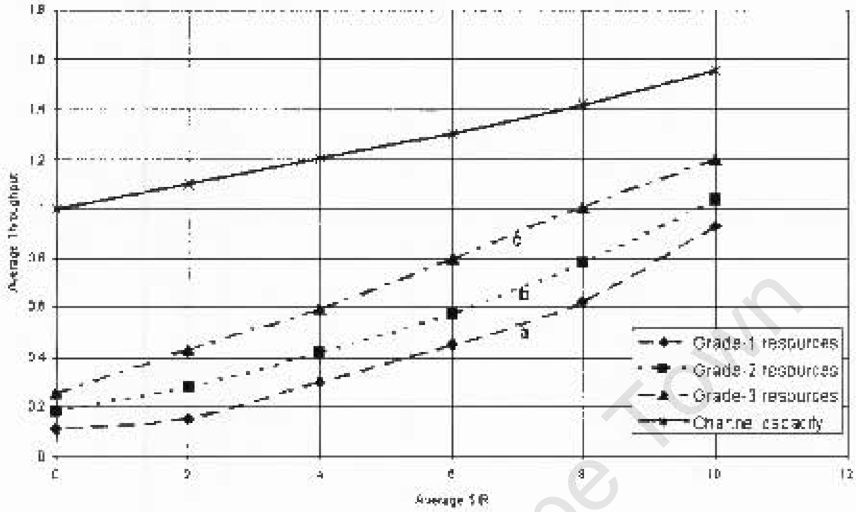

Figure 4.12: Simulation results showing the throughput of different resource grading schemes

The result in Figure 4.12 shows different grading of radio resources. The adaptive modulation is an example of grade-1 with a resource and it is denoted by 'a'. The AMC is an example of grade-2 with two resources and it is denoted by 'b'. The adaptive CDMA falls into grade-3, which consists of three resources and it is denoted by 'c'. Given the maximum transmission time of a codeword as time t, the throughput performance of the adaptive algorithms is examined for grade-1, grade-2 and grade-3 sets of transmitter parameters as shown. It is observed that the increase in the number of available resources at the transmitter lead to increased throughput.

4.9.3 Radio Resource Management Algorithms for Downlink System

So far, the modulation level and processing gain have been set. Obviously, it is difficult for the mobile user to change the number of users in a cell, so this algorithm is actually intended for downlink transmission. However, if we delete the part of adjusting the number of users, the algorithm can be exploited for uplink transmission as well. Figure 4.13 presents the flow

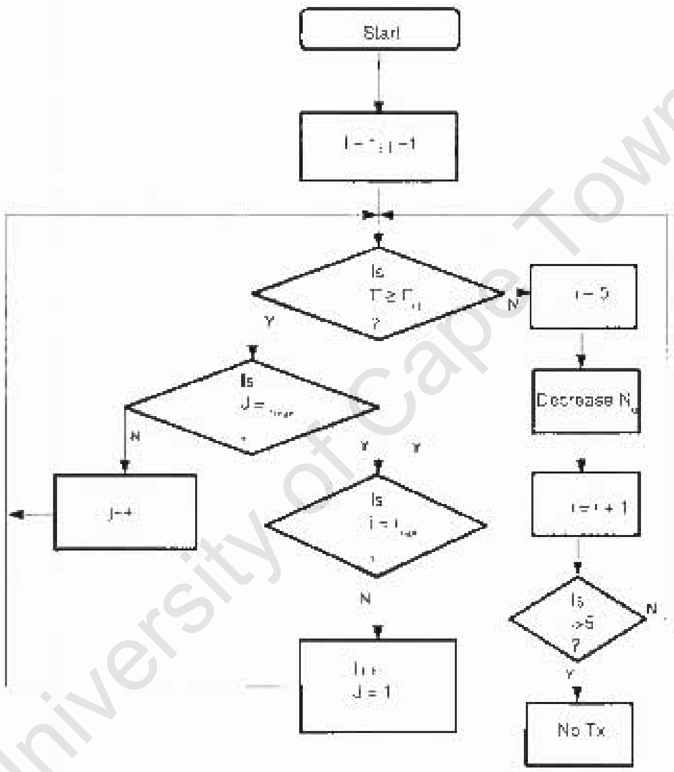

Figure 4.13: Flowchart of Adaptive CDMA Algorithm for Down-link Transmission

chart of setting MCS, processing gains and adjusting the number of users. The algorithm compares the current channel SINR, Γ with the minimum threshold Γ_{min} first. If it is smaller than Γ_{min}, it means that the target BER cannot be guaranteed, even if the largest G_p and highest MCS is assigned. This situation is tagged the *critical section* in this work. In Chapter 3, it is indicated that CDMA systems can also influence the channel situation by changing the number of users, N_u. In this case, the algorithm decreases the number of users to increase the SINR. This process will be repeated at most five times to make the channel SINR larger than Γ_{min}. If the channel SINR is still too small for transmission after the number of users has been adjusted for five times, then a *no transmission* (No-tx) command is automatically activated by the algorithm. In this thesis, the situation is tagged *beyond critical condition*. It is clear from the flowchart that transmission may be performed if a user enters into critical condition but absolutely no transmission if a user is beyond critical condition. The users in a critical condition are expected to maintain their QoS though the throughput may not be guarantee.

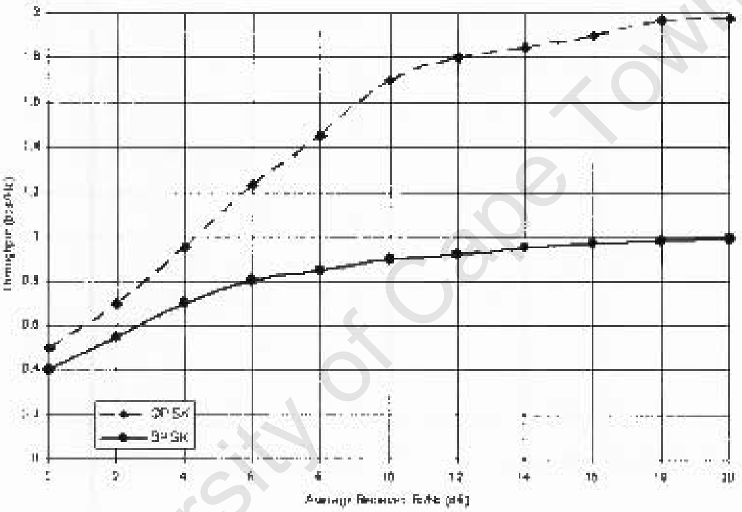

Figure 4.14: Impact of Modulation level

4.9.4 Impacts of Radio Resources on adaptive CDMA Algorithm

This Section evaluates the impacts of some radio parameters on the adaptive CDMA algorithm using simulations. The algorithms have been designed to guarantee a certain transmission QoS (here BER is set to be 10^{-3}) and to get as much throughput as possible. The algorithm assigns settings based on the instant channel situation. The impacts of some pa-

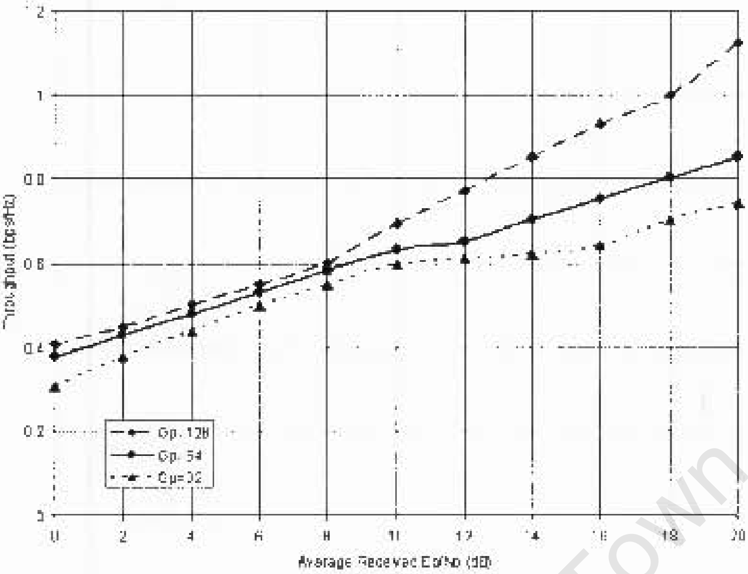

Figure 4.15: Impact of processing gain

rameters assigned by the algorithms during the simulation period are shown in Figures 4.14, 4.15 and 4.16. The Figures present the impacts of modulation levels, rate and the processing gain for the transmission respectively.

The number of bits that can be transmitted with each transmission can be increased with the modulation level, M. With BPSK and QPSK for instance, 1 and 2 bits can be transmitted over each symbol, respectively. The throughput in bps/Hz, defined as the ratio of the number of information bits transmitted successfully to the total number of bits transmitted and multiplied with the transmission rate normalized by the bandwidth, is plotted in Figure 4.14. It is observed that with the increase in modulation level, the throughput increases. Figure 4.15 plots the throughput for the CDMA scheme as a function of the average received E_b/N_0 with parameter G_p. It is observed that the throughput increases with an increase in G_p. Figure 4.16 plots the throughput for the CDMA scheme as a function of the average received E_b/N_0 with parameter R. Also, the throughput for the CDMA scheme is increased as the coding rate increased.

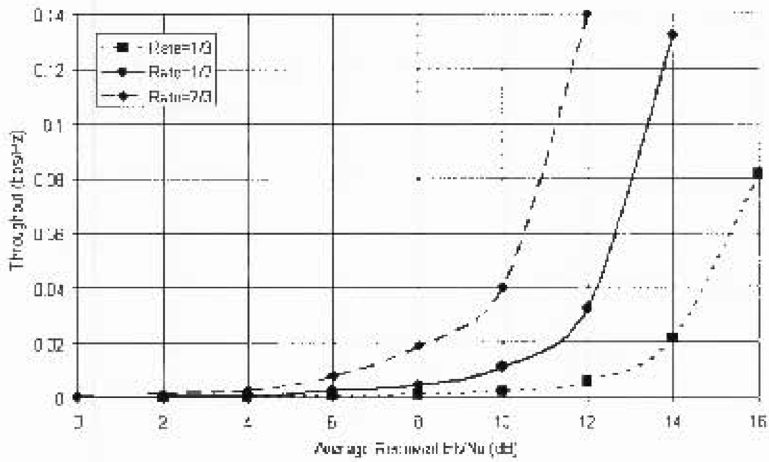

Figure 4.16: Impact of Rate

4.9.5 Percentage Usage of Radio Resources in adaptive CDMA System

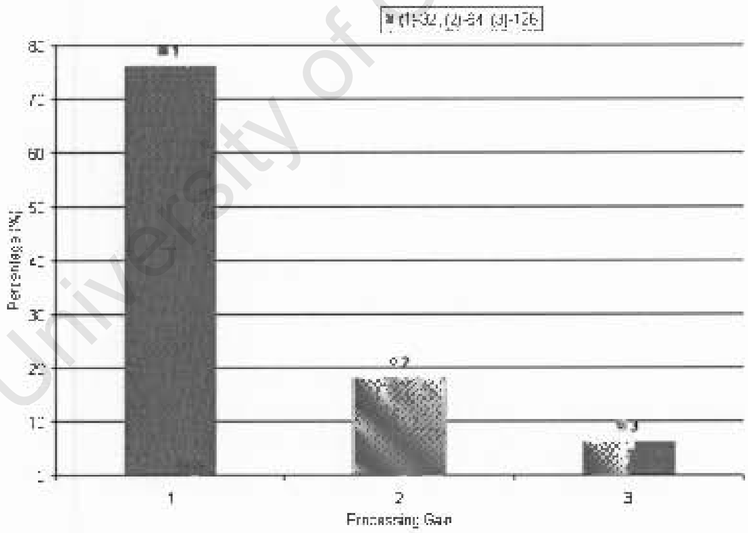

Figure 4.17: Percentage of processing gain used

The percentage contributions of the parameters and values in terms of their usage in adaptive CDMA system are measured. Figures 4.17, 4.18 and 4.19 shows the percentage of the

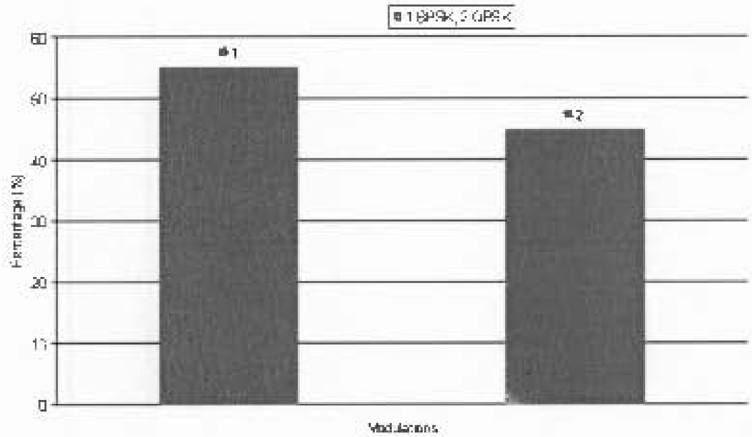

Figure 4.18: Percentage of modulation used

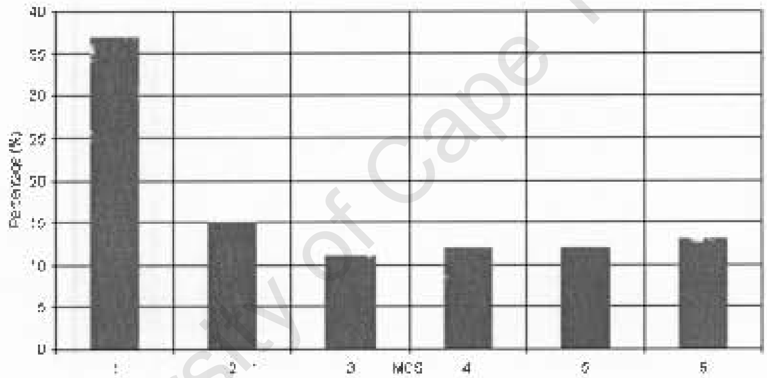

Figure 4.19: Percentage of each MCS used

usage of the processing gain G_p, modulation M, as well as modulation and coding schemes, MCS during the simulation. From Figure 4.17 it is found that our algorithm assigned a processing gain of 32 for 75.33% of the transmissions, a processing gain of 64 for 18.12% of the transmissions, and a processing gain of 128 for 6.55% of the transmissions. Figure 4.18 show the percentage of usage of each of the modulation levels during the simulation. BPSK was used for 56.31% and QPSK for 43.69% of the transmissions. Figure 4.19 presents the percentage of usage of each of the MCS. In our simulations, 36.87% of the transmissions exploited MCS1, 15.50% MCS2, 10.81% MCS3, 11.97% MCS4, 11.97% MCS5 and 12.88%

MCS6. All settings can change with the channel situation.

4.9.6 Performance Comparison of Fixed Radio Resources and Adaptive CDMA System

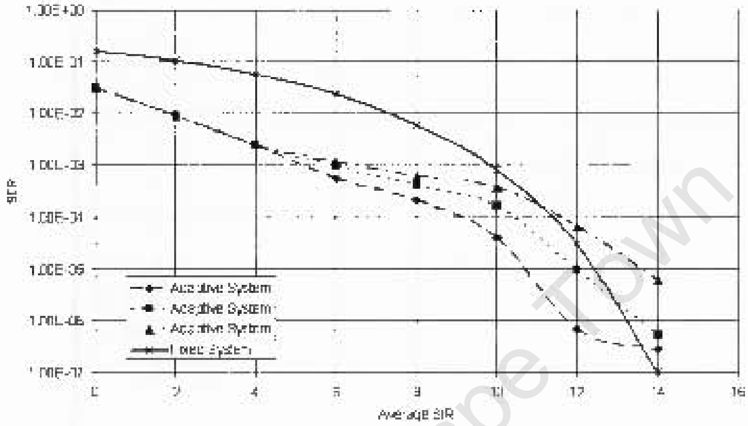

Figure 4.20: Comparison of Fixed and Adaptive System

For constant BER operation, the BER and the average normalized throughput against average reference SIR are shown in Figure 4.20 with different values of control parameter. The BER curves of the proposed scheme flatten when it reaches the adaptation range, achieving the constant BER purpose. This is because the switching thresholds of the proposed scheme make the instantaneous BER relatively insensitive to the variation of instantaneous SIR in the adaptation range. Different control value gives different adaptation range and different BER level at the flatten position. At high SIR, higher throughput is traded with lower than necessary BER. On the other hand, at low SIR, the BER is kept low at the expense of lower throughput. With this situation, adaptive is said to be robust to channel conditions and can achieve both performance and throughput at the same time.

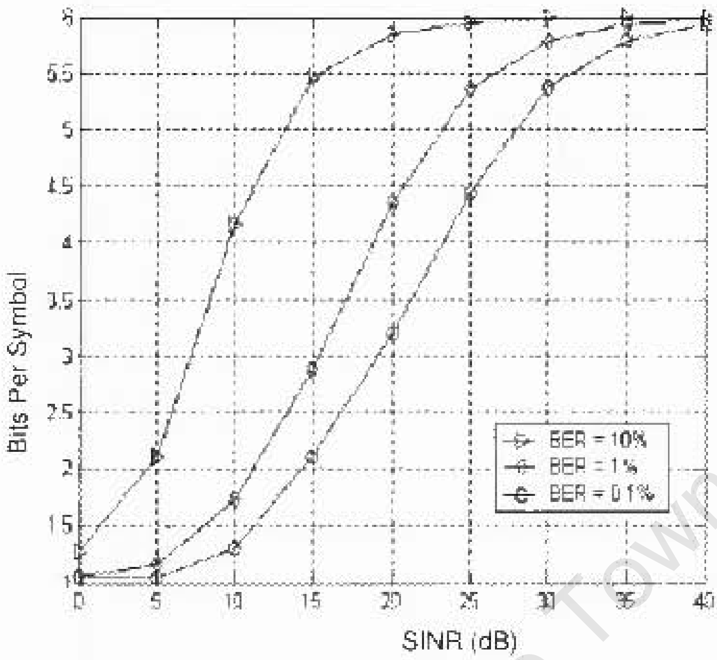

Figure 4.21: Spectral Efficiency of Ideal System at Different BER Target

4.9.7 Impacts of spectral efficiency on BER

In the previous section, the impact of spectral efficiency on channel quality is examined. In this section the impact of spectral efficiency on the performance of adaptive CDMA algorithm is examined directly. To examine this, three sets of switching levels corresponding to SINR for the modulation and coding schemes is assumed. These BER target for QAM are 0.1%, 1% and 10%. The SINR ranges corresponding to the three different BER targets have been described. The first thing to note about the BER performance and spectral efficiency of adaptive system is that no non-adaptive scheme shown provides better performance while simultaneously providing better spectral efficiency. In other words, AMC provides the best combination of energy and spectral efficiency of any of the modulation schemes. This is to be expected. While fixed schemes either achieve good spectral efficiency or good energy efficiency but not both, AMC increases spectral efficiency without sacrificing performance. Figure 4.21 shows the throughput performance of the three references BER for ideal situation. While Figure 4.22 shows the throughput performance of our algorithm. It is observed that

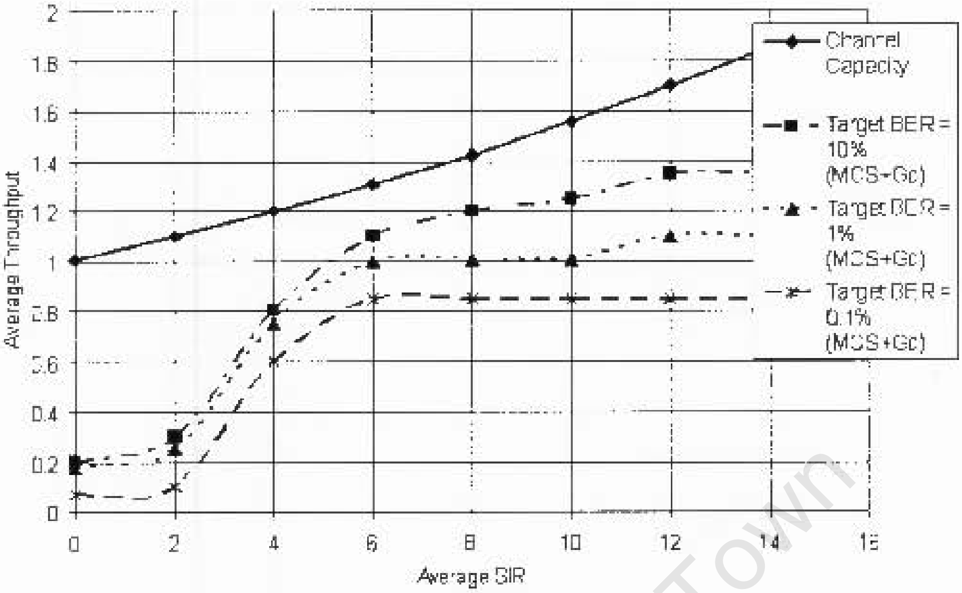

Figure 4.22: Comparing BER target on the Proposed System

as the target BER is increased, the spectral efficiency is also increased. Thus a trade off can be made between performance and spectral efficiency by changing the BER target and thus switching levels. That is $BER = 10^{-1}$ has better spectral characteristics than systems with $BER = 10^{-3}$ and $BER = 10^{-6}$ respectively.

4.10 Chapter Summary

In this Chapter, an adaptive algorithm for radio resource management is proposed. Adaptive CDMA algorithm for satellite channels is developed. These algorithms assign modulation and coding parameters in accordance with the instantaneous channel situations. The objective of these algorithms is to get as much throughput as possible under the condition of guaranteeing a certain transmission quality. These algorithms have been explained in detail and the simulation results have been given. The simulation results indicated that significant throughput can be obtained by the algorithms. The algorithm is developed with references to other existing adaptive algorithms. The impacts of some parameters on the algorithm have been investigated. The proposed adaptive CDMA algorithm for satellite systems is a resources-control algorithm. The difference between the algorithm and the other adaptive CDMA algorithm for satellite channels is that this algorithm is developed so that the processing gain is prioritised over the modulation and coding scheme. Moreover, the user parameter is employed for the downlink system. Unlike many algorithms, which adopt a 'no transmit' method when the transmit power is beyond the minimum threshold, in this work, user parameter is evoked whenever a 'critical condition' occurs. The critical condition is initiated in this thesis and it is defined as the situation when the transmit power is beyond the minimum threshold.

Chapter 5

Conclusions and Recommendations

In this thesis successful algorithms for adaptive CDMA modulation for Rice-lognormal channels have been proposed. These algorithms have been implemented in broadband direct spectrum - code division multiple access (DS-CDMA) satellite networks for the optimisation of scarcely available radio resources and for the realisation of next wireless generation in satellite system. Good results have been obtained in terms of QoS and throughput gains given a target BER of 10^{-3}. The diversity technique and rate compatible punctured turbo-code (RCPT), which have been found to give improved throughput performance in a direct sequence (DS) CDMA are exploited. Also, power control error (PCE) and the Rice K factor have been exploited at the channel settings. Beside the in-depth descriptions and identifications of the satellite network requirements, the following are the major activities that have been performed in this thesis.

The channel and system model have been presented. First, the channel characteristics were discussed based on its categorization, namely the large-scale and small-scale fading channels. Large-scale fading is mainly represented by the path loss, and is caused by long-distance propagation. Small-scale fading presents the effect of multipath transmission. Subsequently, channel model considered was briefly described. This model is based on the Rice-lognormal distribution. The influence of Rice K factor and power control error (PCE) on channel models were discussed and validated. It is found by simulation that as the K factor decreases, there is a degradation in the system performance when the PCE is held constant. Similarly, performance degradation occurs as the PCE is increased when the K factor is held constant.

The strategy and work procedures of adaptive modulation algorithms, which is unique have been outlined. It has been indicated that there are two aspects to the activity of adaptive modulation: the *modulation and coding scheme (MCS)*, and *channel*. The channel situation determines the adaptivity of the modulation parameters. It has been proposed in this thesis that investigating adaptive modulation is different from investigating traditional modulation. In traditional modulation, basic modulation technology is studied and combined with new technologies or modified for improvement. In adaptive modulation, it is prerequisite to study the relationships within parameters in the same domain and between the parameters in another domain. Therefore, to build an efficient adaptive modulation algorithm, it is required to first carry out the relationship study. Also, relationships among the CDMA modulation parameters are analyzed together with the parameters in MCS for the design of adaptive CDMA algorithms.

The development of adaptive staging and grouping has been put forward. It is indicated that the adaptivity is dependent on the number of available resources. It has also been said that to manage the radio resources efficiently, grouping of the available radio parameter is important.

Parameter of interest in coding and modulation systems is the throughput. This is a measure of the total information rate being successfully transmitted and received over the multiple-access network. Throughput formulas for CDMA systems have been derived and parameters influencing the throughput of these systems have been indicated. The throughput formula for CDMA systems in terms of bit rates indicate that the bit rate of CDMA systems is linearly proportional to the coding rate and transmission bandwidth, logarithmically proportional to the modulation level, and inversely proportional to the processing gain. Following the discussion of parameters influencing throughput, the parameters influencing QoS are investigated.

The basic structure and characteristics of the AMC systems have been described. Different categories of modulation systems have been described. Simulations, which investigate the gains of different adaptive schemes in terms of the throughput have been carried out. The influence of the throughput on BER is considered as well. A detailed description of the gains of AMC systems is given and compared to adaptive CDMA systems.

Algorithms for radio resource management have been proposed. Adaptive CDMA algorithms for satellite channels have been designed. These algorithms have been evaluated by simulations and good results have been obtained. The algorithms assign modulation parameters suitable for the instantaneous channel situation. Perfect channel estimation is assumed in our algorithms. The objective of these algorithms is to get as much throughput as possible under the condition of guaranteeing a certain transmission quality. In the adaptive CDMA algorithm for satellite channels, prioritised processing gains over MCS and a solution to 'critical section' at the downlink transmission have been introduced. Simulation results show that our algorithm achieves an improved spectral efficiency over the traditional AMC and adaptive CDMA for satellite channels.

Radio resources have been evaluated based on percentage of the parameters and values in term of their usage in adaptive CDMA system. This measurements is inevitable for the radio resources management. Thus the amount of the resources used during transmission and the energy saved could be determined.

In this thesis, a prioritized adaptive CDMA algorithm has been examined for DS-CDMA satellite networks. It is concluded that this proposed algorithm improves the throughput at different level of resources available to the system. Thus the algorithm that has been described has potential to improve reliability, availability, performance and robustness. These improvements are due to radio resource management algorithms. This optimised method of radio resource management has helped to increase capacities and improve performance of wireles systems.

5.1 Recommendations for Future Work

Adaptive modulation algorithms have been proposed successfully in this dissertation. The following recommendations are given for future work.

It should be noted that adaptive modulation and coding is a powerful technique to improve spectral efficiency over fading channels, and can also be used to meet the different delay, BER, and data rate requirements of different types of media. Shannon theory offers some valuable insights into good adaptive strategies, although these strategies do not always work well in practice. Much work remains in developing good adaptive strategies especially on satellite channels, for multi-access and broadcast systems. For multiuser systems adaptive modulation can be combined with other adaptive resource allocation policies such as dynamic channel and base station allocation. Adaptive joint source and channel coding strategies that combine adaptive compression with adaptive modulation may also lead to good performance in time-varying channels.

Channel estimation is an important part of any adaptive system. The channel information is the input for the adaptive scheme. However, almost all the work done on adaptive systems (including our work) assume perfect channel estimation at both transmitter and receiver. Some work has been presented in literature where channel estimation at the receiver side was considered. In this type of estimation, the transmitter gets the feedback of the channel information from the receiver. The need for the transmitter itself to estimate the channel is important because the channel state estimates at the transmitter can be different from the channel state estimates at the receiver. They can be modelled by their joint distribution with the true channel state information.

In this work as well as other related works, adaptive rate compatible puncturing turbo codes (RCPT) system has been carried out on both systematic bits and non-systematic bits. However, it has been mentioned that it would be better to puncture only the non-systematic bits to obtain a higher rate. The uniform random puncturing scheme saturates in a lower rate than the uncoded system at high signal-to-noise ratio (SNR) for a given target error probability, while the non-systematic random puncturing scheme approaches the uncoded scheme at high SNR. The performance of the non-systematic puncturing scheme, and/or a hybrid system with uniform random puncturing at low SNR and non-systematic random puncturing at high SNR ought to be investigated to determine which scheme to use for a particular situation. The adaptive RCPT can also be extended to fast fading, where one codeword spans multiple fading blocks, and compared to the non-adaptive scheme.

Further study is recommended on the influence of the Rician K and SNR on the relative gains of our adaptive algorithms. Correlation among the transmit and receive antennas will be an interesting factor to combine into the adaptive algorithms. Three techniques are indicated as solutions to problems in the time, frequency, and space domain. The degree of dependency of the three techniques in these domains is a subject for further study.

Appendix

A-1: Turbo Decoding Algorithm

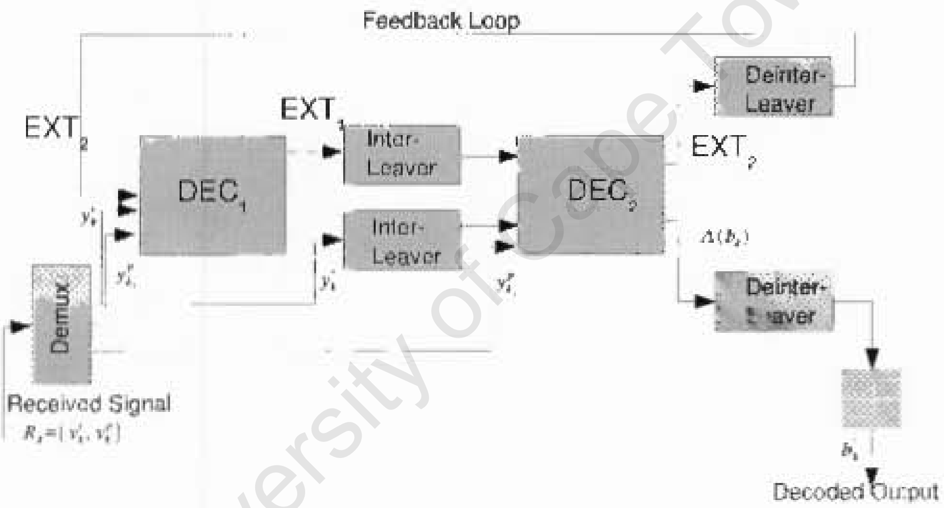

Figure 5.1: Turbo Decoding

The turbo decoder shown in Figure 5.1 inputs a sequence of code values $R_k = \{y_k^s, y_k^p\}$ from the demodulator. The turbo decoder DEC_1 decode sequences from ENC_1, and DEC_2 decode sequences from ENC_2. Maximum A Posteriori (MAP) is considered in each of the decoder. DEC_1 inputs sequence of systematic values y_k^s and the sequence of parity values $y_{k_1}^p$ of ENC_1. DEC_1 outputs sequence of soft estimates EXT_1 (*extrinsic information*) of the transmitted data bits b_k. The interleaver is applied and DEC_2 inputs the interleaved values y_k^s with the sequence of parity values $y_{k_2}^p$ from ENC_2 along with EXT_1, provided by

DEC_2. EXT_2 is estimated from the transmitted data sequence b_k. EXT_2 is then fed back to DEC_1 and the procedure is repeated in an iterative manner.

A-2: The MAP/Log-MAP Algorithm

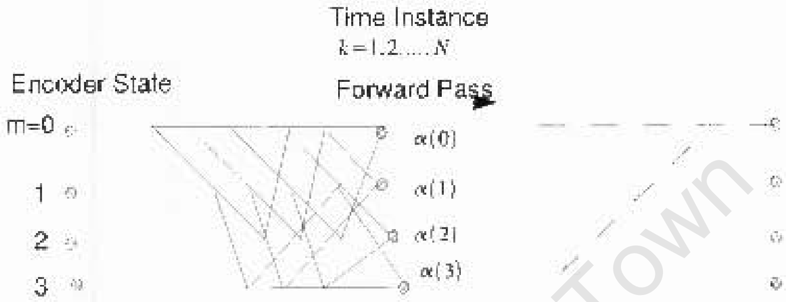

Figure 5.2: Forward Pass - α State Probabilities

The modified Bahl algorithm is commonly referred to as the Maximum A Posteriori (MAP algorithm), and achieves soft decision decoding by making two passes of a decoding trellis. Passes occurs in the forward direction, and in the backward direction, as shown in Figures 5.2 and 5.3 [75, 102].

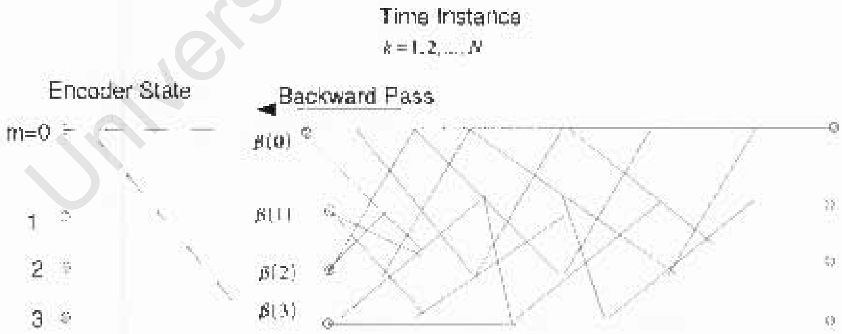

Figure 5.3: Backward Pass - β State Probabilities

Figure 5.4 illustrates the process further for the simple four state RSC code with trellis connectivity defined by the generator polynomial $G = \{7, 5\}$. The trellis is traversed in

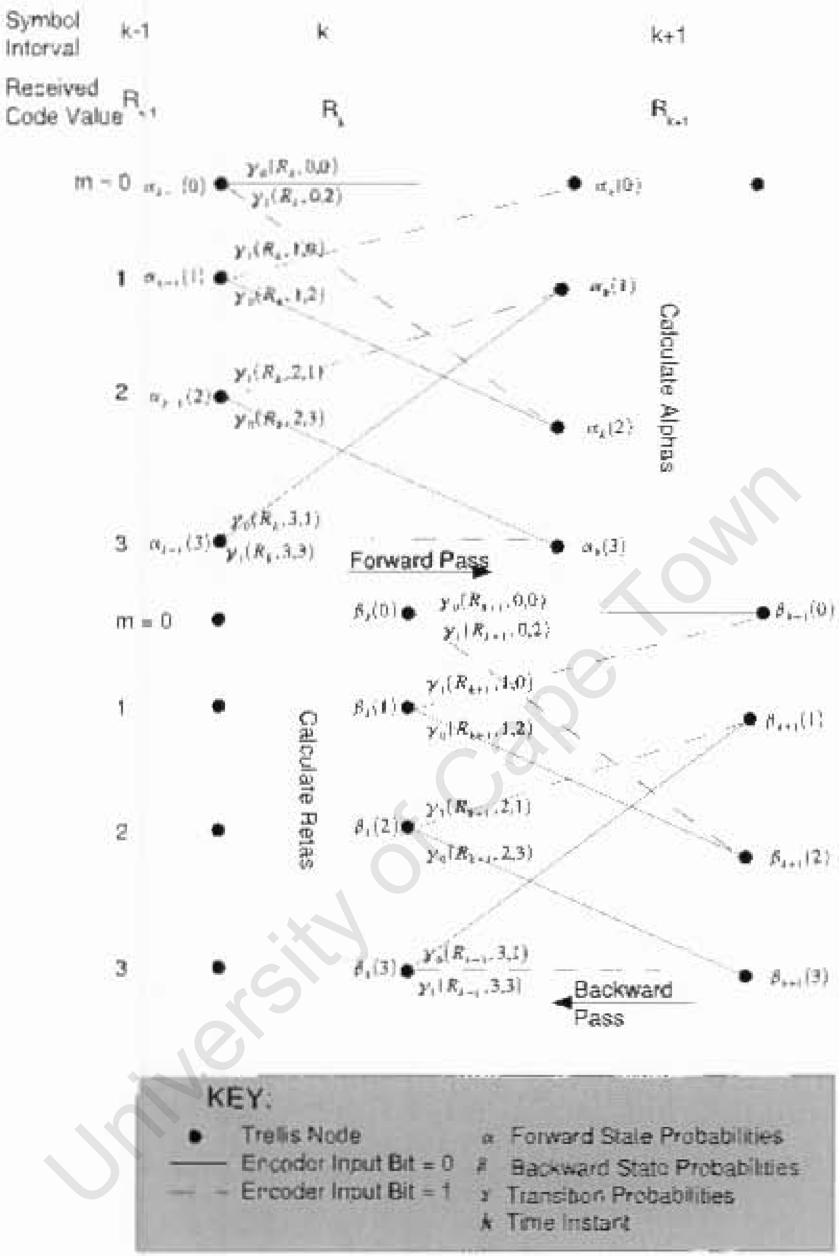

Figure 5.4: Illustration of State Probabilities

the forward direction. The operation at each state probability is carried out by multiplying the state probability at the previous node $\alpha_{k-1}(m')$ with the branch transition probability, $\gamma_{k-1}(m',m)$, given the received code pair $R_k = \{y_k^s, y_k^p\}$. This is expressed as follows [75, 102]:

$$\alpha(m) = \frac{\sum_{m'} \sum_{i=0}^{1} \gamma_i((y_k^s, y_k^p), m', m)\alpha_{k-1}(m')}{\sum_{m} \sum_{m'} \sum_{i=0}^{1} \gamma_i((y_k^s, y_k^p), m', m)\alpha_{k-1}(m')}, \tag{5.1}$$

where m is the current state, m' is the previous state and i is the data bit ('0' *or* '1') corresponding to each branch existing node.

Then the trellis is traversed in the reverse direction and expressed as follows:

$$\beta(m) = \frac{\sum_{m'} \sum_{i=0}^{1} \gamma_i((y_{k+1}^s, y_k^p), m', m)\beta_{k+1}(m')}{\sum_{m} \sum_{m'} \sum_{i=0}^{1} \gamma_i((y_{k+1}^s, y_{k+1}^p), m', m)\alpha_k(m')}, \tag{5.2}$$

The transition probability for each branch between nodes is given by the equation:

$$\gamma_i((y_k^s, y_k^p), m', m) = p((y_k^s, y_k^p)|b_k = i, m', m).q(b_k = i|m', m).\pi(m|m'), \tag{5.3}$$

A-3: The Log Likelihood Probabilities, $\Lambda(b_k)$

The soft estimate of the transmitted data bit b_k is represented as a log likelihood ratio (LLR), $\Lambda(b_k)$ as showned in Figure 5.1. It is calculated as follows [75, 102]:

$$\Lambda(b_k) = \ln \frac{\sum_{m} \sum_{m'} \gamma_1((y_k^s, y_k^p), m', m)\alpha_{k-1}(m')\beta_k(m)}{\sum_{m} \sum_{m'} \gamma_0((y_k^s, y_k^p), m', m)\alpha_{k-1}(m')\beta_k(m)}, \tag{5.4}$$

where $\Lambda(b_k)$ represents the probability that the current data bit is a '0' (if $\Lambda(b_k)$ is negative) or a '1' (if $\Lambda(b_k)$ is positive).

A-4: The Max Log-MAP Algorithm

MAP algorithm is not usually implemented for reasons of computational complexity. Hence there is a need to simplify the equation above using the logarithmic values rather than their actual probabilities. The logarithm of $\gamma_i((y_k^s, y_k^p), m', m)$ for the AWGN channel is taken [75, 102]:

$$p(y_k^p|b_k = i, m', m) = \frac{1}{\sqrt{\pi}N_o}.exp\left(-\frac{1}{N_o}[y_k^p - x_k^p(i, m', m)]^2\right), \tag{5.5}$$

$$p(y_k^s|b_k = i) = \frac{1}{\sqrt{\pi N_o}}.exp\left(-\frac{1}{N_o}[y_k^s - x_k^s(i)]^2\right), \tag{5.6}$$

where x_k^s and x_k^p are the transmitted systematic and parity bits, respectively, corresponding to the current branch at time instant k. y_k^s and y_k^p are the received systematic and parity values, and N_o is the noise power spectral density. The logarithm of the equation for $q(.) = 1$ can be written as [75, 102]:

$$\ln \gamma_i[(y_k^s, y_k^p), m', m)] = \frac{2y_k^s x_k^s(i)}{N_o} + \frac{2y_k^p x_k^p(i)}{N_o} + \ln Pr(m|m') + K. \tag{5.7}$$

Then

$$\ln \alpha_k(m) = \ln\left[\sum_{m}^{'}\sum_{i=0}^{1} exp(\ln \gamma_i[(y_k^s, y_k^p), m', m)] + \ln \alpha_{k-1}(m'))\right]$$

$$-\ln\left[\sum_{m}\sum_{m}^{'}\sum_{i=0}^{1} exp(\ln \gamma_i[(y_k^s, y_k^p), m', m)] + \ln \alpha_{k-1}(m'))\right], \tag{5.8}$$

The multiplications terms have been reduced to exponential terms but this is also computationally intensive. However, we can approximate the sum of a series of log terms by considering only the maximum log value [75, 102]:

$$\ln[exp(\xi_1) + \cdots + exp(\xi_n)] \approx max_{i\in\{1..n\}}\xi_i. \tag{5.9}$$

This is simplify as:

$$\overline{\alpha}_k(m) \approx max_{(m',i)}\{\overline{\gamma}_i[(y_k^s, y_k^p), m', m)] + \overline{\alpha}_{k-1}(m')\}$$

$$-max_{(m,m',i)}\{\overline{\gamma}_i[(y_k^s, y_k^p), m', m)] + \overline{\alpha}_{k-1}(m')\} \tag{5.10}$$

and

$$\overline{\beta}_k(m) \approx max_{(m',i)}\{\overline{\beta}_i[(y_{k+1}^s, y_{k+1}^p), m, m')] + \overline{\beta}_{k+1}(m')\}$$

$$-max_{(m,m',i)}\{\overline{\beta}_i[(y_{k+1}^s, y_{k+1}^p), m, m')] - \overline{\alpha}_k(m)\} \tag{5.11}$$

The log-likelihood probability of each bit, $\Lambda(b_k)$, is then given approximately by:

$$\Lambda(b_k) \approx max_{(m,m')}\{\overline{\gamma}_1[(y_k^s, y_k^p), m', m)] + \overline{\alpha}_{k-1}(m') + \overline{\beta}_k(m)\}$$

$$-max_{(m,m')}\{\overline{\gamma}_0[(y_k^s, y_k^p), m', m)] + \overline{\alpha}_{k-1}(m') + \overline{\beta}_k(m)\}. \tag{5.12}$$

Now, $\Lambda(b_k)$ is divided into three components to allow iterative decoding to occur, the *extrinsic*, the *a priori*, and the *systematic* component.

By separating the components of $\Lambda(b_k)$, we have:

$$\overline{\gamma}_i'(y_k^p|m',m) = \ln p(y_k^p|b_k = i, m, m') + \ln q(b_k = i|m, m') \tag{5.13}$$

Further, we obtain:

$$\begin{aligned}\Lambda(b_k) \approx \{max_{(m,m')}\{\overline{\gamma}_1'[(y_k^s, y_k^p), m', m)] + \overline{\alpha}_{k-1}(m') + \overline{\beta}_k(m)\\ + \ln p(y_k^s|b_k = 1) + \ln Pr(b_k = 1)\}\\ -\{max_{(m,m')}\{\overline{\gamma}_0'[(y_k^s, y_k^p), m', m)] + \overline{\alpha}_{k-1}(m') + \overline{\beta}_k(m)\\ + \ln p(y_k^s|b_k = 0) + \ln Pr(b_k = 0)\}\end{aligned} \tag{5.14}$$

Finally, this can be arranged as:

$$\begin{aligned}\Lambda(b_k) \approx max_{(m,m')}\{\overline{\gamma}_1'[(y_k^p), m', m)] + \overline{\alpha}_{k-1}(m') + \overline{\beta}_k(m)\}\\ -\{max_{(m,m')}\{\overline{\gamma}_0'[(y_k^p), m', m) + \overline{\alpha}_{k-1}(m') + \beta_k(m)\}\\ + \frac{4y_k^s}{N_o} + L(b_k).\end{aligned} \tag{5.15}$$

The first two terms form the extrinsic component, the third term is the systematic component, and the fourth is the a priori component. It is the extrinsic component which is passed from one MAP decoder to the other, where, after interleaving, it forms the a priori input term of the new decoder [75, 102].

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