

**COMPARISON OF CALL DROP MINIMIZATION TECHNIQUES IN
WCDMA WIRELESS COMMUNICATION NETWORK**

BY

NDIHIBE, PATRICK IFEANYI

REG. NO: PG (UD) M.ENG/06/49

**DEPARTMENT OF ELECTRONICENGINEERING
UNIVERSITY OF NSUKKA**

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APPROVAL PAGE

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**A THESIS SUBMITTED IN PARTIAL FULFILLMENT OF THE
REQUIREMENT FOR THE AWARD OF DEGREE OF MASTER OF
ENGINEERING (M.ENG) IN THE DEPARTMENT OF ELECTRONIC
ENGINEERING, TELECOMMUNICATION OPTION, UNIVERSITY OF
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CERTIFICATION

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I, **NDIHIBE PATRICK IFEANYI**, a postgraduate student in the Department of Electronic Engineering, University of Nigeria, Nsukka declare that the work imbibed in this thesis is original and has not been submitted by me in any part or in full for any other diploma or degree of this or any other institution.

DEDICATION

This work is dedicated to God Almighty, the author and finisher of my life, my amicable wife **Mrs. Amaka A. Ndihibe** and my parents **Chief P.O.C. Ndihibe** and **Mrs. C.C. Ndihibe**.

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Ndihibe, Patrick Ifeanyi

February, 2016

ABSTRACT

One of the research challenges in cellular networks is the design of an efficient model that can reduce callblocking probability and improve the quality of service (QoS) provided to mobile users. One most important parameter for assessing the quality of service (QoS) in a network is the probability of dropped calls, which has so far not been deeply studied within the context of established cellular networks. Call/packet dropping refers to the event described as the termination of calls in progress before either involved party intentionally ends the call. There are numerous drop call causes in cellular networks with majority of them occurring in the Um interfaces mainly due to lack of radio resources created by electromagnetic causes and user mobility (i.e. handover). Another important contributor of drop call rate is the traffic load in which, the call arrival rate and holding time play significant roles. Drop call probability is defined as the probability that a call is terminated due to one or all of the above-mentioned causes and is basically estimated from drop call rate by applying the Poisson probability distribution function. This work performs deep analysis and evaluates the comprehensive differences between the various techniques employ to reduce the effect of call drop algorithms in Wireless Code Division Multiple Access (WCDMA) networks. MATLAB simulation model of a simple WCDMA system with radio resource algorithms is implemented and analyzed to make a clear comparison between the algorithms in terms of outage probability, bit error rate (BER) and channel capacity which does not have an exact limit in WCDMA technique. The results showed that the hard handoff scheme offers a better outage probability amongst the call drop minimization schemes considered and the implementation possibility of all the schemes depends on their individual performances and network demands.

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CHAPTER ONE

INTRODUCTION

1.1 Background of Study

The drop-call probability is one of the key performance indicators (KPI) used by various mobile phone service operators for measuring quality of service (QoS) [1]. Its minimization is the typical result of many optimization procedures applied to several service aspects as: the maximization of service coverage area and of network usage, the minimization of interference and congestion, the optimum traffic balancing among the different frequency layers (e.g., 900 and 1800 MHz in the European GSM standard) [2].

Call drop generally refers to the phenomenon of call/packet dropping in both voice and data networks. Call/packet dropping refers to the event described as the termination of calls in progress before either involved party intentionally ends the call [3]. Call drop-out is a measure of the ability of a mobile network to maintain a call until it is terminated [3]. The call drop-out rate is measured as the amount of call drop-outs as a percentage of calls. The lower the percentage, the better is the QoS.

Wireless networks involve radio and wire-line links as well as switching hardware and software, and data base operations. However, call drop rate is mainly determined by the radio resources in the network. These resources translate mainly into the plethora of radio channels all of which share a common bandwidth. One reason for call dropping is when

the mobile user moves out of a wireless network range. An active call cannot usually be sustained across a different company's network (as it is impossible for calls to re-route themselves over the traditional network system while in progress) [4]. This scenario results in the termination of the call due to inability to sustain the signal between the mobile user and the original network. Another common reason is when a mobile user enters an area that has no wireless signal presence, interrupted, interfered with, or jammed. Judging from a network perspective, this is similar to leaving the coverage area. Occasionally calls are dropped upon handoff – between cells, within the same provider's network [1, 2]. One possible reason is a traffic imbalance between two cell sites when the new cell site is at capacity and cannot accept additional traffic from the call trying to "hand in". Another reason is wrong network configuration, which renders one cell site "unaware" of the cell the mobile user is attempting to hand off to. If the mobile phone cannot find an alternative cell to provide the handshake, then the call is lost. Co-channel and adjacent channel interference can also be responsible for dropped calls in a wireless network [3, 4]. Neighbouring cells with same frequencies may interfere with each other to deteriorate the quality of service and cause dropped calls. However, the reduction in co-channel interference depends on the amount of sectoring a cell uses [4]. A cell is usually partitioned into three 120° sectors or six 60° sectors and as many as six sectors per cell have been used in practical network services. Transmission problems are also a common cause of dropped calls – resulting from a faulty transceiver (TRX) within the base station [4]. At the receiver's end, calls may be dropped if a mobile phone loses battery power and abruptly stops transmitting. Sun spots and solar flares are rarely

blamed for causing interference that leads to dropped calls [5]. The increasing demand of cellular network services with the corresponding increase in network resources constitutes another source of call drop. This capacity related issue has forced network operators to struggle in order to provide stringent QoS to network users. Experiencing frequent call drop is one of the most common customer complaints received by wireless service providers. The providers have attempted to address the complaint in various ways, including expansion of the customers' network coverage, increased cell capacity, offering compensations to individual dropped calls, etc. This research work presents a comparative analysis of call drop minimization techniques in cellular networks.

1.2 Statement of Problem

Presently in Nigeria, the service quality of some of the mobile wireless network operators has been following a deteriorating trend with the increase in the number of subscribers. There has been an increase in the complaints by mobile users on poor service quality, frequent call drops, echo during radio conversation, poor interconnectivity to and from licensed networks, distortions, network congestions, among other factors. As a result, customer's dissatisfaction is increasing, which in turn leads to low customers loyalty and high churn rate. In some cases, operators are implementing new features in their networks without performing proper network planning which is also a reason for poor QoS.

Hence, the need to provide solution to the enumerated problems necessitated this current research work which aims at performing comparative evaluation of some of the

techniques employed to ameliorate the effects of call drops and improve the end users satisfaction rate with the good quality and grade of service.

1.3 Aim and Objectives

The main aim of this research work is to carry out performance analysis and comparatively evaluate the techniques used to minimize call drops in Wideband Code Division Multiple Access (WCDMA) wireless communication systems. The objectives of this work include;

- To determine the causes of call drops and its effects in a WCDMA wireless communication system
- To evaluate the techniques used to minimize the call drops, these include power control scheme, hard handoff and soft handoff schemes
- To perform comparative analysis of these call drops minimization techniques using simulation
- To make recommendations for planning and optimizations

1.4 Significance of the Study

The delivering of both good quality of service (QoS) and Grade of Service (GoS) in any competitive mobile communication environment is a major factor to reducing subscribers' churn rate. Therefore, it is important for wireless mobile network operators to ensure stability and efficiency by delivering a consistent, reliable and high-quality end user (subscriber) satisfaction. This can only be achieved by conducting a regular network

performance monitoring and optimization as it directly impacts the quality of the offered services and hence user satisfaction. The outcome of this research will help the network operators to know and select the best possible method of reducing call drops in their network and hence improve their quality of service.

1.5 Scope of Study

The fact that in a country like Nigeria, the overwhelming improved characteristics of WCDMA systems has made most new major service providers to adopt the CDMA technique. This research work focused mainly on enhancing the performance of WCDMA networks. The scope of this research work is limited to the comparative analysis of the techniques used in the minimization of call drops in a WCDMA wireless communication system. The theory and technical features of WCDMA, its advantages and performance improvement techniques on a general basis were also not left out in this work. The focus is to carry simulation evaluation of some of the existing techniques and to use the result to make constructive judgment on the best possible technique to be employed by the operators.

1.6 Organization of the work

This research work is systematically presented in five chapters. Chapter One introduces this research by elaborating on the background, statement of problem, objectives, justification of the research and the scope of the research.

Chapter Two presents different related works on ways of minimizing call drops in a wireless communication networks.

Chapter Three presents the research methodology. It contains materials and theories used in the realization of the objectives of the research.

Chapter Four presents the system implementation and analysis; the evaluation is done through simulations

Chapter Five is the conclusion and recommendation. It contains summary of achievement, the recommendations for further research and contribution to knowledge.

CHAPTER TWO

LITERATURE REVIEW

2.1 Review of Related Works

The main aim of this thesis is to study different schemes to maximize system capacity and minimize call drops in a WCDMA network. Call drop is one of the most disturbing problems in mobile communications, it is the most important Quality of Service (QoS) indicator for a mobile carrier, hence over the years, several proposed models or techniques for reducing call blocking probability exist in literature either for reducing new call blocking, or reducing handoff blocking or reducing both [1]. Whatever the case, there is a tradeoff between reducing handoff probability and new call blocking probability.

In their work in [1], the researchers x-rayed the numerous drop call causes in cellular networks with majority of them occurring in the Um interfaces mainly due to lack of radio resources created by electromagnetic causes and user mobility (i.e. handover). According to the researchers, another important contributor of drop call rate is the traffic load in which, the call arrival rate and holding time play significant roles. The researchers show that available radio resources and their utilization as well as traffic

intensity related parameters play significant roles in call drops. More work has been carried out to discover the trends of these drop-call probability factors in a live GSM network. Starting with data obtained from the network, mean values were computed for the number of channels, utilization factor, and then the call arrival rate and call duration. At least, their results show a general trend that is in line with the basic principles. The result can be used as a good guide for evaluating and optimizing an operating GSM network. In this case, the results show the network under study as a well established network. The result can further be used for drop-call probability model simulation for further confirmation.

In [6], the M/G/∞ queuing model is used to assess the uplink capacity of CDMA networks. The researchers also present a technique for computing the outage probability of the network. These classical papers have focused on rigid traffic models that do not consider elastic or best effort traffic, whose bit rate dynamically changes. The importance of modeling call dropping and its impact on the Erlang capacity in cellular networks in general has been emphasized by several authors [1, 6, 7]. An analytical approach to modeling single cell (including single cell clusters) is presented [7]. The researchers use statistical estimates from the Visafone telecommunication company operating in Italy to predict call dropping. They also show numerically (without simulation) how the traffic parameters affect the system's performance.

A dropped call probability model, derived in an experimental context is proposed [8]. The researchers improve on the traffic model in [7] and extend the model to capture the

impact of multicells on CDMA networks. The researchers also perform simulation of the proposed model to carefully examine the traffic behaviour and how they affect the system's performance. The traffic data is exploited in two ways. First, a study (and collection) of daily call drop statistics, including the cell index base station and dropped call rate is carried out. Second, a formulation of an analytical model from a theoretical and practical point of view is made. Furthermore, studying call dropping behaviours as a function of other network parameters (e.g. traffic load, call duration, etc.) would aid the optimization of the system's performance and guarantee excellent quality of service delivery as well as improved revenue. Concerning the tradeoff between call dropping and the network traffic, the authors discovered that the traffic demand in the network exceeded the available network capacity and this phenomenon occurs when the network at capacity or nearly full.

The researchers presented an empirical case study on the focus of the performance evaluation and optimization of GSM mobile network in Asaba Region, Nigeria [9]. The researchers specifically employ a combination of essential KPIs such as dropped call rate, call setup success rate and outage call rate to examine overall QoS and GoS performance of the investigated GSM network. The results after network optimization showed significant performance improvement in terms of call drop rate, call set up success rate, and call block rate across. Sachs et al. presented a framework for integrating different access systems into a multi-access system and for selecting the best suitable access network for users [10]. They proposed a utility-based approach for evaluating different

access allocation choices, which is based on user and network policies, the performance of access bearers, and the availability of access resources. Hasib and Fapojuwo presented an analysis of common radio resource management scheme for end-to-end QoS support in heterogeneous wireless networks [11]. Gelabert et al. presented a Markovian approach to Radio Access Technology (RAT) selection in heterogeneous wireless networks [12]. They developed an analytical model for RAT selection algorithms in a heterogeneous wireless network comprising GSM/EDGE and UMTS. The proposed algorithm selects just one RAT for each call. Song and Zhuang proposed a load-sharing call admission control scheme for voice and data services in an integrated cellular/WLAN network [13]. In the proposed scheme, voice calls are preferably admitted to the cellular network, whereas data calls are admitted into both cellular and WLAN.

According to Lee, blocking probability, dropping probability, blocking and dropping in different types of traffic, force termination probability required grade of service (GoS), uplink and Downlink load factors, effective bandwidth (BW) allocation among users, load control active connection and others, these factors increase service quality and come under admission control (AC) based on network planning and power control [14]. A problem in wireless communication is limitedly available radio-frequency spectrum. The intelligent base station assigns resources to mobile stations intelligently for power control and power adjustment to tackle the near far problem. In addition, it makes smart handoffs between base stations. Traffic distribution parameter includes user mobility, cell size of the wireless link, cell capacity, network load, scheduling at the base stations,

handoff, and location management. To maintain quality of service (QoS) the traditional approach based on the Erlang-B formula given by Erlang a Danish pioneer of tele-traffic theory in telecommunication networks.

Researchers compute call blocking and dropping probabilities based on Complete Sharing (CS) policy [15]. Next generation cellular systems demand of multimedia services like audio, video conference, Web service with guarantees high data rates and Quality of Service (QoS). To have such high data rates, new broadband wireless systems have introduced. For example, the 3rd Generation Partnership Project (3GPP) introduces High Speed Downlink Packet Access (HSDPA) in (3.5G specifications of Release 5) cellular system, which is extension to 3G Universal Mobile Telecommunication System (UMTS). The data rate of HSDPA theoretically support up to 14.4 Mbps, which is seven times more than data rate offered by the 3G Universal Mobile Telecommunications System (UMTS). HSDPA improve the WCDMA performance for downlink packet traffic. The multimedia demand for bandwidth in wireless networks is much more so it is important to allocate system bandwidth efficiently among connections with different QoS requirements.

This work, presents a comparative overview of drop-call minimization techniques in cellular networks. Cellular system operators want best system utilization so as to increase the number of users, revenue and later want to receive better QoS. The important goal of QoS is to reduce call drop this can be achieved by reserve bandwidth for handoff mobile users, efficient Call Admission Control (CAC) and Bandwidth Reservation (BR)

schemes, cell sectorization, radio resource management, various diversity schemes, equalization techniques and so many more.

Furthermore, studying call dropping behaviours as a function of other network parameters (e.g. traffic load, call duration, etc.) would aid the optimization of the system's performance and guarantee excellent quality of service delivery as well as improved revenue. Also, network optimization engineers should make effort on regular basis to improve the service quality and capacity of operational networks, as well as developing and deploying new ones in order to meet customer demands. Moreover, carrying out regular optimization remain essentially the only way to keep track of the network by looking deep into network operation statistics and collecting/analyzing drive test data.

2.2 Existing Methods for Preventing Call Drops

Quality of Service (QoS) provisioning in wireless networks is a challenging problem due to the scarcity of wireless resources and to support various integrated services with a certain QoS requirement in wireless network, resource provisioning is a major issue. Radio Resource Management (*RRM*) plays a major role in Quality of Service (*QoS*) provisioning for wireless cellular networks. Specifically, *RRM* techniques along with the network planning and air interface design determine the *QoS* performance at both the mobile user's level and the network level.

RRM policies are composed of frequency and time channels, transmit power, and access to base stations in order to efficiently control the allocated resources to different mobile users with the objective of maximizing a specific function such as network resource utilization, or total network revenue, subject to some constraints such as the maximum call blocking and dropping rates, and/or the minimum signal to interference ratio. The efficient design of *RRM* techniques has a direct impact on the performance of the mobile users in general and on the overall network performance in particular. For instance, the allocated transmission power for a mobile user has a positive effect on the *QoS* for that user, but it negatively affects the interference level for other mobile users which as a result degrades their *QoS* level. *RRM* strategies are not subject to the standardization process, so that they can be a differentiation feature among equipment vendors and operators.

Different *RRM* techniques are proposed for managing the radio resources for wireless communication systems. These techniques can be classified into three sets: Frequency and time allocation, which concerns channel allocation, scheduling, and bandwidth reservation schemes; power allocation, rate allocation and congestion control, which manage the allocation of mobile users' transmission powers and their allocated transmission rates, and reacts to system congestion scenarios; call admission control (*CAC*), base station assignment, and handoff algorithms, which handle the mobile users' access to the wireless communication system [16]. These different sets interact together as shown in Figure 2.1 to maintain the mobile users' provisioned *QoS*

parameters, taking the network loading conditions, network revenue, and other network performance objectives into consideration.

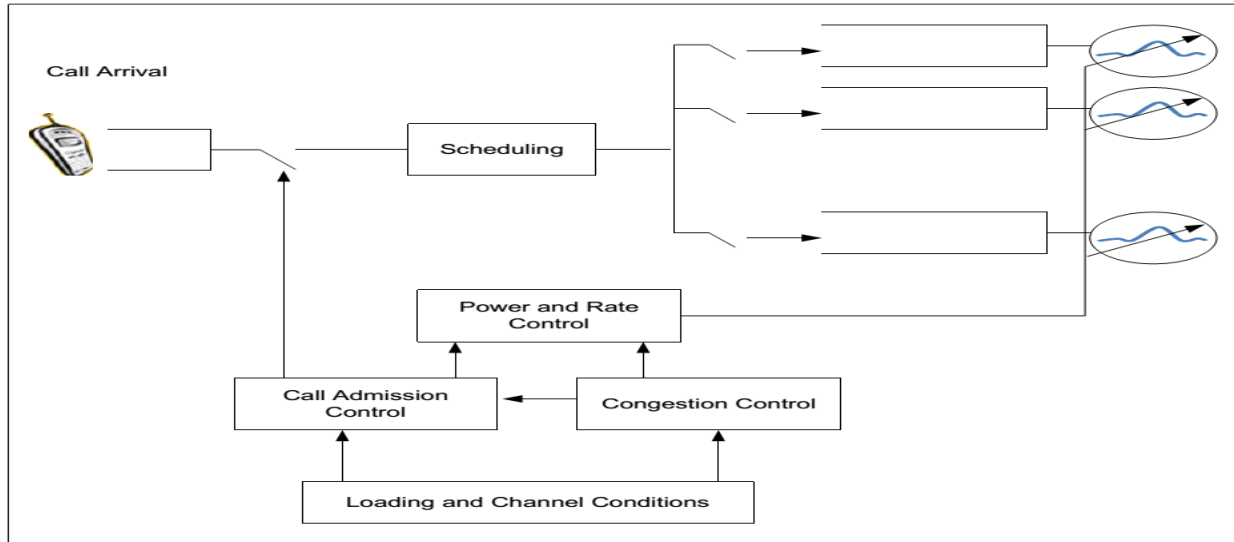


Figure 2.1: Components of Radio Resource Management [16]

2.2.1 Call Admission Control

From the network perspective, admitting a call (new or handoff) has a reward. This reward comes from utilizing the network resources for a certain amount of revenue. However, for such a reward there is also a potential penalty, particularly at high congestion levels, which can be translated as degradation in the offered QoS level to the existing mobile users' calls. Therefore, a balanced network state, which maximizes resource utilization and revenue while maintaining the provisioned QoS parameters is the objective of CAC schemes. CAC has been widely investigated in wireline networks as a QoS provisioning and congestion control tool. Different approaches of CAC design and performance analysis, particularly in the context of broadband integrated service digital network ($B-ISDN$) based on asynchronous transfer mode (ATM) technology, have

been investigated [17]. However, *CAC* in wireless networks is more challenging due to the unique characteristics of wireless networks such as wireless channel quality variation, handoff requirements, and bandwidth limitations. Due to the growing popularity of wireless communications, *CAC* has been receiving special attention during the last two decades since it plays a central role in *QoS* provisioning in terms of signal quality, call blocking and dropping probabilities, packet delay and loss rate, transmission power, and transmission rate. In first and second generation wireless cellular systems, *CAC* schemes have been generally developed for a single class of service which mainly considers voice application [18]. However, the introduction of data services and multimedia services in 2.5G and 3G wireless systems, respectively has necessitated the development of more sophisticated *CAC* schemes to cope with the stringent *QoS* requirements of these applications. Multi-class *CAC* schemes are more challenging than single-class *CAC* schemes due to the need for considering service prioritization, fairness, and resource sharing policies [19]. *CAC* schemes can be classified as centralized or distributed. Each of these *CAC* design options has its advantages and disadvantages. For example, centralized *CAC* schemes are implemented by a central network unit such as the Mobile Switching Center (*MSC*) in 1G and 2G cellular systems or the *RNC* in 3G cellular systems. This central unit manages the admission policy in the whole network. Centralized *CAC* schemes are more efficient than distributed *CAC* schemes due to the global information availability but their complexity is high which prevents their use in real systems. Distributed *CAC* schemes are simple, scalable, and more reliable since they are implemented in each network base station. The amount of available information for

distributed *CAC* schemes is limited when information exchange between network base stations is not enabled. Various algorithms and schemes have been developed for *CAC*. The purpose of an admission control algorithm is to decide at the time of call arrival, whether or not a new call should be admitted into the network. Some of *CAC* schemes are:

(a) Guard Channel Schemes

In this scheme, a portion of channel is reserved for handoff calls and whenever a channel is released, it is returned to the common pool of channels [20]. Whenever a mobile user travels from one area of coverage or cell to another cell within a call's duration the call should be transferred to the new cell's base station. Otherwise, the call will be dropped because the link with the current base station becomes too weak as the mobile recedes. This seamless ability of transference from one base station to another during the duration of a call is known as handoff procedure [21]. During handoff, the user needs to release the currently allocated channel and acquire a channel from another base station without the call being dropped. The base station without Guard Channel Schemes may not have channels available, leading to call drops. Guard Channel Schemes make sure that a portion of channels are reserved for handoff calls and call drops do not occur during handoff requests.

The main drawback in this strategy is that it increases the new call blocking probability and under-utilizes the radio resources, far out-weighting the reductions it achieved in handoff blocking probabilities. In other words, the cost of the classical method of

employing guard channels to decrease the call dropping rate is the increase in the call blocking rate. Since subscriber mobility changes in time, the number of handoff attempts in each cell is subject to fluctuations, making static assignments (or periodic update) of a given number of guard channels inefficient. Therefore, only slight gains in the utilization can be achieved by this scheme. Furthermore, it does not give any thought as to whether any handoff channel are required at any given moment or if so, by how many.

(b) Queuing Priority Scheme (QPS)

The handoff call is queued if there is no available channel in destination cell when MS send its request to BS. If a channel in destination cell is unavailable before crossing handoff area then call termination occur. If there is number of calls in queue then system follow First in First out (FIFO) [22]. Queuing of handoff requests when there is no available channel can reduce the dropping probability but at the expense of higher new call blocking. If the handoff attempt finds all the channels in the target cell occupied it can be queued. If any channel is released it is assigned to the next handoff waiting in the queue and not to a new call.

(c) Reserved Channel Scheme (RCS)

RCS dynamically gives priority to every function. Functions like new call attempts and handoffs channel allocations. In normal condition, highest priority is given for handoffs calls and then for new call attempts, however it is possible to set your own priority as system requirements. Handoffs reserved channels uses fixed channel allocation (FCA)

scheme and remaining channels using dynamic channel allocation (DCA) scheme, help the network to reduce probabilities of new call blocking probability, handoff-dropping probability, call forced termination probability and, call not completed probability. An increase in system performance has been observed by varying threshold value to optimize the parameters in different traffic conditions [22]. Figure 2.2 shows the flow chart used to implement a prioritized reserved channel scheme

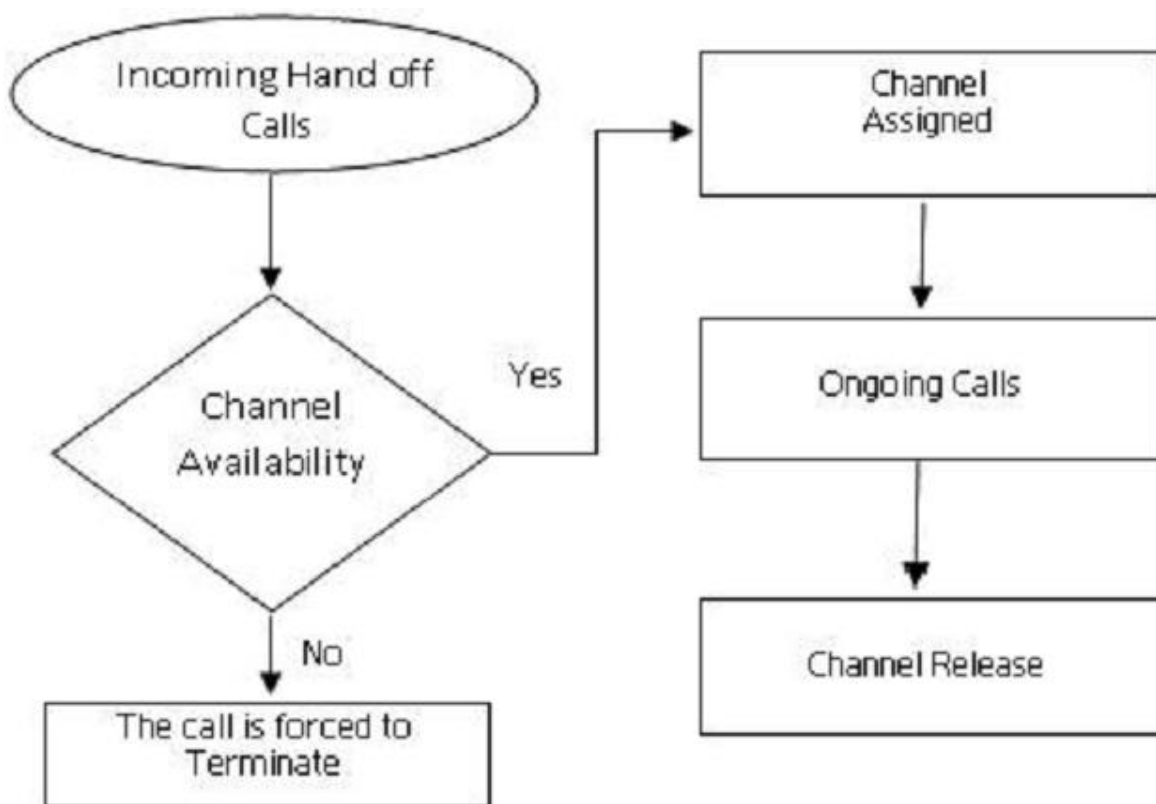


Figure 2.2: Flow chart of prioritized reserved Channel Scheme (PRCS) [22]

2.2.2 Bandwidth Reservation Scheme

Any bandwidth reservation scheme provides guarantees QoS and low Call Dropping Probability (CDP) due to fixed bandwidth allocation exclusively for handoffs, thus

priority given over new connections. However, it seriously decreases Bandwidth Utilization (BU) and cause high Call Blocking Probability (CBP) for new calls. For each service, the QoS targets have to be set and naturally met. QoS specified by speech coverage and blocking probability only [23]. As, [15] proposed bandwidth reservation scheme reduces Call Dropping Probability (CDP), Call Blocking Probability (CBP), and improves Bandwidth Utilization (BU) at the same time. The researchers consider two types of traffic in system; one is new calls and other handoff calls these further divided in to four traffic categories: class-1 handoff, class-2 handoff, class-1 new call, and class-2 new call. To create a balance between CDP and CBP to improve BU in a cell, the priority order is as follows: class-1 handoff has highest priority, class-1 new call, class-2 handoff, and finally class-2 new call has the lowest priority in all. The purpose of bandwidth reservation scheme is to reserve different bandwidth to each class. Giving four bandwidth thresholds, t_1 , t_2 , t_3 , and t_4 whereas ($t_1 \geq t_2 \geq t_3 \geq t_4$) as shown in Figure 2.3.

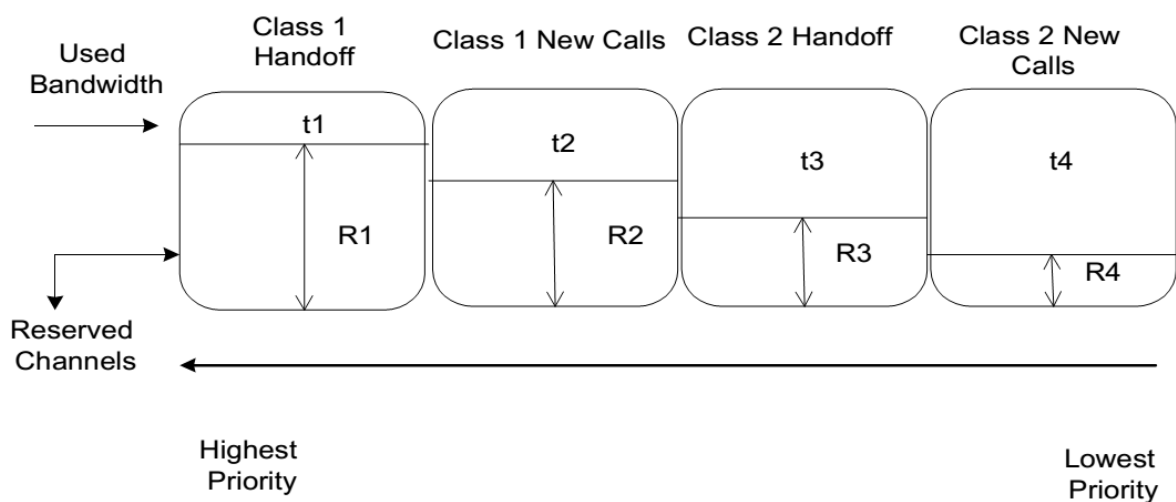


Figure 2.3: Priority base bandwidth reservation [15]

Either the bandwidth represents the cell bandwidth that is currently in use by new calls or handoff calls. The reserved bandwidth portion reserved for future calls and shared with all type of incoming calls. In reservation scheme t_1 , t_2 , t_3 , and t_4 are thresholds for class-1 handoff, class-1 new call, class-2 handoff call, and class-2 new call, respectively. As reserved bandwidth of the corresponding category decreases, the priority level decreases. For example, according to priority assignment for class-1 handoff has highest priority; it means more bandwidth should be reserved for future incoming class-1 handoff connections. It also means if resources equally divided among classes then for highest priority class threshold value t_1 should be smallest.

2.2.3 Reducing Call Drop using Bit Error Rate (BER) reduction

One of the important reasons for call drop outs is high Bit Error Rate (BER). In general, many existing wireless systems set a threshold BER before a call is dropped. A new call is admitted if and only if its QoS constraints are satisfied without jeopardizing the QoS constraints of existing calls in the network. QoS is usually defined according to the BER in digital transmission [24]. For example, the QoS requirement for voice users is usually expressed as a BER less than 10^{-3} in order to guarantee quality of communication. Thus, if we improve the BER for the system, the problem of call drop out can be reduced. BER quantifies the reliability of the entire radio system from bits in to bits out. In general, the BER is the percentage of bits that have errors relative to the total number of bits received in a transmission, usually expressed as ten to a negative power. Hence BER is expressed as in equation (2.1) according to the authors in [24]

$$BER = \frac{e}{N} \quad \text{2.1 [24]}$$

Where, e = number of errors and N = total number of bits received

With a strong signal and an unperturbed signal path, this number is small and can be considered insignificant. Bit-error-rate performance is usually depicted on a two dimensional graph. The ordinate is the normalized signal-to-noise ratio (SNR) expressed as E_b/N_0 : the energy-per-bit (E_b) divided by the power spectral density of the noise (N_0), expressed in decibels (dB). The abscissa is the BER, a dimensionless quantity, usually expressed in powers of ten. In order to create a graph of BER versus E_b/N_0 , series of points are plotted. Each of these points requires us to run a simulation at a specific value of SNR or E_b/N_0 . A typical plot of BER vs. E_b/N_0 can be shown in Figure 2.4.

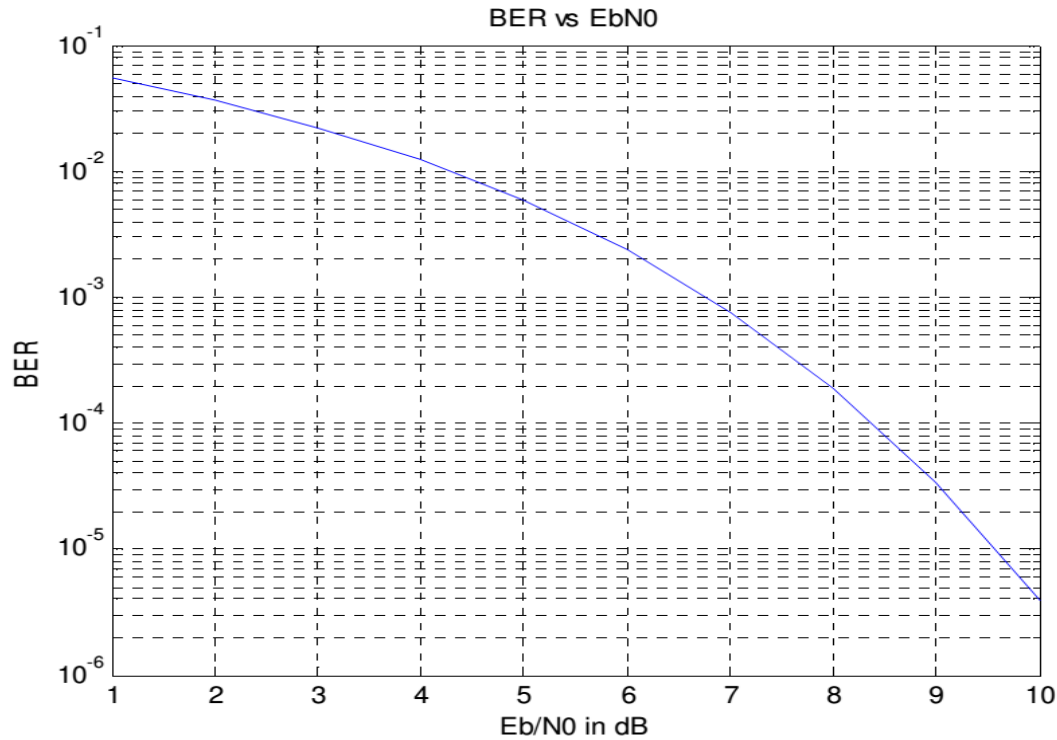


Figure 2.4: Typical plot for BER vs. E_b/N_0 [24]

The authors improve the BER curve i.e. for same E_b/N_0 , to get less BER, this they achieved by using a new signal processing subsystem at the receiver section of a wireless system this improved the end- to-end performance of the system. This in turn will help in the reduction of call drop out.

2.3 Universal Mobile Telecommunications System (UMTS)

The UMTS architecture is briefly introduced and a more detailed overview of UTRAN is given, including essential network elements and the interfaces between them. Furthermore, various aspects of the network are discussed, such as network capacity, radio resource management, etc.

2.3.1 System Architecture

UMTS evolved from GSM (Global System for Mobile Communications) and utilizes the well-known architecture of GSM, but also introduces some new elements. As shown in Figure 2.5, the network elements are grouped by functionality into the UMTS Terrestrial Radio Access Network (UTRAN), that handles all radio related functions, and the Core Network (CN). The CN is the long-range network that transports user data to its destination. It contains an amount of switching systems (like Mobile Services Switching Center (MSC) for interworking with location databases), as well as gateways to other networks, like the Public Switched Telephone Network (PSTN) or the Internet. The CN supports two types of switching: circuit and packet switching. Both parts of the CN use the same Radio Access Network (RAN). The Gateway Mobile Switching Center (GMSC) provides edge functionality within a PSTN for circuit-switched calls. It terminates the PSTN signaling and traffic formats and converts them to protocols used in mobile networks. For mobile calls, it interacts with the Home Location Register (HLR) to obtain routing information. The Gateway GPRS Support Node (GGSN) supports the edge routing function of the GPRS (General Packet Radio Service) network. The GGSN performs the task of an Internet Protocol (IP) router to external packet data networks, e.g., Internet. The interface between CN and RAN (Iu-interface) is divided into an interface for the circuit-switched part of the CN (*IuCS*-interface) and an interface for the packet switched part (*IuPS*-interface). The RAN contains two types of nodes: Radio Network Controller (RNC) and Node B (often called base station), and is responsible for all radio related tasks like radio resource and connection management. The RNC controls resource management for one or more Node Bs, that are connected to

it. The RNC is the service access point for the services UTRAN provides to the CN, e.g. management of connections to user equipment (UE). The interface between RNC and Node B is called the *Iub*-interface. The UE, which will be called mobile terminal (MT) in the following, is connected to a Node B via the *Uu*-interface. The Serving GPRS Support Node (part of CN) keeps track of the location of an individual MT and performs security functions and access control. A Node B supplies one or more radio cells and converts the data flow between *Iub* and *Uu* interfaces. Besides the main transfer functionality, a Node B performs some basic radio resource management operations, like inner loop power control, and handles the radio measurements for handover decision, load estimation in the cell, or admission control as requested by the RNC. System information messages are distributed by Node B according to the schedule given by the RNC. The coordination between Node B and RNC is performed on a master-slave basis.

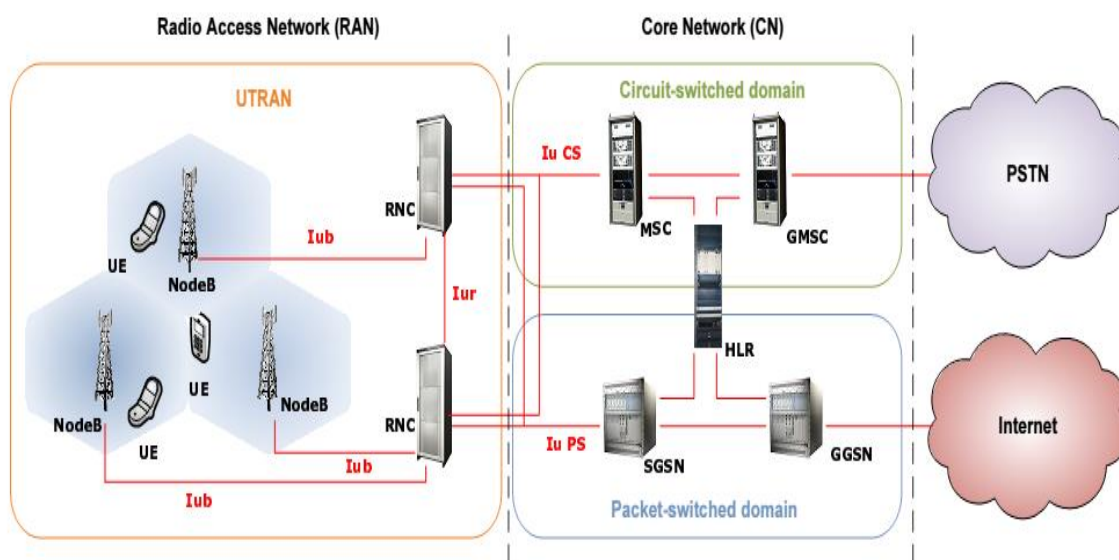


Figure 2.5 : UMTS Architecture [25]

2.3.2 WCDMA Concept

Wideband Code Division Multiple Access (WCDMA) is a multiple access technology which is used in the Universal Mobile Telecommunication Service (UMTS) to provide 3rd generation services in the growing demand of mobile communications. User information bits are spread over a wider bandwidth by multiplying the user data bit with quasi-random bits (called chips) derived from CDMA spreading codes in order to support very high bit rates (up to 2 Mbps). Here the main parameters of WCDMA are presented [2]:

- ✓ WCDMA uses the Direct Sequence Spread Spectrum (DSSS) technique to spread the user information over wider bandwidth.
- ✓ It uses 5 MHz carrier bandwidth.
- ✓ The chip rate is 3.84 Mcps.
- ✓ It supports Bandwidth on Demand i.e. variable data rates for different types of user.
- ✓ It supports both Frequency Division Duplex (FDD) and Time Division Duplex (TDD) methods.
- ✓ FDD: Separate 5 MHz frequency band is used for the uplink and for the downlink.
- ✓ TDD: Only one 5 MHz frequency band is time shared between uplink and downlink.

- ✓ WCDMA operates on the mode of asynchronous base stations so that there is no need for a global time references. WCDMA employs coherent detection of uplink and downlink based on the use of pilot signals.
- ✓ It employs smart adaptive antennas to increase the capacity and the coverage.
- ✓ It supports the handover between the GSM and the WCDMA.
- ✓ WCDMA uses the rake receiver to adopt the multipath diversity of the transmitted signal.
- ✓ It can operate both soft and hard handover.

WCDMA can also works with GSM networks. It can support hard hand off when it works with the GSM network and uses the soft and softer hand off when it works within the WCDMA network .The use of rake receiver makes it more efficient against the multipath diversity of the wireless channel. It can transmit high data rate traffic up to 2 Mbps in an indoor environment. The main characteristics of WCDMA pointed out in the table 2.1.

Table 2.1 Parameters of WCDMA [2]

Multiple access method	DS CDMA
Duplexing Method	FDD and TDD
Chip rate	3.84 Mcps
Frame length	10 ms
Service multiplexing	Multiple services in one connection.
Data rate	Up to 2 Mbps
Detection	Coherent detection with pilot symbols.
Multiuser detection , smart antennas	Supported by the standard.
Hand over	Soft and handoff

This section discusses the fundamental WCDMA concepts, like spreading and scrambling, power control, and soft handover.

(i) Spreading As mentioned above WCDMA is a common technique used in the UTRAN air interface to distinct the user's physical channels in both uplink and downlink. Data symbols of users are made separable by multiplying them with spreading (also called channelization) codes. This process is called spreading, because the original narrowband signal is spread over a wider frequency band depending on the system chip rate, i.e., the number of chips per time interval. In UMTS, the chip rate is 3.84 Mcps, which corresponds approximately to a system bandwidth of 5 MHz. After spreading has taken place, each data symbol is represented by a number of chips. The ratio between the chip rate and the bit rate is called the spreading factor. It implies the number of chips that are used to spread one data symbol. The spreading sequence is known at both the transmitter and the receiver. The data can be recovered at the receiver by correlating the received signal with the spreading sequence of the data stream to be recovered. Different spreading sequences should be (at least approximately) mutually orthogonal (with zero cross-correlation). To achieve orthogonality between codes with different spreading factors, the tree structured orthogonal variable spreading factor (OVSF) technique can be used.

(ii) Scrambling In addition to spreading, the transmitter also performs the scrambling operation. Scrambling is done after spreading, in order to randomize the chip sequence resulting from spreading, and to identify the source of each signal

(MT, Node B). Randomization prevents the reception of correlated signals, which would lead to significant performance losses due to interference. Scrambling codes are quasi-random chip sequences. Figure 2.6 shows the schematic illustration of scrambling and spreading in WCDMA, where DPDCH denotes Dedicated Physical Data Channel.

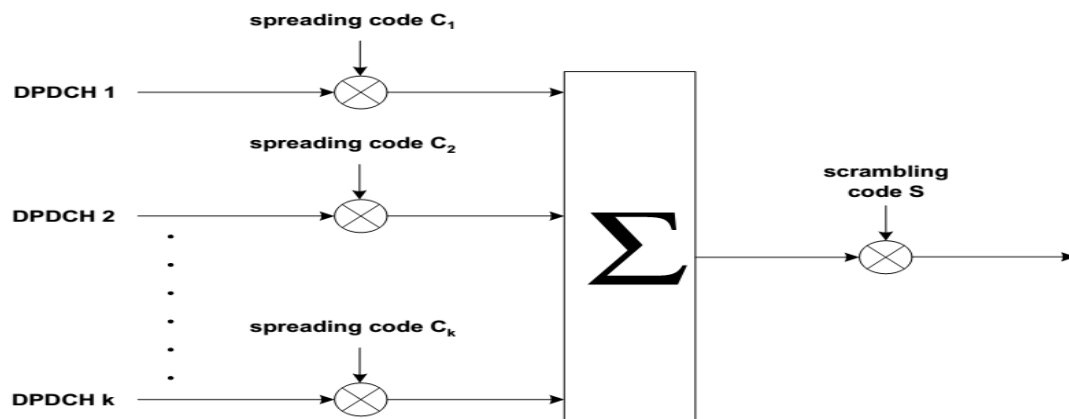


Figure 2.6: Scrambling and Spreading in WCDMA [2]

2.3.3 Differences between WCDMA and Second Generation Systems

The main differences between the third generation and the second generation mobile communications are described below. GSM is the most popular second generation mobile communication systems. The second generation air interfaces is mainly developed for the voice services while the third generation air interfaces is developed for the multimedia applications, voice, video, streaming etc. The requirements of the 3rd generation mobile communication are listed below [2]:

- The 3rd generation mobile system has to support data rates up to 2 Mbps
- It has to support bandwidth on demand

- It has to support multiple services on a single connection
- The 3rd generation system should support better QoS from 10% frame error rate to 10^{-6} bit error rate
- It has to coexist with the second generation system like GSM
- Support of asymmetric uplink and downlink traffic
- High spectrum efficiency
- Co-existence of FDD and TDD modes.

The table 2.2 list a few difference between WCDMA and GSM

Table 2.2 Difference between WCDMA and GSM [2]

	WCDMA	GSM
Access technology	CDMA	TDMA
Bandwidth	5 MHz	200 KHz
Power control frequency	1500 Hz	2 Hz
Hand over technique	Soft and softer handoff	Hard hand off
Frequency reuse factor	1	More than 1
Quality control	Radio Resource management algorithm	Network Planning
Packet data	Load based packet scheduling	Time slot based scheduling with GPRS
Downlink transmit diversity	Supported for improving downlink capacity	Not supported by the standard
Supported data rate	Support high data rate up to 2 Mbps.	Support low data rate 12.8 kbps for voice

2.3.4 Radio Resource Management

The capacity of a cell in a WCDMA network essentially depends on the orthogonality and number of spreading codes used. Perfectly orthogonal codes guarantee that the different physical channels do not cause mutual interference. In this case, the capacity of a radio cell is determined by the number of mutually orthogonal codes. However, in a

real world system, the combination of spreading and scrambling code does not yield a fully orthogonal sequence, which means every user causes some interference to other users. Thus, the wireless interference in a WCDMA system is the main limiting capacity factor. Therefore, efficient management of radio resources is the most important task in such a system. The group of control mechanisms, which are responsible for utilization of the radio interface is called Radio Resource Management (RRM). Their tasks are to supply optimal coverage, to maintain the planned capacity, to adequately engineer system resources to meet quantitative performance objectives in terms of Grade of Service (GoS) requirements, and at the same time to provide connection quality which meets some given Quality of Service (QoS) characteristics. GoS and QoS concepts have different viewpoints. While the QoS concept views the situation from the customer's perspective, the GoS concept considers the network aspects. In case a user requests a service from a system where all channels are currently occupied, congestion occurs. GoS is expressed in relation to how frequently the system congestion is permitted to exceed some predefined value. The traditional measure used in conjunction with congestion is the call blocking probability – the probability that a user cannot establish a call. Call blocking might occur when all user channels are busy or due to insufficient network capacity. The call is not accepted and is either rejected or allowed to wait in a queue (depending on the type of the system). Obviously, the larger the value of GoS, the better is the service. Since the traffic volume is continuously growing, to maintain the GoS value within reasonable limits, the network is initially sized to have a much better GoS than the recommended one. Quality of Service (QoS) is defined in ITU-T Recommendation E.800 as the collective

effect of service performance, which determines the degree of user satisfaction regarding undesired effects such as call dropping, packet delays, etc., concerning a particular service.

By means of sophisticated RRM algorithms, like load control, power control and handover the systems can be kept in a stable operational state for all traffic and radio environment conditions.

2.3.5 Techniques for Radio Resource Utilization

All functions for handling the radio interface resources of UMTS can be expressed by the term Radio Resource Utilization (RRU). Those, which are relevant for this thesis, are described below.

(i) Power Control

A mobile communication radio environment is constantly and rapidly changing, e.g. due to fast and slow fading, as well as due to the interference from users in the cell. The received ratio of bit energy to noise power spectral density, E_b/N_0 is an important parameter in communication systems. If the received E_b/N_0 drops below an acceptable level and does not improve during a predetermined period of time, the call will be dropped. Thus, for an efficient operation of the interference limited WCDMA network it is crucial that the transmit power of all users is controlled. A group of measures is therefore introduced within the RRU to minimize the interference level in the system.

They are summarized as power control (PC) and consist of open-loop PC, inner-loop PC (also called fast-closed-loop PC), and outer-loop PC in both uplink and downlink. Open-loop PC makes initial estimates of the transmit powers needed in the uplink and the downlink for each MT, before it accesses the network, and is only important for the initialization of the call. The estimation is based on path loss calculations in the downlink direction. The other two loops are active for the complete duration of the call. The target of the inner-loop PC in WCDMA is to keep the E_b/N_0 constant. Due to the fast feedback loop (1500 Hz) this is fairly successful. It means that for a chosen service, given channel conditions, and a required BLER (block error rate), the received power on the radio channel divided by the interfering power is approximately constant [15]. The outer-loop power control is needed to keep the quality of communication at the required level by setting the target carrier-to-interference ratio C/I (average received power of the useful signal to that of all relevant interfering signals) for the fast-closed-loop power control both in the uplink and the downlink. The frequency of the outer loop power control is typically 10-100 Hz.

(ii) Handover Control

Another algorithm for radio resource utilization of WCDMA is the soft handover (SHO). SHO has been developed to provide a seamless user migration from one cell to another. SHO is different from traditional hard handover. For hard handover a decision is made on whether to handover or not, and the MT communicates with only one Node B. However, in WCDMA systems, the user can get two or more simultaneous connections

with different Node Bs. Hard handover happens at a distinct time, while soft handover lasts for a period of time. Moreover, in SHO the user has a gain from diversity. The diversity gain provided by SHO is called macro diversity. This means that one or more (up to 3, usually) additional signals can be received at a time, which offers protection against fast fading by selecting the best signal or combining the signals in a suitable way. The user transmits the same signal to all Node Bs it is connected to. Furthermore, the Node Bs transmit their received signals to the RLC (Radio Link Control), which combines them or selects the best one. If the signal strength on one radio link rapidly changes, the RNC instantly switches to a signal from one of the other Node Bs with significantly better quality. This prevents an interruption in data flow. Macro diversity can be exploited for decreasing the uplink interference, which leads to a higher system capacity.

(iii) Call Admission Control

Since WCDMA allows many users to transmit simultaneously within the same frequency band, admitting a new call will always increase the interference level in the system, reducing the cell capacity for new calls and even leading to dropping of established connections. The objective of the Call Admission Control (CAC) is to admit or reject new or handover call requests, based on the actually free capacity, the current traffic load situation in the system and the desired QoS of the new calls as well as calls in progress. The CAC is proactive, its goal is to prevent unacceptable degradation of QoS and to achieve a trade-off between system capacity and user satisfaction. Compared to the

straightforward CAC algorithms in TDMA systems, where the system capacity is fixed (predefined number of radio channels), in cellular networks with WCDMA radio interface it is much more difficult to predict the effect of an admitted new calls on the active calls due to uncontrollable fading variations and wireless interference.

(iv) Congestion Control

Even with an efficient admission control algorithm an overloaded situation in the system may occur. When an overload is encountered, the output powers are rapidly increased by the fast closed-loop power control until one or several transmitters have reached their maximum output power. The connections unable to achieve their required quality are considered useless and are only adding interference to the system. This is an unacceptable system behavior. Hence, a procedure to remove the congestion is needed. The method is called congestion or load control, and its objective is as quickly as possible to monitor the traffic load in the network, detect traffic overload, and react in order to get the system back to a feasible traffic load, which is defined by the radio planning. This can be achieved e.g. by slightly degrading the QoS in the overloaded cell during the time it takes to resolve the congestion

2.4 Traffic Concepts and Models

Teletraffic theory quantitatively studies the relationship between communications systems, traffic and performance. Queuing theory, a subset of teletraffic theory, deals with the mathematical modeling of traffic and the analysis of the statistical behavior of

telecommunication systems [26]. The basic goal is to provide reliable methods for determining the cost-effectiveness of various network sizes and configurations. In the following some fundamental concepts of queuing theory will be reviewed.

2.4.1 Mathematical Background: Basic Model

Queuing systems model processes, where users generate traffic flows, which can either be circuit-switched (speech services) or packet-switched (data services). An example of a queuing system is illustrated in Figure 2.7 and can be described as follows: Service requests enter the system in order to be processed [26]. If the required resources are available, the requests are admitted and leave the system after successful handling. In case a new request arrives and finds all resources busy, it enters a queue and waits until sufficient capacity becomes available to serve its request. The standard notation to identify the main elements that define the structure of a queuing system is the *Kendall notation* $A/B/N/n/S/X$, where A denotes the distribution of the inter-arrival times and B the distribution of the service times; N specifies the number of servers, n the maximum number of waiting requests in the finite case (number of servers plus the capacity of the queue), S is the number of users (passive and active) and X defines a queuing discipline.

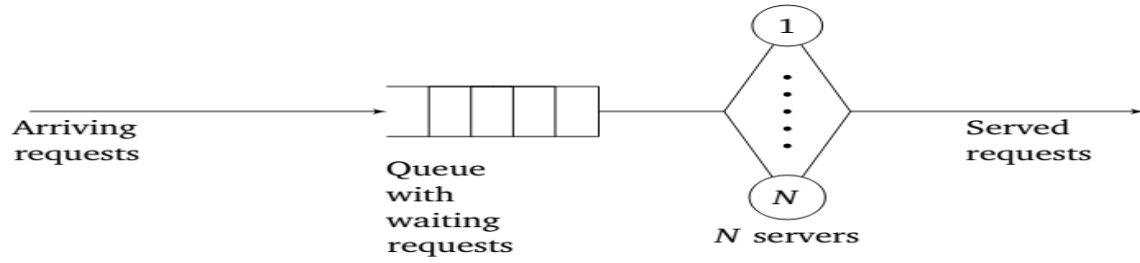


Figure 2.7: A queuing system with arrival and departure of service requests [26]

2.4.2 Birth-Death Process

Certain types of queuing systems can be modeled by birth-death processes. A birth-death process is a process, where the population with size i (the number of currently active users) may increase or decrease according to certain rules [27]. Specifically, when the population size is i , the only possible state transitions are from i into $i - 1$ (death) or from i into $i + 1$ (birth). The following state-transition diagram (Figure 2.8) is called a birth-death Markov chain. Directed branches represent transitions between the states.

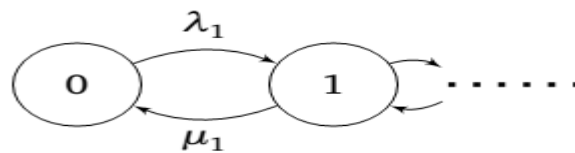


Figure 2.8: Birth-Death process [27]

The birth-transition intensity is described by $\lambda_i \geq 0$ for $i \geq 0, i \in \mathbb{N}$ (user arrival rate) and the death transition intensity is designated by $\mu_i \geq 0$ for $i > 0, i \in \mathbb{N}$ (user service rate). Teletraffic theory relies on the concept of stochastic processes. It is usually assumed that inter-arrival times are independent and identically distributed [28]. In most cases, the inter-arrival time distribution is assumed to be exponential, in which case the arrival

process is a homogeneous Poisson process. In Figure 2.9 a realization of a Poisson process is shown. Poisson arrivals occur completely at random in time.

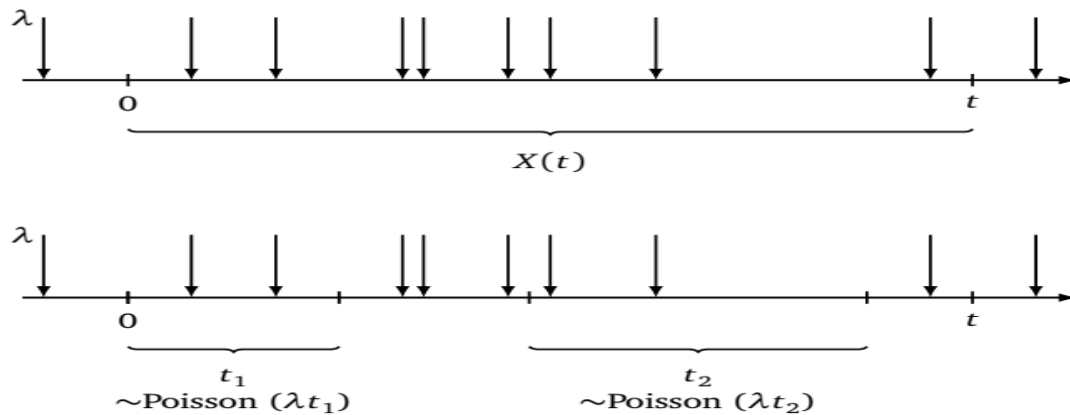


Figure 2.9: Poisson Arrival [28]

Mathematically, the process is defined as a collection $X(t) : t \geq 0$ of random variables, where $X(t)$ is the number of events that have occurred up to time t . A Poisson process is a pure birth process and is characterized by λ (i. e. number of events per time unit) such that the number of events $X(t)$ in a finite time interval of length t follows a Poisson distribution with parameter λt . This relation is given by [28] as

$$P\{X(t) = n\} = \frac{(\lambda t)^n e^{-\lambda t}}{n!} \quad (2.2) \quad [28]$$

A Poisson process has several important properties:

Stationarity: The distribution function of the number of observations in a time interval depends only on the length of the interval and not on its position in time

Memorylessness: A Poisson process $X(t)$ satisfies the Markov property *memorylessness*, which states that

[illegible]

for any choice of time instants $t_i, i = 1, \dots, n$, where $t_i < t_{i+1}$.

This should be understood as follows: the future state of the process at time t_{n+1} is independent of its past history (earlier time instants $t_1, \dots, t_{n-1}$) and depends solely on its present state at time t_n (this follows from properties of the exponential distribution).

Ordinariness: means that the normalized probability of the occurrence of more than one event in an very small interval h goes to 0:

$$\lim_{\eta \rightarrow 0} \frac{\partial}{\partial \eta} \left(\frac{1}{\eta} \int_0^\infty \frac{d\lambda}{\lambda^2} \left[\frac{1}{\lambda} \left(\frac{1}{\lambda} - \frac{1}{\lambda + i\eta} \right) \right] \right) = 0 \quad (2.4) [28]$$

2.4.3 Markovian Queuing Systems

The traffic intensity in the network, which is defined as $\rho = \lambda/\mu$ may change over time. However, for practical calculations of the system performance measures, here the interest is in traffic in statistical equilibrium, which means that the traffic intensity is rather stable during the period of observation. Furthermore, the traffic process must be observed for a time interval sufficiently large to neglect the effect of a transient period, assuming we start with an empty system [26]. This is in agreement with conditions during busy hour (the busy hour is the 60-minute period during the day when

the traffic volume reaches its maximum) and therefore allows for approximating the actual traffic by means of a stationary traffic intensity model.

2.4.4 M/M/1 System

The simplest queuing system model is $M/M/1$, a single-server queue with randomly distributed service requests. The defining characteristics of the system are:

• The time intervals between successive events are exponentially distributed, with $\lambda_i = \lambda$.

• The service time is an exponentially distributed (Markovian) random variable, with $\mu_i = \mu$.

• The number of arrivals during a finite time interval follows a Poisson distribution.

• The arrival process and the service process are independent of each other.

The capacity of the queue is infinite and the service discipline is First-In-First-Out (FIFO). The $M/M/1$ queue is a birth-death process with the birth rate and death rate constant, which is easy to deal with mathematically. A very useful relation between $E(S)$, the long-term average number of users in a system, $E(W)$, the long-term average time a user spends in the system (waiting time plus service time) and λ , the average number of users arriving at the system per unit time is given by *Little's law* [28]:

$$E(S) = \lambda E(W) \quad (2.4) \quad [28]$$

Where $E(S)$ is the average number of users, $E(W)$ is the average time a user spends and λ

is the average number of users arriving at the system per time

The above mentioned properties include the Markov (memoryless) property, which makes it possible to describe the state of the system at arbitrary points in time with a single variable, which holds the number of users in the system. Markovian systems can be directly mapped to a continuous time Markov chain (CTMC). The CTMC for $M/M/1$ is depicted in Figure 2.10.

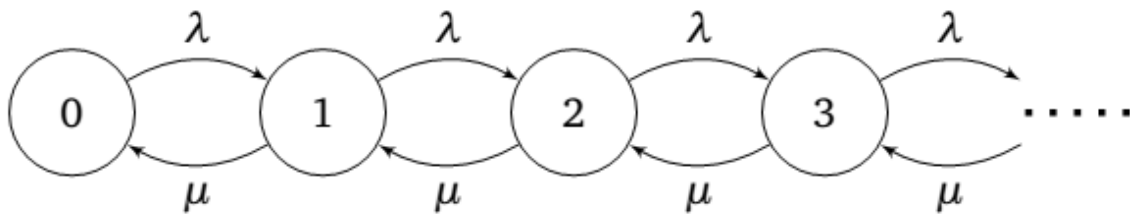


Figure 2.10: M/M/1 Process [28]

A system transition from state to state is described by a Markov process [29]. The process starts in state 0 (the system is empty) and moves successively from one state up to another with rate λ , and down with rate μ . A Markov chain is completely defined, when all states and state transition rates are given. This concludes the description of the $M/M/1$ queuing system.

However, up to now we have not yet studied the behaviour of the queue, since the steady-state probabilities p_i must be determined. The steady-state probability is the probability of observing the system in state i at a random point in time. This probability does not depend on the states the chain was in before the current state. If cut between two arbitrary states i and $i + 1$, $i \geq 0$, the transitions from each side of the cut to the other are the state transitions $i \rightarrow i + 1$ and $i + 1 \rightarrow i$.

2.4.5 Blocked-Call-Cleared – Erlang B

In a truncated system, when a particular user requests service and all of the servers are already in use, the user is denied access to the system and thus blocked. This type of truncation is called Blocked-Call-Cleared (BCC), see Figure 2.11. The following assumptions are made:

- ✓ There are memoryless arrivals of requests, implying that all users may request service at any time.
- ✓ The number of arrivals during any time interval is Poisson distributed
- ✓ The duration a user occupies a server is exponentially distributed, so that longer calls are less likely to occur than shorter calls (there is no limit for the call length, but the distribution is not heavy-tailed).
- ✓ There is a finite number of servers available in the network
- ✓ Requests are not allowed to queue if there is no server available when the request arrives.
- ✓ Blocked requests (or calls) are lost.
- ✓ Lost calls are not retried.

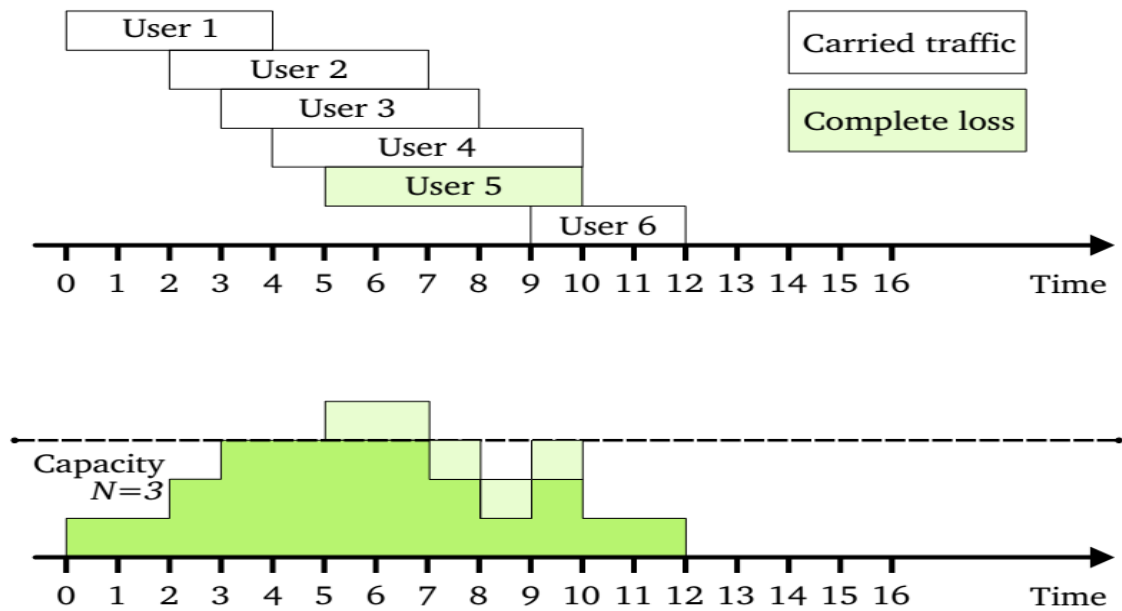


Figure 2.11: Example of Blocked Call Cleared Model [28]

The Erlang-B formula in equation (2.5) is used to compute hard blocking as a function of the number of available channels and the offered load. The equation is based on analytical probability theory and can be used when the following assumptions are satisfied.

- i. All call attempts are Poisson distributed with exponential service time
- ii. Blocked calls are cleared (BCC) in the system and that the caller tries again later

$$B_N = \frac{A^N / N!}{\sum_{n=0}^N (A^n / n!)} \quad (2.5) \quad [29]$$

Where, B_N = probability of blocking,

A = offered traffic in Erlang,

N = number of channels in the cell, and

i = number of busy channel

Thus, there is no queuing and no retry for unsuccessful calls with the Erlang-B model.

2.4.6 Extended Erlang B Model

The Extended Erlang B Model uses the same formula and assumptions as Erlang-B model except that a percentage of callers retry their calls until they are serviced. This model is commonly used for standalone trunk groups with a retry probability such as a MODEM pool. The model is as expressed in equation (2.6)

$$B_k = \frac{A^k / k!}{\sum_{i=0}^N (A^i / i!)} \quad (2.6) \quad [29]$$

Where, k is the number of busy channels, N is the number of servers (trunks), A is the traffic density in Erlang, and B_k is the blocking probability.

The Extended Erlang B traffic model is used by telephone system designers to estimate the number of lines required for public switch telephone network connections (trunks) or private wire connections and takes into account the additional traffic load caused by blocked callers immediately trying to call again if their calls are blocked. It therefore, allows retry of unsuccessful calls and assumes infinite population of callers (sources). This traffic model may be used where no overflow facilities are available from the trunk group being designed.

2.4.7 Erlang-C Model

In the Erlang C Model, the system is designed around the queuing theory. The caller makes one call and is held in a queue until answered, so the formula expresses the waiting probability. Just as the Erlang B formula, Erlang C assumes an infinite population of sources, which jointly offer traffic of A Erlangs to N servers. However, if all the servers are busy when a request arrives from a source, the request is queued. An unlimited number of requests may be held in the queue in this way simultaneously. This formula calculates the probability of queuing offered traffic, assuming that blocked calls stay in the system until they can be handled. This formula is used to determine the number of agents or customer service representatives needed to staff a call centre a specified desired probability of queuing. The blocking probability or the delay probability (waiting probability) is given in equation 2.7.

$$P_w = \frac{\frac{A^N}{N!} \left(\frac{A}{N} \right)}{\sum_{i=0}^{N-1} \frac{A^i}{i!} + \frac{A^N}{N!} \left(\frac{A}{N} \right)} \quad (2.7) \quad [29]$$

where,

A is the total traffic offered in units of erlangs

N is the number of servers; i is the number of busy servers

P_w is the probability that a customer has to wait for service

It is assumed that the call arrivals can be modeled by a Poisson process and that call holding times are described by a negative exponential distribution. A common use for Erlang C is modeling and dimensioning call center agents in a call center environment. It can also be used to determine bandwidth needs on data transmission circuits.

2.4.8 Engset Formula

The Engset formula is used to determine the probability of congestion occurring within a telephony circuit group. It deals with a finite population of S sources rather than the infinite population of sources that Erlang assumes. The formula requires that the user knows the expected peak traffic, the number of sources (callers) and the number of circuits in the network. Engset's formula given in equation (2.8) is similar to the Erlang-B formula; however one major difference is that the Erlang's equation assumes an infinite source of calls, yielding a Poisson arrival process, while Engset specifies a finite number of callers. Thus Engset's equation should be used when the source population is small. But for population sources greater than 200 users, extensions or customers, it becomes similar to Erlang-B model.

$$P_n = \frac{A^n \binom{S}{n}}{\sum_{k=0}^n A^k \binom{S}{k}} \quad (2.8) \quad [29]$$

where,

A = offered traffic intensity in Erlangs, from all sources

S = number of sources of traffic

N = number of circuits in group

P_b = probability of blocking or congestion

In the traditional flat cellular networks, the Erlang-B model is generally used to describe the limitation in physical resources independent of the quality experienced on the radio interface (i.e. hard blocking). With the increasing complexity of CDMA cellular networks, the required assumptions are no more valid and the Erlang-B formula is found to overestimate the capacity. Thus, to properly account for the quality of service experienced at the BS, soft blocking should be modeled and evaluated

2.5 QoS-Based Key Performance Indicators (KPIs)

In telecommunication engineering, and in particular teletraffic engineering, the quality of voice service is specified by two measures [29]: the Grade of Service (GoS) and the Quality of Service (QoS). In the ITU-T Recommendation E.800 [30], QoS is defined as the collective effect of service performance, which determines the degree of satisfaction of a user of the service. How the end-user perceives the service quality, i.e. his satisfaction with the service as a function of the measured parameters, is highly dependent on the service level (QoS parameters) he is accustomed to. QoS is a measure of the Key Performance Indicators (KPIs). GoS is a subset of QoS and it is the mechanism for controlling the performance, reliability and usability of a telecommunications service. The smaller the value of GoS, the better is the service. A telephone service is available if the GoS is better than a specified percentage. Availability is the QoS characteristic that represents the proportion of time when

satisfactory service is available. According to ETSI's standards [31] QoS KPIs can be categorized in five phases during service use from the customer's point of view:

1. Network Availability: The probability that a service is available when the end user requires it, is called availability.

2. Network Accessibility: Probability that the user performs a successful registration on the network.

3. Service Accessibility: Accessibility has to do with the end users being able to set up a call and access radio resources. If the user wants to use a service, the network operator should provide him as fast as possible access to the service. Includes KPIs that describe how successful is the service access (e.g, call setup rate) as well as KPIs that describe setup time (i.e, call setup time).

4. Service Integrity: This describes the QoS during service use. Includes KPIs that describe the quality of the service.

5. Service Retainability: Retainability covers the ability to keep up a call; either in accordance with or against the will of the user. It Includes KPIs that count the time that a service can retain (Drop Call Rate or call holding time).

The service quality performance indicators consider in this work cover the different phases of QoS [31] aspects during service use from the customer's point of view. They are here defined by:

(a) Call Setup success rate (CSSR): The measure of the service accessibility provided by GSM network is the call set-up rate (CSSR). This indicator measures the ease in which calls are established or set up. It analyses the ability to successfully establish voice communications between two ends, a mobile network terminal and a fixed network terminal, and the ability of networks to maintain this call during a pre-established period of time. Mathematically, CSSR describes the ratio between the calls successfully connected to the system and the overall number of call attempts [31]:

$$CSSR = \frac{N_{success}}{N_{attempt}} \quad (2.9) [31]$$

where $N_{attempt}$ is the number of call attempts and N_{block} is the number of blocked calls

(b) Drop Call rate (DCR): The measure of the service retainability, which describes the termination of services in accordance with or against the will of the end user is the CDR. It is defined by the ratio between the number of dropped calls and the overall number of call releases [31].

$$DCR = \frac{N_{drop}}{N_{release}} \quad (2.10) [31]$$

where N_{drop} is the number of dropped calls and $N_{release}$ is the overall number of call releases

(c) Outage rate (OutR): This indicator measures service integrity, which is the QoS during service use. OutR describes the ratio between the number of calls which, although normally released, perceived an unacceptable outage time lasting more than 5% of the

duration of the call, and the overall number of call releases (that is both the normal and the abnormal releases [31].

$$N_{outage} = \frac{N_{outage}}{N_{total}} \quad (2.11) [31]$$

where *N_{outage}* is the number of calls with unacceptable outage time.

(d) End points Service Availability (ESA): This indicator measures service availability, which is the QoS during service use and it is appraise from the view point of what the customer get from their end. Endpoints are defined as the interface between the customer and the equipment providing access to the service. ESA is defined as the percentage of time a usable call can be established and maintained between two end points. The measurement of ESA is described by the ratio in equation (2.12) [31].

$$ESA = \frac{N_{usable}}{N_{total}} \quad (2.12) [31]$$

CHAPTER THREE

RESEARCH METHODOLOGY

3.1 Overview

This chapter presents an in-depth description of the system environment and performance metrics utilized throughout this thesis. Furthermore, the simulation(s) and benchmarks utilized for performance evaluation in each of the techniques in this thesis are described. This work concentrates on the performance evaluation and comparison of various existing techniques that help to reduce the effect of call drop in a wideband code division multiple access (WCDMA) network.

The performance evaluation is done using MATLAB simulator, the mathematical model and system parameters of a WCDMA communication network will be derived and implemented. This work focuses on comparison and performance evaluation of power control, handoff schemes (Hard and Soft) as some of the techniques to minimize call drop. The comparison was made under the condition of multi-traffics WCDMA system. The evaluation of the QoS and performance was done through a programming simulation to analyze the relationship between signal to noise ratio SNR and probability of outage in terms of bit error rate BER. Another simulation process is done to show the relationship between SNR and capacity of the network in the case of single cell traffic and multi cell traffic cases. Control admission algorithm is generated to add or reject a new user based

on certain policy into the network and this needs some analysis to evaluate the impact of the decision algorithm on the system performance.

3.2 System Model

In this model power is considered to be the only limited resource and other resources such as spreading codes and buffering capacity are assumed to be available. A single caller is assumed to request a single service. The Base Station is assumed to be located at the centre of a cell with an Omni-directional antenna. The network cells are divided into a number of manageable groups. Each group is controlled by one *RNC* which logically manages the radio resources of these cells and it also acts as a gateway to other communication systems such as the Public Switching Telephony Network (*PSTN*) and the Internet (see Figure 3.1). The model was implemented in a MATLAB simulation environment. User requests are processed on a "first come, first served" basis.

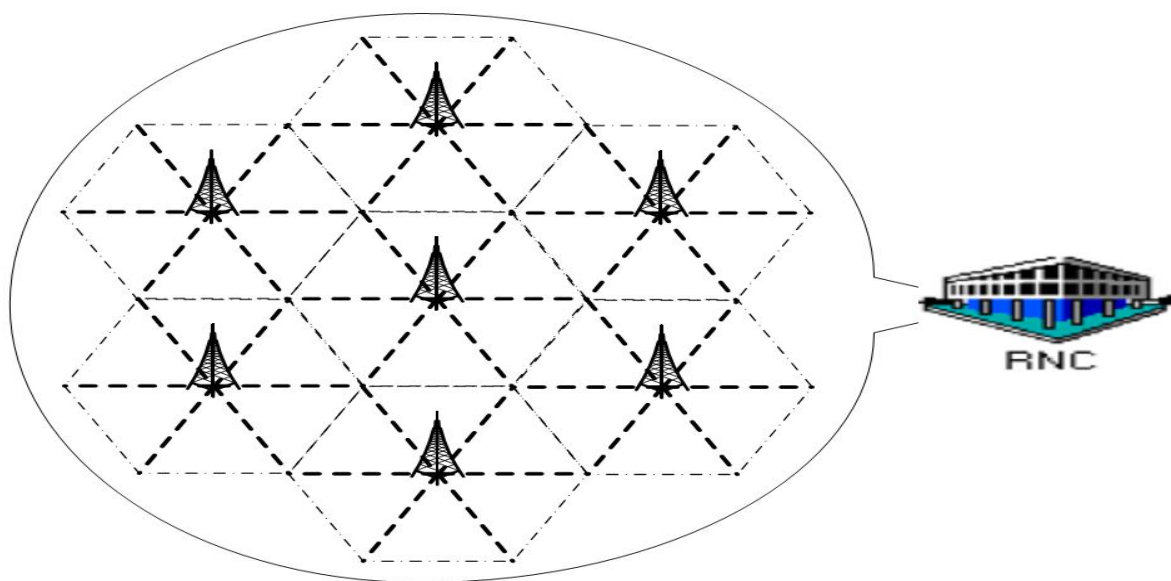


Figure 3.1: Network Architecture [2]

The cell radius is 300m. Mobile users move freely over the network coverage area. When a mobile user moves out of its current serving cell coverage area and enters the coverage area of an adjacent cell, a handoff protocol is executed to transfer the mobile user connection to the base station of the moved to cell. The decision of accepting or rejecting a request is based on the QoS profile attached to the request, the maximum power available in the system and the QoS requirements of users being served. In the uplink direction the total interference measured at the base station is used as the load indicator. In the downlink direction the only power level that needs to be considered is that emitted by the base station. The cost of a connection is computed according to the following formulae.

$$C_{i,j} = \frac{P_{i,j}}{P_{max}} \cdot \frac{1}{W} \cdot \frac{I_{i,j}(t)}{I_{i,j}(t) + 1} \quad (3.1)$$

Energy to noise ratio is represented by $\frac{P_{i,j}}{N_0}$. Inter-cellular interference is not taken into account in this model. The chip-rate is represented by W and $I_{i,j}(t)$ represents the sum of interferences exerted by existing users at a given time within the same cell. The central limit theorem is used to model $I_{i,j}(t)$ as a Gaussian process with zero mean and a given variance σ^2 . The channel gain at a given time $h_{i,j}(t)$ follows a Rayleigh distribution. $C_{i,j}$ is the cost (power per bit) for maintaining connection i in an interference limited environment. The total average power required by connection i operating at bit-rate $R_{i,j}$ is given by:

$$P_r = P_t \times \frac{G_r}{L} \quad (3.2)$$

3.2.1 Propagation Model

The relationship between the received power P_r and the transmitted power P_t is given by:

$$P_r = P_t G \frac{10^{\frac{L}{10}}}{L} \quad (3.3)$$

Where L is the path loss, $10^{\frac{L}{10}}$ accounts for the loss due to slow shadowing, G being a normal variate with zero mean and $\frac{1}{L}$ represents the gain, with an exponential distribution of unit mean, due to fast fading. The path loss is expressed as:

$$L = 128.1 + 37.6 \log_{10} r \quad (3.4)$$

Where r (in meters) represents the distance between the mobile and the base station. Furthermore the shadowing standard deviation is assumed to be 5dB.

3.2.2 Number of Channels

The number of channels is calculated using the Erlang B formula for loss probability. The calculation is done with Erlang B calculator. The formula is given as follows:

$$P_B = \frac{\frac{A^N}{N!}}{\sum_{n=0}^N \frac{A^n}{n!}} \quad (3.5)$$

Where, k is the number of busy channels, N is the number of available channels, A is the traffic density in Erlang, and B_N is the blocking probability.

This shows that drop-call probability decreases as the number of channels increases.

3.2.3 Channel Utilization

The channel utilization depends on the total traffic, call arrival rate and it is given as:

$$U = \frac{A(1 - B_N)}{N} \quad (3.6)$$

3.3 Traffic Parameters

In this section, the basic principles of some of the key drop-call probability related traffic parameters will be discussed. They are as follows:

3.3.1 Call Arrival Rate

Call arrival rate, λ , refers to the traffic offered expressed as the number of call attempts per unit time [12], which in this case is given as:

$$\lambda = \frac{A}{B_N} \quad (3.7)$$

To relate call arrival rate to the performance of a network, the term grade of service (GOS) denoted by B is used. The GOS can be mean proportion of time for which congestion exists, or probability of congestion or blocking probability, or probability that a call will be dropped due to congestion. It is defined as:

$$P_d = \frac{\lambda \mu}{\lambda \mu + 1} \quad (3.8)$$

From (3.8), it is obvious that drop call-probability varies inversely with call arrival rate, that is, drop-call probability decreases as call arrival rate increases. This leads to the deduction that system performance improves as the traffic entering the system increases.

3.3.2 Call Duration

Call duration is another parameter that can affect the quality of service in a cellular network, hence it is considered when planning the network. Call duration or mean call holding time is defined as the time a mobile station takes to complete a call connection. Mathematically, call duration is given by

$$D = \frac{1}{\lambda} \quad (3.9)$$

Where λ = call arrival rate λ = traffic intensity in Erlangs,

Thus call arrival rate varies with call duration the same way it varies with drop-call probability. Thus drop-call probability increases with a decrease in call duration.

3.3.3 Drop-Call Probability

The drop call probability is given as:

$$P_d = \frac{(\lambda \mu)^n}{n!} e^{-\lambda \mu}, \quad \lambda \mu \geq 0 \quad (3.10)$$

Here, λ is the drop-call rate, t the call duration, while Y is a random variable that counts the number of drops and n is the confirmed calls dropped. This is a Poisson Probability function with a discrete variable which counts the number of dropped calls. The number of dropped calls is calculated from the relation:

$$P(Y=n) = \frac{\lambda^n e^{-\lambda}}{n!} \quad (3.11)$$

The probability of occurrence of the call dropping event (drop-call probability) based on the above formula is thus calculated using the Poisson mass probability function from Microsoft Excel.

3.4 Call Admission Control

When a new call arrives in the system, it needs to check whether to accept the call or not. At first the system has to examine whether the new call is going to degrade the quality of the ongoing calls or the planned coverage area. If it attempts to make degradation in the system, then the system should block the call. In order to maintain the required quality of service of the new incoming call, there are three parameters that have to be checked: required SIR, inter cellular interference, intracellular interference. Based on these parameters the system admits the call in a selective way that does not affect the ongoing calls. This decision making part of the UMTS network is called the call admission control (CAC). It belongs to the radio resource management functionalities of the RNC in a WCDMA based UMTS network.

The call admission control strategy adopted for this work is power based admission control. In this scheme, as soon as a new user seeks admission into the system, the power control module makes the total power distribution of already existing users available to the call admission module, as input for its decisions. The call admission module now estimate the increase in the receive/transmitted power in the cell and in the neighboring cells expected if the new user were to be admitted, and determine whether such increase is acceptable. Priority is given to handoff request, since users are more sensitive to dropped calls than blocked calls.

Let P_{tot} be the total power of the existing users, as estimated by the power control module and P_{th} be the power threshold set in the system.

Let ΔP be the increase in power levels provoked by the new user seeking admission and B_{max} be the maximum overload of non-real time traffic class.

Accept a new call request.

If it is a handoff call, then

If $P_{tot} + \Delta P \leq P_{th}$

Admit the request

If $P_{tot} + \Delta P > P_{th}$

Degrade the existing non-real time users as per their specifications

If still $P_{tot} + \Delta P > P_{th}$

If $P_{\text{tot}} + \alpha P < B_{\text{max}}$

Admit the request

If $P_{\text{tot}} + \alpha P > B_{\text{max}}$

Reject the request

If it is a new call

If $P_{\text{tot}} + \alpha P \leq P_{\text{th}}$

Admit the request

If $P_{\text{tot}} + \alpha P > P_{\text{th}}$

If $P_{\text{tot}} + \alpha P < B_{\text{max}}$

Admit the request

Otherwise

Block the request.

The Service Degradation Descriptor (SDD) proposed in [15] was adopted in this work.

The SDD is a number between 0 and 5; the larger the SDD is the more willing is the user to be degraded, and eventually dropped. The call admission control algorithm is present in a flow chart as shown in figure 3.2.

The algorithm is illustrated with the flow chart below:

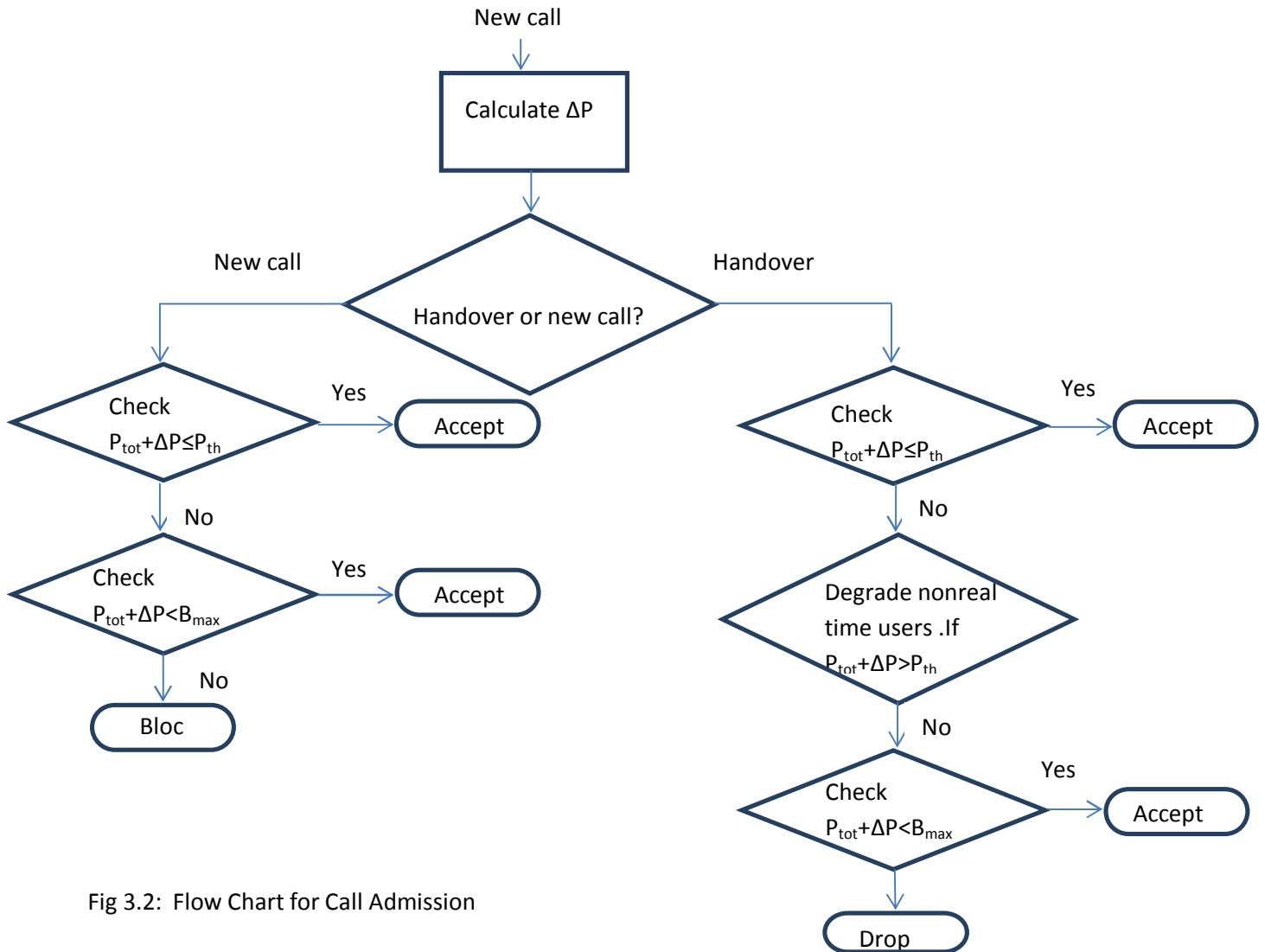


Fig 3.2: Flow Chart for Call Admission

3.5 Power Control

The Power control is an important feature of any CDMA based cellular system. Without proper adjustment of power control the system could be downgraded. There is some common phenomenon in WCDMA system like fading of the signal, intracell interference and random nature of the wireless channel that deteriorate the signal quality to achieve

the required QoS. And the most important issue is the near far problem that is mobile stations that are closer to the base station can dominate the mobile stations that are far away from the base stations. In order to mitigate these problems power control is necessary in WCDMA. Power control mechanism makes the power level of each transmitter at a certain level that it could not make interference to others transmission. As a result the capacity and also the throughput of the system are increased. The power control adopted for this work is the SIR-based power control. To manage the power control mechanism WCDMA uses two different type of mechanism:

(a) Open Loop Power Control (OLPC)

(b) Closed Loop Power Control (CLPC)

Open Loop Power Control (OLPC): Open loop power control adjusts the power mainly in the uplink of the communication. In this technique the User Equipment (UE) estimates the transmission power based on the received pilot signal strength from the base station. It also gets the information of the maximum allowed transmitting power. So that the UE estimates how much power it needs to increase to maintain the required QoS. Figure 3.3 illustrates the OPLC applied in the uplink. In this case the UE estimates the transmission signal strength from the received power level of the pilot signal from the base station and make the adjustment of the transmitting power in a certain level that is inversely proportional to the pilot signal strength. Consequently the stronger the received pilot signal, the lower the UE transmitted power.

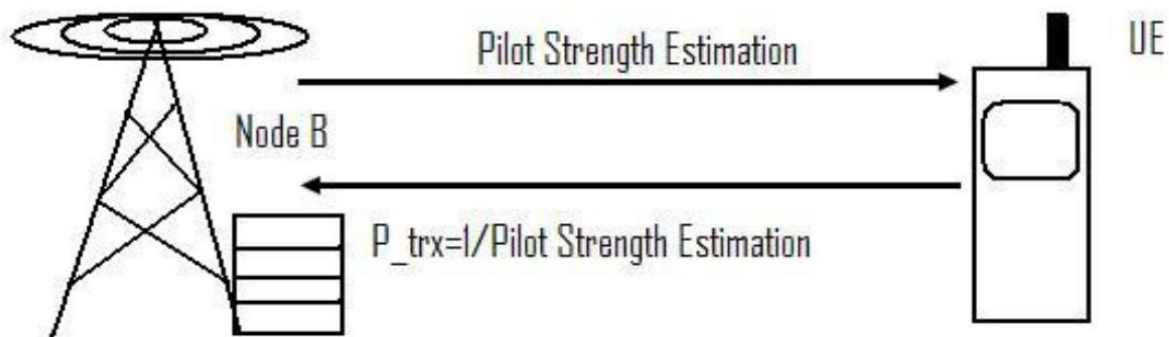


Figure 3.3 Open Loop power Control [32]

Close Loop Power Control: Closed loop power control adjusts the power level of the radio signal when radio connection has already been established. Its main task is to compensate the effect of the rapid changes of the in the radio signal strength. It is used both in the uplink and the downlink. Figure 3.3 illustrates the operation of a closed loop power control (CLPC) in WCDMA. In this case the Node B commands the UE to increase or decrease the power level of the signal in order to maintain the required QoS. The Node B periodically examine the power level of the signal and sends commands to the UE at a rate of 1.5 KHz (1500 cycles per second) to upgrade or downgrade its power level in a step size of 1, 2 or 3 dB. The Node B take this decision with the help of the received SIR of the signal and compare it with the predefined threshold. There are four parameters that are closely related with the decision of power control. The signal strength, received SIR, frame error rate (FER) and bit error rate (BER).

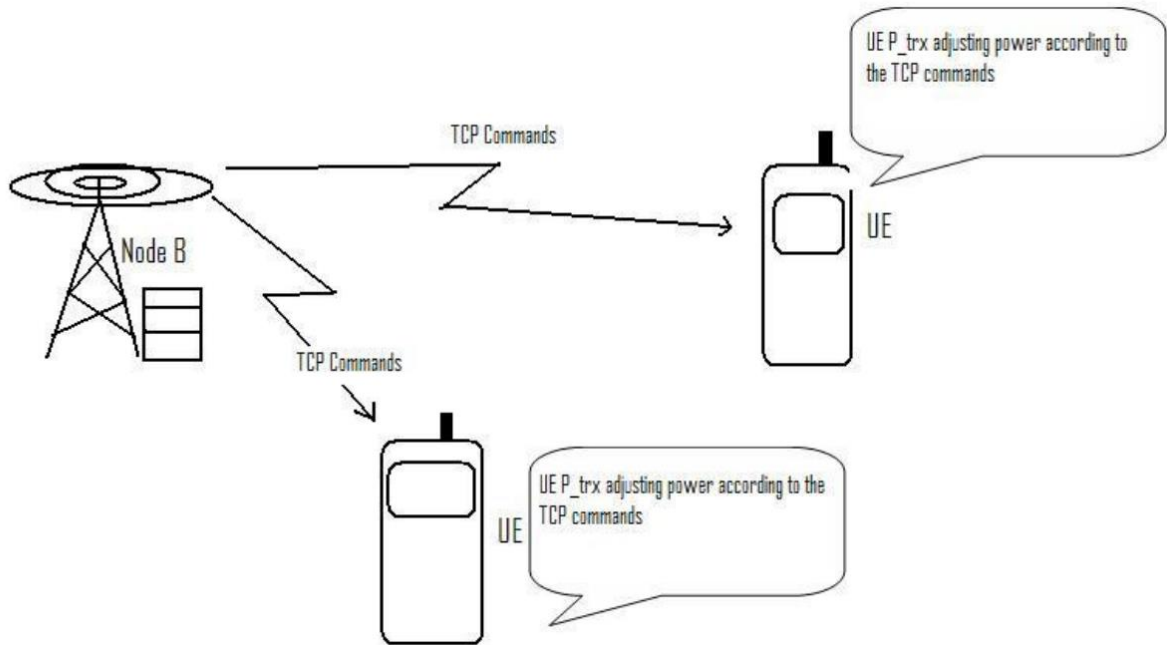


Figure 3.4: Closed Loop Power Control [32]

3.5.2 SIR-Based Power Control

The goal of the power control is to assign minimal transmit power to the forward and reverse link such that every user experiences maximum signal-to-interference ratio. For a user equipment requesting access to the system, the received SIR is estimated and compared with the target SIR. A request for increase in the transmitted power is sent to the mobile if the estimated SIR is below the target SIR, while a request for decreasing the transmitted power is sent to the mobile if the estimated SIR is above the target SIR.

The SIR experienced at each measurement point is estimated as follows:

$$SIR = \frac{P_{rx} - N}{I} = \frac{P_{rx} - N}{I_{intra} + I_{inter}} \quad (3.12)$$

Where I = Intra cell interference(I_{intra}) + Inter cell Interference(I_{inter})

The transmitted power of each mobile station is adjusted at each iteration to maintain target signal to interference ratio, $SIR_{k,target}$ for class k. The new power control level is executed every 10ms and the new power level is evaluated as

$$P_{k,new} = P_{k,cur} \frac{SIR_{k,target}}{SIR_{k,cur}} \quad (3.13)$$

where $SIR_{k,cur}$ is the current SIR experienced by the mobile station of class k. The actual SIR experienced by each user, are evaluated after each power control iteration. Power request that cannot be satisfied may result in outage. To compensate for power control error, the SIR target value is taken to be 1dB greater than the required value. After each power control iteration the actual *SIR* values experienced by the each user is evaluated. If the *SIR* is lower than the target value, $SIR_{k,target}$, the system increases its transmitted power provided sufficient Base Station Power is available. If the system resources are insufficient, that call is dropped. The users with the lowest SDD values and Real-Time services are considered first. The users with high SDD values and Non-Real Time services are considered last. If a user has a *SIR* value higher than the transmit power allocated to that user is reduced. The Downlink Power Control mechanism is summarized in a flow chart as shown in Figure 3.5. Note that call termination due to user movement out of the cell is considered a completed call.

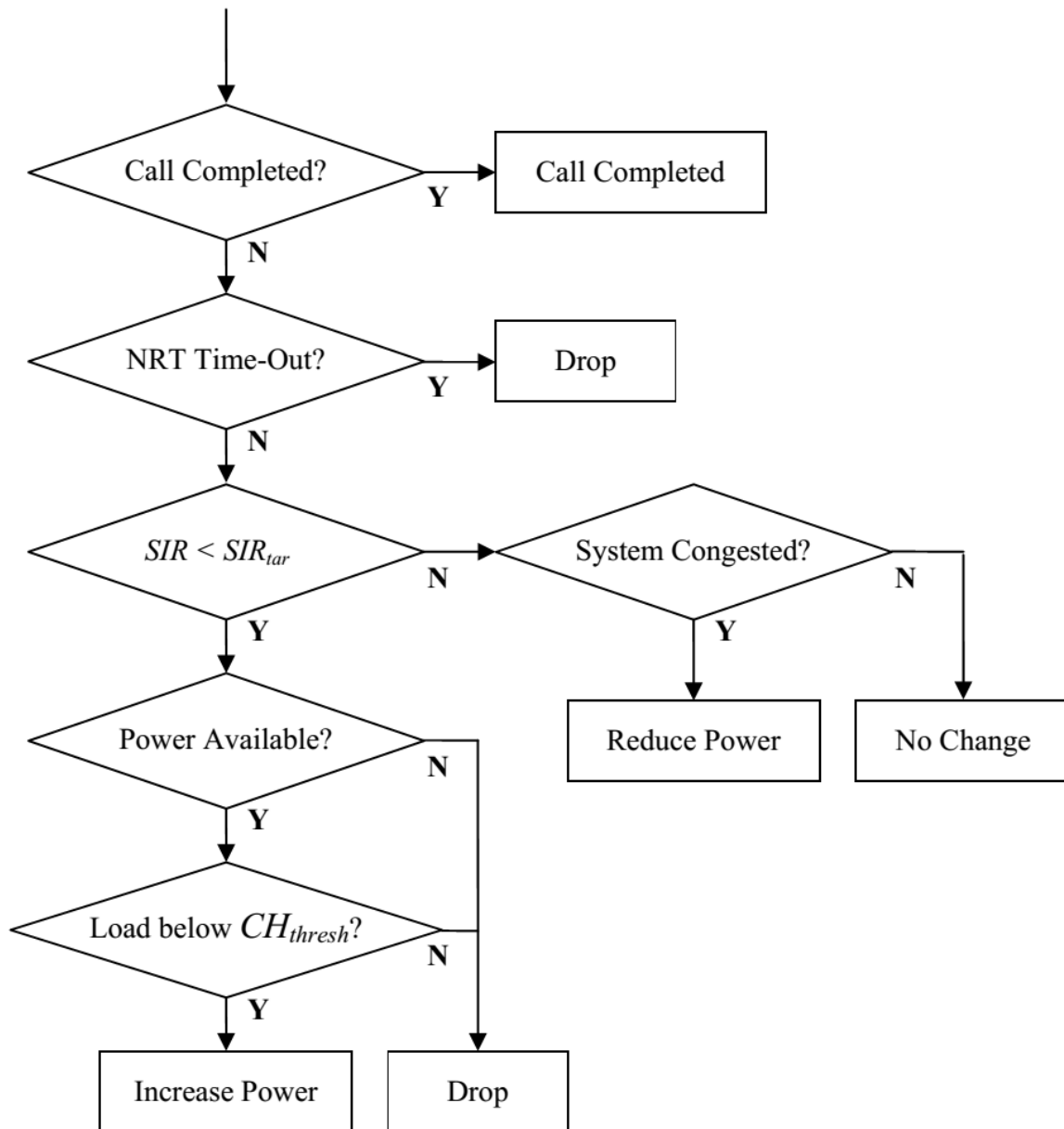


Figure 3.5: SIR-Based Power Control Mechanism

3.6 Handoff

Whenever a mobile user having an active call crosses a cell boundary, the active call necessitates transfer from its current base station to another one in order to avoid call discontinuation. This process is known as a handoff and concerns the transfer of the current communication channel of an active call from one base station to a new one.

If the new base station has unoccupied channels the handoff call is assigned to one of them. When all channels in the new base station are in use, there are two possibilities: handoff failure which leads to call dropping, or handed off call queuing while waiting for possible channel availability by another call completion or handoff in the new cell. Generally, a handoff scheme's objectives are to minimize call dropping rate while not significantly increasing call blocking. The handoff process operates in the three steps i) Measurement ii) Decision and iii) Execution.

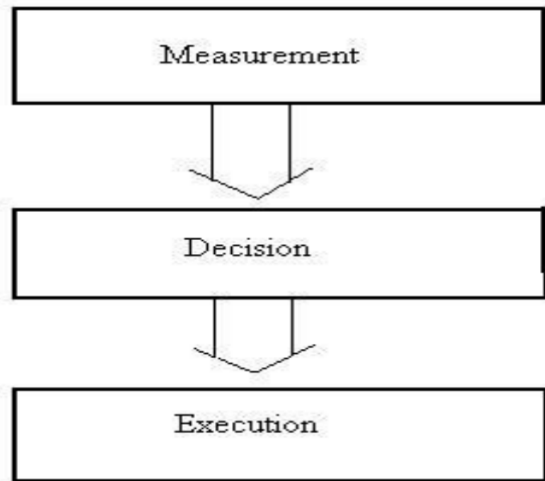


Figure 3.6: Handoff Process

i) **Measurement:** The measurement task in hand over is important for the system performance. In this phase the system measures the signal strength as it is vary drastically due to the fading and the signal path loss. And the excess of the signal strength in the Node B causes the overall system downgrade which is not desired. For the handover purpose the UE continuously measures the signal strength concerning the neighbouring

cells and reports to the RNC. According to the 3gpp technical specification TS 25.331, the measurements of the UE categorised into different groups.

- Inter frequency measurements include the signal strength in the downlink of the communication with different frequencies.
- Intra frequency measurements include the signal strength in the downlink of the communication with same frequencies.
- Inter System measurements means the downlink signal strength in other radio access network like GSM.
- Traffic Volume measurement, measures the uplink traffic of the network.

According to the above measurement the UE periodically reports to the RNC.

ii) **Decision:** Getting the all measurement information from the previous step the system asses the required QoS for the connection and compare it with the requested QoS. Depending on this comparison the system take the decision whether the handover will be occurred or not.

iii) **Execution:** In this phase the Serving RNC checks whether the reports generated from the measurement phase trigger any criteria. If it satisfies the triggering criteria then it executes the handover.

The general procedure for the handover is illustrated in figure 3.7. In this example the decision making criteria is based on the pilot signal strength, which is reported by the UE. The following term and parameters are used in this handover algorithm.

- ✓ Upper threshold: The maximum allowed signal power level according to the required QoS.
- ✓ Lower threshold: The lowest minimum signal power level for maintaining the connection with QoS.
- ✓ Handover Margin: It's a predefined value at which the signal power of the neighbouring cell starts exceeding the power of signal of its own cell.
- ✓ Active Set: It's a list of signal branches through which UE simultaneously connected with UTRAN.

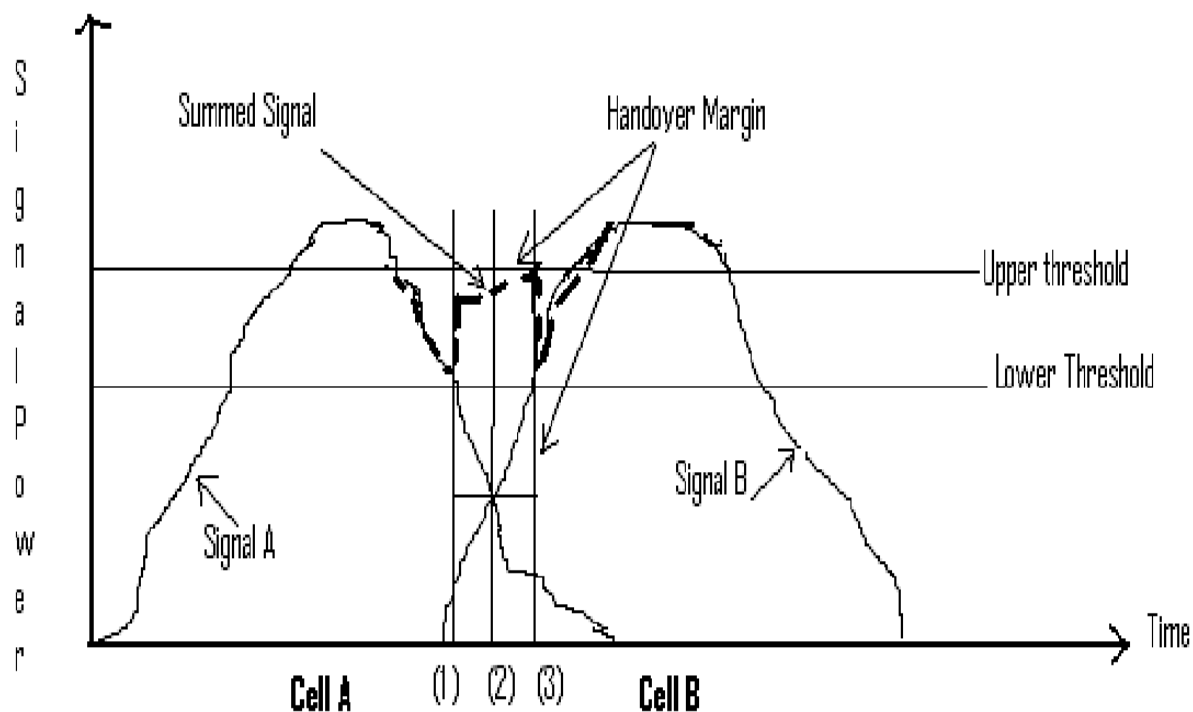


Figure 3.7: Handoff Control [32]

Let us assume that in this example an ongoing UE moving from Cell A to Cell B. As the UE moves towards the Cell B, the signal strength with Cell A going lower approaching to

the lower threshold. In this stage there must be a handover which is indicated in the figure with 1 2 and 3. 1. In the first steps the signal of the Cell A equals to the lower threshold. This is reported to the RNC by the UE and by this time the RNC also gets the information from the neighbouring Cell B that this signal is improving so that the RNC place that signal in the active set. Upon this event the UE has two simultaneous connections to the UTRAN. As a result we get summed signal of Cell A and Cell B. 2. In this stage the RNC noticed that the signal of the Cell B started improving than the signal from Cell A. RNC marks this point as a margin point for handover calculation. 3. The strength of the signal B becomes equal and started better than the lower threshold so that the signal B can carry on the ongoing call. On the other hand the summed signal exceeds the upper threshold. As a result the RNC deletes the Signal A from the Active set.

3.7 Congestion Control

Even with efficient Admission Control, congestion could still be caused, mainly by users moving from one area within the cell to another area. When affected by congestion the output powers are rapidly increased by the fast closed-loop power control until one or several transmitters are operating at their maximum power. The connections unable to achieve their required *SIR* levels are considered to be useless and are only adding interference to the system. Therefore a procedure to reduce congestion by removing such users is required. Congestion can be controlled with several methods.

- Lowering the bit-rate of one or several services that are insensitive to increased delays.
- Performing inter frequency handoff.

• Removing useless connections.

The same measurements used for Admission Control can be used for Congestion Control. But these measurements have to be updated regularly because the considered values change rapidly during heavy congestion. Once the connections to alter are identified Congestion Control schemes can be utilized.

CHAPTER FOUR

IMPLEMENTATION AND PERFORMANCE EVALUATION

4.1 Model Implementation

This chapter presents the simulation and the simulation results for this work. Chapter three presented the model development. The system model considers the effect of Additive White Gaussian Noise AWGN and the effect of Rayleigh and/or Nakagami multi path fading. The specific simulated environment and the performance metrics used to measure and compare performance are also described. It also looked at the performance comparison of some of the resource allocation schemes that directly impacted on the call drop probability. The performance evaluation was done using MATLAB simulator, the comparison was made under the condition of multi-traffics WCDMA system. The simulation was conducted using the parameters given in Table 4.1. The performance metrics considered during the simulation were presented in the following section.

4.2 Simulation Testbed

This section describes the simulation testbed used for the simulation of the physical layer of a wireless communication system designed to evaluate the selected call drop minimization techniques for a Wideband CDMA (WCDMA) system. The simulation was implemented in MATLAB using the parameters and mathematical models discussed in chapter three, the simulation was done in MATLAB platform to take advantage of the

built-in functions that perform various signal processing applications like filtering and Fast Fourier Transformation (FFT) and the superior array handling capability. The simulated wireless communication system (Figure 4.1) consists of a transmitter, a receiver and a channel. The transmitter consists of a burst builder (which has the actual payload), modulator, up-sampler and filter. The receiver on the other hand, consists of a matched filter, down sampler, demodulator and a BER calculator. The noise added can be modeled as AWGN. The channel should take fading effects into consideration. Data is transmitted in a frame by frame basis over a time varying multipath channel. Receiver design incorporates rake diversity combining. Additive White Gaussian Noise (AWGN) is added at the front end of the rake receiver. Multiple Access Interference (MAI) is generated in a structured way rather than treating it as AWGN, then the various call drop minimization schemes are evaluated and the results obtained were analyzed.

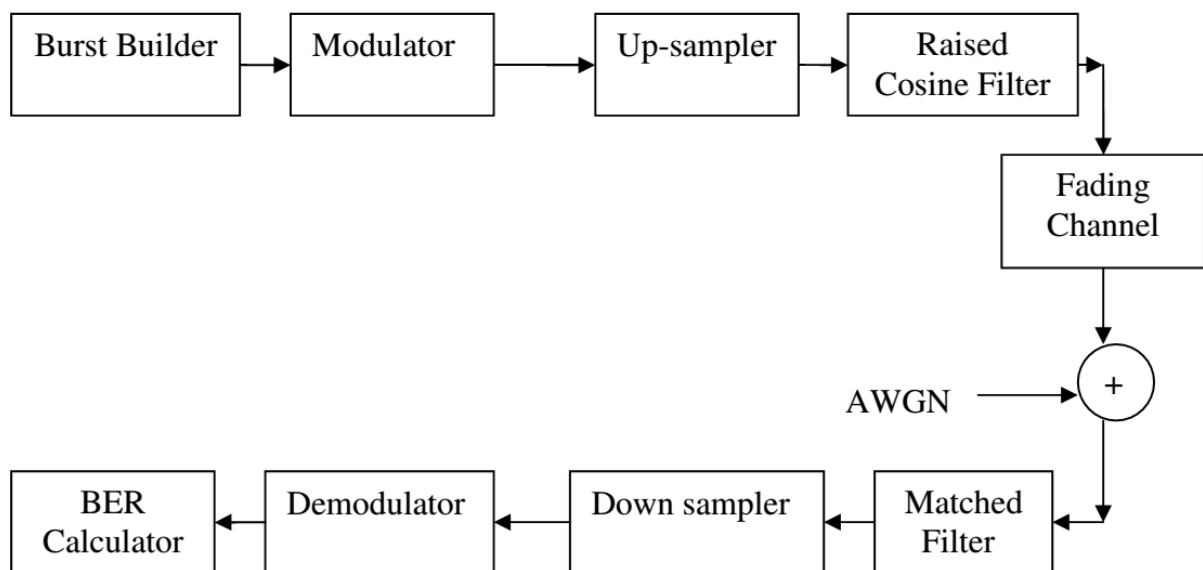


Figure 4.1 Physical Layer simulation of a WCDMA Communication System

The information or the payload (digital version of the baseband signal) transmitted at the physical layer of the communication system is in the form of bits. If voice is to be transmitted, then it is first converted to bits, modulated, up-sampled and passed through a raised cosine filter and then transmitted. But before modulating, sometimes a burst is created. The burst consists of fixed guard bits and payload. The fixed guard bits are at the beginning and the end of the burst to avoid errors due to overlapping between the two adjacent bursts. The guard bits are discarded or removed after demodulation. The information bits coupled with guard bits are input to the modulator. The base station receiver receives signals from all the mobile station transmitters and demodulates them. The signals from different MS follow different channels to reach the base station as discussed in chapter three. MAI is implemented by generating the signals for a number of interfering MS within the system. Each interfering user has its own control channel and one data application. Each of the interfering mobile station generates its transmitted frame in the same manner as the desired user. However these frames are not aligned in time with each other or with that of the desired MS.

4.3 Performance Measures

System availability is the main performance metric considered in this work. This is a very important parameter that is fundamental in system analysis. The outage probability is one of the performance measures adopted for this work which is described as the probability that the signal to noise ratio (SNR) at the output is below a given threshold value (γ_{th})-outage threshold value. The outage threshold is a limit value for the SNR that measures

the quality of service, QoS. This QoS parameter simulated and analyzed as outage performance metric since it can be applied for different QoS requirements. In all cases the target is to increase the performance of the system, by decreasing the outage probability and thereby reducing the blocking probability and dropping probability. In order to simulate the system, the parameter values in table 4.1 are configured into the system.

Table 4.1: Simulation Parameters

Parameter	Value
Chip rate, W	5MHz
BS thermal noise, N_o	-174dBm
Data rate, R_k	144Kbps,253Kbps,384Kbps
Voice activity factor, v_k	0.65, 1
Signal bit energy over noise spectral density, E_b/N_o	3 - 7dB
Multi user detection factor, β	0.78
Transmitting antenna gain, G_{tx}	11dBi
BS antenna height, Δh_b	15m, omnidirectional
MS transmitted power signal	21-30dBm
Maximum cell load, n_{max}	0.8
Values of other-cell interference, I_{other}	$5 e^{-19}mW$ - $6e^{-18}mW$
Offered load, α_k	0.2 – 2.0 Erlang
Loading factor, L_k	0.05, 0.1, 0.15, 0.2, 0.25
Maximum number of states, i_{max}	4
Number of service class, k	3

4.4 Power Control Implementation and Analysis

The steps of operation in the SIR-based power algorithm are listed below as discussed in chapter three.

1. The mobile stations measure the observed value of the SIR at each iteration and compare it with the preset lower and higher critical threshold values.
2. If the observed SIR is smaller than the lower critical threshold, then the mobile sends a power up command to the base station. The first power update command is interpreted as a fixed step modification; however, the step size is adapted dynamically if successive feedback commands request additional change in the power level in the same direction.
3. If the observed threshold is between the lower and the higher critical threshold values, then the mobile does not send any control signal to the base station.
4. The increment step size is chosen larger than the decrement step size. This ensures that mobiles in outage can quickly come out of outage.
5. Once the mobiles are out of outage and not in the buffer region, the smaller decrement step size brings the mobile into the buffer region. As pointed out in step (3), when the mobile is in the buffer region, it does not send power change commands.

The pseudo-code for the simulation is as follows. The complete Matlab code is included in Appendix A.

Pseudo-code

- 1. Initialize number of iterations*
- 2. Initialize number of mobiles*
- 3. Generate uniformly distributed vector of mobile-to-base station distance*

4. Generate vector of link-gains from each mobile to the base station based on a log-normal distribution with zero-mean and standard deviation = 6 dB

5. Allocate initial power for each mobile at the base station

6. Initialize $SIR_{threshold} = -14$ dB

7. for $i = 1$ to iterations

for $j = 1$ to mobiles

- Calculate $SIR_{observed}$ at each mobile
- Compare $SIR_{observed}$ with $SIR_{threshold}$
- if ($SIR_{observed} < SIR_{threshold}$
- Increment outage counter
- Adjust (increase) power by fixed step
- else
- Adjust (decrease) power by fixed step
- end

end

8. Calculate the outage percentage using outage counter and the number of mobiles

9. Plot outage percentage versus number of iterations

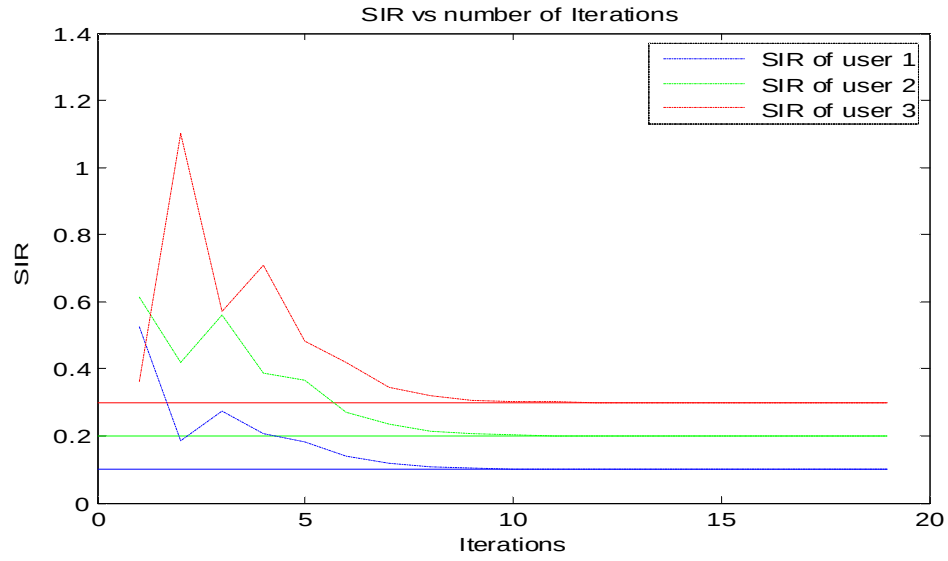


Figure 4.2: The plot of SIR against the number of iterations

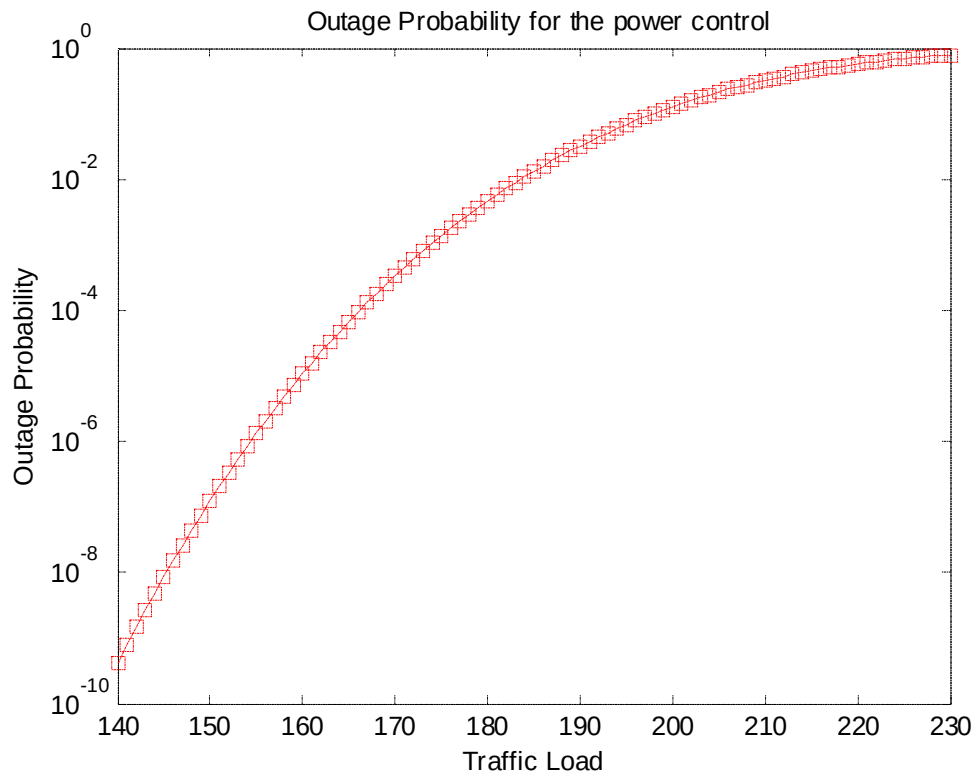


Figure 4.3: Outage probability of SIR-based power control

Figure 4.2, shows the SIR of the users against the number of iteration while figure 4.3 shows the outage probability of the power control. The SIR-based power control algorithm when implemented in the system was able to reduce the outage probability and thereby reducing call drops according to equation (3.10) and improving the QoS of the system as against a WCDMA system without power control. In such a system, the transmitter transmits equal power to all the users since it does not have any information about their power requirement thereby introducing more interference in the system which in turn leads to poor QoS of the system. Equally, as seen in figure 4.4 the plot of a WCDMA network with power control showed improved system capacity than the system with equal power allocation. From (3.8), it is obvious that drop call-probability varies inversely with call arrival rate, that is, drop-call probability decreases as call arrival rate increases. This leads to the deduction that system performance improves as the traffic entering the system increases. The improved system capacity implies that more users can be accommodated and thereby reducing call drop in the system.

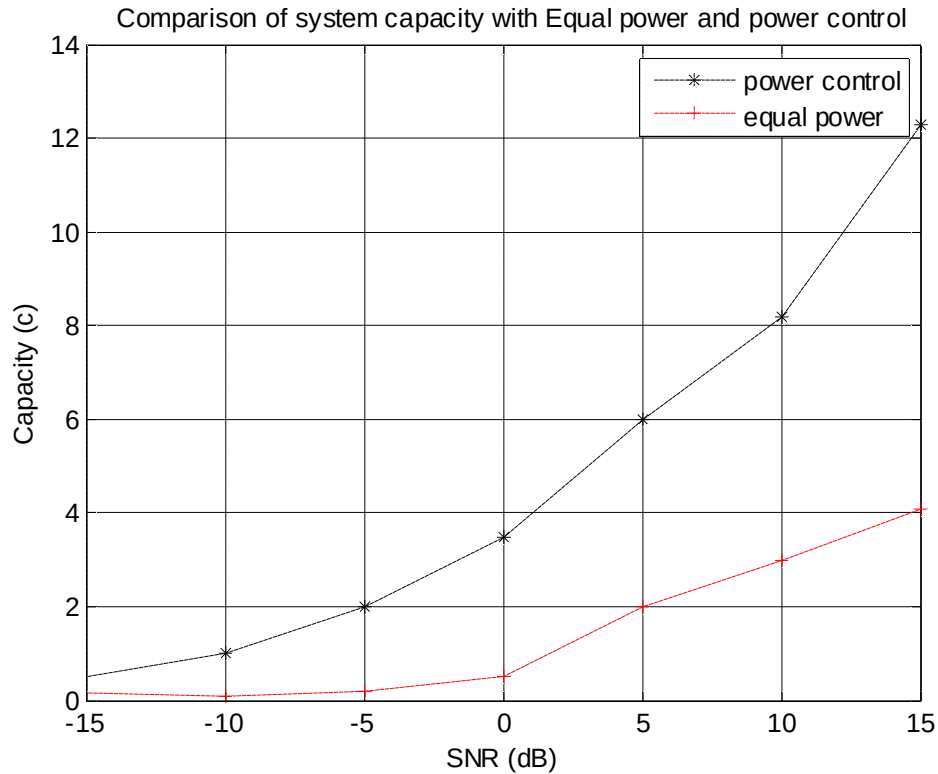


Figure 4.4: Comparison of system capacity

4.4 Handoff Implementation and Evaluation

The WCDMA network is simulated for situations without handoff channel (No priority Scheme (NPS)) and that with a reserved channel (Reserved Channel Scheme (RCS)) for handoff. The service intensity in a system is $(\lambda + \eta)$. Service intensity for handoff call is η and for cell with no handoff is λ . This simulation considered that the total number of channels $s=10$ both for NPS and RCS. As mention above NPS has no reserved channels, therefore the number of users in a cell is fixed. On the other hand, there is scenario in RCS with reserved channels (C_h). For all the cases, there is mean call holding time for handoff call is $\frac{1}{\mu} = 3\text{minutes}$ and mean call holding time for call with no handoff is $\frac{1}{\mu_h}$

also 3minutes. The pseudo-code for the simulation is as follows. The complete Matlab code is included in Appendix B.

Pseudo-code

Step1: Define Total available channels in system, Total reserved channels for Handoff.

Step2: Define Call service intensities 'mu' and 'eta' to calculate value of λ_o , and offered traffic.

Step 3: If $ch = \text{reserved channel}$ then, $\lambda_{handoff} = 0$ else $\lambda_{handoff} = 0.2 * \lambda_o$.

Step 4: Calculate call arrival intensity and initial blocking probability of system.

Step 5: Run the loop for 1000 seconds to get different points of blocking probabilities and dropping probability.

Step 6: Sum the blocking probabilities to get total blocking probability and dropping probability.

Step 7: Find the probabilities of call not completed which depends on call blocking probability, and call force termination probability that depends on dropping probability.

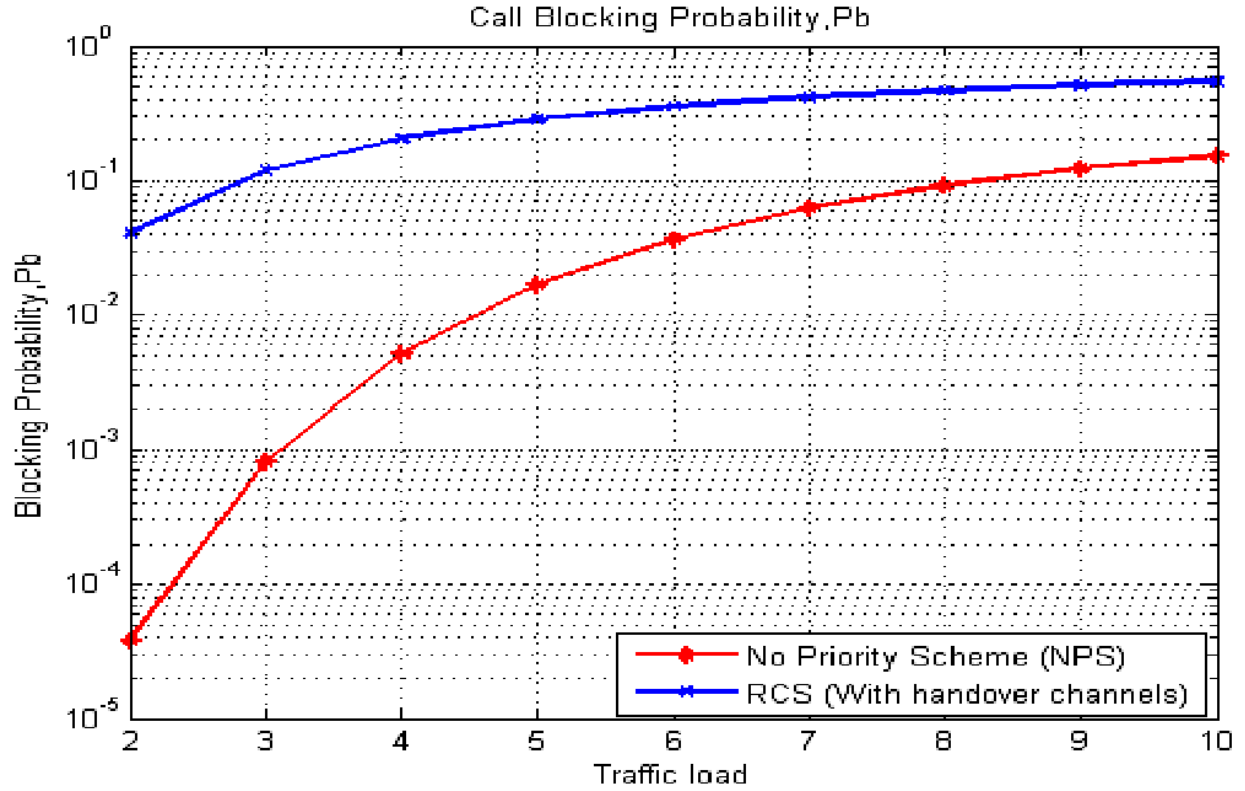


Figure 4.5: Call Blocking Probability for Handoff

In case of RCS, as the number reserved channels decrease the blocking probability of the system decreases gradually as shown in Figure (4.5). It means when there is a decrease in the number of reserved channels, decrease in new calls probability can be seen and few more channels are available (from equation 3.9) to accommodate new calls in system, but at the same time, there is an increase in dropping of handoff calls. In case of NPS, blocking probability P_b remains same due to lack of reserved channels. From simulation, results showed above both in NPS and RCS cases blocking probability P_b increase with increase in number of busy channels. However, the greatest advantage in RCS scenarios is that they have guaranteed channel allocation for handoff users, this in turn bring about reduction in blocking probability thereby leading to a reduction in call drop probability.

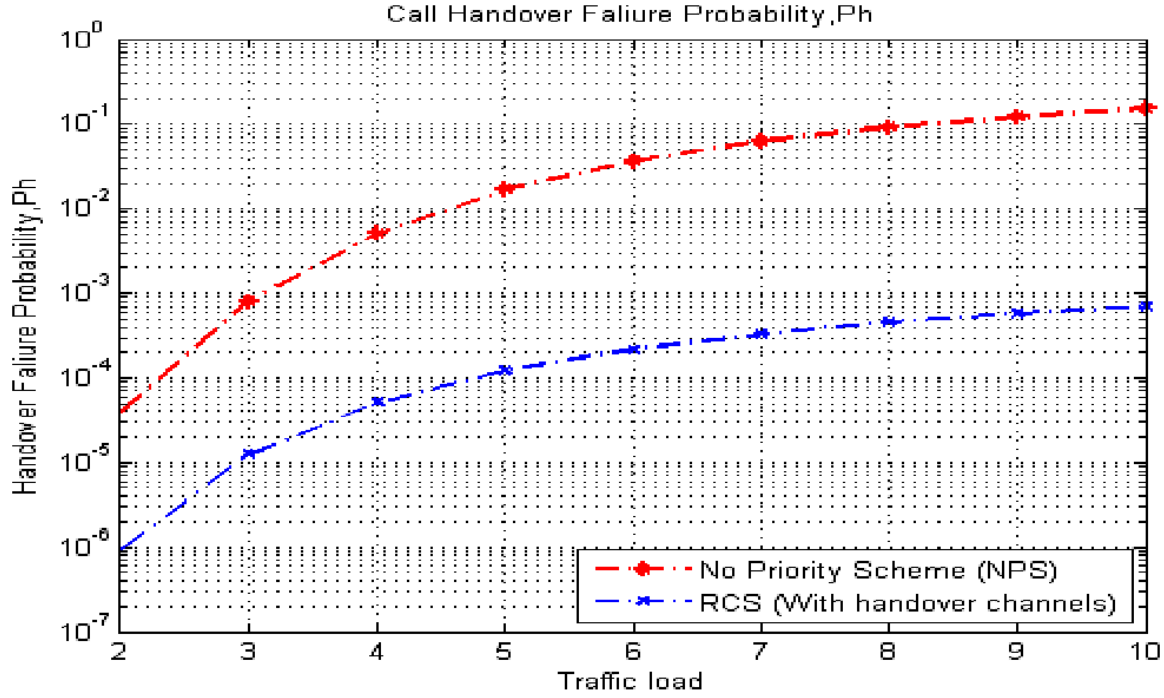


Figure 4.6: Handoff Failure Probability

In case of RCS, decreasing the number of reserved channels call dropping of handoff calls probability of system increase gradually that can be seen in Figure (4.6) clearly. It means decreasing number of reserved channels we have increase in handoff failure probability P_h of calls, so we face the situation of call drop due to unavailability of guard channels. In case of NPS, P_h remains same due to lack of reserved channels. From results of the simulation showed above for both NPS and RCS cases, handoff failure probability P_h increase with increase in number of busy channels. This implies that for call drop probability to be reduced in a system that adopts the RCS scheme, the reserved channels should not be reduced this will enhance the handoff success rate and in turn reduce the outage probability.

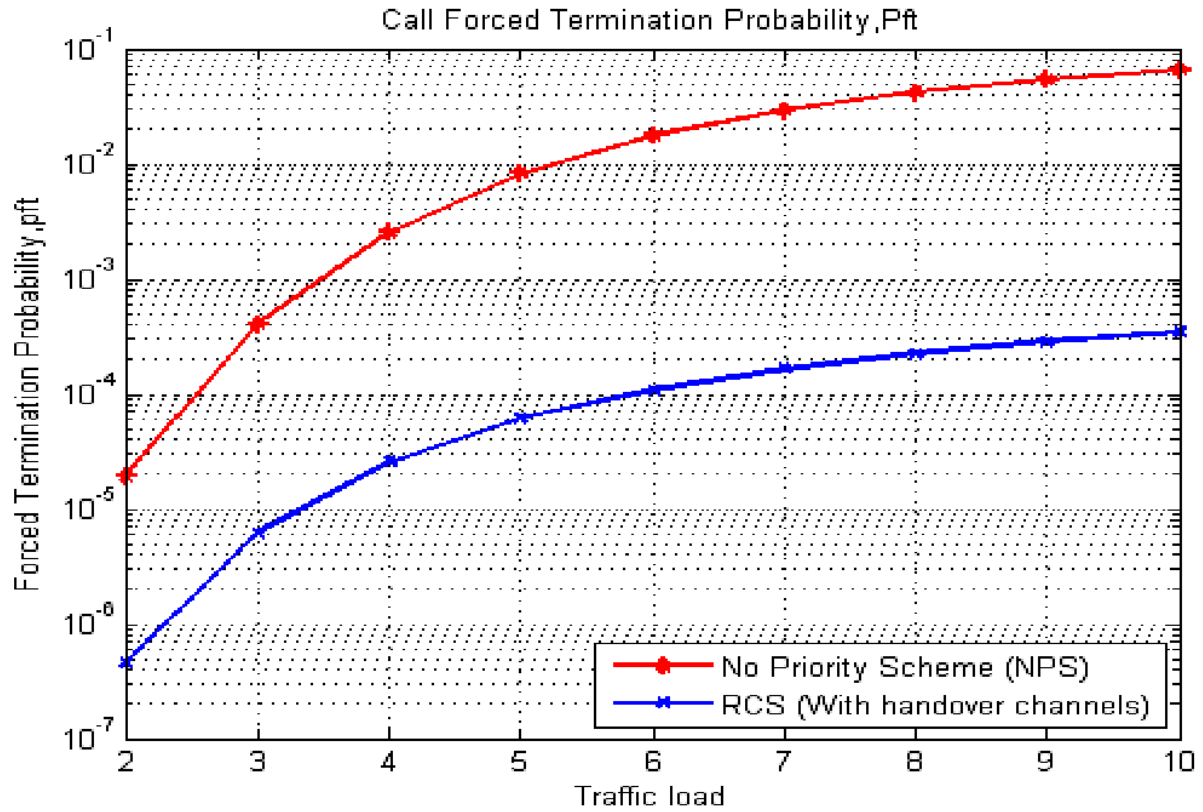


Figure 4.7: Forced Termination Probability for the Handoff Scheme

Force termination P_{ft} means a handoff failure, when a call returns to its original call originating cell after successful handoff to neighboring cell. Call force termination probability depends on occurrence of P_h . In case of RCS, decreasing the number of reserved channels Force termination probability of system increase gradually as shown in Figure (4.7). It means that, when we decrease number of reserved channels we have increase in Force termination probability P_{ft} of calls. The results for P_h and P_{ft} are almost same because it depends on P_h . From simulation showed above both for NPS and RCS cases, P_{ft} increase with increase in number of busy channels.

RCS should be use in cell where the number of user is high in system as number of users is high relative mobility increases and the chance of handoff call are more. Handoff failure reduced in RCS as compare to NPS. Accommodating handoff call in RCS plays important role in guaranteed call completion that increases significant GoS in cellular networks

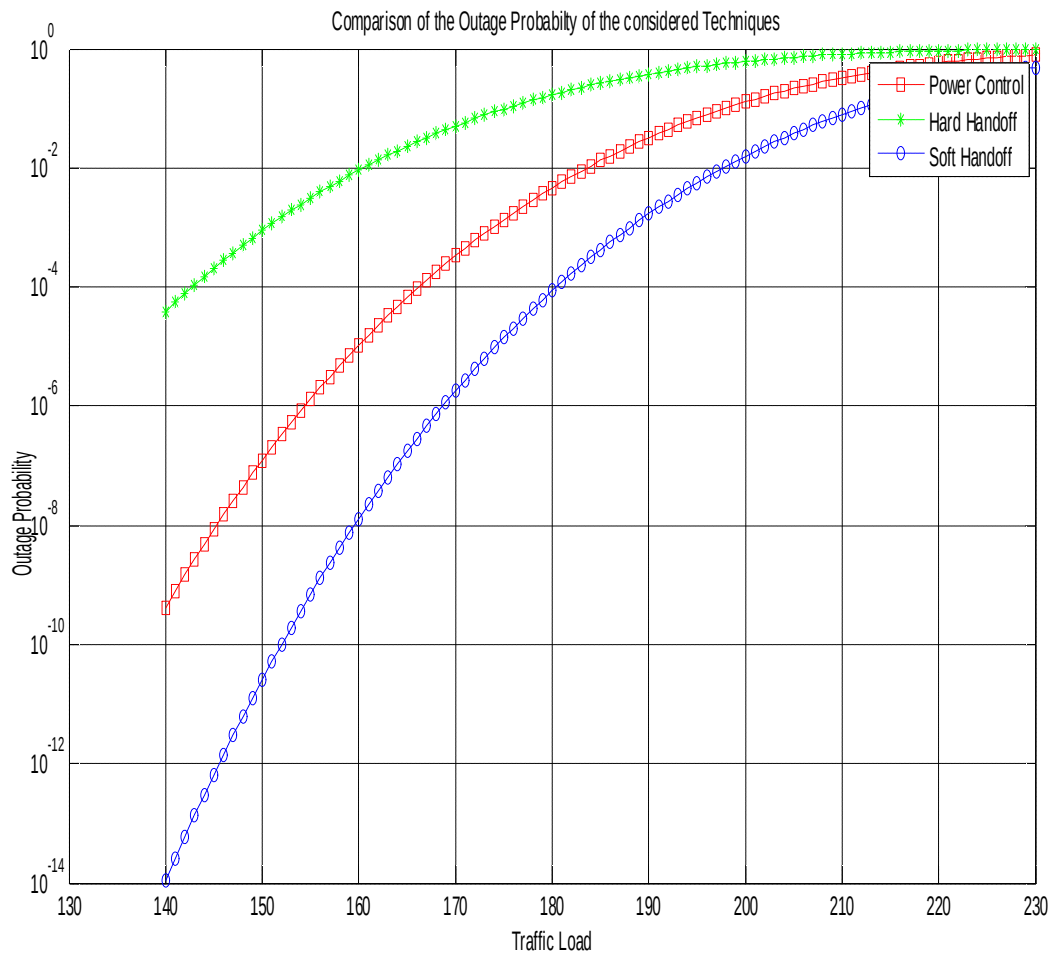


Figure 4.8: Comparison of the Outage Probability of the Various Techniques

Figure 4.8, shows the outage probability of the various techniques considered in this work, the figure shows the extent that the various techniques have succeeded in reducing

the outage probability in a WCDMA network. The outage probability as discussed is an important measure of the call dropping probability of the network and a decrease in the outage probability means a decrease in call drops and an improvement in the QoS of the network.

CHAPTER FIVE

CONCLUSION AND RECOMMENDATION

5.1 Conclusion

This thesis presented the call drops problem on the bases of the availability of radio resources and the schemes that improves the QoS of WCDMA network. This work carried out the comparative analysis of the techniques that directly reduce call dropping in a wireless communication system. This work analyzed the performances of resource allocation schemes, such as the SIR-based power allocation and handoff schemes through simulation to get better QoS and minimum blocking and dropping of calls. Each of the above mentioned schemes are implemented separately to check the variations in terms of call admit, outage probability, blocking probability and call handoff rejection. The cell bases call admission control algorithm calculates bandwidth according to the priority level given to arrival call requests. It has also been proved within the scope of this thesis that reserved channel schemes (RCS) based algorithms performs more efficient in high traffic and multi class environments. The resource allocation techniques have been designed to achieve certain objectives in terms of QoS and user needs. The implementation possibility of all the schemes depends on their individual performances and network demands.

5.2 Recommendation

Several other issues have been encountered during the course of the thesis work, which can be studied further.

- ✓ Further research could be done to eliminate the interference from the mobile stations in its own cell.
- ✓ Interference from the neighboring cells could be eliminated.
- ✓ The performance of different algorithms can also be compared on the basis of metrics like BER and FER
- ✓ Taking all the QoS parameters into consideration for efficient radio resource management algorithm.
- ✓ Areas like performance comparison of these combined power control algorithms, power control in MIMO and adaptive antenna systems and power control in WCDMA can be further explored in the light of this thesis and the results can easily be extended also to them.

5.3 Contribution to Knowledge

In this research work, comparative evaluation of call drop minimization techniques for a WCDMA communication system was carried out. The results obtained showed that the resource allocation techniques have been designed to achieve certain objectives in terms of QoS and user needs. The outcome of this research will help the network operators to

know and select the best possible method of reducing call drops in their network and hence improve their quality of service, since it is important for wireless mobile network operators to ensure stability and efficiency of their services by delivering a consistent, reliable and high-quality end user (subscriber) satisfaction.

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APPENDIX A

MATLAB CODE FOR POWER CONTROL

```
% in this program mobiles are uniformly distributed
within%%%%%%%%%
% the cell to allow varying power for the Downlink, each
mobile%%%%%%%%%
% then transmitte according to the received power from
base station.%%

clear all;
close all;
clc;
%%%%%%%%%
%%%%%%%%%
%%in this cell we define the system constant involved in
calculations%%
iterations = 1;
mobiles = 10;
dmin = 200;
R = 1000;
k = 3;
n = 4;
%%%%%%%%%
%%%%%%%%%
%here the DBPA code start to measure SIR of each mobile and
calculate the%
%outage percentage depending on that SIR_threshold= -14
dB%%%%%%%%%

counterinit('outage_counter');
counterset('outage_counter', 0);

for i = 1:iterations;
    distance = unidrnd (R,1,mobiles);

    power_initial = -73 ; % initial power for all mobiles
near to BS

    for j = 1:mobil
```

```

    if distance <= dmin;

        power (j) = k* (dmin/R)^n;
    else
        power (j) = k* (distance (j)/R)^n ;
    end

    SIR_observed (j) = (10*log10 (power (j)));
    SIR_threshold= -14;

end
    if SIR_observed < SIR_threshold
        counterinc('outage_counter');
    end
    disp(counterval('outage_counter'));
end

outage_percentage =
((counterval('outage_counter')/mobiles)*100);

plot(outage_percentage,mobiles)
title('Outage Probability vs Number of Mobiles - Distance
Based Power Control Algorithm')
xlabel('Number of mobiles in cell')
ylabel('Percentage of mobiles in outage')

delta_P = 3;
gain = lognpdf(distance,0,6);

for i = 1:iterations;

    for j = 1:mobiles;

        SIR_observed_1 (j) = (10*log10 (power (j)));

        if (SIR_observed_1 < SIR_threshold);

            outage_1 (i) = (mean(SIR_observed_1 <
SIR_threshold))* (j);
            outage_counter_1(i) = outage_counter_1(i-1) +
outage_1 (i);

```

```

        SIR_increament (j) = SIR_observed_1 (j)+
delta_P;
    else
        SIR_decreament (j) = SIR_observed_1 (j) -
delta_P;
    end

end

end

outage_percentage_1 = ((outage_counter_1 (i)/mobiles)*100);

plot(outage_percentage_1,mobiles)
title('Fixed Step Power Control Algorithm')
xlabel('Number of Iterations')
ylabel('Percentage of nodes in outage')

```

APPENDIX B

MATLAB CODE FOR HANDOFF

```
close all; clear all; clc; tic
%% System parameters <<<
J = 4; % number of subcarriers
K = 3; % number of users in the system
Rd = 1; % original data rate(bps).
Rc = Rd*J; % highest chip rate(cps), as in DS-CDMA.
Rs = Rc/J; % basic data rate on each subcarrier (sps), the
same as Rd.
Td = 1/Rd; % original data symbol period,
Tc = 1/Rc; % chip period
Ts = 1/Rs; % symbol period on subcarrier
df = Rs; % subcarrier difference, distance between
subcarriers, Not Used
Eb=1; % the bit energy of each user
%% Simulation Parameters <<<
% These parameters should be changed before simulation
EbN0dB = 0: 5: 30; % Eb/N0, in dB
NerrDes = 100; % the desired number of error to accumulate
NumMax = 1e4; % the maximum number of transmitted data
symbol
Navg = 1e2; % for a given EbN0, run 'Navg' times for
averaging
%% Simulation
% creat file to save the simulation result
FileName = ['SB_MCCDMA_J4_K3_indepRayleigh',
clockString(5)]; % the name of the file
fip = fopen(FileName,'w'); % creat the new file and open
fprintf(fip,'%d \t %d \t ',length(EbN0dB),EbN0dB);
fclose(fip); % close the file
% Spreading Code Set Selection
H4 = hadamard(4); % WH4 codes
C = [1 1 0; 1 -1 0; 1 0 1; 1 0 -1]; % MC-FCDMA system
H4 = normColumn(H4); % normalization
C4 = normColumn(C); % normalization;
for codeSelection = 1:2 % select different code set in each
loop
disp(['codeSelection =',num2str(codeSelection)]); % show
some text in the commanding windows
```

```

switch codeSelection
case 1
Code = H4; %
case 2
Code = C4;
otherwise
error('error in [switch codeSelection]'); %
end
%initialize parameters
EbN0 = 10.^(EbN0dB/10); % Eb/N0 in linear scale.
ber_bpsk= 1./4./EbN0; % theoretic BPSK BER in Rayleigh
fading.
Num_sym = ceil(NerrDes./ber_bpsk); % Number of bits to
send in one iteration
Num_sym = min(Num_sym,NumMax); % the transmitted bits
actually
BER = zeros(K,length(EbN0)); % average BER of each user,
row k = user k.
% Pb = zeros(K,length(EbN0)); % BER during one iteration
%% transmitting
for inc = 1:length(EbN0) % snr loop begin
disp(['EbN0 = ', num2str(EbN0dB(inc))]); % show some text
in the commanding windows
%=====
=====
for it=1:Navg % number of iterations for the same EbN0, for
averaging
for k = 1:K % for different users
% generate source bits
Bit(k,:) = BIN01(Num_sym(inc),0.5); % user k info bits
% spreading the data with corresponding codes
sd(k,:) = SPREADING(Bit(k,:),Code(:,k)); % spreading data
end;
% superpose and S/P convertor
sdall = sum(sd,1); % superpose, CDMA
sp = reshape(sdall,J,length(sdall)/J); % S/P conversion
Jx()
% add channel effect
% independent fading on subcarriers (selective fading)
% independent fading on different OFDM symbols
% + AWGN in frequency domain
fading = randn(4,Num_sym(inc))*sqrt(1/2)+sqrt(-
1)*randn(4,Num_sym(inc))*sqrt(1/2);

```

```

r_mp = sp.*fading; % add independent Rayleigh fading effect
on subcarriers
N0 = Eb/EbN0(inc); % one-side noise power spectrum density
sigma = sqrt(0.5*N0); % noise standard variance
noise = randn(size(sp)).*sigma+sqrt(-
1)*randn(size(sp)).*sigma; % awgn
r = r_mp+ noise; % add awgn
% <<<<fft & fft can be ignored !!!>>>>
% SUD OR MUD -----
--
% different SUD including MRC,EGC,ORC will be considered
% MUD will also be used; different SUD/MUD methods are
selected by 'flag'
% 1 -- MRC (SUD)
% 2 -- EGC (SUD)
% 3 -- ORC(ZF) Orthogonality restoring combining or zero-
forcing
% 4 -- MLD(MUD) maximum likelihood detection
flag = 4; % to choose different EQ methods
switch flag
case 1 % MRC of different subcarriers
if it ==1 && inc ==1; disp('MRC'); end;
req = r.*conj(fading); % frequency domain equalization,MRC
for k =1:K; % for different users; only demodulate user 1.
!!!!!!!!!!!!!!
sink(k,:) = DeSPREADING(req,Code(:,k)); % user k
Pb(k,inc) = sum(sink(k,:)~=Bit(k,:))./Num_sym(inc); % BER
for user1,one iteration
end;
% case 2 % EGC
% if it ==1 && inc ==1; disp('EGC'); end;
% req = r.*conj(fading)./abs(fading); % EGC
% for k =1:K; % for different users; only demodulate user
1. !!!!!!!!!!!!!!!
% sink(k,:) = DeSPREADING(req,Code(:,k)); % user k
% Pb(k,inc) = sum(sink(k,:)~=Bit(k,:))./Num_sym(inc); % BER
for user1,one iteration
% end;
% case 3 % ORC
% if it ==1 && inc ==1; disp('ORC/ZF'); end;
% %req = r.*conj(fading)./(abs(fading).^2); % ORC
% req = r./ fading; % ORC
% for k =1:K; % for different users; only demodulate user
1. !!!!!!!!!!!!!!!

```

[illegible]

```

%=====
=====
% plot simulation and analysis results
%=====
=====
% Analysis results
gammaBdB = min(EbN0dB):0.1:max(EbN0dB);
gammaB = 10.^(gammaBdB/10);
Pba = 0.25./gammaB; % BPSK BER in Rayleigh channel
L = 4; % the maximum diversity
for l = 1:L
mu = sqrt(gammaB./(1+gammaB));
pe = [];
for kk = 0:l-1
pe(kk+1,:) = nchoosek(l-1+kk, kk).*(0.5*(1+mu)).^kk;
end;
pes = sum(pe,1);
Pbas(l,:) = pes.*(0.5*(1-mu)).^l;
end
figure(1);
semilogy(EbN0dB,ber(1,:,2),'ko-
','Linewidth',2,'MarkerSize',9);
hold on;
grid on;
semilogy(EbN0dB,ber(2,:,2),'bd-
','Linewidth',2,'MarkerSize',9);
semilogy(EbN0dB,ber(3,:,2),'r+-
','Linewidth',2,'MarkerSize',9);
semilogy(EbN0dB,ber(1,:,1),'k<--
','Linewidth',2,'MarkerSize',9);
% semilogy(EbN0dB,ber(2,:,1),'b*--
','Linewidth',2,'MarkerSize',9);
% semilogy(EbN0dB,ber(3,:,1),'rs--
','Linewidth',2,'MarkerSize',9);
% semilogy(gammaBdB,Pba,'k--','Linewidth',3);
semilogy(gammaBdB,Pbas(1,:), 'b:', 'Linewidth',1);
semilogy(gammaBdB,Pbas(2,:), 'r:', 'Linewidth',1);
semilogy(gammaBdB,Pbas(3,:), 'm:', 'Linewidth',1);
semilogy(gammaBdB,Pbas(4,:), 'k:', 'Linewidth',1);
xlabel('\gamma_0 (dB)'); ylabel('Bit Error Probability');
% title('BERs for MC-CDMA BPSK system in flat Rayleigh
Fading Channel');
title('BERs of MC-CDMA(Indep Rayleigh Fading) MLD K =3');
legend('User1','User2','User3','WH4');

```

[illegible]

APPENDIX C

MATLAB CODE FOR OUTAGE PROBABILITY IN CALL ADMISSION

```
% W=bandwidth,R-data rate,P=voice activity factor,f=other
x=cell
% interference factor,Sigma=standard deviation for Log-
normal shadowing
% propagation model
% InvMhu=average call time,lambda =average call arrival
rate
W=1230000;
R1=4800;
R2=9600;
n=0.1;
EbOverNo=10^(0.35);
p=0.5;
F=0.55;
Beta=(log(10))/10;
Sigma=10^(0.25);
InvMhu1=linspace(1,25,100);
Lambda1=100;
ko1=((W/R1)*((1/n)-1))/(EbOverNo);
ko2=((W/R2)*((1/n)-1))/(EbOverNo);
a=exp(((Beta*Sigma)^2)/2);
b=exp((Beta*Sigma)^2);
c=p*(1+F);
d1=sqrt(Lambda1*InvMhu1);
e1=1./d1;
f=sqrt(c);
g11=((ko1.*e1)-(d1.*c*a))/(f*b);
g12=((ko2.*e1)-(d1.*c*a))/(f*b);
Z11=erfc(g11./sqrt(2))/2;
Z12=erfc(g12./sqrt(2))/2;
figure(1)
hold on;
semilogy(InvMhu1,Z11);
grid on
ylabel('Outage Probability');
xlabel('Traffic Load');
figure(2)
hold on
```

```

semilogy(InvMhu1,Z12);
grid on
ylabel('Outage Probability');
xlabel('Average Call Time (Min)');
InvMhu2=3;
Lambda2=linspace(10,800,200);
d2=sqrt(Lambda2.*InvMhu2);
e2=1./d2;
g21=((ko1*e2)-(d2*c*a))/(f*b);
g22=((ko2*e2)-(d2*c*a))/(f*b);
Z21=erfc(g21/sqrt(2))/2;
Z22=erfc(g22/sqrt(2))/2;
figure (3)
hold on;
semilogy(Lambda2,Z21);
grid on
ylabel('Outage Probability');
xlabel('Traffic Load');
figure (4)
hold on
semilogy(Lambda2,Z22);
grid on
ylabel('outage Probability');
xlabel('Traffic Load')

```

APPENDIX D

MATLAB CODE FOR COMPARING THE OUTAGE PROBABILITIES

```
clc;
clear all;
ko1=230.4;
ko=230.4;
bet=log(10)/10;
ex=exp((bet*1.5)^2);
ex1=exp((bet*2.5)^2);
ex2=exp((bet*0)^2);
ex3=exp((bet*2.0)^2);
for y=140:230
    pout(y)=qfunc((ko-(y*sqrt(ex)))/(sqrt(y)*ex));
    pout1(y)=qfunc((ko1-(y*sqrt(ex1)))/(sqrt(y)*ex1));
    pout2(y)=qfunc((ko-(y*sqrt(ex2)))/(sqrt(y)*ex2));
    pout3(y)=qfunc((ko-(y*sqrt(ex3)))/(sqrt(y)*ex3));
end
semilogy(pout, '-sr');
hold all;
semilogy(pout1, '-*g');
hold on
semilogy(pout2, '-o');
hold on
semilogy(pout3, '-dk');
grid on
hold all
title('Comparison of the Outage Probabilty of the  
considered Techniques');
ylabel('Outage Probability');
xlabel('Traffic Load')
legend('Power Control', 'Hard Handoff', 'Soft Handoff', 'Call  
Admission')
```