Performance Analysis of a Hybrid Topology CDD/TDD-CDMA Network Architecture

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DECLARATION

I declare that this research report is my own work. It is being submitted in partial fulfilment of the requirements for the degree of Master of Science in Engineering at the University of the Witwatersrand, Johannesburg. It has not been submitted for any degree or examination in any other University.

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ABSTRACT

Code division duplexing (CDD) has steadily garnered attention in the telecommunication community. In this project report we propose a physical layer implementation of CDD that utilizes orthogonal Gold codes as the means of differentiating transmission directions, in order to implement an ad-hoc networking infrastructure that is overlaid on a standard mobile networking topology, and hence creating a hybrid networking topology. The performance of the CDD based system is then comparatively assessed in two ways: from the perspective of the physical layer using point-to-point simulations and from the perspective of the network layer using an iterative snapshot based simulation where node elements are able to setup connections based on predefined rules.

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3G	Third Generation Mobile
3GPP	Third Generation Partnership Project
4G	Fourth Generation Mobile
AMPS	Advanced Mobile Phone Service
BER	Bit-error Rate
BPSK	Binary Phase Shift Keying
BS	Base Station
BTS	Base Transceiver Station
CDD	Code Division Duplexing
CDMA	Code Division Multiple Access
CDMA2000	Trademark name for Multi-Carrier CDMA
cdmaOne	Trademark name for Interim Standard Number 95
D-AMPS	Digital Advanced Mobile Phone Service
DSSS	Direct sequence Spread Spectrum
EIRP	Effective Isotropic Radiated Power
FDD	Frequency Division Duplexing
FDMA	Frequency Division Multiple Access
FH-SS	Frequency Hopping Spread Spectrum
GSM	Global System for Mobile Communication
ICI	Inter channel interference
IP	Internet Protocol

IS-95	Interim Standard Number 95
ISI	Inter symbol interference
ISM	Industrial, Scientific and Medical band
MAI	Multiple Access Interference
MS	Mobile Station
MUD	Multi-user detector
ODMA	Opportunity Driven Multiple Access
OFDM	Orthogonal Frequency Division Multiplexing
PDC	Personal Digital Cellular
PN	Pseudo-noise
SDMA	Space Division Multiple Access
SNI	Signal-to-interference Ratio
SNR	Signal-to-noise Ratio
TACS	Total Access Communications System
TDD	Time Division Duplexing
TDMA	Time Division Multiple Access
TD-SCDMA	Time-division synchronous CDMA
UMTS	Universal Mobile Telecommunication Service
UTRAN	UMTS Terrestrial Radio Access Network

1 INTRODUCTION

The introduction of the dissertation provides a prelude to the world of mobile telecommunications. The chapter discuses how the market along with the expectation of the user has changed and how this has had an impact on mobile cellular networks globally. The chapter then moves on to the evolution of mobile cellular network. In it, the evolution of mobile cellular is discussed: from the 1st generation mobile technologies to the current mobile network technologies that exist today. The chapter then takes a look at current trends towards the future of mobile cellular networks. Afterwards, the problem statement is then presented: The limitations of current mobile cellular technologies that motivate the research are then outlined. Then, the results from the project are then briefly highlighted. The chapter then concludes with an outline of the project report.

1.1 Evolution of Mobile Cellular Networks

Mobile cellular communications has changed the face of society forever. Personal communication has now become a way of life that the average mobile user has now become accustomed to. The average user has come to expect certain services and standards of services from their mobile devices. As the expectation of the average user changes, so does the market response to services being provided, and in turn there is a ripple effect that affects one of the most critical components of any type of communication network, the access component.

As users move away from voice-centric services to more data/voice centric services, the access network has to adapt to the needs of the consumer by providing a more efficient means of transporting the information that the user requires. For wireless communication, there is a need to make more efficient use of the limited radio spectrum, in order to provide more media rich services.

The changing requirements of the consumer is not the only driving factor that has forced mobile networks to evolve. Within recent years there has been a tremendous growth in mobile network usage. As the number of users increase, it becomes difficult to accommodate them due to constraints, namely the limited radio spectrum. Mobile cellular networks could use a wider radio spectrum to handle larger amounts of traffic. However, regulatory bodies tightly control the limited radio frequency spectrum and the allocation of radio spectrum has become an expensive and complex ordeal. As a consequence, there exists the drive to use more spectral efficient techniques to accommodate a larger number of subscribers and support more media-rich content. Evidence of this drive is the evolution of mobile cellular network technology.

Mobile cellular technology has evolved from being purely analogue to becoming purely digital systems in terms of how information is carried and interpreted. First generation systems were almost dedicated to voice traffic only. Examples of such networks are Advanced Mobile Phone Service (AMPS) and Total Access Communication System (TACS). AMPS is a North American standard that used the 800 MHz radio band. The system was also used in Australia, New Zealand and a few countries in South America. TACS, on the other hand, is a standard that came out of the United Kingdom [11].

The advent of second generation mobile cellular network was a watershed moment in the evolution of the mobile cellular network. They were mostly purely digital systems and had much higher user capacity than their predecessors. There were four main standards that were classified as second generation systems: Global System for Mobile Communication (GSM), cdmaOne or Interim Standard 95 (IS-95), Digital AMPS (D-AMPS) and Personal Digital Cellular (PDC). Of the four standards, GSM (a European standard) became the most widely adopted standard.

Although the time of second generation networks seemingly draws near to an end, the systems, especially GSM, are still in operation and will continue to be in operation for some time before they are phased out by the next generation of mobile cellular systems. Third generation systems are currently being established world wide. There are three main standards: Universal Mobile Telecommunications System (UMTS), CDMA2000, and TD-SCDMA.

In Eastern Asia, third generation networks have been in operation for some time. In Japan, third generation networks have been in operation since 1999. A proprietary version of UMTS was rolled out in Japan by NTT DoCoMo before they migrated to the European

Telecommunications Standards Institute (ETSI) standard version. UMTS is expected to be installed in many countries that currently use GSM networks because UMTS has been designed to facilitate this migration. South Africa has witnessed this migration where two competing GSM network providers have overlaid UMTS networks to provide more media rich services. CDMA2000 has gained a foothold in the Eastern Asia as well. CDMA2000 is the next logical step for network providers that currently use IS-95.

1.2 The Future of Mobile Cellular

As the mobile world gets accustomed to the 3rd generation mobile networks, the movers and shakers of the industry have begun to look forward, beyond 3G to 4G. The fourth generation network is not clearly defined as yet, however one thing is certain, some of the major goals that the fourth generation of communication networks must achieve are higher user capacities, ability to supply the user with media-rich services and offer access to the network whenever, wherever. For these goals to be achieved, one of the current visions of a 4G network is the amalgamation of many radio access networks, wired access networks, backhaul networks and an increase in the amount of logic in user equipment. As a consequence, there will be no single network known as a 4G network. A 4G network may also be merely the standardisation of the seamless interoperability of currently existing communication networks.

Although 4G has not yet been defined, there is already vast amount of research into access network possibilities and the core network infrastructure. Currently, the 4G development initiatives tends towards an all IP core network and will continue to develop on the access technologies used in third generation mobile networks. Yet another 4G vision is in defining a new air interface for wireless components of the network where the wired and the wireless components are integrated with an all-IP core network. Ignoring the core network and the wired network, the wireless access network has many interesting candidates for the air interface; however, the one that has garnered the most attention is Orthogonal Frequency Division Multiplexing (OFDM).

The question is: Where will Code division duplexing (CDD) fit in the future of mobile communications? There is a simple answer to that however: CDD will be able to fit easily into the extensions of current technology and hence will be able to fit easily into the development of fourth generation mobile networks.

1.3 Problem Statement

The ideal radio environment would be one where the attenuation of a radio signal can be described by free space path loss. Unfortunately this is not the case. The typical environment is hostile to radio transmissions. For a cellular mobile network, the typical radio environment ranges from business areas to residential areas to rural areas; all of which are non line-of-sight environments and all of which have path loss exponents of four or greater in comparison to free space loss with a path loss exponent of two. For example, an IS-95 base station transmitter with a specified transmission power level of 40 dBm can cover an area of 600 kilometres radius in a free space line of site model for a path loss 146 dB; but can only cover an area of 2.74 kilometre radius in realistic radio environment. As a consequence, despite higher transmit powers, coverage remains a limiting factor for a mobile cellular radio access networks.

On the other hand, areas with high user densities (typical of dense urban and urban environments) tend to suffer from user traffic congestion in which users are unable to make calls. In other words, there is not enough capacity to support the high number of users in an area. This can be easily solved by adding more micro and pico cells. However, there are financial expenditure implications for such a solution.

There seems to be inherent benefits in using an ad-hoc overlay to the standard cellular infrastructure to provide a means of creating an adaptable network topology. The potential lies in the fact that the hybrid topology could have capacity and coverage improvements. The postulate is as follows: The ad-hoc overlay can extend coverage range and can alleviate congestion.

The aim of this research is to investigate the possibility of capacity and/or coverage improvements by way of using an ad-hoc overlay on standard CDMA cellular infrastructure. Thus looking at the issues that are associated with such hybrid networks. The project proposes a new duplexing method known as code division duplexing. CDD is a means by which this ad-hoc overlay can be implemented. Along with using code division duplexing, time division duplexing coupled to create an architecture that could have capacity and coverage advantages. The project aims to investigate suitable code parameters for code division duplexing and explain the impact on multiple access interference and as a consequence the impact on coverage and capacity.

The African context is one of the driving factors behind the motivation of this project. The project aims to tackle the challenges that are present in modern mobile cellular networks, in that they are mostly designed for a Eurocentric market or a North American market. Therefore the structure of these network architectures, from the ground up, is designed to fit in the topologies and demographics of the aforementioned markets.

1.4 Performance of CDD

A CDD system based on its definition will not perform as adequately as a standard a direct sequence spread spectrum system, due to the increase amounts of multiple access interference. As a consequence, two schemes are proposed in this project to mitigate the effects of multiple access interference. Based on the scheme that closest approximates the performance of a direct sequence spread spectrum system, a simulation was setup to demonstrate the performance of a hybrid topology architecture. Simulations showed that a hybrid topology could increase the signal to inference ratio of a base station but could decrease the average signal to inference ratio amongst mobile station. However, in all cases, it was shown that a hybrid topology would allow for coverage to nodes that were not serviced by the base station.

1.5 Project Report Organisation

Chapter 2 starts off with basic introduction to CDMA and highlights that the multiple access scheme is used as the basis of current 3rd generation mobile technologies. The chapter then proceeds to explain FDMA, TDMA and CDMA in greater depth. A discussion on spread spectrum techniques then follows. Highlighted are the two key spread spectrum techniques: frequency hopping spread spectrum and direct sequence spread spectrum. In section 2.1.3, direct sequence spread spectrum is focused on. Within this section, the concept of the spreading factor is introduced along with a basic schematic of a DSSS system. The chapter moves on to the explanation of pseudo-random noise sequences. The section highlights how the codes are generated and the properties that they have. The section then progresses to an explanation of Gold Codes - how they are generated, and the properties that they entail. Orthogonal codes are then discussed. The section traverses the theory of three orthogonal code types: Walsh Codes, Orthogonal Gold Codes and Smart Codes. Section 2.2 of the chapter then briefly reviews multiple access interference. The section tackles the topic by explaining the interference limited nature of CDMA DSSS and gives a general expression of the bit-error performance of the system. The section then moves on to discussing multiple user detection and how it is a key feature in maximising the performance of CDMA systems. The concept of code division duplexing is then introduced. Here, it is highlighted that the concept was introduced as a peer-to-peer networking facilitation by P. J. Chitamu [17]. Time division duplexing is then described in section 2.6, as means of separating uplink and downlink. In the section, TDD is compared with frequency division multiplexing. The advantages and disadvantages of the duplexing scheme are then discussed.

In chapter 3, projects that have a related concept to the proposed project are discussed. One of the first projects discussed here introduces the concept of “mobile-assisted data forwarding” (MADF). In MADF, the network load is balanced by re-routing traffic in an ad-hoc fashion. In a similar vein, the next system described also re-routes traffic from congested cells to less congested cells. The iCAR system does this by way of using relay stations that use the ISM band. The UCAR system is another system that utilises the ISM band to relay data. In this project, data is re-routed by IP tunnelling over the 802.11b radio interface on MS units. Projects that are based on ODMA (a 3GPP extension to UMTS) are then reviewed. The ODMA specification describes the ability of mobiles to hop (or route traffic through) on another mobile and the investigations into ODMA look at the coverage and capacity benefits. Afterwards, a brief review is conducted into the theoretical capacity of hybrid networks.

In chapter 4, the basic CDD system design is described. Here it is explained that orthogonal Gold codes are used to implement the code division duplexing. The chapter then proceeds to define CDD and to explain the receiver and transmitter structure diagrammatically. Section 4.2.3 then describes the link budget for the proposed architecture. Section 4.3 continues to elaborate on the design of CDD on the MAC layer. At this point, the TDD frame structure is proposed. In section 4.4, the call setup and routing are briefly discussed. There the section describes a billing oriented call setup and the transmission strategies, in routing, that exist: One hop-One Slot, Multi Hop-One Slot, One Hop-Multi Slot and Multi Hop-Multi Slot.

Chapter 5 then describes the tools and procedure used to evaluate CDD. Here it is explained how MATLAB was used to implement a point-to-point CDD links. The setup here is the implementation of the transmitter and receiver structures in the previous chapter. Then with the specified parameters, CDD transmissions were simulated and the results are represented in graphical form and discussed.

In chapter 6, a description of how the coverage and capacity simulation was setup is presented. The chapter introduces the design of the simulation by describing the node objects that are used in the simulation. The assumptions that are used in the simulation are then outlined. Procedures and tools are then described.

The results of the project are highlighted and explained in chapter 7. In this section result relating to the CDD link simulation and the Coverage/Capacity simulation are discussed.

In chapter 8, areas of further research, which are closely related to this field, are discussed and assessed to see how they can be further investigated: The chapter describes explicit diversity gain, where nodes act as mirrors for packet based hybrid networks; and then explains the potential further research will present for this field. The next section then suggests that multi-user detection for CDD is an area that requires thorough investigation. The next section goes on to describe the complexity of routing strategies hybrid networks and how further insight into this field can be used to increase the efficiency of hybrid networks. The next section goes on to describe channel assignment, while suggesting how further investigation in this topic will prove useful for hybrid networks in terms of efficiency. Finally, the chapter suggests further investigation in power control within hybrid networks is key path to increasing its efficiency.

The conclusion summarizes the results as well as the concepts that are introduced. It also outlines how these new findings can be used.

2 BACKGROUND

Code division duplexing is a CDMA based technology that extends the concept of using spreading codes to differentiate the direction of data in terms of its origin and destination. In this section, the focus is on the essential concepts that are needed to have an understanding of a basic CDMA system with CDD. The various multiple access schemes are highlighted to shed some light on CDMA itself. Direct sequence spread spectrum systems are then briefly discussed, seeing that the proposed CDD architecture is based on the form of direct spread spectrum modulation. Following that, the challenges that face CDMA radio systems are highlighted namely the radio environment itself and multiple access interference. Finally, the conceptual building blocks of the proposed system are then identified.

2.1 Code Division Multiple Access

Over the past ten years, there has been a phenomenal growth in mobile network subscribers. Along with regulatory activities that control radio frequency spectrum allocations, one of the consequences of the staggering increase in subscribers is the drive to find the most efficient mechanism of using the limited radio frequency spectrum. Code Division Multiple Access (CDMA) is, spectrally, the most efficient of the three multiple-access techniques used predominantly for modern commercial mobile networks. As a consequence CDMA was selected as the basis of the radio interface in 3G specifications [5].

As mobile telecommunications developed, the use of CDMA is foreseen to continue to play a dominant role in future communication systems.

2.1.1 Multiple Access Schemes

Before CDMA became the primary candidate for current mobile communications, there were two other dominant access techniques that were used primarily in older mobile networks and are still used today. Most first generation mobile networks, for instance, AMPS relied on Frequency Division Multiple Access (FDMA) as the means of differentiating[1] users (as

[1] This is the means of identifying each unique user.

shown in Figure 2.1). FDMA provides user access by dividing the scarce radio resource into physical frequency bands known as channels. In the case of AMPS, each user channel is exclusively assigned two channels which are used for uplink and downlink communications. In other words, AMPS is a Frequency Division Duplexing (FDD) system. Once a voice circuit is setup (which uses one user channel) the mobile and the base station both transmit continuously for the duration of the call. To minimise interference from each of the adjacent narrowband channels (30 kHz in the case of AMPS), FDMA systems require very steep roll-off radio filters.

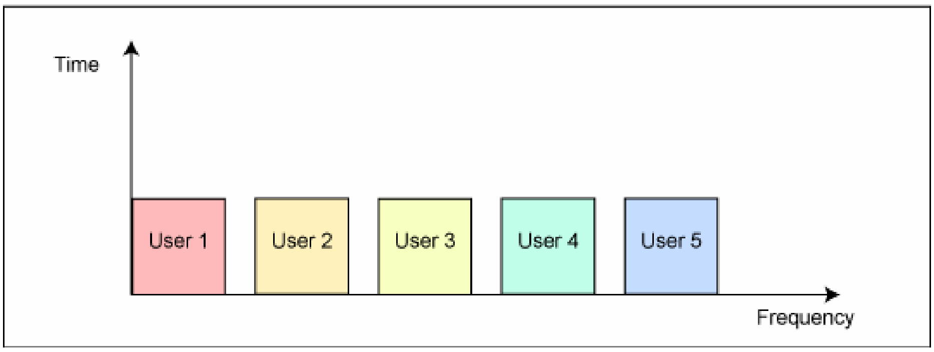

Figure 2.1: Graphical representation of FDMA - Users distinguished by frequency

The use of Time Division Multiple Access (TDMA) based networks ushered in the era of second generation mobile networks. The most popular of the second generation networks, GSM, is a TDMA/FDD network. The use of TDMA allowed more than one user to occupy the entire bandwidth of the physical channel but only for a short non-overlapping period of time, and consequently sharing the bandwidth while being distinguished in time (as shown in Figure 2.2). This aforementioned time period forms a logical channel and is most commonly referred to as a *timeslot*. TDMA was more efficient in the sense that it allowed more users to be able to seemingly occupy the radio resources at the same time, in comparison to FDMA where a single user is dedicated a physical channel and the moment it is unoccupied the entire physical channel is wasted. Logical channels thus further increased the number of users that can be accommodated in the network. In addition, a set of physical channels is dedicated for the uplink and a set dedicated for the downlink. This is employed in GSM networks and hence it is referred to as a hybrid TDMA/FDD system.

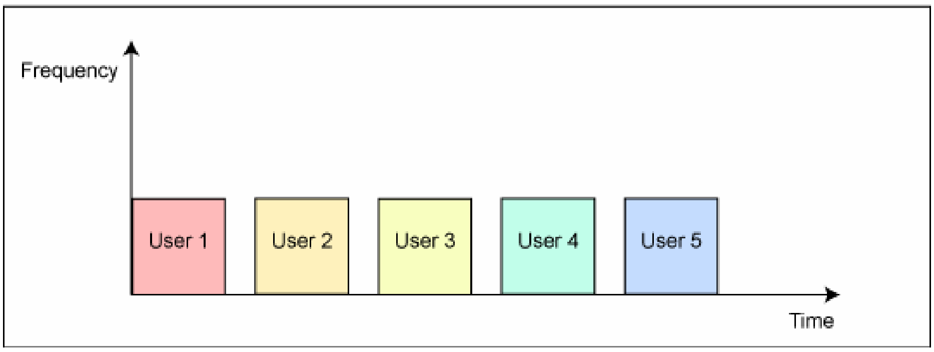

Figure 2.2: Graphical representation of TDMA - Users distinguished in time

There are some distinct advantages that made TDMA systems a more obvious choice over FDMA systems. Transmission with the base station is not continuous. Consequently, user equipment consumed less power than their FDMA counterparts. Along with the power saving attributes was the simplification of handovers due to the discontinuous transmissions and enhanced link control. However, there are some intrinsic disadvantages that are associated with TDMA systems. There is a need for high synchronisation overhead because of the discontinuous nature of transmissions. In addition to this, there is a need for guard times to ensure that timeslots do not overlap and this results in larger overheads overall in comparison to an FDMA system [27].

Despite the overwhelming popularity of TDMA based networks due to the success of GSM, CDMA based networks had a place in the second generation mobile network era, and CDMA would eventually prove itself to be the prime candidate for the access mechanism for next generation of mobile networks – 3G networks. Originally known as IS-95, cdmaOne is based on (as the name suggests) CDMA which is used to differentiate users. In a CDMA network all users use the radio resource at the same time, which implies that users are transmitting information on the same frequency band at the same time. In order to separate transmissions intended for particular users from the barrage of other transmissions, each user is assigned a unique code (as shown in Figure 2.3). This code is used to convert the user data into a wide bandwidth waveform. The same code is used by the user themselves to sift out the information intended for themselves.

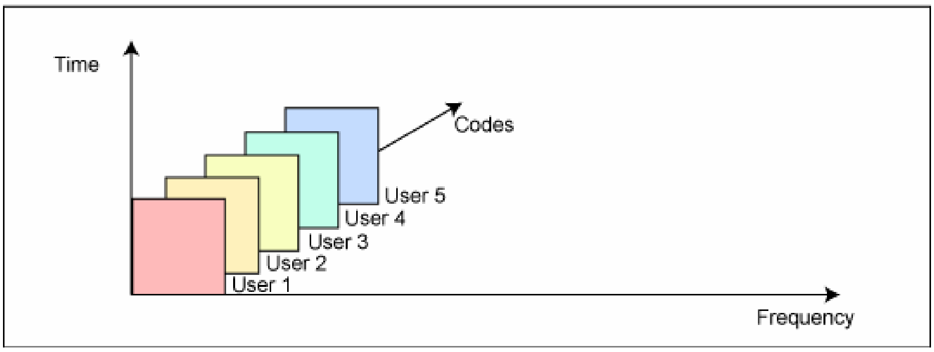

Figure 2.3: Graphical representation of CDMA - Users distinguished by codes

As the name suggests, space division multiple access (SDMA) differentiates users in space by directing radio energy to each user's position. An example of this multiple access scheme is *cell sectorisation* in the sense that energy from an antenna is directed to focus on a specific area. To implement an optimal SDMA system, adaptive antennas capable of emitting a precision beam of specified width would be needed. These adaptive antennas must also have the ability to keep track of each user (antenna) with accurate positioning. Currently, such adaptive antennas do not yet exist. Nonetheless, current beam-forming adaptive antenna arrays are sufficient enough to provide considerable interference reduction in conjunction with other multiple access techniques.

The principle access techniques in wireless systems are FDMA, TDMA, SDMA and CDMA. However, as modern wireless systems continue to evolve, it becomes apparent that the combination of these schemes will lead to significantly more spectrally efficient mobile cellular systems. One such example of this is Multi-Carrier CDMA (MC-CDMA). MC-CDMA is the combination of Orthogonal Frequency Division Multiplexing (OFDM) and CDMA. In OFDM, a high rate data stream is divided up in to many low rate sub-carriers that have a symbol bandwidth smaller than the bandwidth of the channel which has a constant gain and a linear phase response, thus reducing the effect of inter-symbol interference. As proposed in [13], the large number of sub-carriers require a digital implementation of multi-carrier modulation in which rectangular pulse shaping with guard-times implemented using Discrete Fourier transforms. In MC-CDMA, a single data symbol is spread across the multiple carries, resulting in a processing gain with frequency diversity.

2.1.2 Spread Spectrum Modulation

The concept behind spread spectrum modulation is to transform a signal of limited bandwidth to a signal with a much wider bandwidth. In other words, spread spectrum communications involves sending information using a far larger bandwidth than that required by the information being sent. The two most common forms of modulation techniques used in CDMA communication are Direct Sequence Spread Spectrum (DSSS) and Frequency Hopping Spread Spectrum (FSSS). Both these techniques use a pseudo-noise (PN) sequence (also known as a pseudo-random code) to expand a signal into a larger bandwidth in which the resulting spread spectrum signal has uniform energy over that bandwidth (see Figure 2.4).

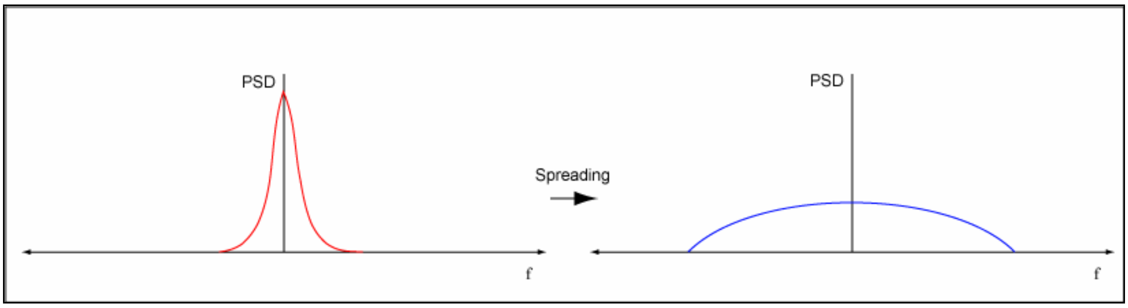

Figure 2.4: Illustration of Spreading Spectrum Modulation

There are properties of spread spectrum modulation that make it extremely attractive for the mobile radio environment. Its inherent resistance to interference is the most notable property. Narrowband interference only affects a fraction of the spread spectrum signal and, therefore, can be easily removed by notch filtering. In a similar fashion, frequency selected fading is thwarted because only a small fraction of the spread spectrum signal is exposed to fading. Spread spectrum systems can also exploit the multi-path components using a RAKE receiver. The RAKE receiver has the ability to combine resolvable copies of the signal to obtain a more accurate estimate of the received signal. Another significant advantage of spread spectrum communications, in respect to mobile networks, is the simplification of frequency reuse planning due the fact that all users share the same frequency band.

Of the two modulation techniques, DSSS is the more significant in third generation mobile communications. Both UMTS Terrestrial Radio Access Network (UTRAN) and CDMA2000 are based on DSSS modulation. In UTRAN, information is spread over a 5 MHz bandwidth and hence UMTS is known as a Wideband CDMA system (WCDMA). On the other hand, CDMA2000 was designed to use multiple carriers of 1.25 MHz, in which information is spread over and hence CDMA2000 is known as a multi-carrier DSSS system.

Although DSSS is more significant than FH-SS in terms of this research, there are few popular FH-SS based systems. Bluetooth is a short range ad-hoc networking protocol that uses FH-SS as its modulation scheme.

FH-SS attains the expansion in bandwidth by (pseudo-) randomly changing the carrier frequency of the information signal across a wider bandwidth, in which the frequency being hopped to is determined by a pseudo-random code. The rate of change (hopping) of the carrier frequency can be fast or slow. Fast frequency hopping implies that the carrier frequency is being changed more than once in the duration of a single symbol; and slow frequency hopping implies that more than one symbol durations must occur before the carrier frequency is changed.

On the other hand, the process of spreading in DSSS is achieved by multiplying each bit of a data sequence by a pseudo-random code, known as the spreading code. The spreading code is a binary sequence in which the bit period is smaller that the bit period of a bit in the data sequence.

2.1.3 Direct Sequence Spread Spectrum

As stated earlier, a DSSS system expands the baseband data by each bit of a data signal with a pseudo-noise sequence and with an orthogonal code sequence in commercial mobile systems. In other words, each bit of the baseband signal is represented by the complete spreading sequence.

Each bit of the spreading sequence is known as a “chip” and the spreading factor is the number of chips that are used to represent a bit and can be represented as mentioned earlier. The spreading sequence is at higher bit (chip) rate than the baseband signal and hence requires a larger bandwidth.

$$Spreading\ Factor = \frac{Chip\ Rate}{Symbol\ Rate} \tag{2.1}$$

The higher the spreading factor is, the higher the processing gain will be. In a similar fashion, a higher processing gain implies that there is greater ability to recover the user information from the summation of the other user transmissions. In other words, the higher the processing gain the lower the bit-error rates will be. However, there is a trade off for increasing the processing gain. Seeing that each data bit is now represented with more chips at a constant chip rate, it therefore implies that the user data rate will be reduced by the factor of the processing gain increase. So if the processing gain increases by a factor of two, then the user data rate will decrease by a factor of two.

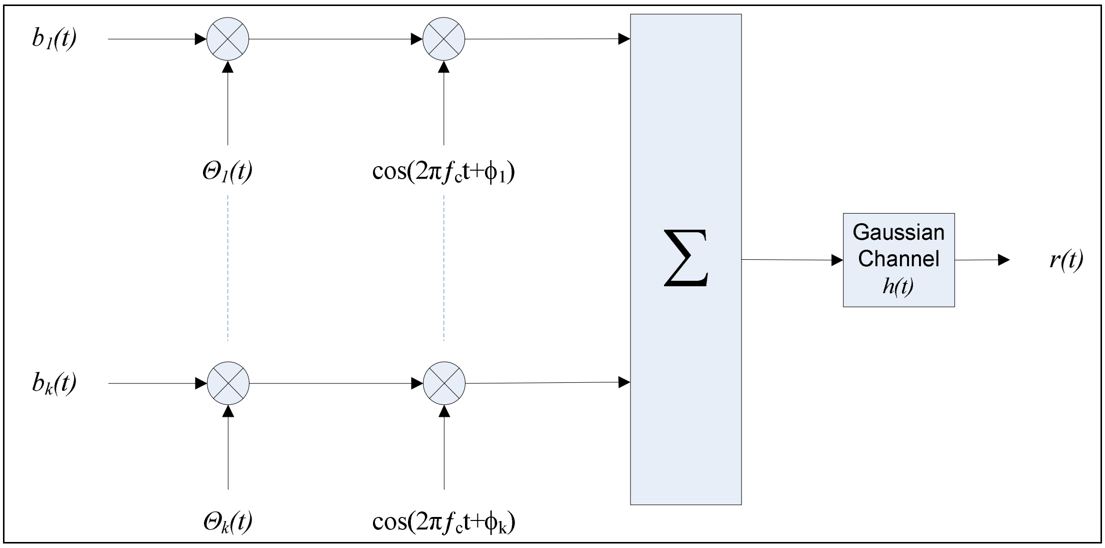

Figure 2.5: Schematic diagram of a DSSS system with k users

Figure 2.5 shows k independent users whose data is modulated with direct sequence spread spectrum modulation. The data stream for each user is spread by a chip sequence $\Theta_k(t)$, where the chip rate is higher than the bit rate of the user data stream. Each user is assigned a unique chip sequence, which is used to later demodulate the data for that user. Each user utilises the same carrier frequency and transmits simultaneously over the Gaussian channel.

The data stream of the k^{th} user, $b_k(t)$, can be defined as

$$b_k(t) = \sum_{m=0}^{M-1} b_k(m)\Pi_{T_b}(t - mT_b) \ . \tag{2.2}$$

Here $b(m) \in \{1, -1\}$, in which m represents the m^{th} bit in the series of bits and where $b(m)$ is multiplied by a rectangular pulse, Π_τ, with a period of T_b (the symbol period). Thus making $b_k(t)$ a bipolar bit sequence.

The data sequence is spread by $\Theta_k(t)$, and modulated such that the output waveform is the chip sequence represented with binary phase shift keying. The post Gaussian channel representation of the combination of each of the user transmissions at the receiver is represented as:

$$r(t) = \sum_{k=0}^{K-1} \sqrt{P_k} b_k(t) \Theta_k(t) \cos(2\pi f_c) + n(t) . \qquad (2.3)$$

Here P_k is the received power of the k^{th} user and $n(t)$ is the additive white Gaussian noise of the channel.

At the receiver side, $r(t)$ is then used to extract the information for a specific user. This is achieved by multiplying $r(t)\cos(2\pi f_c)$ with the spreading sequence, $\Theta_k(t)$; and is implemented with a conventional matched filter receiver. The output of the receiver can be represented as

$$w_k(m) = \tfrac{1}{T} \int_{mT}^{(m+1)T} r(t) \Theta_k(t) \cos(2\pi f_c) dt . \qquad (2.4)$$

At this point $w_k(m)$ still requires addition processing before it can represent the m^{th} symbol from the k^{th} user. The output symbol for the k^{th} user can be represented as

$$b_k'(m) = decision(w_k(m)) = \begin{cases} +1 & if \ \ w_k(m) \geq 0 \\ -1 & if \ \ w_k(m) < 0 \end{cases} . \qquad (2.5)$$

In equation 2.5, $w_k(m)$ is the subject of the decision function that determines the m^{th} bit for the k^{th} user.

2.2 Multiple Access Interference

Sharing the spectrum in DSSS CDMA systems isn't without its consequences. One of the largest obstacles in DSSS systems is the dealing with multiple access interference. Succinctly, in DSSS systems, interference is caused by the presence of other users. The more users that are introduced to the system, the more the interference will be for a single user.

Considering a single cell with perfect power control, where all the users in the cell transmit with a power *aS* and the power received by the receiver is *S*; and let the total number of users be *K*. Therefore, within this cell, the total amalgamated power being transmitted is *KS*, and hence the total amount of interfering power being received by a single user is *(K-1)S*. Therefore the signal-to-noise ratio for a single user can be expressed as

$$\frac{S}{(K-1)S} = \frac{1}{(K-1)}. \tag{2.6}$$

Based on the equation 2.6, it can be demonstrated that as the number of users (*K*) increases, the signal-to-noise ratio (SNR) falls exponentially. Using Shannon's theorem, it can, therefore, be shown that the capacity of the communications channel will decrease if the SNR decreases. The relationship between capacity and the signal-to-noise ratio is described as

$$C = B \times \log_2\left(1 + \frac{S}{N}\right). \tag{2.7}$$

In fact, it is shown in [27] that the bit-error performance, where thermal noise is not a factor, is stated by equation 2.8:

$$P_e = Q\left(\sqrt{\frac{3N}{K-1}}\right), \tag{2.8}$$

where, N is the number of chips that are used to spread a single bit; K is the number of users in the system (a single cell) and P_e is the probability of an error.

The equation implies that as the number of users increase the probability of error increases and in a reciprocal manner, as the processing gain increases, the probability of error decreases.

2.2.1 Multi-User Detectors

To summarise, the performance of a DSSS CDMA system decreases as the interferences due to other users increases. There are many proposed techniques for the reduction of multiple access interference (MAI). The use of transmission sectors (also known as sectorisation[1]) is one of the simple means of reducing the number of users in an area, and consequently reduces the MAI. The use of multi-user detectors (MUDs) is another means of negating the effect of the MAI. Current MUDs are more complex than the conventional receiver, which are correlation receivers based on a matched filter. Research into MUDs has gained significant popularity since the advent of the use of DSSS systems in commercial mobile communication systems.

The radio channel degrades the spreading codes such that they are no longer perfectly orthogonal. As a consequence, the conventional receiver performance can be drastically reduced. In conventional receivers, such as a matched filter receiver, the decision statistic is hard, which implies that the decision is only based on the signal that has been received within the symbol period. As a consequence, using such receivers require the system to be perfectly synchronised. This type of user detection is not optimum due to the fact that conventional receivers ignore information that can be extracted outside of the symbol period, namely information on the interference coming from overlapping symbols [22].

It was shown in [24] that in order to combat that problem, the information for each user must be jointly-extracted in order to maximize the performance of correctly extracting information for a single user. This technique is known as the optimum multi-user detection based on the maximum likelihood rule. In optimum detection, it is assumed that the receiver has information on each user's spreading code sequence, time delays, phase shifts and received signal amplitudes, such that an m^{th} bit intended for the k^{th} user can be expressed as

[1] Sectorisation provides orthogonality in space.

$$b' = \begin{bmatrix} b_0(1) & \cdots & b_0(M) \\ \vdots & \ddots & \vdots \\ b_{K-1}(1) & \cdots & b_{K-1}(M) \end{bmatrix}. \tag{2.9}$$

Provided that transmitted bits are independent and are exposed to the same probability distribution then b' is used to maximise the likelihood of

$$P[r(t) \mid b'] = Ce^{-\frac{1}{2\sigma_n^2}\int_0^T \left[w_k(t) - \sum_{k=0}^{K-1} b'_k(0)\sqrt{P_k}\Theta_k(t) \right]^2 dt}, \tag{2.10}$$

where C is a constant, $w_k(t)$ is the m^{th} bit of the k^{th} user at time t and σ^2 is the noise power.

Maximising this function can be implemented using the Viterbi algorithm to optimise the equation by dynamically selecting b' [22]. However, as the number of users increases the complexity of detection increases exponentially; because of this, this form of MUD is not practical for use in a commercial mobile system. Research in suboptimum multi-user detectors garner much of the attention that is paid to MUDs. Mainly due to the attributes that they can possibly possess. In other words, there may exist a suboptimum multi-user detector that is able to approach the performance of an optimum MUD, while approaching the simplicity of a standard correlation receiver.

2.3 Radio Environment

In the same way that signals in wired networks experience attenuation or a drop in voltage level between source and destination, radio signals also experience propagation attenuation. In fact, the radio environment is much more hostile in comparison with wired networks. As a consequence, the radio environment (or radio channel) is not as reliable as copper or optic fibre. The radio environment is considered to be hostile because of the effects of three types of hindrances: path loss, slow fading and fast fading. In the drive towards improving mobile cellular communication, these characteristics must be overcome in order to optimise a radio communications link.

2.3.1 Path Loss

When transmitting a signal from an isotropic antenna, the signal emanates as a spherical waveform. As the waveform emanates, the strength of the signal reduces as the distance of the signals propagation increases. The amount of power that emanates with the signal remains constant, but as the signal travels the spherical waveform grows in size. The amount of power remains constant over the entire sphere but is now distributed over a larger surface area, and thus the effective power of the signal reduces for a constant area. This form of propagation loss is known as *free space loss* and can be represented as an inverse square relationship:

$$P_r = P_t \left(\frac{\lambda}{4d} \right)^2 . \tag{2.11}$$

However, in the radio environment, there are other factors, like building losses and atmospheric absorption, which have an effect on the attenuation of the radio signal. The combined effect of these factors is known as *path loss*. The contribution of these additional factors is significant to the extent that the relationship between the distance and the received power is no longer an inverse square relationship, but becomes a larger inverse exponential value, usually between three and five. This value is referred to as the path loss exponent. The expression for the received power can then be expressed, in terms of decibels, as

$$P_r = P_t - 10\alpha \log_{10} \left(\frac{d}{d_0} \right) . \tag{2.12}$$

where α is the path loss exponent and d_0 is the distance in which P_t was observed. Therefore the path loss model can be expressed as

$$L = 10\alpha \log_{10} \left(\frac{d}{d_0} \right) . \tag{2.13}$$

The aim of defining a path loss model is to predict the coverage area of a radio transmission. In other words, the path loss model intrinsically represents a radio channel and thus can be used to design a mobile cellular network in terms of geographic coverage. Because of this significance, there are many empirical models that aim to approximate radio propagation loss as realistically as possible. One of these models is addressed later in section 2.3.4.

2.3.2 Slow Fading

When the radio signal is obstructed by an object in the environment, the effect of the signal loss is referred to as *slow fading* and is more commonly known as *shadowing*. The overall effect of the shadowing is modelled statistically and is known as the log-normal shadowing law. It defines the probability of a mobile unit being in the presence of the radio shadow. Slow fading is usually compensated for in the link budget (as the log-normal fade margin) in order to achieve specified coverage reliability. It is done such that mobile stations are able to get a satisfactory signal at cell edges, where slow fading is most devastating; and therefore bolsters the probability that a call can be made without failing. When slow fading is considered, path loss is then defined as

$$L = 10\alpha \log_{10}\left(\frac{d}{d_0}\right) + \chi , \tag{2.14}$$

where χ is the component of loss[1] in decibels due to shadowing.

[1] Log Normal Fade Margin

Table 2.1 contains log normal fade margins, χ, for required coverage reliability:

Table 2.1: Required fade margin for a single cell [12]

Coverage Reliability (P_{rel})	χ dB	
	σ dB = 8 dB (No power control)	σ dB = 2.5 dB (Perfect Power control)
90%	10.25	3.20
95%	13.16	4.11
98%	16.43	5.13

The log normal fade margin, χ, can be defined as the product of the standard deviation of the path loss and the ratio of the mean and the standard deviation which is given by $Q^{-1}(1-P_{rel})$ where P_{rel} is the coverage reliability. The standard deviation, σ, is estimated by calculating the root mean square error from the actual path loss data versus the expected path loss.

2.3.3 Fast Fading

Fast fading is another form of attenuation that is primarily caused by multi-path components of the desired signal. These multi-path components are caused by reflections and diffractions because of obstacles in the environment. The multi-path components are delayed either in time or undergo frequency modulation. The components which are delayed by an odd number multiple of half of the wavelength will cause cancellations and components that are delayed by an even multiple of half of the wave length will bolster the signal. The result is the rapid fluctuation of the radio signal, hence the term *fast fading*[1]. The frequency modulated copies are mostly caused by the diffraction of the radio signal off the edges of obstacles. In the worst case, fast fading can cause attenuation of up to 30 dB [12]. In CDMA based networks, fast power control is one of the techniques used to combat the adverse

[1] This is also known as Rayliegh Fading.

effects of fast fading. Fast fading is the primary cause of burst errors in radio communications. Burst errors occur to bits that are in close proximity to each other in time. Therefore, to combat these burst errors, bits are interleaved. In other words, bits are shuffled around in an effort to reduce the impact of burst errors.

2.3.4 Empirical Propagation Models

Propagation modelling is a key feature in designing a mobile cellular network. A propagation model is meant to represent the radio environment, which could be urban, suburban or rural. In a best effort to model path loss, path loss models have been proposed that are based on empirical radio propagation data. Hata, Carey, Elgi, Longley-Rice, Bullingtom, Lee and Cost are just some of the more popular propagation models [5]. All of these models use unique assumptions, and as a consequence each model must be fully understood before it can be applied. In a similar vein, each of these models has its own unique constraints and advantages. For instance, a model may suite a dense urban setting and will be able to give a reasonable approximation of the environment. On the other hand this model cannot be used as a valid approximation for radio propagation in a rural environment. Therefore an understanding of the morphology is also necessary such that the most appropriate model can be applied.

Geographic morphologies are usually categorised in four groups, dense urban, urban, suburban and rural. A dense urban area is characterised as being a dense business district with buildings above 10 floors in height. An urban area is similar to the dense urban area except that in this case the buildings heights are in the range of five to ten floors in height. A suburban area is characterised by the mix of residential and business sites, in which it mainly consists of one to two floored structures. Finally, rural areas are characterised by sparsely populated open areas where building structures do not normally exceed two floors in height.

2.3.4.1 The Hata Propagation Model

One of the most popular models of the empirical models is the Hata model. It, along with modifications of it, is widely used amongst mobile cellular network operators. The Hata pass loss model is based on the empirical technical report by Okumura [27]. The model defines the path loss in an urban setting, although it can be adjusted to satisfy the criteria of a suburban or rural setting. One of the constraints of the model is that it is invalid for distances less than one kilometre. The model is defined as:

$$L_H = 69.55 + 26.26\log(f) - 13.87\log(h_b) - a(h_m) + (44.9 - 6.55\log h_b)\log R\,, \tag{2.15}$$

where f is any frequency between 150-1500 MHz, h_b is the height above ground level in metres of a base station, h_m is the height above ground level in metres of the receiver antenna, R is the distance from the base station in kilometres and L_H is the path loss in decibels. Finally, $a(h_m)$ is the correction factor for the effective mobile antenna height in two morphologies based on the urban morphology: small to medium cities and large cities. The full definition of $a(h_m)$ is defined as:

$a(h_m) = (1.1\log f - 0.7)h_m - (1.56\log f - 0.8)$ dB, for small to medium size cities.

$a(h_m) = 3.2(\log 11.75h_m)^2 - 4.97$ dB, for large cities, where $f \geq 300$ MHz.

The model can be corrected for suburban and open rural areas as:

$L_{H-suburban} = L_H - 2[\log(f/28)]^2 - 5.4$ dB, for suburban areas and,

$L_{H-rural} = L_H - 4.78(\log f)^2 + 18.33\log f - 40.94$ dB, for rural areas.

2.3.5 Power Control

In CDMA systems, power control is essential in keeping the signal-to-noise ratio at a tolerable level in order to maximise the number of simultaneous users. One of the problems particular to CDMA systems is what is known as the *near-far effect*. This is when a received transmission power of a mobile station that is close to the base station is far greater than the received power of a mobile station that is farther away. In this scenario, the mobile station that is farther away will have a significantly lower signal-to-interference plus noise ratio because of the interference due to the mobile station that is closer to the base station. As a consequence, the power of the mobiles must be controlled in such a way that the power received at the base station from both mobiles is approximately the same. In addition to the near-far effect, radio conditions do not remain constant and the position of mobile stations change randomly, power control must be adaptive to respond to the ever-changing radio conditions. Power control schemes must also have a high dynamic range seeing that variation between mobile stations can be in the range of 80dB [12] [22]. Power control must combat the effect of fast fading. Fast fading can cause attenuation of up to 30 dB in the worst case. At the same time, the occurrence of fast fading becomes function of the speed of the mobile station. As consequence power control is fast in modern CDMA systems due to the rapid fluctuation of the signal-to-noise ratio.

Open loop and closed loop power control are the two types of power control employed in modern CDMA systems. In open loop power control, the mobile station uses the received signal from the base station to adjust its power level. This is a quick and simple technique[1] and normally takes effect at the point of network access. However, in some CDMA systems (namely FDD systems), the uplink and downlink channels are separated in frequency. This implies that the path loss model is different for uplink and downlink and thus open loop power control alone is insufficient. As a consequence, the base station measures the mobile station's received power and sends a power control command for the mobile to reduce or increase its power – this is closed loop power control. In UMTS there is a further distinction in closed loop power control: inner loop and outer loop power control. In the inner loop, power control is fast to compensate for signal fluctuations, whereas in the outer loop, power control is slow in order to guarantee the desired power level for a target bit-error rate.

It is important to have an effective power control algorithm. In modern 3G CDMA systems, power control is administered by simply reducing the power or increasing the power with a power control command. Even though in real systems power control is not perfect, power control commands are transmitted in range of thousands of times per second to maximise the system performance. Typically during power control, the received power at the base station resembles a log-normal distribution in which the standard deviation of the signal-to-noise ratio is between 1 dB and 2 dB [22].

[1] This technique is also known as automatic gain control.

2.4 Code Sequences

In modern mobile communication systems, coding schemes are essential for bolstering the information against the harsh radio environment. In CDMA radio there are number of coding schemes that are employed, some being more sophisticated than others. One of the most basic coding schemes are block concatenation schemes in which bits are appended to a block of bits in a data stream in order to add redundancy[1]. The outcome is the ability to detect or detect and correct errors. This can be done at any layer of the communication stack and is usually done at the physical link layer in CDMA systems before spreading.

Convolution coding is another scheme for reducing the error rates in mobile radio. In contrast to block coding, convolution coding spreads the redundant information across the data stream. With convolution coding, long streams of data can be continuously coded without grouping them into discrete blocks to add redundancy [20].

Source coding is another application layer coding scheme usually used to compress video and audio. A video or audio coding scheme, usually referred to as a multimedia CODEC, is designed to be error tolerant in such a way that the media is still presentable but may have glitches (for example lost frames), rather than being totally inaccessible.

In this section there is a focus on spreading code sequences. Typically, spreading implies that the processing gain of a data bit is increased however spreading codes can provide other functions. Intrinsic to CDMA systems, spreading codes are used for channelization. Spreading codes are also used for scrambling to provide added security to the radio channel. Spreading code sequences are chosen based on their auto-correlation properties and cross-correlation properties. Codes that are preferred have lower cross-correlation peaks and low auto-correlation peaks when not in alignment. From henceforth, some of the spreading code sequences, used in modern spread spectrum mobile communications, are discussed.

[1] Redundant information is also referred to as parity-check information. Redundant information is used to check for errors or to check and correct errors.

2.4.1 Pseudo-noise Sequences

In DSSS systems, Pseudo-noise (PN) sequences provide the functions of spreading baseband signals across a wider bandwidth, scrambling, synchronisation and access (the differentiation of logical user channels).

PN sequences, also known as *pseudo-random sequences*, are binary sequences that resemble a randomly generated sequence over a period; also the autocorrelation of a PN sequence resembles that of the autocorrelation of band-limited white noise. In fact, after the spreading of signal is complete, the energy distribution along the frequency domain is uniform and resembles Gaussian white noise. PN sequences can be generated from basic simple feedback shift registers which are easily implemented in hardware, and thus sequences can be reproducible at the receiver. This generation method is a form of linear recursion, whereby each element of the sequence is calculated from *m* elements that precede it. The *m* elements are also referred to as *stages*. These stages can be represented in polynomial form. For example, x^4+x+1 represents a four-stage feedback shift register (shown in Figure 2.6). The example presented (x^4+x+1) forms an m-sequence, therefore, the length of the sequence is 2^4-1 bits or 15 bits long.

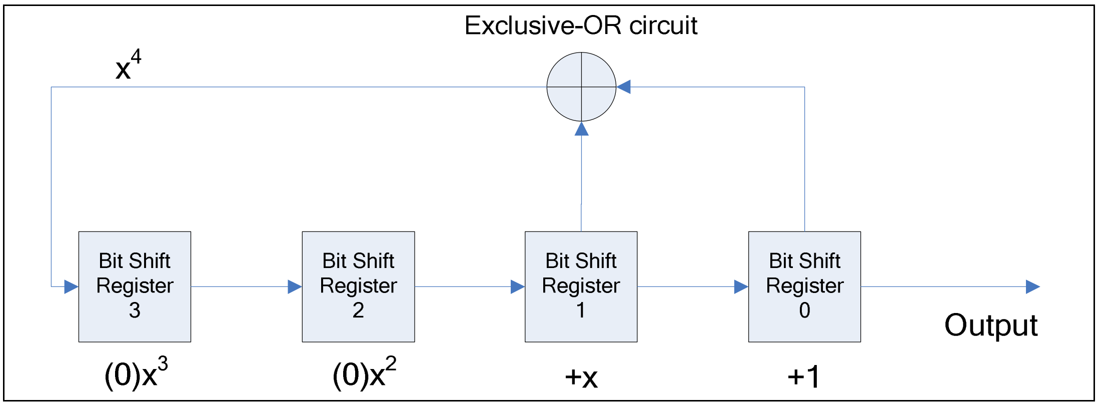

Figure 2.6: Linear Feedback Shift Register Circuit of 4 bits (x^4+x+1)

One of the characteristics of PN sequences is that, on average, it has an equal amount of ones and zeros throughout the sequence. A feedback shift register with *m*-stages can produce a PN sequence with a maximum length of 2^m-1 bits. Maximum length sequences are referred to as m-sequences and are used over non-m-sequences for spreading because of the auto-correlation properties. While codes are being aligned (whilst not yet in perfect alignment), m-sequences have very low auto-correlation peaks in comparison with non-m-sequences which have higher auto-correlation peaks.

2.4.1.1 Gold Codes

Because of the excellent autocorrelation properties of PN sequences, they are used for synchronisation. However, PN sequences can have pronounced cross-correlation maxima. Gold codes are PN sequences that have more favourable cross-correlation properties with slightly reduced autocorrelation performance. Gold codes are created from two parent PN sequence generators with favourable cross-correlation properties. One of the sequences is set to an offset and the resulting bits are combined by way of a logic XOR operation. Like PN sequences, the maximum length of Gold codes is 2^m-1 bits, where m represents the number of stages in both of the parent PN sequence generators.

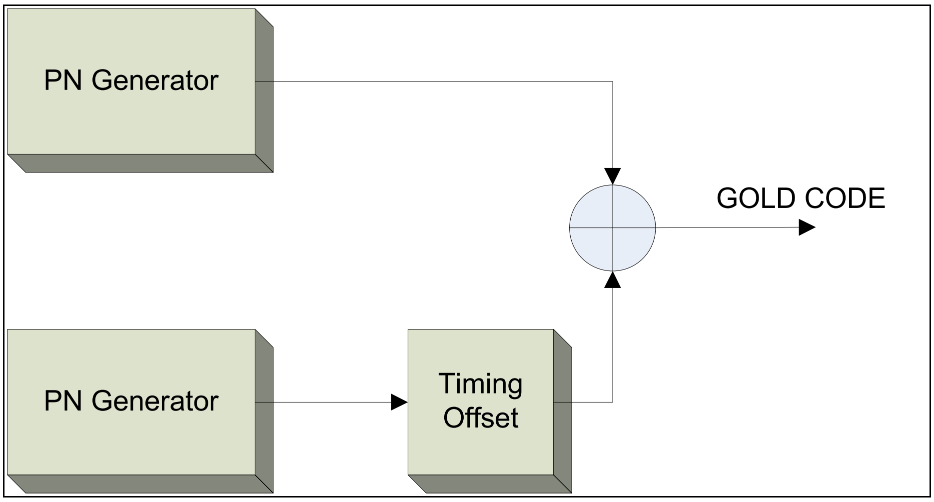

Figure 2.7: Gold Code Sequence Generation

The cross-correlation between any two Gold codes well yield either -*t(m)*, -1, or *t(m)* - 2, where *t(m)* is defined as follows:

$$t(m) = \begin{cases} 1+2^{\frac{m+1}{2}}, m\ odd \\ 1+2^{\frac{m+2}{2}}, m\ even \end{cases} . \tag{2.16}$$

2.4.2 Orthogonal Sequences

The properties of PN sequences are ideal for spread spectrum systems; however, PN codes are not orthogonal and are not as well suited for channel differentiation as orthogonal codes. Orthogonal codes will give a result of zero when cross correlated in perfect alignment. However, not all orthogonal sequences have a uniform spectral density and are used in conjunction with PN sequences in DSSS systems.

2.4.2.1 Walsh Codes

Walsh codes are a set of orthogonal codes defined by J. L Walsh in 1923 [12]. Walsh codes are not only orthogonal but are also normalised, that is when they are auto-correlated in perfect alignment the result is one. Walsh codes are the codes used in most commercial CDMA mobile network systems. In IS-95 (cdmaOne), a fixed length set of Walsh codes consisting of 64 chips is used. For the forward link Walsh codes are used to differentiate user channels. These codes are used to spread the users' data bits. The codes are allocated on call setup. The codes are then used differently in the reverse link in which it is used as a 64^{th} orthogonal modulation scheme[1].

Walsh codes can be generated using Rademacher functions, Hadamard matrices and by exploiting the symmetry of the codes to list a few ways. Code lengths are in powers of 2, hence lengths of 1, 2, 4, 16, 256 chips, etc.

Currently, 3G CDMA systems, namely UMTS, use variable length Walsh Codes. The use of variable length Walsh codes is one of the mechanisms used to control the transmission data rates. Shorter orthogonal sequences increase the symbol rate assuming that the chip rate is constant. Code assignment becomes a critical management issue in systems that use variable length Walsh codes. Longer codes are generated from shorter codes. Codes used within the same cell area, that are related through parentage (or heritage) will have similar chip patterns and information intended for one user can be inadvertently accessible to other users and can cause data stream corruption.

[1] The modulation scheme represents 64 data symbols.

2.4.2.2 Orthogonal Gold Codes

The cross-correlation of some Gold Codes (described in section 2.4.1.1) can yield a value of -1. From this result, it was realized that Gold codes could be modified in order to make them orthogonal. The Gold Codes are padded with '0' such that the cross-correlation result is zero. Due to this padding the length of orthogonal Gold codes is defines as 2^m, where m represents the number of stages in the feedback registers used to define the PN m-sequence parents.

More interestingly, all the padded gold sequences are orthogonal to each other, resulting in $2m$-1 orthogonal Gold codes, provided that all $2m$-1 orthogonal Gold codes are parented from the same pair of PN m-sequences.

2.4.2.3 Smart Codes

The concept of smart codes, as defined in [29], is a code sequence in which the auto-correlation (when the code is perfectly aligned with the start and ending chips) yields a value of 1; and yields a value of 0 when shifted. Another property of a Smart Code sequence is the cross-correlation with another smart code yields a zero for all τ where $\tau \leq \tau_c$ and τ_c is the cross-correlation window that yields a zero value.

Smart codes can be complementary orthogonal code that consists of two sets of codes C and S. The smart code sequence has a length of L chips and the code components C and S have code length $L/2$ chips. The result of the correlation between C and S yields a value of 1 when the codes are perfectly aligned ($\tau = 0$). The codes are not aligned in time ($\tau \neq 0$) then the correlation between the codes is 0.

The number of smart codes that can be used in a system is limited. As a consequence, smart codes are not suitable for a multi-user access system but instead smart codes make an ideal candidate for code division duplexing systems where codes differentiate only the uplink and downlink of a communication system.

Based on above mentioned auto-correlation property, using smart codes completely eliminates multi-path copies of the same sequence at the receiver. In addition, Rayleigh fading and random frequency modulation due to the multi-path components will not be observed by a moving mobile station and hence will not affect the received signals.

Likewise, multi-user interference is also suppressed because of the cross-correlation properties of these smart codes.

2.5 Code Division Duplexing

Code division duplexing (CDD), proposed in [30], is an access mechanism that differentiates data transmission direction. Like FDD and TDD where transmissions can be separated into uplink and downlink channels on different frequencies and different timeslots respectively; uplink and downlink can be separated by spreading codes.

In [17], CDD was proposed as a physical layer solution to providing an ad-hoc overlay to a standard cellular infrastructure, whereby CDD is the basis of the access mechanism that differentiates mobile to mobile transmission from mobile to base station transmissions (using unique spreading codes that identifies the direction). In other words, CDD facilitates (on the physical layer) a peer-to-peer networking setup that still allows for a star network to be layered beneath the peer-to-peer network. There are three categories of directions which are characterized in Table 2.2:

Table 2.2: Code Assignment in CDD [17]

Direction	Code Designation
Mobile Station to Base Station	Code 1
Base Station to Mobile Station	Code 2
Mobile Station to Mobile Station	Code 3

The CDD spreading code is added to ensure that (peer-to-peer) transmissions are perceived as noise to devices in which the messages were unintended for. The proposed coding scheme is shown in Figure 2.8:

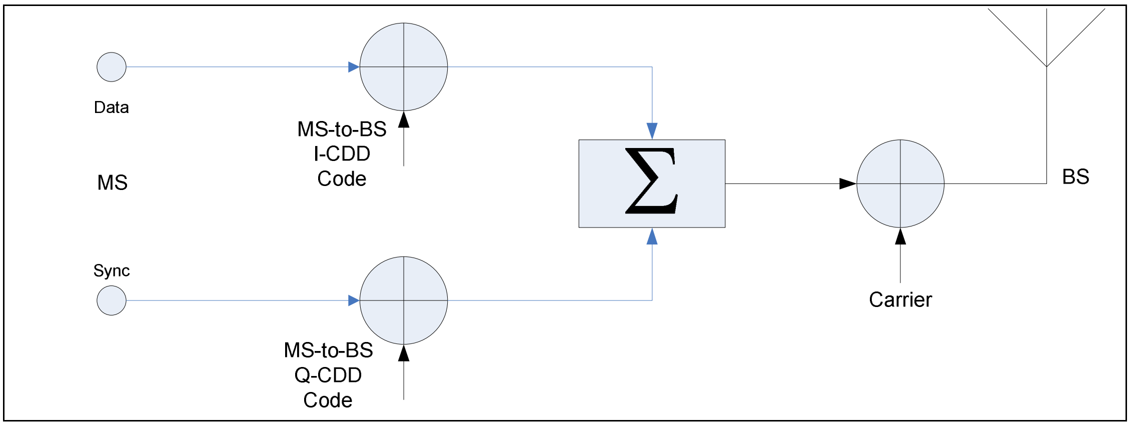

Figure 2.8: Coding For CDD [17]

2.6 Time Division Duplexing

Duplexing refers to the means in which information is sent and received. A half duplex system suggests that information cannot be sent and received at the same time, but instead are mutually exclusive operations. A full duplex system, therefore, refers to a system where information can be sent and received at the same time.

The most popular forms of duplexing are time division duplexing (TDD) and frequency division duplexing (FDD). In FDD the uplink (forward channel) and the downlink (reverse channel) are separated on different frequency carriers. On the other hand, in TDD the downlink and the uplink share the same frequency channel. Within that allocated spectrum, timeslots are defined and each timeslot is assigned an upload or download slot. In the truest sense TDD systems are half-duplex systems where as FDD systems have the capability to be truly full-duplex. However, TDD timeslots are small enough for communications in the system to be perceived as full duplex communication.

2.6.1 Advantages of TDD

The mode of duplexing that is being focused on in this research is time division duplexing because of some of its intrinsic properties. As mentioned earlier, TDD communication modes uses one frequency band; therefore, frequency reuse planning is simplified, unlike in FDD systems which use two bands with different propagation and path loss properties. In a similar respect, power control is simplified as well. Seeing that TDD systems use the same frequency for the uplink and downlink, frequency selective characteristics can be estimated for both uplink and downlink. As a consequence, only open loop power control is required in a TDD-CDMA system.

The simplicity of implementing asymmetric data rates in TDD systems is another attractive property. Slots can be dynamically allocated for the uplink or downlink to allow for change in throughput required in either direction. For example, in a voice conversation, equal timeslots can be allocated for uplink and downlink, whereas in a web-surfing session more timeslots will be allocated to the downlink assuming that a web-surfing session requires a smaller uplink capacity (in which considerably less data needed for control and requests versus content data). In comparison to FDD, the uplink and the downlink frequency bands are symmetrical. As a consequence, the uplink band will be under utilised during the use of downlink oriented applications, thus effectively wasting the scarce radio resource.

In addition to the ease of controlling data rates, TDD systems have been shown to consume less power over their counterpart, FDD systems [18]. Moreover, TDD-CDMA terminals are cheaper to produce *en masse* because they do not require the hardware complexity, to implement the duplexing mode [19].

2.6.2 Disadvantages of TDD

Despite the advantages that make TDD systems well suited for modern day mobile communications, they possess disadvantages that must be considered before selecting TDD mode as the dedicated duplexing mode. TDD systems are known for their lack of coverage due to the critical timing misalignments that occur when TDD frames are received over large distances. Essentially, FDD is more suited for wider area coverage than TDD.

Power pulsing from TDD also causes a noticeable interference, both audible and visible, depending on the electromagnetic equipment within range of the TDD mode radio. However, this physical effect is not detrimental in any respect. In fact this is the same problem that GSM faces as well, and it has not thwarted the rapid popularisation of the mobile system.

In TDD systems, timeslot allocation schemes are not ideal. In real TDD systems timeslots cannot be simply assigned to uplink and downlink based on the user’s need. In such an instance MS units in close proximity may interfere due to the problem of overlapping uplink and downlink timeslots.

TDD systems suffer from inter-cell and intra-cell interference. Outlined in [11], from the perspective of inter-cell interference, downlink and uplink from adjacent cells can share the same timeslot. Likewise, within a cell one user may be using one slot as an uplink slot while another nearby user requires the slot as a downlink slot. This problem can be addressed if the system is perfectly synchronised and all users in the systems use the same symmetry. This problem can also be addressed with the use of code division duplexing as introduced in section 2.5.

2.7 Summary

To summarise, CDMA is the access mechanism that is used as the basis of code division duplexing. Other access mechanisms exist but CDMA possess attributes that make it preferable to be used as a mobile cellular network access method. Direct sequence spread spectrum is one of the types of CDMA, and is the one mostly used in CDMA radio access networks. It is highlighted here because it forms the basis of the proposed CDD design. Multiple access interference is then described to clarify one of the major obstacles that a spread spectrum multiple access system faces; and thus, multiple access interference will also have a detrimental effect on the performance proposed CDD design. The features of a hostile radio channel (path loss, fast fading and slow fading) are then addressed. These are dealt with in order to highlight another obstacle that must be considered when designing any radio system. Power control and the near-far effect are then addressed in order to illuminate another challenge that direct sequence spread systems (like the proposed CDD design). It is then highlighted that pseudo-noise sequences are used for spreading, scrambling, synchronisation and channelization. Orthogonal codes, on the other hand, are used mostly for channelization. At this point CDD was introduced where it was revealed that the duplexing scheme is suitable for use as a peer-to-peer access mechanism. Finally the advantages and disadvantages of TDD were described. One of the key points here is that TDD systems facilitate asymmetric data rates, ease coverage planning, and simplified power control. The proposed CDD design aims to take advantage of the benefits of a TDD based system by incorporating the duplexing mechanism in the design.

With a basic understanding of the theoretical background that is needed for a the design of a spread spectrum system, the report will next address past projects that have dealt with similar designs and aims, more specifically, the use of a hybrid network topology to provide an increase in the performance of a mobile cellular network.

3 RELATED WORK

The CDD-TDD CDMA system aims to create a peer-to-peer network that still uses the infrastructure of the mobile network. In doing so it is necessary to learn from the projects that are similar: projects that look at the performance of multi-hop or peer-to-peer networking that are overlaid on a standard cellular infrastructure (such overlaid networks are sometimes referred to as hybrid networks). The common concept amongst the highlighted projects is that some form of peer-to-peer relaying outside of a direct connection to the base station can potentially increase the overall performance of a standard mobile cellular network. The peer-to-peer relaying is used in fundamentally different ways, such as load balancing, also, it can be used to find most efficient path to the base station or it can simply be used to set up a peer-to-peer connection between nodes.

One of the fundamental differences between the systems that are described and the proposed system is that the proposed system attempts to achieve this hybrid networking topology from the physical layer by way of implanting it through code division multiplexing.

3.1 Code Division Duplexing

In [29], the concept of CDD is being applied to orthogonal frequency division multiple access. Here it is being used purely to differentiate uplink and downlink instead of being used as a means of enabling peer-to-peer communication. The project proposes the use of a unique set of codes that are referred to as smart codes (as explained in section 2.4.2.3) that are designed for the purpose of duplexing.

3.2 Mobile-Assisted Data Forwarding

The concept of "mobile-assisted data forwarding" (MADF) was presented in [32]. MADF is an approach to add an ad-hoc overlay on a fixed cellular infrastructure. The MADF system demonstrated that there is a means of dynamically allocating channels reserved from the network, and then using those channels to re-route traffic from an area with high congestion of traffic (referred to as hot spots) to an area of low traffic congestion. A forwarding agent could be a repeater that is not connected to the cellular infrastructure or another mobile itself.

The discussion in [32] focused on the use of TDMA and Aloha for data transmissions. The level of traffic was measured by sending test packets to ascertain the response time of the base station. Congestion was determined by a specified threshold value. There is another threshold value that represents what congestion has dissipated and when the system should return to a normal routing scheme. In the presence of congestion, free traffic channels from neighbouring cells would be made available to mobiles in congested areas. It was shown that with MADF both the Aloha system and the TDMA system had data throughput improvements.

MADF is similar to the concept proposed by this research in that the scheme is implemented such that it can be used in a homogenous radio access network.

3.3 Heterogeneous Radio Access Networks

Heterogeneous radio access networks are radio networks that utilise more than one type of radio access interface protocol. For instance, a global mobile phone network is usually a heterogeneous radio access network, whereby in cities it would utilise the GSM access network and in areas lacking GSM coverage it would revert to the satellite link interface. Heterogeneous radio access networks have garnered attention with the advent of the popularisation of 802.11 protocols inter-working with standard mobile networks. In addition, there is the school of thought that the 4G mobile network will have a heterogeneous radio access network, seeing that software defined radio will allow for mobile units to seamlessly switch between radio access networks.

3.3.1 Integrated Cellular Ad-Hoc Relaying

Like MADF scheme, the Integrated Cellular Ad-Hoc Relaying (iCAR) system provides an ad-hoc overlay on a standard cellular infrastructure. The iCAR system, which was first proposed in [10], provides an ad-hoc overlay to a standard star topology cellular network. This is achieved by deploying ad-hoc relay stations (ARSs) that operate in the Industrial, Scientific and Medical (ISM) band (2.4 MHz). The ARSs use the ISM band for ARS-to-MS and ARS-to-ARS communications. The ARSs also contain a second cellular radio interface that is used to communicate with BTSs. Consequently, the iCAR system requires that each MS has two radio interfaces. MSs in highly congested cells will have calls made routed through ARSs to a BTS that is situated in a less congested cell. In other words the iCAR system is primarily designed to accommodate load balancing and load alleviation in a mobile system. The dynamic ability to balance the traffic in a mobile network also gives the network the ability to accommodate more traffic and consequently more users.

There are, however, draw backs to such a system that could make it financially infeasible. The complexity of radio planning increases because the two radio interfaces, which operate at different frequency bands, have different radio channel characteristics. Another significant draw back to using such a system is the additional cost of the relay stations. Therefore, an iCAR system would be most suitable in a case where there exists a network that is already heavily saturated and is in need of load balancing.

3.3.2 Unified Cellular and Ad-Hoc Network

In a similar fashion, the hybrid network topology, presented in [9], also takes advantage of the ISM band to provide an ad-hoc overlay to a standard topology cellular network. The architecture proposed in this case is based on the synergy of two wireless technologies, CDMA2000 and IEEE 802.11b. This would also require mobile terminal to have two radio interfaces. Unlike the iCAR system, the Unified Cellular and Ad-Hoc Network (UCAN) architecture uses the 802.11b radio interface to relay data (via IP tunnelling) from MSs (with weak signal strength to the BS) to MSs with a strong signal strength connection with a BS. In comparison to UCAN, iCAR uses added infrastructure, in the form of the ARSs, to implement the ad-hoc overlay. Despite the lack of added infrastructure, the system displayed an increase in throughput of a cell while at the same time increasing a single user's throughput by up to 310% [9].

Like the iCAR system, radio coverage planning becomes more complex in UCAN. Both radio interfaces of the system work on different radio frequencies, and therefore channel characteristics are different for both. As a consequence, it also becomes difficult to plan the coverage areas for political and regulatory purposes. Along with coverage issues, the concern of the usage of the limited power supply of the mobile station arises. The two interfaces on the device will be constantly active most, if not all, of the time. It therefore means that MSs will have to be fitted with more efficient battery technology. This leads into the discussion of the cost of the MS units. Each MS unit would be expected to be more expensive seeing that it would have to physically contain two radio interfaces and a more efficient battery.

3.4 Opportunity Driven Multiple Access

Within the 3GPP specification there is an extension to UMTS Terrestrial Access Network Time Division Duplexing (UTRAN-TDD) mode known as Opportunity Driven Multiple Access (ODMA). The specification itself has been withdrawn by the standardization body because it was deemed to be of no further interest [1]. ODMA is inherently a hybrid topology protocol, because it allows for mobile stations to either communicate with the base station or with another mobile station in the vicinity. In ODMA the radio link can be broken down into a number of smaller hops where MS units are used to relay the data. The MS unit must have the computing ability to calculate the optimum route to achieve the minimum path loss for the end-to-end link. This is one of the disadvantages of ODMA, and is probably one of the reasons why 3GPP lost its interest. Another disadvantage of ODMA is the complexity of the signalling needed to establish a route. In ODMA, MS units do not communicate independently of the base station, .i.e. no peer-to-peer calls are setup, however ODMA can be easily extended to do so. Although ODMA has been abandoned by the standardization body there is research that demonstrates the benefits of the specification.

It has been shown in [25] and [26][1] that in ODMA based TDD-CDMA networks, there is an increase in the capacity of the network and that simple ODMA will lead to a reduction of the overall power consumption of user equipment. The project also showed that multi-hop communication amongst peers resulted in a reduction of power consumption. In that work it was highlighted that power control must become decentralised, and that there is no guarantee that interference will be minimised, but rather, emphasising that there will be cases where mobiles are drowned out because of their neighbours' signal power causing excessive interference. In such instances, their recommendation is to revert to a star topology. Seeing that power control is autonomous where SNI is not easily controlled, they devise a scheme for admission control. Admission control in a hybrid topology network can no longer be based on a simple parameter like the number of active calls. They assume that all calls can

[1] ODMA has also been shown in [26] to extend the coverage limit of a cell in UTRA TDD.

be made and that mobiles that are transmitting at too high a power level are removed based on a power level prediction metric. They described the routing protocols that they employed to be direct routing, simple ODMA and interference based ODMA.

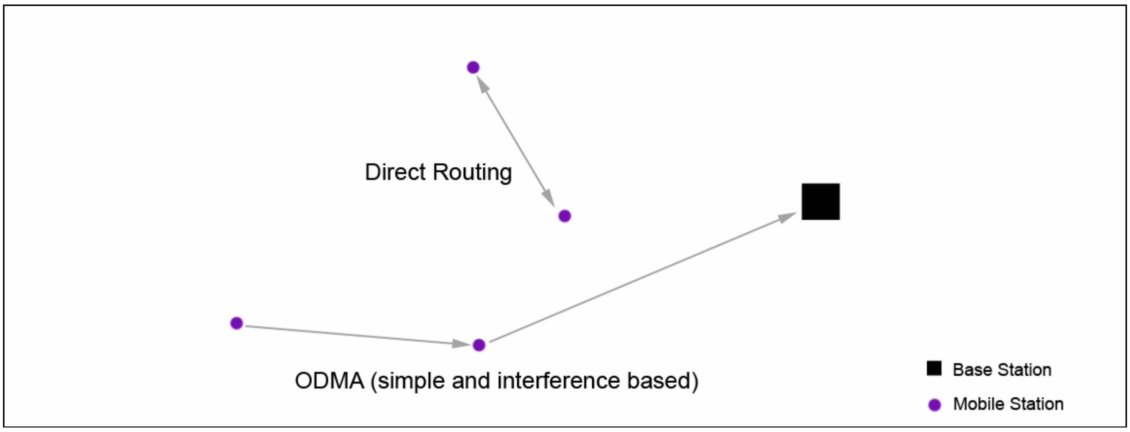

Figure 3.1: Illustration of routing protocols in ODMA

In direct routing[1] an MS will attempt to connect directly to another MS provided that the target MS is within the handover range of the BS administering the cell (shown in Figure 3.1). Simple ODMA is based on the 3GPP standard in which MS nodes route data based on achieving the path with the lowest path loss. In interference based ODMA nodes route data as in simple ODMA, however, after power control has taken effect the path loss metric is adjusted to take in consideration the interference at each MS node. As a consequence, data will be routed based on the modified path loss values. To ascertain the performance of the system, a simulation environment was implemented. The simulation involved three scenarios. The first scenario consisted of four base stations placed at the corners of a square area of 100 m^2. The second scenario involved seven base stations in a hexagon of maximum radius 50 metres. Each base station was located at the corners of the hexagon and one was placed in the middle. The third scenario consists of 20 base stations in an indoor office model, in which the indoor office path loss model, as specified by the 3GPP, was used. Table 3.1 summarises the performance of their system when there is a 1:1 ratio of local and non-local traffic.

[1] Direct Routing is the same as an inter-cell call which is later on described in section 6.3.2

Table 3.1: Supported users per BS for various scenarios for 1:1 ratio of local to non-local traffic [25]

Scenarios	Number of supported users/BS		
	Direct	ODMA	ODMA int
Square, pg=12.9 dB	4	5.5	8
Square, pg=15.9 dB	5	7.5	11.5
Hex, pg=12.9 dB	4.5	6.1	8.5
Hex, pg=15.9 dB	6	9	13.5
3GPP, pg=12.9 dB	1.5	3	4
3GPP, pg=15.9 dB	2	5	6

In Table 3.1, it can be seen that interference based ODMA outperforms the other routing mechanisms.

In [15] a medium access and radio resource management for FleetNet[1] is presented. In their paper it was highlighted that ODMA presents difficulties in terms of power control, radio resource management and synchronisation. Thus, they present a new approach for challenges mentioned earlier in order to achieve ad-hoc networking with a standard cellular infrastructure. Seeing that the network is decentralised, there is a need for a synchronisation scheme to accommodate this topology. It is assumed that there are nodes with access to the Global Positioning System (GPS) and that the nodes will be able to do a coarse time synchronisation. In order to achieve a more accurate synchronisation amongst nodes, each node transmits a synchronisation sequence at the beginning of each transmission. Since each node then has to evaluate the accuracy of the synchronisation burst. As a consequence, there is an addition parameter (known as the ‘sync-level’) that qualifies the level of synchronisation. Nodes with access to GPS are deemed to have the highest sync-level and

[1] FleetNet is an ad-hoc network that utilises UTRA TDD. It aims to provide voice and data communications for vehicles in an ad-hoc manner, namely “car-to-car”.

nodes whose clocks are running freely are deemed to have low sync-level. Based on this, nodes will set their time base based on the node with the highest sync-level. The connectivity protocol proposed dictates that only one station can occupy one timeslot. Each node connects to the target nodes at the same time in the code domain. For instance, MS1 is only allowed to use timeslot 1 but within that timeslot it communicates to MS2 and MS3. While doing so, MS1 can allocate different data rates to each connection by allocating multiple codes to each connection. For radio resource allocation they use a Reservation ALOHA (R- ALOHA) scheme, where reservation is implemented with in-band signalling (piggybacking). They showed that the delay of using R- ALOHA remains almost constant provided that the load levels do not surpass the saturation point.

In a similar vein to [25], it is shown in [16] that relaying can benefit a standard CDMA cellular infrastructure. They showed that relaying can induce an overall power saving in MS units for low to medium loads. Their investigation also revealed that significant path loss reduction can be achieved by relaying. On the other hand, it was suggested that as the network load is increased, the performance of relaying diminishes and hence at high network loads the standard infrastructure setup is preferable. Unlike [25], a CDMA FDD system was considered for the simulation model. Since ODMA is an extension to UTRA TDD, this project is not strictly ODMA based. The simulation techniques used takes snapshots of a realistic scenario in order to investigate the relaying and its effect on the average transmit power. The simulation consists of numerous snapshots to achieve statistical convergence, and thus more reliable results.

3.5 Capacity in Hybrid Networks

From a more theoretical perspective, it has also been shown that ad-hoc networks that are given infrastructure support will also benefit from an increase in capacity and vice versa, as demonstrated in [28], [2], [4] and [23]. The theoretical analysis provides an insight into the performance of hybrid networks irrespective of the radio access network (i.e. modulation schemes, coding schemes and user multiplexing) and path loss models. In all cases there is the assumption that channels are perfectly orthogonal, namely, the effects of MAI, ISI and inter channel interference (ICI) are ignored.

Reference [28] looked at the theoretical gains that can be attained by adding infrastructure support in terms of the maximum end-to-end data that can be achieved between a pair of nodes. The system model involved nodes that could communicate with themselves as well as the access points associated with the infrastructure. It is assumes that the infrastructure has an abundant amount of bandwidth (which implies that the infrastructure network can never be the bottle to performance). Ad-hoc nodes and access points are randomly distributed in a circular area πR^2, R is the radius of the disc. They assume that there is an available bandwidth to each node of W bits/sec over multiple orthogonal channels in which the N nodes are able to use. They go on to show that the capacity of the hybrid network is $\Theta(W/\log(N))$. This is a significant improvement over the performance of an ad-hoc network with no infrastructure support, which was found to be $\Theta(W/\sqrt{N\log(N)})$. The model, however, suits scenarios in which the access points are randomly place, for instance, a cellular/WLAN scenario. This suggests that the improvements would not necessarily translate to homogeneous RAN hybrid mobile cellular networks. In a homogeneous RAN mobile cellular network, the equivalent to an access point would be a base station. Base stations are normally laid out in a deterministic manner.

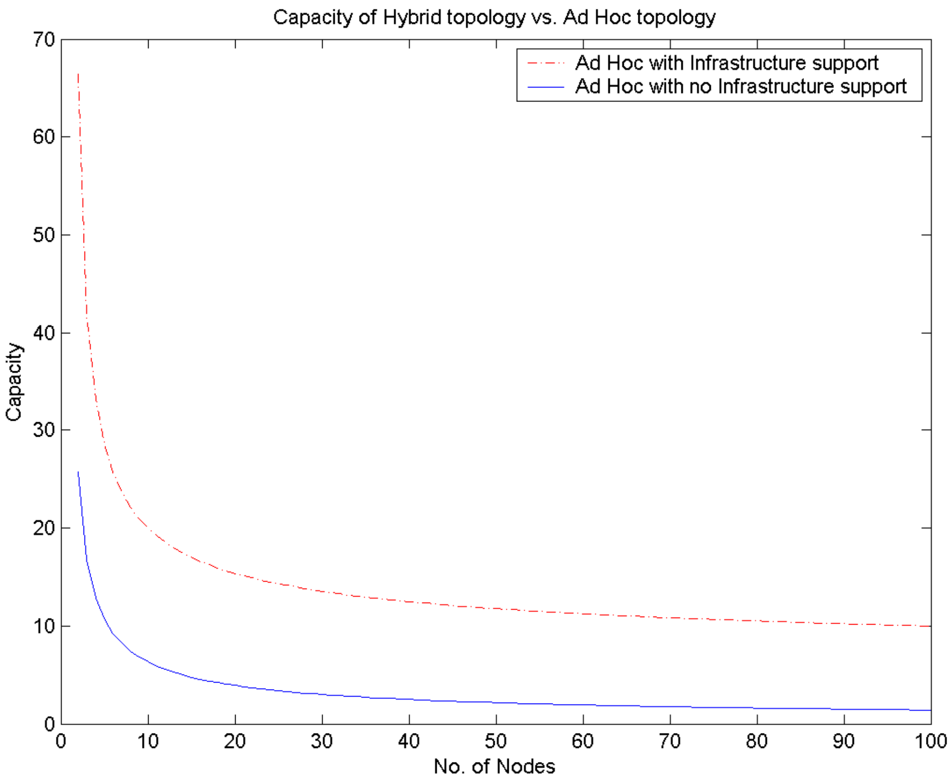

Figure 3.2: Comparison of the Capacity of a Hybrid Topology vs. Ad-hoc Topology with constant band width per node [28]

In [4], the model that they use to assess the performance of a hybrid network consists of access points which have a deterministic layout pattern, more akin of a traditional mobile cellular network. There are n nodes and m base stations in the hybrid network. Each node is capable of transmitting at W bits/sec. Two routing strategies are considered. In the first, if the destination node is in the same cell or within k nearest neighbouring cells, the source node will then forward data in an ad-hoc manner; otherwise it will use the infrastructure. For the second strategy, nodes route via the ad-hoc mode or the infrastructure mode based on a probability 1-p or p respectively. It was found that for the first routing strategy, if m increases at a rate less than the root of n, then the addition of the infrastructure does not provide any benefit. In the instance of the second routing strategy, it was found that the network behaves like a pure ad-hoc network if m increases at a rate less than $\sqrt{\frac{n}{\log n}}$, thus there is no gain from using the infrastructure. If m grows faster than $\sqrt{\frac{n}{\log n}}$, then maximum throughput is proportional to the number of base stations, and is represented as $\Theta(mW)$. The implication of these findings for a mobile cellular network is that in order to benefit from the hybrid topology, a large number of base stations is required in comparison to mobile stations. This is not always feasible, depending on the environment.

In [2], the capacity of a hybrid network was analysed using a protocol and a physical model. Both models consist of n nodes in a disc of unit area. In the protocol model, no node can be a sender and a receiver at the same time and each sender-receiver pair has associated with a region of interference, where the condition for a successful transmission is when neither the send nor the receiving node is not in the region of interference of another sender-receiver pair of nodes. In the physical model there is path loss and noise such that the criteria for successful transmission is that the signal-to-interference plus noise ratio must be over a specified value. Using this network model it was shown that, if power control is employed, a hybrid wireless network allows for better scaling of network capacity. It is also shown that the capacity of the hybrid network is improved with power control (in comparison with no power control).

3.6 Summary

To summarise, the synergy of the two networking topologies promises to provide capacity and even coverage benefits. From a simulation stand point, the benefits are evident in that the capacity of the system increases. From a theoretical stand point the benefits are in the scaling of the network. In this project, a novel physical layer implementation of a homogeneous radio access network hybrid topology architecture using code division duplexing was proposed to differentiate between peer-to-peer traffic and traffic between the network and the mobile station, which, to knowledge, has not been done before being proposed in [17].

This chapter addressed projects that looked at the use of a hybrid topology as a means of improving the performance of a cellular mobile network. In particular, the ODMA based projects presented in [25] and [26] are closely related to the proposed CDD design in the sense that they employ a TDD based homogeneous radio access network to implement a hybrid topology mobile cellular network. In [25], the authors showed that the number of users that the base station could support would increase. They also pointed out that the autonomous power control does not guarantee interference reduction and that localised interference becomes a problem for MS nodes.

Now that previous projects have been looked at, the next chapter will present the design of a CDD system. Some elements adapted from these past projects help to bolster the proposed design, which will address the physical layer to the networking layer. The proposed routing rules for the CDD/TDD CDMA system are based on the simple ODMA routing rule which selects the best path based on the path loss to the base station (also used in [25]). The method of simulation selected was based on taking multiple snapshots of the environment, which was also used in [16].

4 CDD SYSTEM DESIGN

A ground-up approach is taken in the design of the proposed hybrid topology CDD network. In this chapter a CDD based CDMA radio network is approached from the physical layer, medium access control to the network layer. The Physical layer is based on the definition of CDD; as a consequence, CDD is analytically defined first. Following that, the physical layer of the system is defined based on the analytical definition. The medium access layer and the network layer are described briefly in order to give details of the considerations for the topology.

4.1 CDD Definition

Let $d_k(m)$ be a sequence of binary data where m represents m^{th} bit in time where $d_k(m) \in \{1, 0\}$ then let $b_k(m)$ be the bipolar representation of the m^{th} bit of $d_k(m)$ where $b_k(m)$ is represented as

$$b_k(m) = 2d_k(m) - 1. \tag{4.1}$$

Then let bipolar bit sequence for the k^{th} user be represented as

$$b_k(t) = \sum_{m=0}^{M-1} b_k(m)\Pi_{T_b}(t - mT_b), \tag{4.2}$$

where M is the number of bits sent in the stream, T_b is the time period of a data bit symbol, and where $\Pi_\tau(t)$ is a unit rectangular pulse defined as

$$\Pi_\tau(t) = \begin{cases} 1 & 0 < t \le \tau \\ 0 & \text{elsewhere} \end{cases} \tag{4.3}$$

Let the Walsh code sequence for the k^{th} user be represented as

$$\Omega_k(t) = \sum_{n=0}^{N-1} W_k(n) \Pi_{T_c}(t - mT_c) , \tag{4.4}$$

where $W(n) \in \{1, -1\}$, T_c is time period of a Walsh code bit symbol, and where N represents the length of the Walsh code for the kth user.

At this point, there is a representation for each user's bit stream and a representation of the Walsh code that will be used to spread and consequently differentiate each user. There is one final piece to the puzzle, and that is the representation of CDD code that will be used for direction differentiation.

Orthogonal Gold codes are used in this system for duplexing. Orthogonal Gold codes make an ideal choice because of the properties they possess. Due to their orthogonal nature, they can be used for channelization and hence used for duplexing. They also have the properties of a pseudo-random sequence, and hence can be used for scrambling. However, unlike PN sequences, orthogonal Gold codes have better cross-correlation properties, whereby the cross-correlation maxima are not as pronounced as PN sequences. Even though orthogonal Gold codes can be used for channelization, Walsh codes are used for this purpose. As mentioned earlier, Walsh codes are orthogonal codes that are primarily used for channelization.

Let the orthogonal Gold code, $C_k(t)$, for the kth user be represented as

$$C_k(t) = \sum_{l=0}^{L-1} G_k(l) \Pi_{T_p}(t - lT_p) \cos(\omega_p t) , \tag{4.5}$$

where $G_k(l) \in \{1, -1\}$, T_p is the time period of a orthogonal Gold code bit symbol and where L represents the length of the orthogonal Gold code sequence. $T_c/T_p \in \{Nat\}$ and $T_b/T_c \in \{Nat\}$, where *{Nat}* represents the set of positive whole numbers. In other words, the orthogonal Gold code can be used to even spread the chips further. T_b is referred to as bit period, T_c is referred to as the chip period and T_p is referred to as the 'pip' period.

The transmission of the k^{th} user of CDD can be defined as follows:

$$s_k(t) = \sqrt{\alpha_k}\sqrt{P_k}\left(b_k(t)\Omega_k(t)C_k(t)\right), \tag{4.6}$$

where $b_k(t)$ represents the user data stream, $\Omega_k(t)$ represents the unique Walsh code of the user and $C_k(t)$ represents the orthogonal Gold code used to provide the code division duplexing to uniquely identify the direction of the transmission. With the definition of a CDD transmission for each user, the amalgamation of *K* user transmissions can be expressed as

$$r(t) = \sum_{k=0}^{K-1} s_k(t) + n(t). \tag{4.7}$$

To retrieve (de-spread) the single user's data stream from $r(t)$, the Walsh code associated with that user and the duplexing code are used to extract the relevant bits stream. The de-spreading process can be represented as follows:

$$U_0 = \int_{m}^{(m+1)T_b} \left[\sum_{k=0}^{K-1} s_k(t) + n(t)\right]\Omega_0(t)C_0(t)dt, \tag{4.8}$$

where U_0 is the decision statistic at the receiver for the m^{th} bit. Assuming that we are receiving the 0^{th} bit of M bits then the integration period for a bit will be represented as $\int_0^{T_b}(\cdot)dt$ instead of as $\int_m^{(m+1)T_b}(\cdot)dt$.

U_0 can further be expanded into:

$$U_0 = \int_0^{T_b}\left[b_0(t)\Omega_0^2(t)C_0^2(t) + n(t)\Omega_0(t)C_0(t) + \left[\sum_{k=1}^{K-1} b_k(t)\Omega_k(t)C_k(t)\right]\Omega_0(t)C_0(t)\right]dt, \tag{4.9}$$

where the desired portion of the signal is represented as

$$D_0 = \int_0^{T_b} b_0(t)\Omega_0^2(t)C_0^2(t)dt\,, \tag{4.10}$$

the contribution due to noise is represented as

$$\eta = \int_0^{T_b} n(t)\Omega_0(t)C_0(t)dt\,, \tag{4.11}$$

and the contribution to MAI is represented as

$$\zeta = \int_0^{T_b}\left[\left[\sum_{k=1}^{K-1} b_k(t)\Omega_k(t)C_k(t)\right]\Omega_0(t)C_0(t)\right]dt\,. \tag{4.12}$$

Therefore, the decision statistic for a single for a single bit is represented by

$$U_0 = D_0 + \eta + \zeta\ . \tag{4.13}$$

4.2 Physical Layer

In designing the CDD system, there are two segments that are considered, the transmitter radio and the receiver radio. The transmitter selects the appropriate CDD code to transmit while the receiver always applies the necessary codes to extract the required information. The concept of using code division duplexing for peer-to-peer communication primarily depends on the changes in the physical layer of a communication protocol stack for the idea to be implemented. With CDD defined in section 4.1, it is now easy to conceptualise the physical layer and how it will slot into the rest of the protocol stack. For instance, the transmitter will have the ability to duplex the user’s information; however the transmitter will need information about the destination which can come from control planes or higher layer information.

4.2.1 MS Transmitter Structure

The transmitter structure depends on higher layer functionality to determine the code required for the correct duplexing direction. The data stream from layer 2 is spread by the user Walsh code (at a higher chip rate). The Walsh code is determined by the control functionality in order to control the channel that the user is assigned. The user is not assigned a static address, as this will limit the flexibility of the system. The addressing can be controlled centrally by the network, and therefore can be dynamically assigned for the network traffic management purposes.

The spread data stream is then multiplied by the CDD orthogonal Gold code which is used to determine the direction of the transmission. The resulting data stream is then modulated and transmitted into the air by a BPSK transmitter.

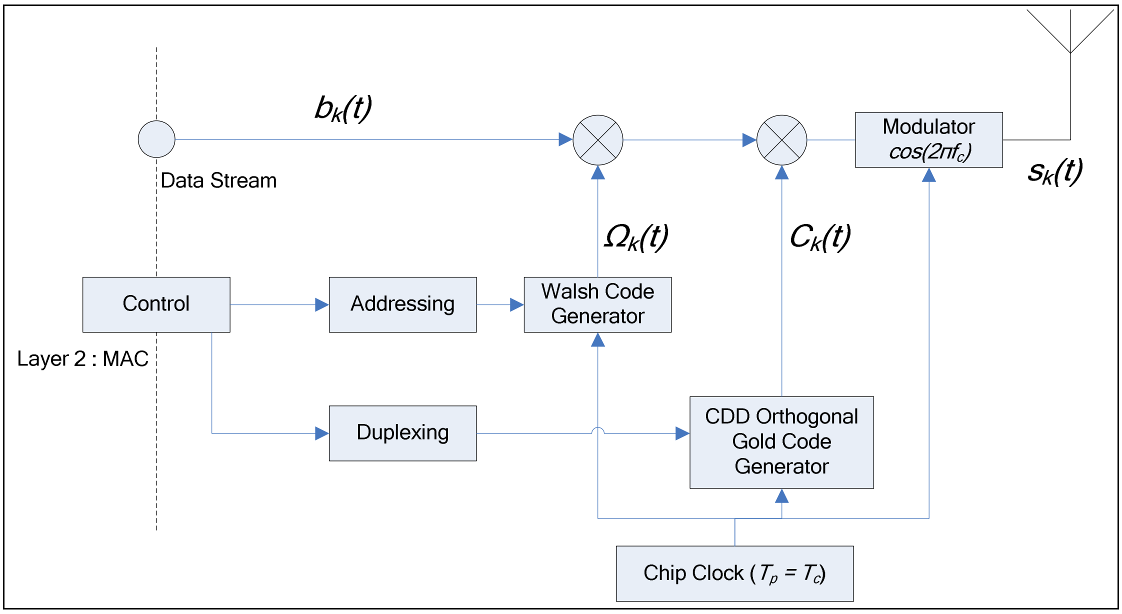

Figure 4.1: MS Transmitter Structure

It is assumed that $b_k(t)$ is the data stream after post processing (interleaving, block forward error correction and convolution coding). It is also assumed that $s_k(t)$ is transmitted over a Gaussian channel.

4.2.2 MS Receiver Structure

Based on the information received from the handshaking protocol, the receiver structure selects the appropriate orthogonal Gold code for the de-spreading of the incoming waveform via the Gaussian channel. The process is linear, so the spreading is not affected by the order in which it is spread. Therefore, the CDD orthogonal Gold code can be used to de-spread the signal first as well as the Walsh code. The resulting data stream is then multiplied with the user Walsh code and then sent to the higher layers for error correction and so on. In both the receiver and the transmitter instances, it is assumed that the 'pip' rate is equal to the chip rate.

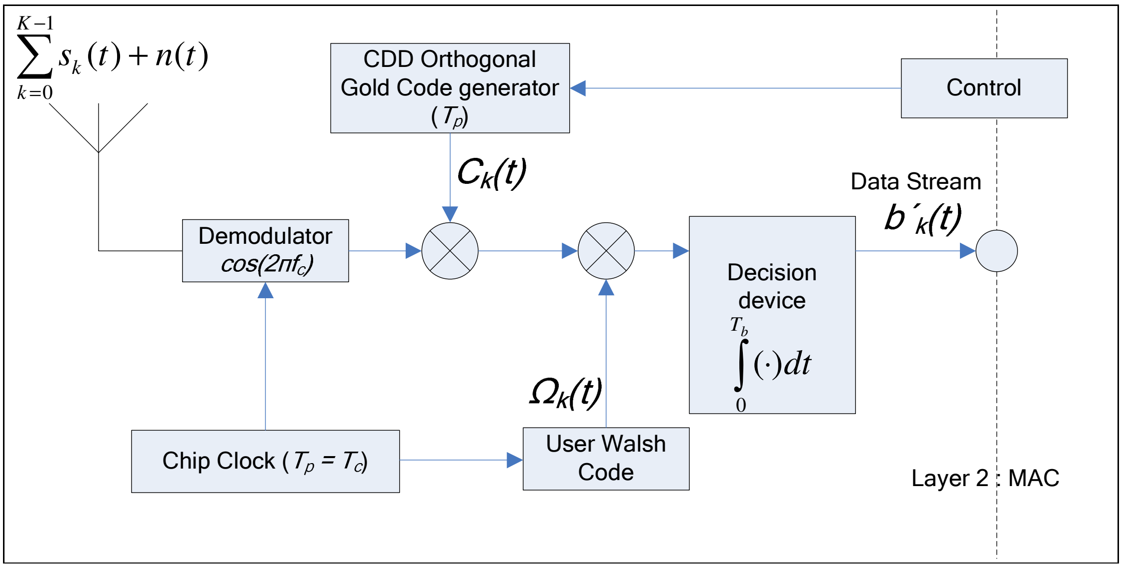

Figure 4.2: MS Receive Structure

4.2.3 Link Budget

The next consideration for the system is the link budget. The maximum allowable path loss is calculated based on the direction of the transmission. Seeing that there are three duplexing directions, there are hence three maximum allowable path loss values. The link budget is used to calculate the maximum allowable path loss and is shown in Table 4.1. It is based the speech service being at a user data rate of 14.4 Kb/s and an operating frequency of 800 MHz. The link budget is based on recommendations for 3rd generation mobile radio found in [5] and [21].

Table 4.1: Link Budget - Speech - Indoor/Pedestrian @ 800 MHz @ Rate: 14.4 Kb/s

Item Label	**Item**	**Units**	**Reverse Link**	**Forward Link**	**Peer2Peer Link**
a	Maximum Transmission Power/Traffic Channel	dBm	23.00	27.00	23.00
b	Body Loss	dB	3.00	3.00	3.00
c	Transmitter Antenna Gain	dBi	0.00	17.15	0.00
d	Transmitter EIRP/Traffic Channel [d=a-b+c]	dBi	20.00	41.15	20.00
e	Receiver Antenna Gain	dBi	17.15	0.00	0.00
f	Transmission Cable and Connector Losses	dB	3.00	3.00	0.00
g	Receiver Noise Figure	dB	5.00	5.00	5.00
h	Receiver Noise Density	dB/Hz	-174.00	-174.00	-174.00
i	Receiver Interference Margin	dB	6.00	6.00	6.00
j	Total Effective Noise Plus Interference Density [j=g+h+i]	dB/Hz	-163.00	-163.00	-163.00
k	Information Rate -10log(14.4Kbs)	dB	41.58	41.58	41.58
l	Required E_b/N_0	dB	4.00	6.00	6.00
m	Receiver Sensitivity	dBm	-117.42	-115.42	-115.42
n	Hand Off Gain	dB	4.70	4.70	4.70
o	Explicit Diversity Gain	dB	0.00	0.00	0.00
p	Log-normal Fade Margin	dB	10.30	10.30	10.30
q	Building Penetration Loss	dB	0.00	0.00	0.00
r	**Maximum Link Path Loss [q=d-m+e-f+o+n-p-q]**	**dB**	**145.97**	**147.97**	**129.82**

4.3 MAC Layer

Information sent from the higher layers of the protocol stack is translated such that the information can conform and be slotted into the logical channels provided. The logical channels are then formatted so that information can then be transported on the physical channels. The MAC layer is responsible for link management (the connecting and releasing of connections) and error management (error detection and maintaining protocol data unit sequence).

4.3.1 Simple TDD Protocol

In terms of the physical channels, the radio resource is divided into the code domain and the time domain. Users are distinguished in the code domain, origin to destination is distinguished in the code domain and uplink and downlink in the time domain. Within the time domain, any user can use any slot for either uplink or downlink at anytime. Unlike physical channels defined in the code domain (which are centrally assigned and administered by the network) MS units can freely access timeslots.

The frame structure of the MAC protocol is based on the 3GPP project FRAMES[1] specification for TDD mode transmission frame structure [21], in which there are 16 timeslots available in a frame; and a frame is 10 milliseconds in length and each timeslot is 625 μs long (see Figure 4.3). The finalized specification decided upon, by the 3GPP, for UMTS UTRA-TDD was a 10 millisecond frame with 15 timeslots rather than 16 timeslots. 16 timeslots was preferred simply because it meant that the length of a timeslot would be 625 μs and not 667 μs which is actually an approximation.

[1] Future Radio Wideband Multiple Access System (FRAMES). It is a standardisation project from the European Advanced Communication Technology and Services initiative (ACTS) which aimed to define the radio access interface for UMTS.

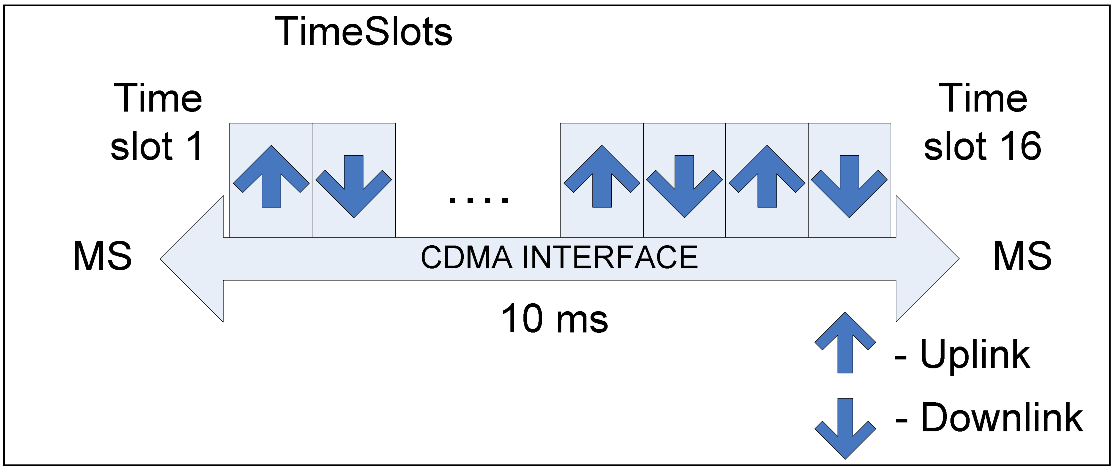

Figure 4.3: Graphical Representation of TDD

Of the sixteen timeslots there is one timeslot always reserved for either the uplink or downlink for a single unique connection link. This ensures that important control information can always be transmitted, even if the application is downstream or upstream based.

Each timeslot can be used on a per frame basis to maintain a mobile-to-mobile connection link. This allows each mobile to have up to 15 peer-to-peer connections, provided that each link is delay tolerant. Doing so is impractical, and thus there must be a timing restriction enforced. Each mobile station is allowed to have a recommended three unique connection links to either three other mobile stations or two mobile stations and the presiding base station of the current cell. Therefore, if all links are in use, two timeslots are reserved for the link, where one is for the downlink traffic and the other for the uplink traffic. In the case where there is a single link to an MS, all the timeslots are dedicated to the connection; in the case where there are two links to an MS, eight timeslots are dedicated to each link; and in the case where there are three links to an MS, eight timeslots are dedicated to one link and four timeslots dedicated to the two remaining links.

4.4 Network Layer

The network layer of a hybrid topology network is considerably more complex than a traditional network. The added overhead, due to routing, on the MS unit and on the network itself is significantly complex. In addition, there are other factors in the networking layer that have to be addressed for this type of topology. These added complexities are possible parts of the reason that hybrid topologies were not incorporated in current 3GPP and 3GPP2 standards. In this section, some of the issues that are associated with hybrid topologies are briefly addressed namely call setup and routing.

4.4.1 Billing Oriented Call Setup

One of the considerations that must be taken into account when designing a hybrid topology network is the call setup and the repercussions on billing. In this section, a billing oriented call setup is proposed. This specific instance looks at a MS-to-MS direct call. It is assumed that call setup along with billing authentication will proceed as normal if the link is between the MS and the BS and visa versa. The aim of the billing oriented call setup is to only establish a call when a mobile is in the presence of BS (i.e. the MS is under network administration). For example if a MS unit is one hop away from the BS, that MS is deemed to be in the presence of the BS, and thus that MS will be able to execute transactions with the rest of the network (see Figure 4.4); otherwise, an MS will remain without service. If the originating and target MS nodes are in range of each other but are unable to contact the network via the base station directly or via a hop to the base station, the MS units will not be allowed to proceed with the call.

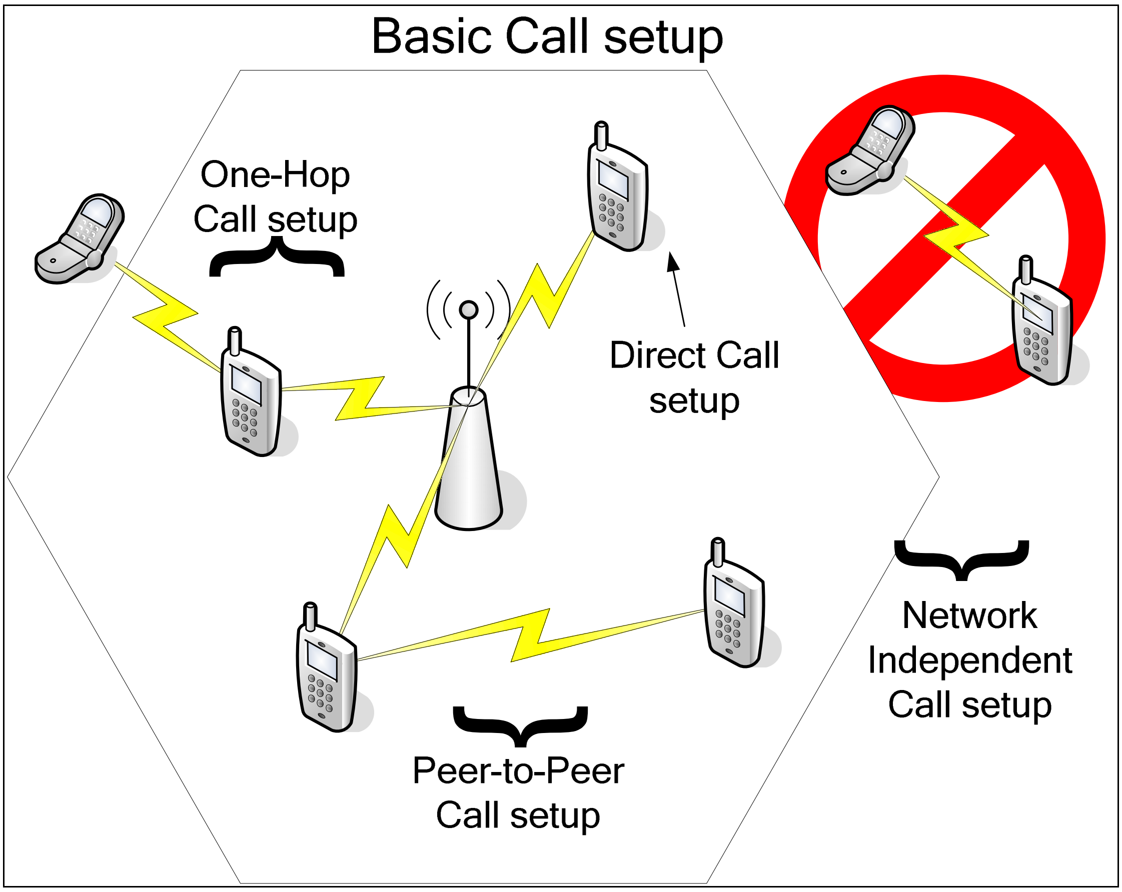

Figure 4.4: Illustration of Call Setup Scenarios Message Sequence Chart

Figure 4.5 summarises the call setup protocol for the system during a peer-to-peer call and is based on the call setup used in GSM (which is in turn based on the call set up in Q.931 [8]). Every MS attempting to instantiate a call must communicate with the base station. The call setup protocol is as follows:

1. The mobile device must access a random channel in order to send a request for a traffic channel that has been assigned by the network. The method by which a traffic channel is assigned shows the traits of a centralised networking topology. However, one of the main purposes behind doing this is to ensure that the mobile device cannot communicate without the permission of the network. If the initiating MS is out of BS radio range, it will send a hop-request. Any other MS within range of the hop-request, with the resources to accommodate the initiating MS will become the proxy for that MS.

2. The network then assigns a channel to the originating mobile. In Network Managed Code Assignment the network will assign a CDD code such that code sharing is minimised (as described in section 5.5.1).

3. The initiating mobile then notifies the network of the type of service it is requesting; in this case it sends a voice service request.

4. All the security features of the impending call are negotiated with the mobile device making the call and the network before the call setup is complete. Therefore, at this point, the mobile sends a security setup request.

5. The network will then respond with security information that includes the cipher to be used and the cipher mode.

6. The originating mobile then sends a call setup request that contains the details of the target mobile.

7. The network will respond to the mobile with information that confirms that the network is prepared for the call, that the number being dialled is in fact a mobile device and also information of the cell location of the target mobile.

8. The MS initiating the call will send out a search request known as a *ping* to commence the hand shaking procedure with the target MS. This ping is only broadcast if the call setup response indicates that the target MS is in the current cell or in a neighbouring cell. The ping contains the address of the originating MS, the address of the target MS and a session ID. The ping is sent for two reasons. The first reason being that there could be a possibility that the target MS has moved out of cell coverage range, but is still within the transmitting vicinity of the MS making the call. The second reason is that, there is no guarantee that the target MS is able to receive even if the target MS is in the transmission vicinity of the MS originating the call. At this stage, the target MS has not yet received any form of transmission. When the network returned the possible whereabouts of the target MS to the originating MS, this was merely a query of the home location register (HLR) database.

9. The originating mobile will notify the network that it has sent a ping.

10. Once the target MS receives the ping, which is sent on a known broadcast channel, the target MS will notify the network (using the information found in the ping) that it has received a ping and the network will assign a channel to the target MS. The target MS will begin the procedure to make a peer-to-peer connection using the orthogonal Gold code assigned by the network.

11. The target MS will then send an acknowledge response to the ping (known as a *pong*) on the channel that has been assigned to both devices. Like the ping, the pong contains the session ID but, unlike the ping, the pong contains the status of the target MS, which could indicate the device is busy or available.

12. The pong will give the initiating MS information of the path loss to the target MS based on the received power of the signal. Assuming that the target MS remains stationary, then the path loss from the originating to the target mobile is the same. This is an advantage of a TDD system, seeing that uplink and downlink have the same path loss characteristics. Provided that the target MS is available and the path loss is tolerable, a peer-to-peer connection will be setup.

However, there stands a chance that the target mobile is not in signal range of the ping. As a consequence, the network continues the process of instantiating the call connection. For the network to continue to do this, the originating MS has to report a ping timeout error back to the network. The network, which still has the session information that was requested from the HLR, can now check the status of the target MS. If the target MS is available, the network will continue the usual call setup protocol, and instantiate the voice mode call through the network.

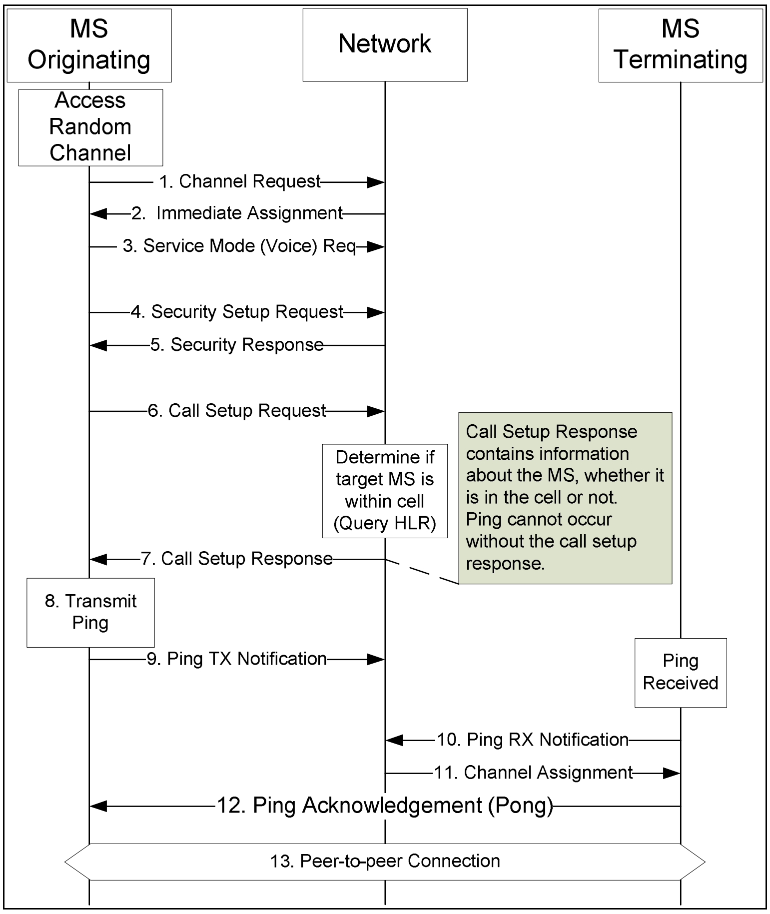

Figure 4.5: Call setup procedure

4.5 Routing

In a mobile network that supports peer-to-peer transmissions, the routing decisions become more complex because of the costs factors associated. Like a traditional data network, route costs include the number of hops to the destination and the total time to the destination. In addition to these factors, there are added factors to be considered in routing such as the signal-to-interference plus noise ratio, battery life and radio resources occupied.

In order to reduce the complexity of the CDD network design, routing is based on path loss. It is assumed that MS nodes have full knowledge of all the other nodes' path loss to the base station and that each MS node has full knowledge of the path loss to every other node. As a result, a route is chosen based on the node that has the smallest path loss to the base station[1]. Also to avoid complex routing table, the network is designed to use a one hop strategy.

4.5.1 Transmission Strategies

Transmission strategies affect the efficiency of the network, and hence they play a critical role in the design of the network. The way nodes connect can have an effect on the coverage and the capacity of the network. Transmission strategies can be defined as the method in which each node attempts to make a connection and how it responds to the network environment.

A hop is a term for a transmission relay point and a slot is the term for the allocated resource for a either a peer-to-peer link or a link to the base station.

4.5.1.1 One Hop-One Slot

In a one hop-one slot strategy, a mobile has the opportunity to make a connection to one other mobile in order to make a connection to the base station. The mobile that provides the route to the base station is the first and only hop to the BS. The mobile providing the route reserves one channel available for the hop. Therefore, any other mobile attempting to use that mobile as a route will be denied resources. The advantage of this strategy is in its simplicity to implement and manage.

4.5.1.2 Multi Hop-One Slot

In a multi hop-one slot strategy, a mobile device has the opportunity to make a connection to the base station by way of one or more mobiles. Each of these mobiles is known as a hop and each of which only provides one slot for routing. Multi hop strategies are expected to have higher coverage capabilities than their single hop counterparts. However, one of the disadvantages of Multi-hop strategies for mobile networks is that as the number of hops increases the complexity of calculating the optimum routes increases exponentially. For example, if a route must be chosen using nine hops and each MS unit has eight neighbours then there are 8^9 possibilities or 134,217,728 possible routes.

[1] This strategy is also employed in ODMA.

4.5.1.3 One Hop-Multi Slot

In a one hop-multi slot strategy, many mobiles may use a single “relay” mobile as a route or hop to the base station. In such a strategy the mobile that acts as the hop must set a limit on the number of connection it can accept. This has to be done to ensure that the routing mobile has reserved adequate communication slots for its own communication to the base station.

4.5.1.4 Multi Hop-Multi Slot

The multi hop-multi slot strategy allows for many mobiles to make many hops to connect to the base station. Each hop or routing mobile device will allow for several connections until the connection limit is reached. Of the strategies mentioned, this one provides the largest degree of freedom in terms of the path that can be taken to a connection point. Such networking environments are complex and become difficult to manage. On the other hand, this strategy has potentially the greatest capacity and coverage advantages.

4.6 Summary

In this chapter, the design for a CDD based mobile cellular network was proposed. For the physical layer, the use of orthogonal Gold codes was recommended to implement the directional CDD codes. In chapter 5, the performance of CDD was then evaluated in a set of point-to-point simulations. In the MAC layer design, the concept of slots is proposed in which a MS unit is able to share its resources. There it was highlighted that, by design, an MS unit is able to share resources with up to two other MS units with resources reserved for communication with the base station. In terms of the network layer design, call setup does not allow mobiles to proceed with call setup if the mobiles have no access to the base station directly or via a hop. The design elements that stem from the MAC layer and the network layer were considered in the coverage and capacity analysis simulations, presented in chapter 6.

5 PERFORMANCE OF CDD

This novel approach to channel duplexing is analysed in this section. Code division duplexing is analysed by setting up point-to-point transmission links that are simultaneously and continuously transmitting. Here, the tools and procedures are outlined along with the results that are associated with the initial simulation setup.

5.1 CDD Simulation: Tools and Procedure

MATLAB was used to create a simulated environment where multiple CDD users would simultaneously and continuously transmit randomly generated data at the baseband level. For each user, modulated data would have noise added to it. Afterwards, CDD receivers would then demodulate the data, where the sent data and the received data are compared to ascertain the bit-error rate.

MATLAB was chosen because of its ease of use for generating mathematical functions, and the wide assortment of pre-written scientific functions. There are some disadvantages in using MATLAB. The object oriented nature of the scripting language is not easy to manipulate. MATLAB is slow; primarily due to the scripting language it uses which is interpreted line by line.

The first step in setting up the simulator was to ensure the accuracy of the base band modulation scheme. Binary phase shift keying (BPSK) is the modulation technique that was selected for its wide spread use in CDMA systems, and its basic simplicity. In order to verify the baseband version of the function, it is used to send data (in the order of 10,000 bits) in the presence of additive Gaussian white noise. The resulting error rates of the function are plotted against the theoretical error rates of the BPSK (see appendix V).

After the BPSK function was verified, the function was used to build a simple direct sequence spread spectrum system in which the effects of channel loading could be seen.

To build a spread spectrum system, many of the components had to be specially written that were not available in MATLAB (see appendix II). One of the initial pieces of the puzzle that had to be defined was a function to generate pseudo random m-sequences. The function was written to accept the polynomial definitions. This in turn generated a PN-sequence. Another function was written to ensure that the sequence generated was indeed an m-sequence of a specific bit length. The input to the m-sequence generator was the number of shift register stages. The number of m-sequence polynomial definitions for a given number of shift register stages was then compared and verified against tables found in [31].

With the ability to create PN m-sequences in hand, the next logical step was to create a function that could generate Gold Codes. As mentioned in section 2.4.1.1 Gold Codes are created from PN m-sequences, and based on the Gold code function, a function to generate orthogonal Gold Codes was also then created. One more significant component was needed for the DSSS test setup: The Walsh code generator. This is used to provide orthogonal user discrimination. Walsh Codes were generated using HARDAMARD matrices. At this point, all the major components required to create a simple test DSSS setup had been attained.

Before the sequence generating functions were used, their validity was first established. Each of the functions were tested by putting inputs that would give known outputs. For the Walsh code generator, the index was given, as well as the code length. The output from the generator was compared with Walsh Code tables in [12]. Even after the results were matched, they were still tested for being orthogonal, i.e. the result of cross-correlation would result to zero when the codes were perfectly aligned. Likewise polynomial definitions were used as the input to the m-sequence generator, and the output compared against tables. The Gold codes were verified automatically once the m-sequences were verified.

Gold codes were created based on the m-sequences that had the lowest cross-correlation peaks. In the case of seven-stage register shift register, 18 m-sequence definitions exist. It was found that m-sequences that were generated with $x^7+x^3+x^2+x^1+1$ and $x^7+x^5+x^4+x^3+x^2+x^1+1$ had the lowest cross-correlation maxima, of magnitude 17, and the lowest cross-correlation minima, of magnitude 23 (see Figure 5.1).

Figure 5.1: Cross-correlation result of two m-sequences

These two m-sequences were used to generate 127 orthogonal Gold codes. Based on the properties of Gold codes, presented in section 2.4.1.1 it is expected that the cross-correlation properties of the orthogonal Gold codes would be inherited from the m-sequences chosen.

Figure 5.2: Cross-correlation result of two orthogonal Gold code sequences

Figure 5.2 shows the cross-correlation of two orthogonal Gold codes (out of the set of 127 codes). It can be seen that there are no maxima that surpass the peak maxima found in Figure 5.1. Figure 5.3 shows the auto-correlation of one of the orthogonal Gold codes, from the set of 127 orthogonal Gold codes, which was selected to be used as one of the three CDD codes.

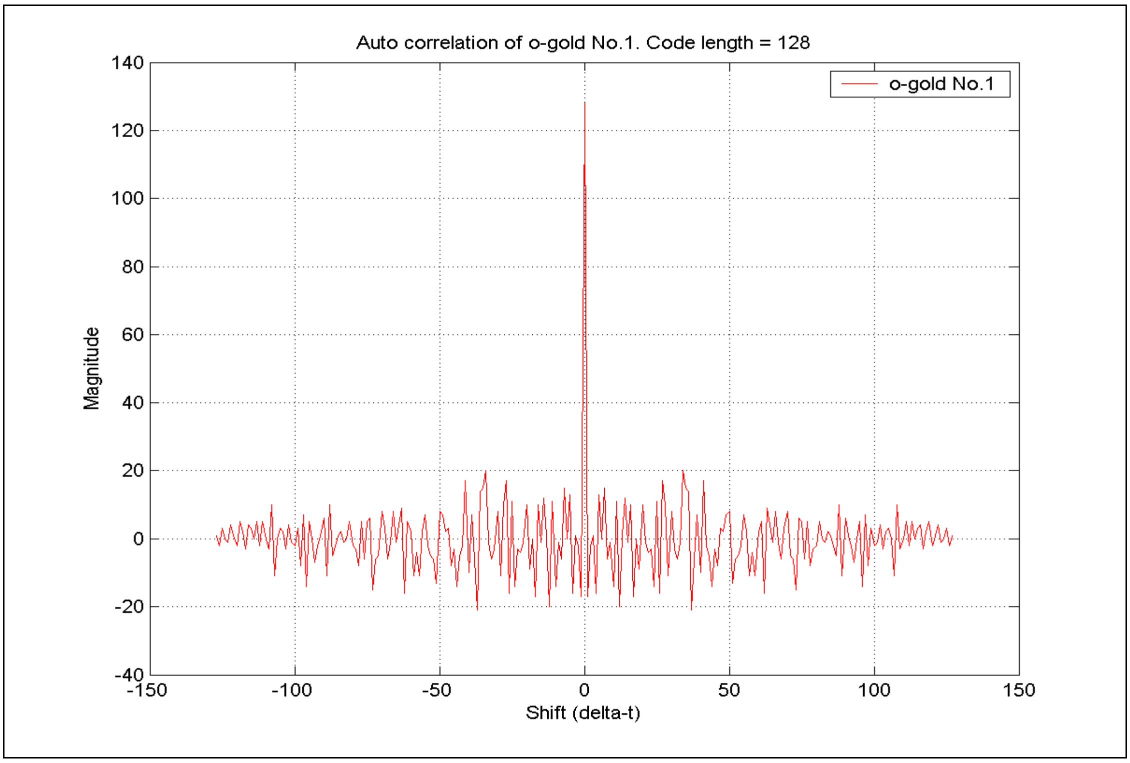

Figure 5.3: Auto-correlation result of an orthogonal Gold code sequence

Before the CDD simulation was done, a DSSS channel loading test was setup with a spreading factor of 64 and signal-to-noise ratio of 4 dB in order to verify at the setup would respond in an expected manner as the channel was loaded using a simple correlation receiver. The results from the DSSS channel loading test setup matched the expected results, i.e. the error rates of the system rose as more users were loaded unto the system. These results closely resembled output from other experiments found in [27] which indicated that the system could now be used to test the concept of CDD and its performance.

5.2 CDD Simulation: Simulation Environment 1

The first test setup was arranged to represent the transmitter structure and the receiver structure that is described in section 4.1. Data was randomly generated and spread using the Walsh Code that has been assigned to that instance of the transmitter. This data stream was then multiplied with the orthogonal Gold code that was used to represent a duplexing direction. The result is then modulated and transmitted. The transmitted signal is then exposed to additive white Gaussian noise. The received signal is then demodulated de-spread to retrieve the user information. In essence, the setup is very similar to the DSSS test environment, but the major difference lies in the fact that users will share codes for their respective duplexing direction.

The aim of the test is to compare the performance in the presence of noise (additive white Gaussian noise) in terms of bit-error rates of six transmission links (six transmitting users and six receiving users) using simple direct sequence spread spectrum and six transmission links using code division duplexing. These six links represented a channel loading of 19%, 9.4%, 4.7% for a spreading factor of 32, 64, and 128 respectively. Of the six transmission link using CDD codes, two links had its traffic defined for the MS-MS direction; two links had its traffic defined for the MS-BS direction and in a similar fashion two links its traffic defined for the BS-MS direction, as shown in Figure 5.4.

The independent variable in this arrangement is the signal-to-noise ratio of the radio channel per user. The simulation was run in an environment from weak signal-to-noise ratios (-15 dB) to strong ratios (15 dB) as to see the effect of MAI on the performance of the system.

The assumptions that were made for this test environment are as follows:

1. The system is perfectly synchronised. All the transmissions are received on time and are demodulated. This also suggests that the transmissions from the other users (who are inherently interferers) are perfectly aligned with the intended user's transmissions.

2. All transmitting nodes are transmitting simultaneously whilst all receiving nodes are receiving simultaneously. The receiving nodes all receiving the same amalgamation of signal powers. This leads logically into the next assumption.

3. Perfect power control was administered such that the power levels received at all the receivers are equal. Each node transmits 1 volt peak for a '1' and -1 volt low for a '0' using baseband BPSK. At the receiver end the signal has only been modified to accommodate the specified signal-to-noise ratio.

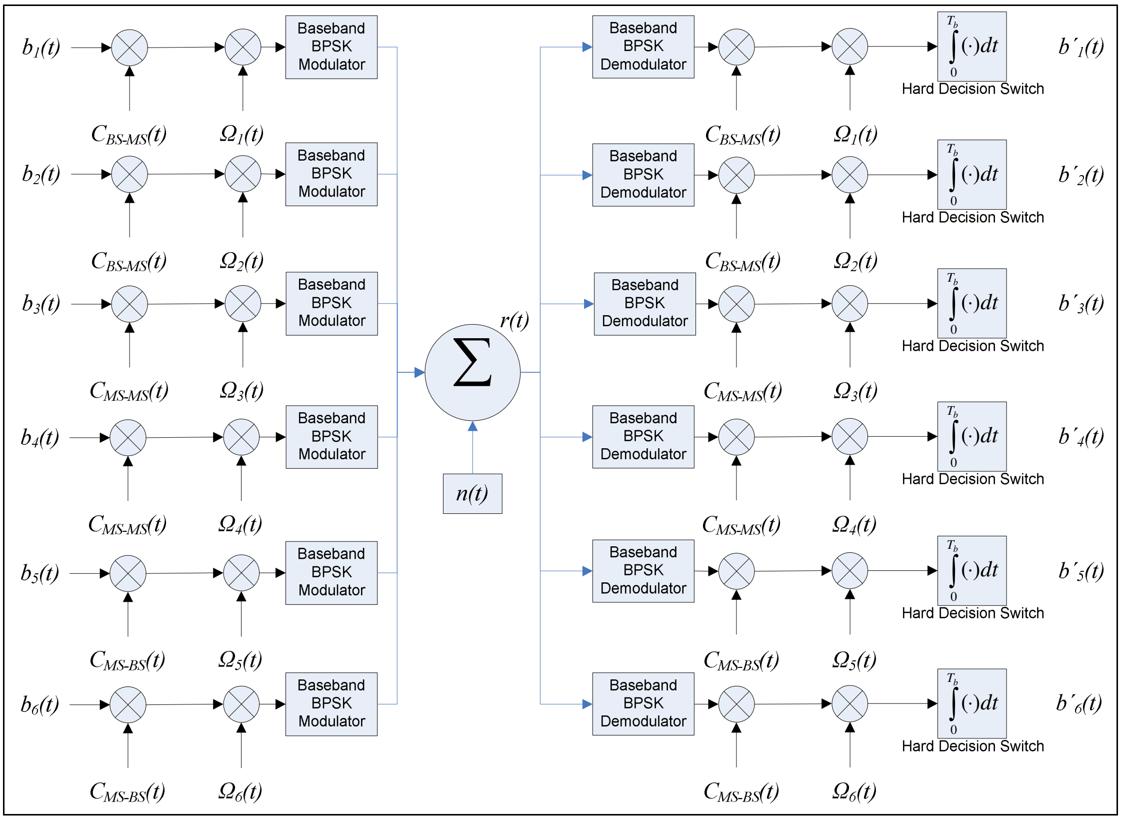

Figure 5.4: Representation of Simulation Environment 1

5.3 Simulation Environment 2

The second setup is similar to the first in terms of the tools used. In this case however, the signal-to-noise ratio is left constant and the number of simultaneous user load is increased. In this simulation, a spreading factor of 64 was used to spread and send 1000 bits. The additive noise from the channel induced a SNR of 4 db. In a similar fashion as the test environment 1, in test environment 2, path loss attenuation is ignored. All the waveforms from the transmitters reach the reference receiver at the same signal strength.

5.4 CDD Performance

Unlike the conventional DSSS system where each individual user has their code, in CDD CDMA, each user will have their specific code as well as a code that is shared. This shared code is the duplexing direction code. For instance, if there are multiple peer-to-peer sessions going on simultaneously then there will be MS units that will be using the same CDD code.

In Figure 5.5, small spreading factor of 32 chips was used for both DSSS and CDD. For small spreading factors, error rates are high because the data signal doesn't have the processing gain to protect it from errors. Therefore, at relatively low signal-to-noise ratios, namely between -15 and -5 dB, the error rates of the two systems are almost identical. However, as the signal-to-noise ratio improves the effect of the multiple access interference becomes apparent. There is a clearly defined performance floor at 5 dB. This is the point the MAI is the significant contributing factor of noise.

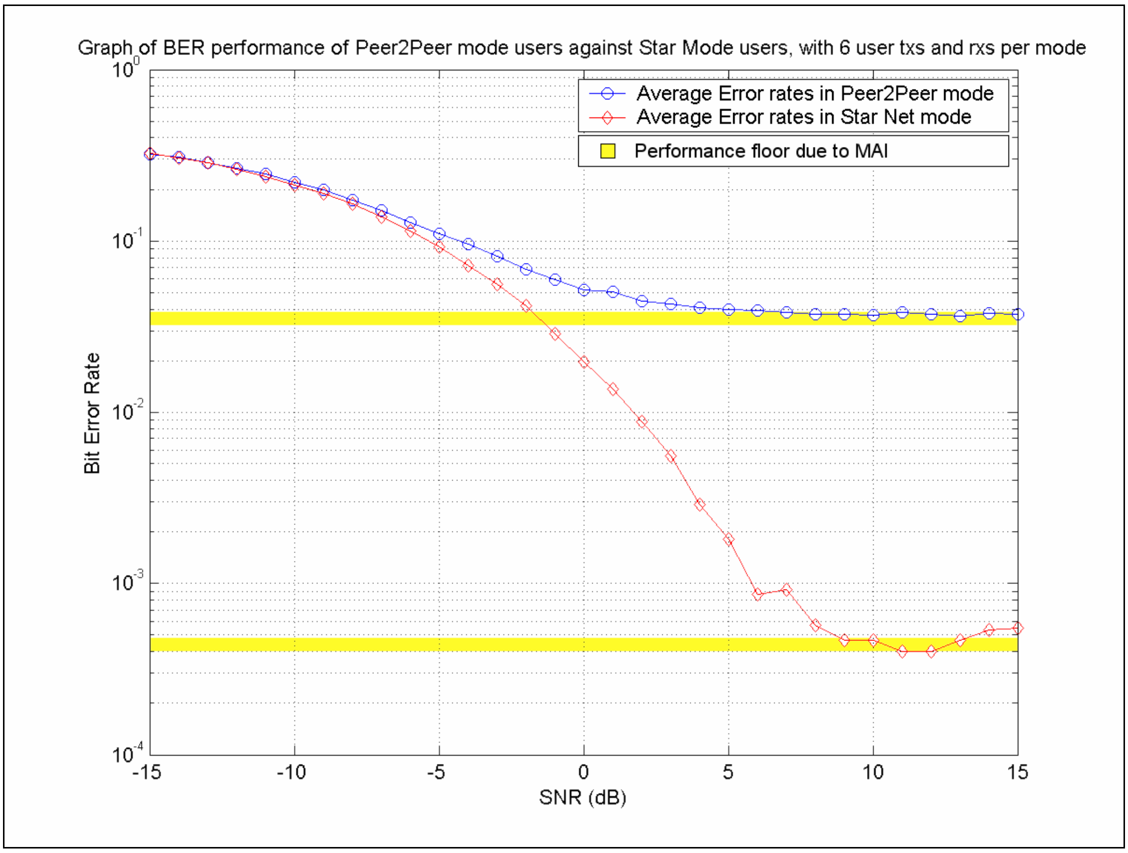

Figure 5.5: BER performance of 6 CDD users against 6 standard users with code length 32

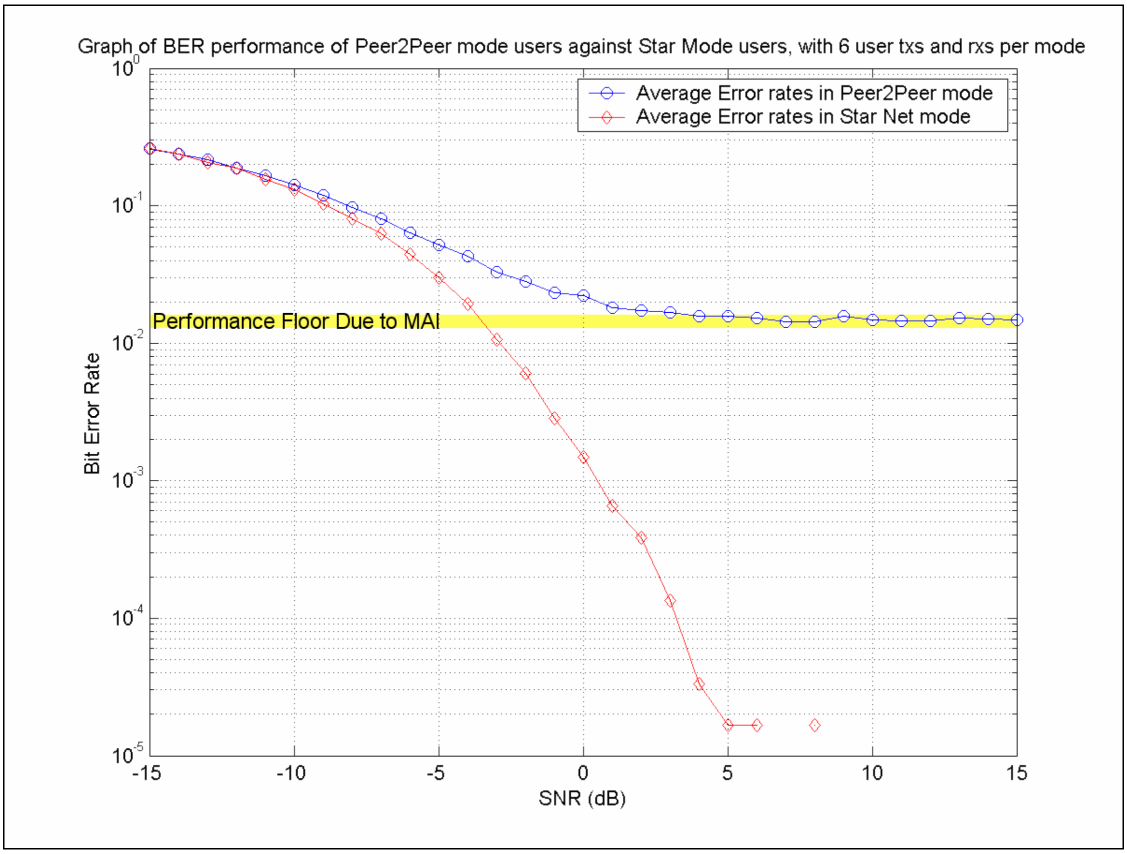

Figure 5.6: BER performance of 6 CDD users against 6 standard users with code length 64

The trend continues in the tests in which all the parameters remained constant except for the processing gain. The larger processing gains improved both systems performance. However, the improvements were not as significant as in the CDD system, when compared with the performance of the system at code length 64 (Figure 5.6). However, when the code length of 128 is used the difference in the two systems performance becomes clear, as shown in Figure 5.7.

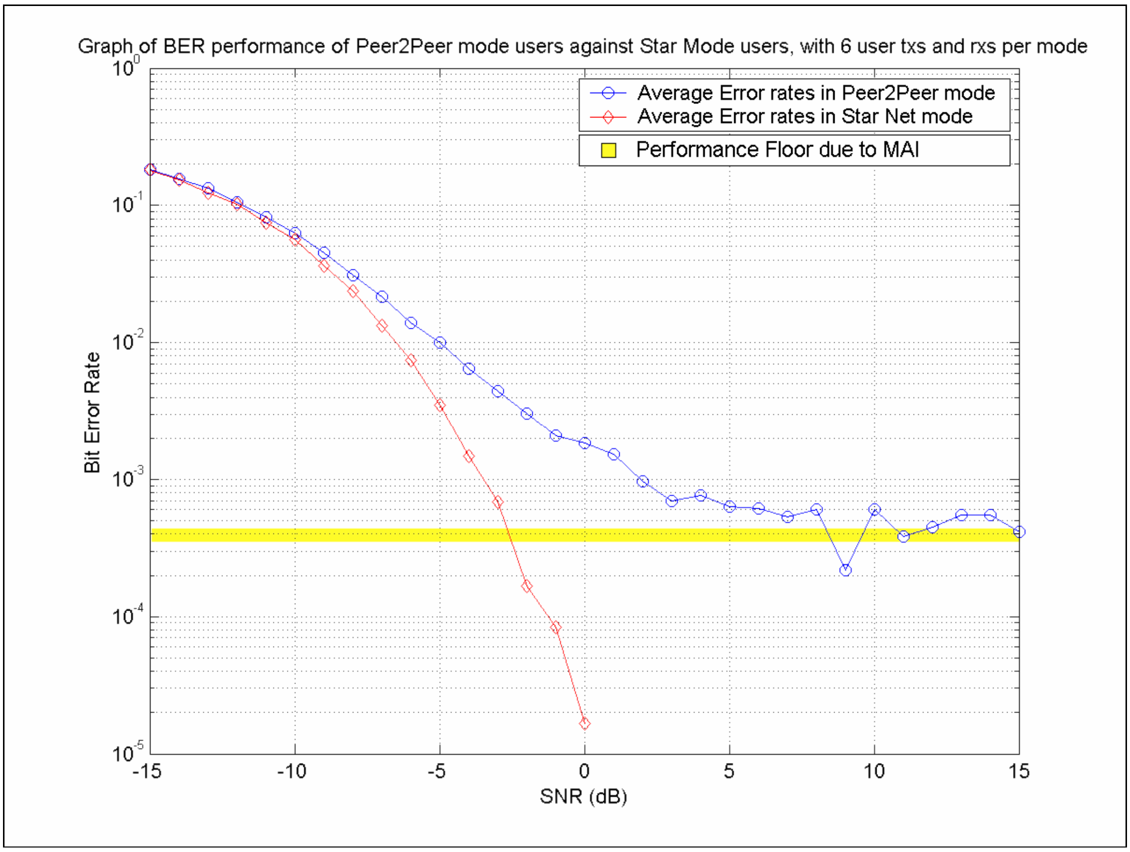

Figure 5.7: BER performance of 6 CDD users against 6 standard users with code length 128

Figure 5.7 shows that in the presence of very little noise the DSSS systems is more efficient, where as the system performance of the CDD mode users experiences a performance floor, which is due to MAI. As the signal-to-noise ratio increases the systems performance is restricted from improving.

In section 5.3, the channel loading experiment was described where the channel was continuously loaded. Referring to Figure 5.8, the bit-error rate of the CDD system is greater in comparison to the DSSS system. As explained in section 5.4, the higher error rates are due to the fact that codes are being shared. Code sharing is avoided in real systems such as UMTS. In the case of UMTS, the Walsh code heritage for multi-rate spreading becomes a significant part of the design. As mentioned, in section 2.4.2.1 codes that have the same parentage or heritage are not used in the same cell. This is because those codes have the same bit patterns and will cause an increase in multiple access interference.

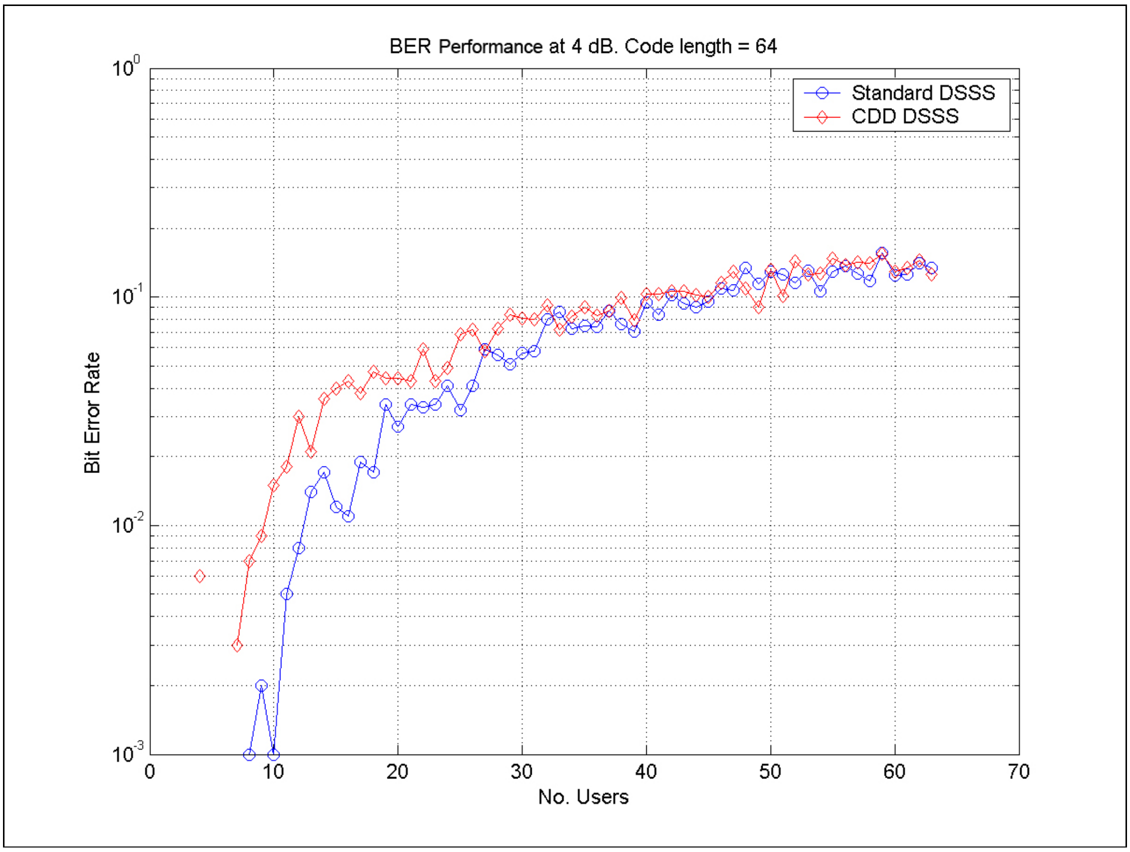

Figure 5.8: Channel Loading Performance of CDD vs. DSSS

Figure 5.8 shows the CDD system performing more poorly than the standard DSSS system. When both systems are loaded with 10 (6.4% channel loading) users whom are transmitting continuously and simultaneously the error rate in the DSSS systems is 1 in 1000 in comparison with an error rate of just under 1 in 100 for the CDD system.

5.5 Multiple Access Interference Reduction for CDD

At this point in the investigation it has been revealed that CDD has an inherent weakness, that is, it is extremely prone to multiple access interference, as shown in section 5.4. The key source behind the high MAI is the code sharing. In order to combat this situation two schemes are suggested: Network Managed Code Assignment and Random Index Code Shifting.

5.5.1 Network Managed Code Assignment

Due to the limitation of sharing codes for CDD, code assignment is one of the techniques to reduce multiple access interference in the system. Instead of merely assigning one code to a duplexing direction, a set of codes can be assigned. This way the number of codes that are shared reduces. Code assignment is facilitated by means of being centrally administered from the network. Codes can be assigned during the hand shaking procedure. Seeing that the hand shaking procedure depends on communication with the base station, codes can be allocated, thus avoiding reuse. However, as the system becomes loaded, the probability of reuse of a CDD code in a code set becomes higher.

Table 5.1: Code Assignment in CDD

Direction	Code Designation	Code Length = 32	Code Length = 64
Mobile Station to Base Station	Code Set 1	11 codes	22 codes
Base Station to Mobile Station	Code Set 2	10 codes	21 codes
Mobile Station to Mobile Station	Code Set 3	10 codes	20 codes

Due to the reason that the codes are managed by the network, there is a higher probability that the performance of the system will be the same as the bit-error performance of direct sequence spread spectrum. Unlike Random Index Code Shifting, described in section 5.5.2, the probability of code sharing only increases when the load on the network surpasses a specified value. Table 5.1 gives an example of how many codes are assigned to a duplexing direction for code length 32 and 64. As the code length increase, so does the number of unique orthogonal Gold codes that can be assigned to the 3 specified duplexing directions.

5.5.2 Random Index Code Shifting

Like network managed code assignment, random index code shifting reduces the amount of code sharing by executing a circular shift on the bit sequence that make up the code that has been assigned to a duplex direction. The index refers to the number of times the code can actually be shifted, therefore if the index is 10 then there are 10 positions that the code can be shifted into.

One of the advantages of this technique is that it is easy to implement in hardware. All modern processors have a basic bit shift command as a part of their rudimentary routines. The operation is a circular bit shift done on a register that contains the CDD code. The random selection of the index is the only provision against medium access collision (which would be code sharing) assuming that all MS units are transmitting simultaneously. The downside to this is that there is the probability of an index clash with another mobile in the network in the vicinity. Nonetheless, random index code shifting does not require the network for administration. However, it does require negotiation with the connecting party so that the party that is being connected to is informed of the index that will be used for the duration of the transmission – where the connecting party could either be a BS or another MS.

The shift distance must be large enough for shifted codes not to align themselves with other multi-path components from other transmission sources. To put this into perspective, if there is a code shift of one or two bits, then it is likely that code sharing would still occur because of an alignment with a multi-path component of another transmission. Then, code sharing would cause an increase in MAI. In a similar manner, the shift is meant to be large enough for a RAKE receiver of an unintended user to reject. A RAKE receiver usually has between three and five fingers (or branches), which means that it is capable of combining uncorrelated multi-path components which are delayed by between three to five chip periods in time. Therefore, for the average RAKE receiver, a code shifted by at least eight chips will not be combined into the signal.

5.5.2.1 Variable Indexing

In Variable Indexing, the number of available indices is worked out by dividing the code length by eight. So if there is a code length of 128 chips then the number available indices would be 16. Therefore, as the code length increases the number of indices increases and the shift distance between each index is eight bits (chips). In the example when there are 16 indices, one index between 1 and 16 is chosen at random. When an index is chosen the duplexing code is shifted by the value of the chosen index multiplied by 8. So, for instance, if five is the value chosen then the code will be shifted by 40 bits (chips). As mentioned in section 2.4.2.2, the code length of an orthogonal Gold code is 2^m one of the reasons for using eight as a factor is the fact that it is easily divisible into the code length of orthogonal Gold codes greater than the length of eight chips. Therefore, as the code length increases the probability of an index collision decreases.

However one of the disadvantages is that the shift codes may not remain orthogonal to other orthogonal Gold codes in which it was orthogonal to before. For instance, code 1 and code 2 are orthogonal, and then code 1 is shifted by an index of 5 and remains orthogonal. However, the same code 1 is then shifted by an index of 6 and is no longer orthogonal to code 2.

From a brute force analysis, in which orthogonal Gold codes were generated, shifted and tested for to see if shifted codes remained orthogonal; it was found that for code length 2^m where $m \geq 5$ all the codes remained orthogonal.

5.5.2.2 Constant Indexing

In Constant Indexing, there are only eight indices available for a code to be shifted into. Therefore, as the code length increases the shift distance increases. For instance, for a code length of 64 chips, each index will result in a code shift of 8 bits, but for code length of 128 bits, each index will result in a code shift of 16 bits.

One of the disadvantages of Constant Indexing is that for small processing gains like 16 and 32 chips, the codes shifts are small (a distance of 2 bits and 4 bits respectively). It is a disadvantage in the sense that one of the aims of choosing a shift distance is such that it is sufficiently larger than the largest time shift finger of a RAKE receiver. However, where a RAKE receiver as well as multi-path is not an issue, then at the processing gains mentioned Constant Indexing still offers eight slots whereas variable offers two slots and four slots respectively.

Another advantage of Constant Indexing is that the codes remain orthogonal to other orthogonal codes. This was demonstrated using a similar brute force test as in the Variable Indexing case.

5.6 Summary

In this chapter, the performance of CDD was assessed. The simulation based on point-to-point links was implemented to compare the bit-error performance of CDD system versus the DSSS system. From the results that were ascertained, CDD does not perform as well as a standard DSSS system in reasonable signal to additive white Gaussian noise ratios. This implies that the multiple access interference thwarts the systems bit-error rate performance in comparison to the DSSS system. The increased multiple access interference is due to the sharing of codes and, as a consequence, two schemes are recommended to reduce the multiple access interference: Network Managed Code Assignment and Random Index Code Shifting. The preferred scheme is Network Managed Code Assignment because there is no code sharing involved and theoretically performs as a standard DSSS system does.

In the next chapter, it is assumed that Network Managed Code Assignment is used in the simulation for the coverage and capacity analysis. Based on this assumption, the bit-error rate performance evaluation is based on the bit-error rate estimation of a DSSS system.

6 COVERAGE AND CAPACITY SIMULATION

In order to compare and contrast the design of the hybrid topology mobile network versus the standard infrastructure network, a simulated environment was developed to assess the performance of the two network topologies. In a similar vein to the simulation presented in [16], a snapshot based simulation is designed and implemented.

6.1 Simulation Design

The simulation represents the ad-hoc network overlay on a fixed cellular CDD/TDD CDMA system. Of the four transmission strategies (described in section 4.5.1) only a one hop single slot[1] and a one hop multi slot strategy was employed, where an MS unit was allowed a maximum of two peer-to-peer connections. In the simulation the network can be loaded by two basic means: via adjusting each node's voice activity and adding network nodes to the simulation area. However, the voice activity was treated as a constant and thus was not varied. In this simulation, higher layer design elements are taken into consideration. The billing oriented call setup is encapsulated in the three routing rules that are described later. As a result, no node is permitted to make an MS to MS link without one of them being in a hops distance away from the base station.

The channel model consists of a single path loss model (described in section 6.1.2) representing an urban radio environment for a single cell. A spreading factor of 256 was used (which results in a processing gain of 24.082 dB). The simulation environment used was a square shape of 6 km^2 area. The simulation area was loaded with 200 MS nodes. The simulation is conducted to ascertain the networks response in terms of signal-to-interference ratio and bit-error rates for both the base station and the nodes on average. In addition, the results about the overall coverage increase are also to be provided from this simulation.

[1] A slot is the ability of an MS to support a single peer-to-peer connection (as defined earlier in section 4.3.1).

6.1.1 Node Objects

A typical cell model consists of two types of nodes, BS node and a number of MS nodes. Both nodes types have a number of common attributes, which can be seen in Figure 6.1. In other words an MS and a BS node are subclasses of a basic transmission node.

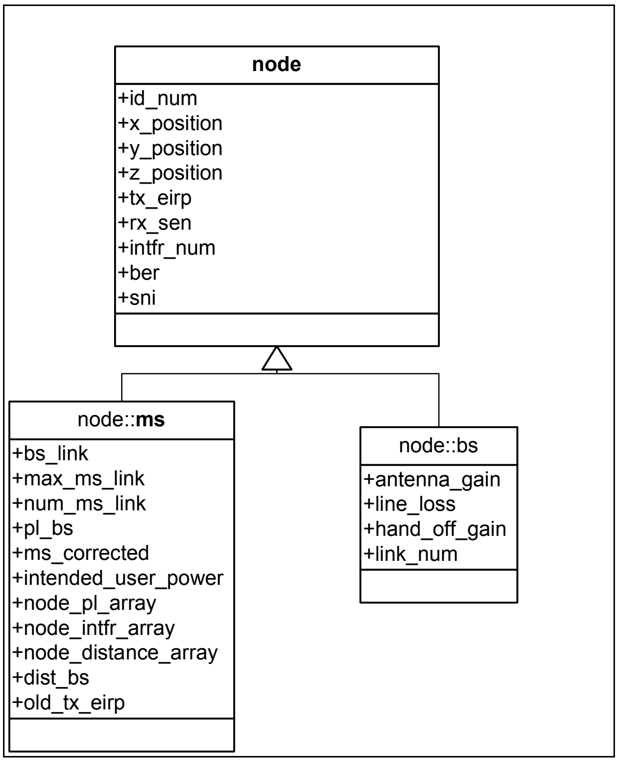

Figure 6.1: Node Class Diagram

6.1.1.1 Node Object Characteristics

As shown in Figure 6.1, the nodes class contains nine attributes that add to the uniqueness of each node. The “id_num” property stores each nodes unique identifier. When an MS node is first initialised, the “id_num” property is randomly selected from values between 1 and 100,000. On the other hand when a BS node is first instantiated the “id_num” property is set to 0. The primary reason for this attribute is to uniquely address all the nodes that have been instantiated in the simulation area. Another reason this is done is to enable nodes to identify an MS or a BS by looking at their “id_num”, in that if the “id_num” is equal to zero then node is a BS, otherwise, the node is an MS.

However, this can also be done by utilising "isa" function[1] to identify the class type of the node. Also contained in the node class are the three coordinate properties that store the longitudinal position, latitudinal position and height above ground with the "x_position", "y_position", and "z_position" respectively. From these coordinates, the distance from one node to any other node is assessed. The values that are most significant to the simulation are the Transmitter EIRP, the Receiver Sensitivity and the Maximum (allowable) Link Path Loss. These values are the combination of other gains and loss that are predefined. The maximum transmission EIRP is stored in "tx_eirp". By default this property is set to 20 dBm for MS units and set to 24 dBm for a BS node. In Table 4.1, item d, the forward link transmitter EIRP is 41.15 dBm. The reason for this difference is the transmitter antenna gain is considered separately from the transmitter EIRP in the BS node object (24 dBm + 17.15 dB). This property is then manipulated for power control. The receiver sensitivity property is represented by the "rx_sen" attribute. This attribute is instrumental in determining whether a communications links has satisfied the maximum allowable path loss. The "intfr_num" attribute is a counter to assist debugging. MS nodes use it to keep track of the number of neighbour MS units within the allowable path loss range. The "ber" and the "sni" are the primary performance indicators. The two indicators are calculated based on the radio environment and the amount of interference being received.

Both the MS and the BS inherit the abovementioned attributes. However both the MS and the BS contain their own attributes that gives then the ability to perform the tasks that are unique to them.

6.1.1.2 MS Object Characteristics

The "bs_link" property of the MS node stores the Boolean value for whether the node has a communications link or not. This value is determined by using the path loss to the node. If the path loss to the node is above the maximum allowable path loss, then the node is deemed outside the coverage area. The "max_ms_link" property contains the value of the maximum number of MS nodes that can connect to the MS node via peer-to-peer links. The "num_ms_link" property contains the value of the number of peer-to-peer links that the

[1] The "isa" function is an inbuilt function found in MATLAB that is used to for object identification.

mobile is currently engaged in. This property is administered by the mobile each time a peer successfully connects, however the MS node will not allow another peer to connect once the "max_ms_link" property and the "num_ms_link" property are equal. The "pl_bs" attribute contains the path loss from the node to the base station. The value is used for assessing the received power from the base station, determining if nodes are in the coverage area and used as a routing decision variable. The "ms_corrected" attribute stores the unique address of the MS node that is currently correcting the power of the MS node whose "ms_corrected" property is being manipulated. If the value of the property is 0, it suggests that the MS node power control commands are from the base station. The "intended_user_power" property stores the receive power of the user in which communication is intended with. Therefore, this value is used to calculate the signal-to-interference ratio relative to that user. The "node_pl_array" is a container of path loss values from the owner to all other nodes in the simulation environment, where a node's id is the index to access the path loss relative to the owner MS node. This property is used similarly to the "pl_bs" property. The "node_intfr_array" is an array containing all the receive power of all the other MS nodes. This array stores milliwatt values and is used in the calculation of signal to inference ratios in conjunction with the "intended_user_power" property. The "node_distance_array" stores the values of distance of all other MS nodes to the owner MS node. Both the "dist_bs" and the "old_tx_eirp" property are used for debugging purposes. The "dist_bs" is used to store the distance the MS is away from the base station; and the "old_tx_eirp" stores the value of the old values of the transmission EIRP and under which routing rule it was changed.

6.1.1.3 BS Object Characteristics

The BS node object contains fewer attributes than the MS node, primarily due to the solitary nature of its role. Although the simulation can be extended to a multiple base station case, only one base station is instantiated in the simulation environment. As a consequence, attributes that can be associated with the base station have been left to be calculated on demand by assessing the status of the MS nodes and the BS.

The "antenna_gain" attribute is used to calculate the maximum allowable path loss and to work out the real transmission EIRP of the base station. It was separated such that it could be used from both the forward link and the reverse link perspective without introducing ambiguity. Both the “line_loss” and the “hand_off_gain” are dB values that are used in assessing the maximum allowable path loss. Their default values are set to the values found in items f and n respectively in Table 4.1. The “link_num” attribute stores the number of MS nodes that have a communications link with the BS node.

6.1.2 Path Loss Model

The path loss model is one of the parameters of the simulation that can be varied but remains fixed. As a consequence, the results from the simulation are based one path loss model. The aim of the simulation is a comparative analysis a topology based on CDD versus the standard star network topology. Thus, the performance of the system in other path loss models can be extrapolated with the results of the simulation. The following model was used in the simulation:

WITS Model (Hata + Free Space): The urban version of the Hata model was used to define the path loss from the base station to the nodes. However, seeing that the model has a limitation whereby the estimation becomes invalid for distances less than one kilometre (as mentioned in section 2.3) the model must be used in conjunction with another model. As a consequence, the model used is the amalgamation of the free space propagation and the Hata propagation model. For distance less than one kilometre the free space model with air and building absorption consideration is used to adjust the free space model. For distances greater than one kilometre the Hata model for urban propagation loss is used; and for distances greater than four kilometres the Hata model for suburban propagation loss is used.

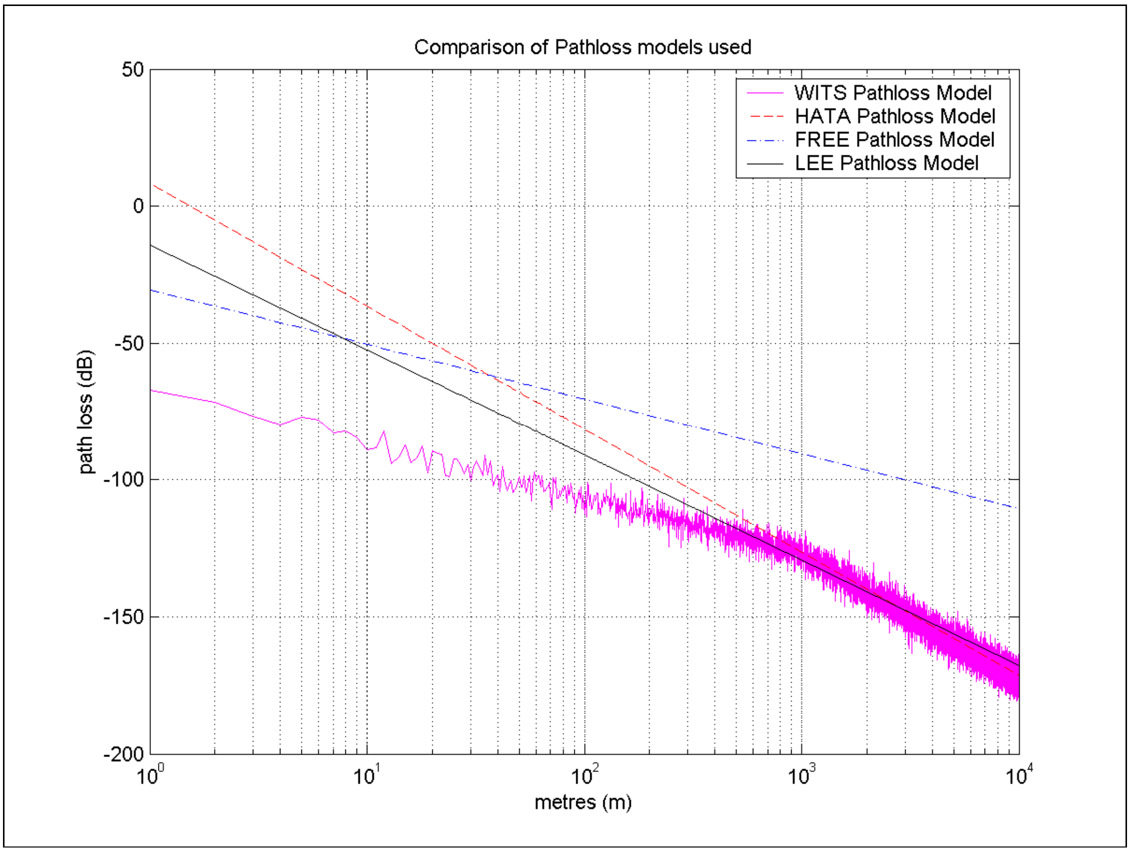

Figure 6.2: Path Loss Model Comparison

Figure 6.2 shows a comparison of the WITS model against other path loss models. The fluctuation in path loss is due to the implementation of statistical log normal shadowing where σ dB = 10 dB which approximates to less than 90% coverage reliability. It can be seen in the figure that the path loss exponent changes from 2 to approximately 4 at 1000 metres; this change is apparent in the change in the gradient of the WITS path loss plot.

6.2 Assumptions

In designing the simulator, there are some assumptions that are made to facilitate the acquisition of specific data elements related to the performance of the CDD hybrid architecture in comparison to the standard star infrastructure. The assumptions made are presented below.

6.2.1 Synchronized system

For the model that involves assessing the bit-error rate of the hybrid system over the standard system, the simulator presents a perfectly synchronous environment. In other words, it assumes that every frame sent and received is received with the correct timing synchronisation established at the receiver. It also suggests that the transmission of an interferer is perfectly in line with the desired user. It is also assumed that the network is perfectly allocating CDD codes to nodes, in order for it to behave like a DSSS system in terms of bit-error performance.

6.2.2 Perfect power control

All the nodes that are within the signal range of the base station will have there EIRP corrected perfectly such that the received signal to the base station from each node is the same. Power control in an actual system is not hundred percent accurate. This is due to the rapidly changing conditions in the radio environment. In the simulation, it is assumed that once the power of the mobile has been controlled, the radio environment remains static; and only distributed autonomous power control will come into effect afterwards; where the concept of a distributed autonomous power was presented in [7].

It is also assumed that for distributed power control, the node initiating a connection will allow its power to be controlled by the target node. The target node will adjust the EIRP of the initiating node based on the path loss between the two nodes, and thus the closer the node is to the target node, the lower the EIRP will be set.

6.2.3 Absence of Fast Fading

As explained in section 2.3.3, fast fading is due to multi-path components causing interference. In the simulation there is only consideration for path loss and log-normal shadowing. As a consequence, MS units close to base station are not prone to any form of loss of reception.

6.2.4 Zero Velocity

It is assumed that the nodes are stationary during an iteration of the simulation. A consequence of this assumption is that nodes do not experience loss due to the Doppler spread. Not accounting for Doppler spread ignores the fact that channel conditions change rapidly with an increase in velocity and it also ignores the frequency shift that a radio signal undergoes when the target mobile is moving.

6.2.5 TDD Factor

The TDD factor is the product of the voice activity and the mean TDD interference activity. The TDD interference probability can be defined as the probability that n of K-1 interferers will be transmitting (and hence adding to the interference) when the kth node is in the receive mode. It is assumed that the probability of being in receive mode is 50%[1] and therefore the probability of being in transmit mode is also 50%. Seeing that each MS node is independent then we can assume that in a point in time that 50% of the interfering MS nodes are transmitting and 50% are receiving. Therefore, only 50% of the nodes are indeed interferers to the kth node. Considering the probability of the kth MS node is also 50% for receiving and transmitting then the average number of interferers relative to the kth node is 25% of K-1. The value for voice activity normally found in literature is 0.375. Thus the resulting TDD factor that is used in the simulation is 0.375 x 0.25 = 0.09375.

6.2.6 Node Distribution

Each node will be randomly distributed in a confined area, with one exemption namely the BS which has a predefined position. The confined area is sufficiently large for the BS not to be able to service all the MS units in the area. The nodes are distributed such that each node has equal probability of being placed within the designated area. As a consequence, the node density is the same throughout the simulation area.

There are other distributions that can be considered. The Normal distribution in an area can be used, where the mean and the variance are set to allow for a higher probability of a node being placed close to the base station and a lower probability of a node being place away from the base station. This would result in a distribution where the user density is decreasing as the node moves farther away from the base station.

[1] This value stems from another assumption in which data rates for uplink and downlinks are symmetrical.

6.2.7 Inter-cell Call Probability

In a cell, there is a chance that an MS node will request to make a call in which the call terminates at another node within the same cell. This probability is termed as the inter-cell call probability. This call probability value is set to 50%. However, the probability of a call being setup on a peer-to-peer basis is smaller. This is due to fact that, there are other constraints that will determine if it is better for the base station to route the call or for the MS units to setup a peer-to-peer connection.

6.2.8 Routing Decision Factors

Routing decisions can be made on the basis of many criteria. For example, if there is a node in need of using one of its neighbours as a relay it may decide to use the neighbour that is nearest to it in distance, nearest to the base station in distance, has the best average BER, has the highest SNR, the least path loss to the base station, the least overall path loss to the base station via a set amount of hops and so on. Routing decision factors along with the routing algorithms is a large topic and is not fully discussed here. Instead, a simple nearest-neighbour selection scheme is used in which the neighbour with the lowest path loss is used as a relay. This means the routing table is based on the path loss between nodes.

6.3 Procedure and Tools

The aim of the setup is to compare and contrast the networking topologies, one being the standard cellular infrastructure setup (star network topology) and the other being the hybrid setup that consists of the ad-hoc overlay on the standard cellular infrastructure. In the standard topology, MS nodes solely communicate with the base station within a cell; whereas in the hybrid topology nodes may communicate with each other.

The resulting aim is to then analyse the key performance parameters of the overall system, which are the capacity from the two topologies in terms of user loads; capacity in out-cell regions; and coverage improvements.

6.3.1 Model Instantiation

For the simulation environment, there are three attributes that are initially defined: the size of the simulation area, the path loss models defined for the area and the number of MS nodes within the area. The base station node was instantiated at x = 3000, y = 3000 and z = 30[1], which was at the middle position of the constrained area. As mentioned earlier, the constrained area is a square of size 6000 metres by 6000 metres. Mobile stations are then instantiated randomly in the constrained area. Due to the random placement of MS nodes, as the number of nodes increases, the nodes get evenly distributed around the BS (see Figure 6.3). In addition to the MS nodes' x-y coordinates, the nodes z-position is also randomised. Each node is given a random height between one and three metres in elevation. The base station, on the other hand, is given a standard height of 30 metres. Each time the environment is initialised, it is unlikely that the nodes will have the same properties in terms of their positions. Once a set of parameters are defined, the environment is instantiated and run for 100 iterations and the results from an instance is recorded so that the average result can be attained.

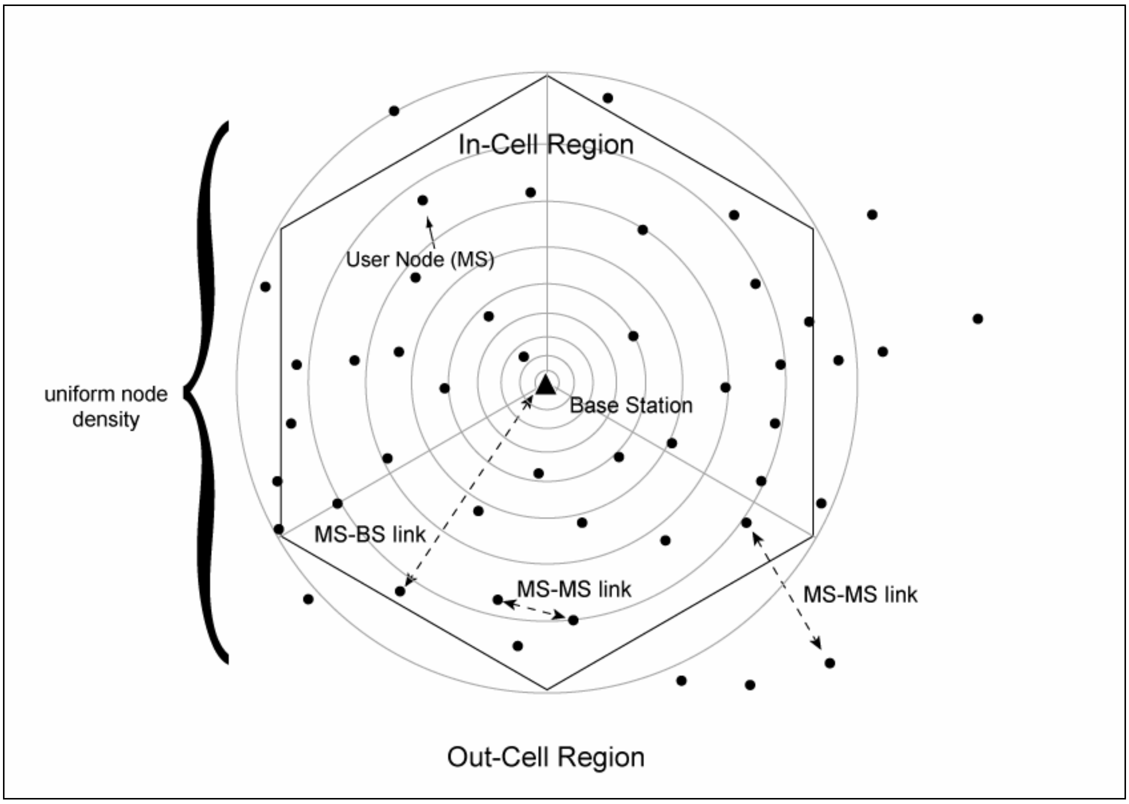

Figure 6.3: Illustration of Simulation Area

[1] 30 metres is the minimum valid height for a base station in the Hata model.

Once the positions of the nodes are defined, the path loss model takes effect. The path loss model used for the simulation remains unchanged for the duration of the simulation. The path loss from the base station to the MS node is dependent on the distance between the two objects and the model that is being used. The path loss also depends on the frequency of the radio transmission as well; however, seeing that it has been predefined to 800 MHz, the frequency is treated as constant. For the simulation, all the models that are employed also employ log-normal shadowing. As a consequence, two nodes that are exactly the same distance away from the base station will have different path losses because of the shadowing statistics. This factor further increases the likelihood of simulation iterations being absolutely unique.

Once the path loss for the nodes has been determined, the BS then searches for the MS that are within its signal range. It achieves this by calculating the maximum allowable path loss from the base station to the node. If the path loss of the node is greater than the maximum allowable path loss then the node is deemed to be outside of the range of the BS or the node is in a radio fallout shadow. The BS then marks all the nodes that are deemed within range by setting the "bs_link" property of the node to true (a value of one). The "bs_link" field facilitates the out-of-cell nodes to find MS units that are candidates for being utilised as a repeater station. The BS then keeps a store of the number MS units that are in range of the BS so that in can then be used later for coverage analysis.

At this point all the nodes are transmitting at their maximum transmission power level. As mentioned in section 2.3.5, this is a large source of interference in CDMA networks. As a consequence, the base station administers simple open loop power control by correcting the transmission EIRP of each node.

There are other parameters that are set before the commencement of the simulation. One of the first parameters that is set is the processing gain. This parameter is used in the estimation of the bit-error rate. The processing gain also gives an indication of the number of users the system can support simultaneously. The next important parameter that is set is the signal to additive white Gaussian noise ratio. This parameter is also instrumental in the bit-error rate calculation for each node. The TDD factor and the Inter-Cell Call Probability are also set.

Another parameter that is set is the maximum number of slots (peer-to-peer connections) an MS node can accommodate.

Table 6.1: List of Controlled Parameters

Parameters
Processing gain Path Loss Model Maximum Number MS Links Signal to AWGN ratio TDD Factor Inter-Cell Call Probability Number of MS Nodes

6.3.2 Routing Rules

After the model is instantiated, it undergoes three phases of execution. The first phase of the execution is to assess the required parameters (BER and SNI) relative to the base station. In the next phase, the parameters are then assessed after simple power control is administered. In the final phase, BER and SNI are assessed once more when the peer-to-peer rules are administered.

There are three basic routing rules implemented:

Rule 1: Nodes that are deemed to be outside of the radio coverage area must attempt to find a node that is within the coverage area such that the node that is outside the coverage area can utilise the node in the coverage area as a relay. The purpose of rule 1 is to provide coverage to MS units that are outside of the coverage area or to provide coverage to MS nodes that find themselves in a critical radio shadow while being in the coverage area. This rule is based on ODMA but only addresses MS nodes that have no connectivity with the base station.

A peer-to-peer link can only be established by an originating MS node:

1. If the target node, in the vicinity, is currently serviced by the base station. This enforces the call setup protocol described in section 4.4.1.
2. If the target node suites the criteria for the allowable path loss between the MS nodes.
3. If the target node can accommodate another peer-to-peer link (provided it is already connected to other peers).
4. If the originating node is not connected to any other MS nodes.

In the situation where there are many candidate target node that satisfy the requirements then path loss from the target node to originating node is used to make the routing decision. The target node with the least path loss to the originating node is selected as the relaying node.

Once the rule is satisfied, the target MS node must ensure that it tracks the number MS nodes it is connected with and it now becomes responsible for controlling the power of the MS node that is outside the coverage area. This simple power control rule is used in all cases. The power control rule states that for a peer-to-peer connection, the target node is responsible for power control for the originating node.

Rule 2: This rule is based on the Inter-Cell call probability. When a MS node attempts to make an inter-cell call the following criteria must be fulfilled:

1. The path loss to the target node must be less than the path loss to the base station for the originating node.
2. The path loss of the target node must be less that the maximum allowable path loss between two MS nodes.
3. Like rule 1, the target node must be able to accommodate another peer-to-peer link (provided it is already connected to other peers).
4. The originating MS node must also be able to accommodate another peer-to-peer link.

Unlike rule 1, rule 2 is only satisfied if the path loss between the target node and the originating node is less than the path loss to the base station from the originating node. In the same manner as rule 1, the counters are adjusted and autonomous power control is executed on the originating node by the target node.

Rule 3: This rule is an Opportunity Driven Multiple Access based rule. The criteria for establishing a peer-to-peer connection is as follows:

1. Provided neighbourhood nodes meet the maximum allowable path loss between nodes, neighbours with path loss less than the path loss to the base station will be selected as the relay.

2. Like rule 1, the target node must be able to accommodate another peer-to-peer link (provided it is already connected to other peers).

3. The originating MS node must also be able to accommodate another peer-to-peer link.

In the same manner as rule 1, once the rule is satisfied the target node will be chosen on the basis of its path loss to the base station if more than one node meets the requirements. In addition, the counters are adjusted and autonomous power control is executed on the originating node by the target node. The combination of rule 1 and rule 3 is essentially the routing strategy used in ODMA.

6.3.3 Performance Evaluation

Performance evaluation of the system was done based on three criteria: Bit-error rate, signal-to-interference ratio and the percentage coverage increase.

To estimate the bit-error rate of a node, a function was used to estimate the expected value. For each node, the power level of the intended signal and the power level of the amalgamation of interference were used as the inputs of the function. The function is based on the Gaussian approximation of the performance of a direct spread spectrum system found in [27] and is defined as

$$P_{ber} = Q\left(\sqrt{\frac{1}{\frac{1}{3N}\left(\sum_{k=1}^{K-1}\frac{P_k}{P_0}\right)+\frac{N_0}{2T_bP_0}}}\right). \tag{6.1}$$

P_0 is the power (at the receiver) of the intended user who is transmitting and P_k is the power received from the kth interferer. The received power from the intended user and from the interferers are attenuated versions of the transmit power with respect to the path loss and the distance between the nodes.

In attaining the results for the SNI at the base station, a reference node was used. The reference node is an exact copy of a randomly chosen node that has a link to the base station whose power has already been controlled by the base station. The reference node has no effect on the SNI of other nodes or the base station. The reference node is segregated in order to prevent it from coming under the influence of other nodes that may affect its transmit EIRP. The assessment used to calculate SNI at the base station is defined as

$$SNI_{dB} = P_{ref} - 10\log\left(\sum_{\substack{k=0\\k\neq ref}}^{K-1} P_k\right) dB. \tag{6.2}$$

Coverage is defined as the ability to give a mobile unit radio access to the network. For the coverage analysis there are three indicators that are used: The overall increase in nodes with network coverage, the ratio of covered nodes by peer-to-peer access to nodes that are not covered at all and the increase in average coverage radius. The coverage increase results are based on the number of mobiles, that were originally not serviced, that are now able to attain radio coverage via peer-to-peer mode as a factor of the population with coverage. The coverage increase is defined as

$$C_{inc\%} = \frac{p+b}{b} - 1, \tag{6.3}$$

where p is the number of shadowed nodes given access via peer-to-peer mode and b is the number of nodes with a direct link to the base station. The second indicator serves the purpose of illustrating what percentage of nodes in radio shadows eventually get coverage via the peer-to-peer mode. The coverage ratio is defined as

$$R_{\%} = \frac{p}{s+p}, \tag{6.4}$$

where p is the number of shadowed nodes given access via peer-to-peer mode and s is the number of nodes that remain shadowed after peer-to-peer rules are administered.

The average coverage radius was evaluated by taking the average distance of the nodes that were originally outside the coverage area of the base station who are currently covered with peer-to-peer access. The expected coverage range of the base station is 2740 metres. This value is evaluated using the WITS pass loss model and the maximum allowable loss on the reverse link. The increase in coverage radius is therefore defined as

$$D_{\%} = \frac{avg\ dist}{2740} - 1. \tag{6.5}$$

6.3.4 Visualising the Environment

In order to verify the simulation, a graphical front end for the environment was developed to represent an instance of the simulation. As shown in Figure 6.4, each node is plotted (dots) and labelled with their "id_num" property; thus representing the Cartesian position but excluding the z-axis information. When a node is clicked upon, its properties are shown in the command window. The properties shown are the id number of the node, the current BER, SNI and transmit EIRP of the node, the past transmit EIRP of the node, the number of peer-to-peer links, the node assisting with power control and the distance away from the base station. The grey circular area is a visual approximation of the coverage of the base station. However, some nodes will get coverage outside of the coverage area (for instance node 39 in the diagram) and some nodes will have no coverage inside the estimated area (for instance node 66). Links that are formed on the basis of rule 1 are labelled with the red line that connects two dots; links that are formed on the basis of rule 2 are labelled with the green line; and links that are based on rule 3 are labelled with the magenta line.

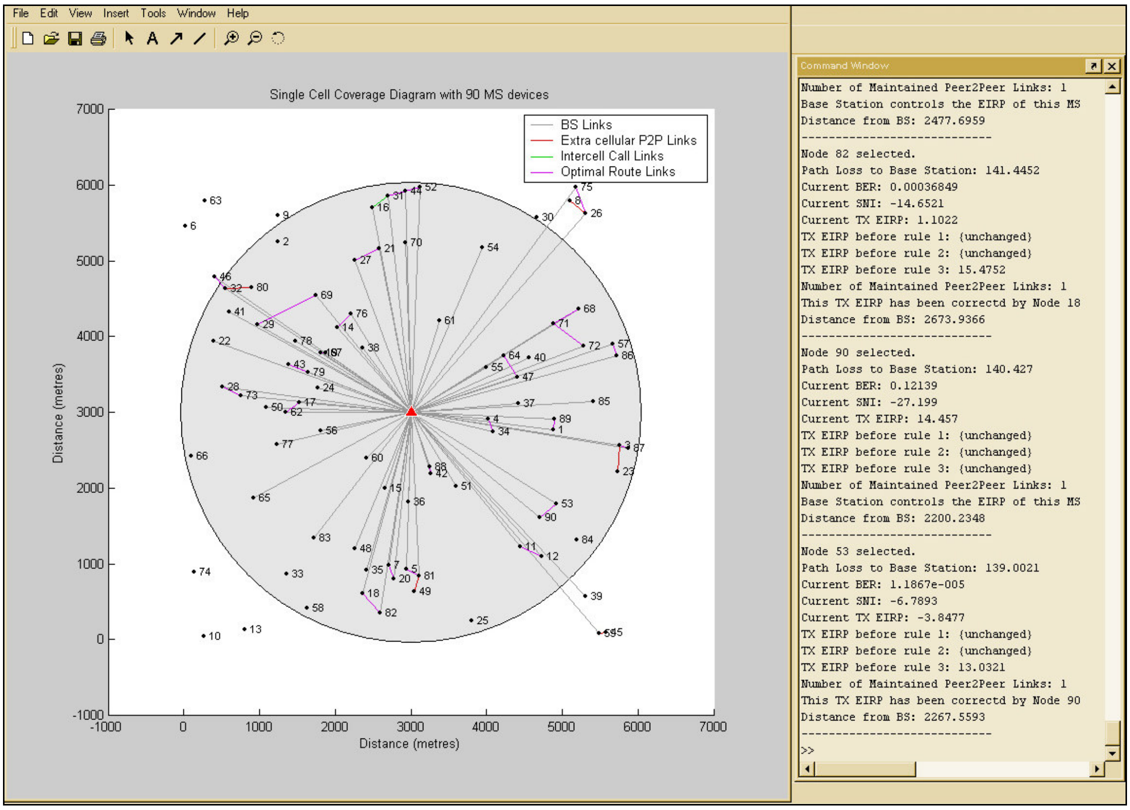

Figure 6.4: Graphical Representation of Simulation Environment

6.4 Summary

In this chapter, the design of the simulation that is used to comparatively assess the performance between the hybrid topology based on the design in chapter 4 is described. Here the data structure of the node elements are described along with the path loss environment that they are placed in. The assumptions about the environment are addressed in order to shed some light on the limitations of the simulation environment. The means by which the simulation evaluates the performance is then described. The results of the simulation are based on average result of 100 iterations for a set number of nodes in the environment. The result of the simulation will lead to a comparison of the performance of the hybrid topology versus standard cellular network topology in one cell. In the next chapter the results of the simulation are analysed and discussed in terms of the performance evaluation described in section 6.3.3.

7 RESULTS

The project presented two types of simulations. The first type of simulation aimed to represent the physical link layer of a code division duplexing system. Based on the aforementioned simulation type, two tests were done. One test based on the performance analysis of six links which involved six transmitters and six receivers. The second test was based on the channel loading where the number of users became the independent variable. The second type of simulation involved a model area in which nodes would employ connectivity rules typical of a hybrid topology network. The aim in this case is to analyse the coverage and capacity changes. In this chapter, the results of these simulations will be discussed.

7.1 Analysis of CDD Performance

Referring to Figure 5.7, there is a clear and apparent limitation of the CDD system. As mentioned earlier, the limitation is due to the multiple access interference. However, there is a reason for it being more significant in CDD than in DSSS. The problem lies in the fact that codes are being shared. Recall equation 4.12, the contribution to MAI, $\sum_{k=1}^{K-1} b_k(t)\Omega_k(t)C_k(t)$ represents the sum of the transmissions *K*-1 users, excluding the intended transmission to the 0th user. Assuming that a set of users is engaged in MS-MS transmissions, another set engaged in BS-MS transmissions and another set engaged in MS-BS transmissions then let there be an MS-MS transmission such that the sum of the transmissions, excluding the intended transmission, can be represented as

$$\sum_{k=1}^{K-1} b_k(t)\Omega_k(t)C_k(t) = \sum_{e=0}^{E-1} b_e(t)\Omega_e(t)C_e(t) + \sum_{f=0}^{F-1} b_f(t)\Omega_f(t)C_f(t) + \sum_{g=1}^{G-1} b_g(t)\Omega_g(t)C_g(t)\,, \quad (7.1)$$

where there are *E* BS-MS interferers, *F* MS-BS interferers and *G*-1 MS-MS interferers. In simulation environment 1 each duplexing direction was represented with a single orthogonal Gold code. Then it follows that

$$\sum_{e=0}^{E-1} C_e(t) = E.C_e(t)$$
$$\sum_{f=0}^{F-1} C_f(t) = F.C_f(t) \tag{7.2}$$
$$\sum_{g=1}^{G-1} C_g(t) = G.C_g(t)$$

$\therefore$

$$\begin{aligned}&\sum_{k=1}^{K-1} b_k(t)\Omega_k(t)C_k(t) = \\ &E.C_e(t)\sum_{e=0}^{E-1} b_e(t)\Omega_e(t) + F.C_f(t)\sum_{f=0}^{F-1} b_f(t)\Omega_f(t)C_f(t) + \\ &(G-1)C_g(t)\sum_{g=1}^{G-1} b_g(t)\Omega_g(t)C_g(t).\end{aligned} \tag{7.3}$$

Substituting in equation 4.9 we get the decision statistic for the MS as

$$U_0 = \int_0^{T_b} \begin{bmatrix} b_0(t)\Omega_0^2(t)C_0^2(t) + \\ n(t)\Omega_0(t)C_0(t) + \begin{bmatrix} E.C_e(t)\sum_{e=0}^{E-1} b_e(t)\Omega_e(t)C_e(t) + \\ F.C_f(t)\sum_{f=0}^{F-1} b_f(t)\Omega_f(t)C_f(t) + \\ (G-1)C_g(t)\sum_{g=1}^{G-1} b_g(t)\Omega_g(t)C_g(t) \end{bmatrix} \Omega_0(t)C_0(t) \end{bmatrix} dt\,, \tag{7.4}$$

the desired portion of U_0 can be simplified as follows

$$D_0 = \int_0^{T_b} b_0(t)\Omega_0^2(t)C_g^2(t)dt\,. \tag{7.5}$$

Since $\Omega_0^2(t) = 1$ and $C_g^2(t) = 1$ then

$$
\begin{aligned}
D_0 &= \int_0^{T_b} b_0(t)dt \\
&= \int_0^{T_b} \sum_{m=0}^{M-1} b_k(m)\Pi_{T_b}(t - mT_b)dt\,, \qquad (7.6)\\
&= T_b b_0(m)
\end{aligned}
$$

then the noise contribution can be represented as

$$
\eta = \int_0^{T_b} n(t)\Omega_0(t)C_g(t)dt\,, \qquad (7.7)
$$

and the contribution due to MAI is then

$$
\zeta = \int_0^{T_b} \left[\begin{bmatrix} E.C_e(t)\sum_{e=0}^{E-1} b_e(t)\Omega_e(t)C_e(t) + \\ F.C_f(t)\sum_{f=0}^{F-1} b_f(t)\Omega_f(t)C_f(t) + \\ (G-1)C_g(t)\sum_{g=1}^{G-1} b_g(t)\Omega_g(t)C_g(t) \end{bmatrix} \Omega_0(t)C_g(t) \right] dt\,. \qquad (7.8)
$$

Let the chip rate be equal to the “pip” rate such that $T_c=T_p$ $\therefore$ T_b/T_c is the processing gain N.

Then let

$$
\Phi_k(t) = \Omega_k(t)C_k(t)\,. \qquad (7.9)
$$

Therefore, we can represent the contribution due to MAI from each duplexing group as

$$
\zeta_e = \int_0^{T_b} \left[\left[E.C_e(t)\sum_{e=0}^{E-1} b_e(t)\Phi_e(t) \right] \Omega_0(t)C_g(t) \right] dt\,, \qquad (7.10)
$$

$$\zeta_f = \int_0^{T_b} \left[\left[F.C_f(t) \sum_{f=0}^{F-1} b_f(t) \Phi_f(t) \right] \Omega_0(t) C_g(t) \right] dt\,, \tag{7.11}$$

and

$$\zeta_g = \int_0^{T_b} \left[\left[(G-1) C_g(t) \sum_{g=1}^{G-1} b_g(t) \Phi_g(t) \right] \Omega_0(t) C_g(t) \right] dt\,, \tag{7.12}$$

where

$$\zeta = \zeta_e + \zeta_f + \zeta_g\,. \tag{7.13}$$

There is a gain proportional to the number of users in each of the duplexing modes. This is the central reason why the error rates in CDD are higher than DSSS for the same number of users. If all users shared the same code, then there would be a gain of K-1 on the interference.

The bit-error performance of the system can be written as

$$P_{ber} = Q\left(\sqrt{\frac{P_0 T_b^2}{2\sigma_\xi^2}} \right), \tag{7.14}$$

then

$$P_{ber} = Q\left(\sqrt{\frac{P_0 T_b^2}{2(\sigma_\zeta^2 + \sigma_\eta^2)}} \right), \tag{7.15}$$

where $\sigma_\xi^2 = \sigma_\zeta^2 + \sigma_\eta^2$

$\therefore$

$$P_{ber} = Q\left(\sqrt{\frac{P_0 T_b^2}{2(\sigma_{\zeta_e}^2 + \sigma_{\zeta_f}^2 + \sigma_{\zeta_g}^2 + \sigma_{\eta}^2)}}\right). \tag{7.16}$$

Using the Gaussian Approximation for spread spectrum CDMA (described in [27])

$$\sigma_{\xi}^2 = \frac{ENT_c^2}{6}\sum_{e=0}^{E-1} P_e + \frac{FNT_c^2}{6}\sum_{f=0}^{F-1} P_f + \frac{(G-1)NT_c^2}{6}\sum_{g=1}^{G-1} P_g + \frac{N_0 T_b}{4}, \tag{7.17}$$

where $\frac{N_0 T_b}{4}$ is the σ_{η}^2 contribution from the additive white Gaussian noise.

So the function for bit-error performance can be written as

$$P_{ber} = Q\left(\sqrt{\frac{P_0 T_b^2}{\frac{NT_c^2}{3}\left(E\sum_{e=0}^{E-1} P_e + F\sum_{f=0}^{F-1} P_f + (G-1)\sum_{g=1}^{G-1} P_g\right) + \frac{N_0 T_b}{2}}}\right) \tag{7.18}$$

$\therefore$

$$P_{ber} = Q\left(\sqrt{\frac{1}{\frac{1}{3N}\left(E\sum_{e=0}^{E-1} \frac{P_e}{P_0} + F\sum_{f=0}^{F-1} \frac{P_f}{P_0} + (G-1)\sum_{g=1}^{G-1} \frac{P_g}{P_0}\right) + \frac{N_0}{2P_0 T_b}}}\right). \tag{7.19}$$

In the interference limited case $\frac{1}{3N}\left(E\sum_{e=0}^{E-1} P_e + F\sum_{f=0}^{F-1} P_f + (G-1)\sum_{g=1}^{G-1} P_g\right) >> \frac{N_0}{2P_0 T_b}$,

where there is perfect power control then

$$P_{ber} = Q\left(\sqrt{\frac{3N}{E^2 + F^2 + (G-1)^2}}\right). \tag{7.20}$$

If all users share the same code then the function can be written as

$$P_{ber} = Q\left(\sqrt{\frac{3N}{(K-1)^2}}\right). \tag{7.21}$$

In comparison with equation 2.8, it is clear that sharing of codes is detrimental to the bit-error performance of a code division system. Figure 7.1 shows the comparison of the two systems when six users are loaded unto the channel (with a processing gain of 128 chips). The error rates in the theoretical plot are lower than the simulation (Figure 5.7) due to the actual correlation of orthogonal Gold codes. Although the error rates are not the same when Figure 5.7 and Figure 7.1 are compared, it can be seen that the behaviour of the system from a theoretical point of view and from the simulated point of view is same. This, therefore, strongly suggests that CDD/TDD CDMA that employs code sharing for each duplexing direction will have reduced performance over a DSSS based system.

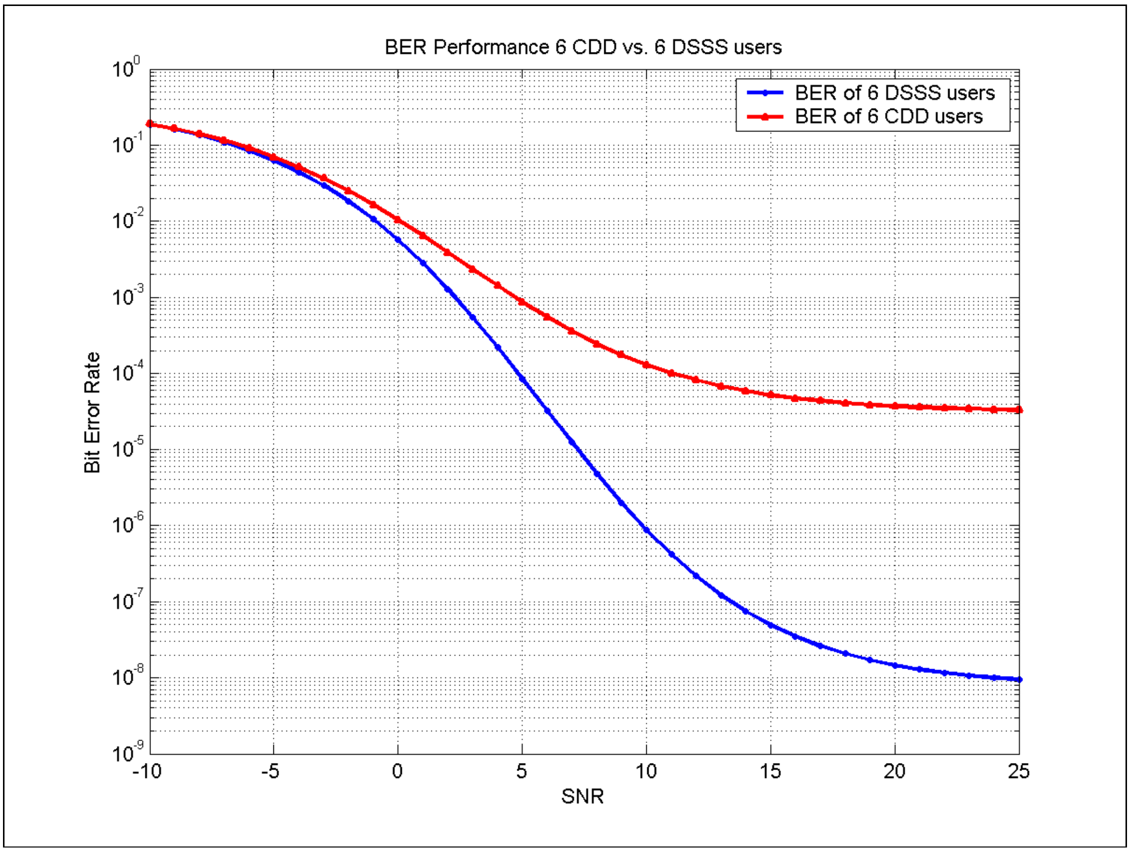

Figure 7.1: Theoretical Performance of 6 CDD users vs. 6 DSSS users

7.2 Coverage Analysis Results

As highlighted in chapter 6, the simulation area is 6 km^2 in size in which one base station presides over the area. Nodes are randomly distributed in the area. Most nodes will get radio service, but a certain number of nodes will be deemed to be in the radio shadow because of their path loss to the base station exceeds the minimum allowable path loss for a radio link. Rule 1 is responsible for giving nodes the initiative to seek a peer-to-peer connection in the absence of a connection to the base station.

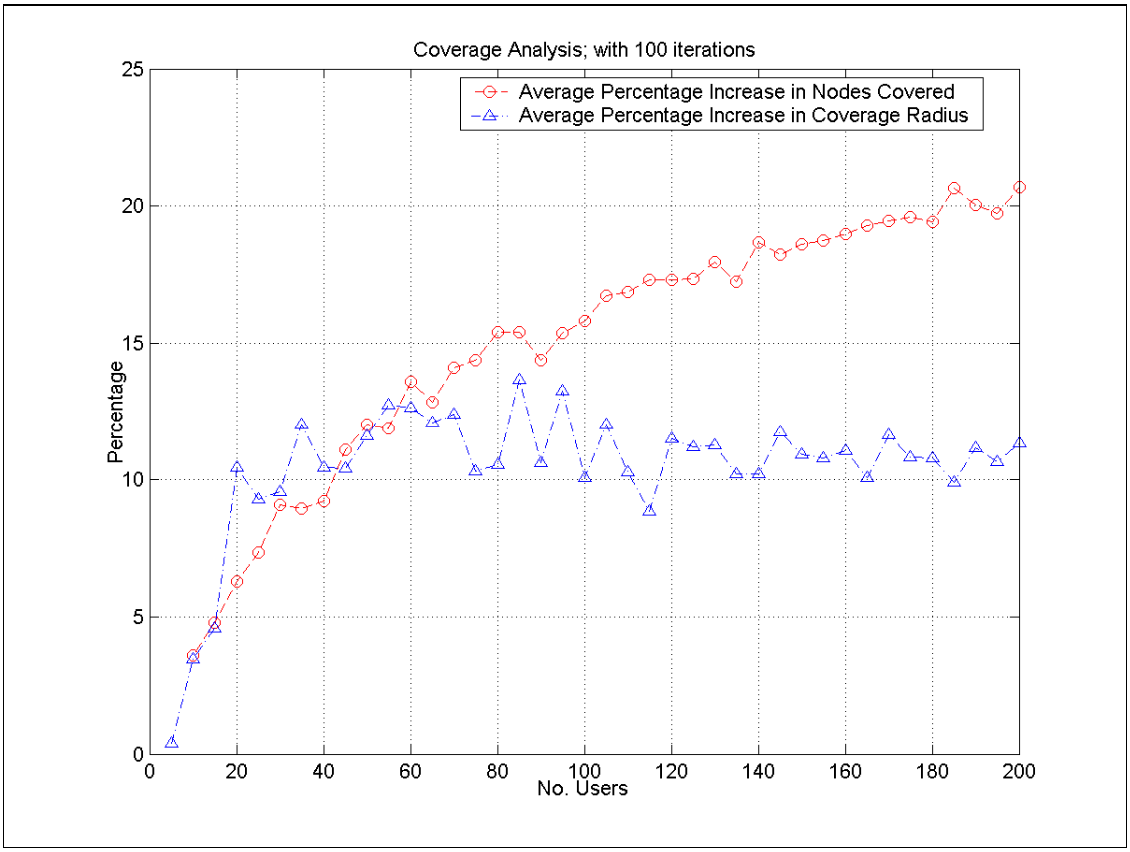

Figure 7.2: Percentage of the Coverage Increase of the Overall Node Population for 1 Slot

Figure 7.2, shows an increase in the overall coverage as the number of nodes increases in the simulation environment. At a population size within the simulation area of 200 nodes[1], the percentage increase on the number of nodes covered is 20.67%. The trend of the increase seems to be asymptotic towards 24%. There is a maximum increase of 13.65% in terms the average radius of the nodes covered by a peer-to-peer connection. This implies that, where as the expected coverage radius of the base station is usually 2740 metres, the new expected coverage radius in a one hop environment is 3114 metres.

[1] 30% channel load. This is calculated using the following: (200 x voice activity) / processing gain

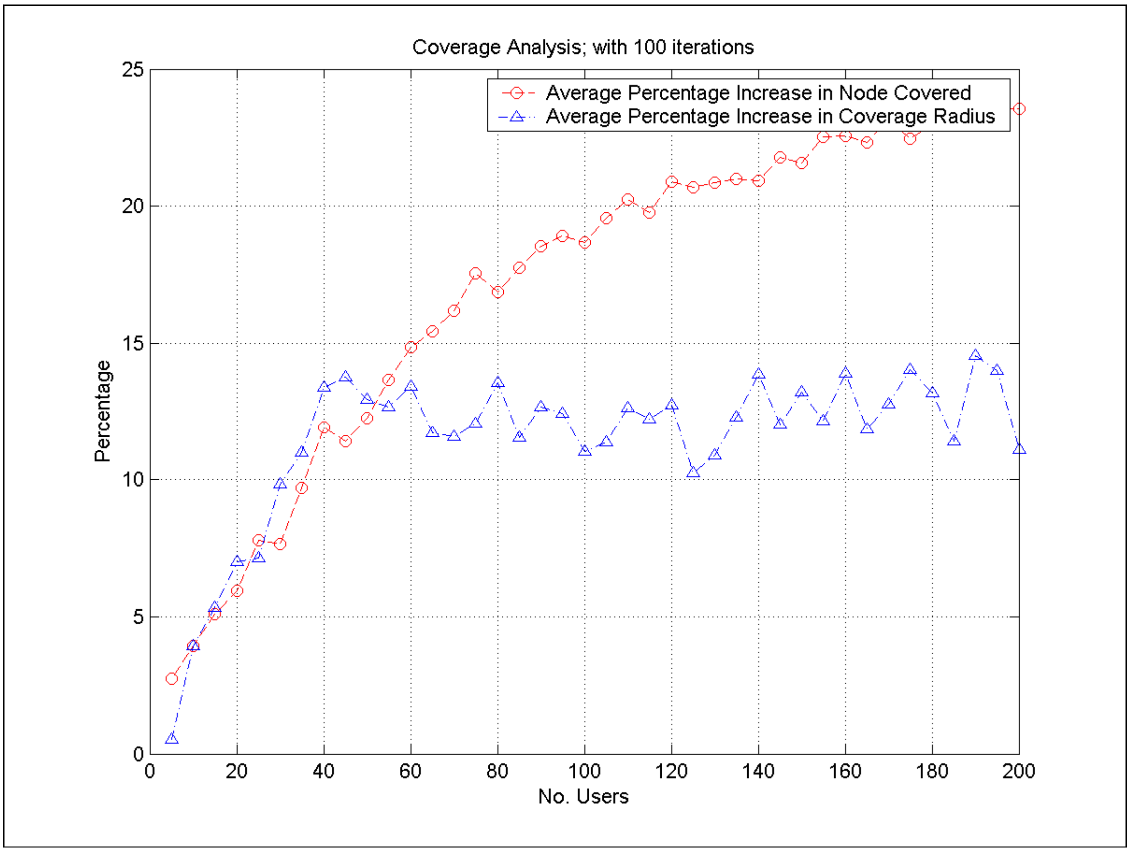

Figure 7.3: Percentage of the Coverage Increase of the Overall Node Population for 2 Slots

Figure 7.3 shows the coverage increase for the instance when nodes are allowed to share resources with two other nodes. Here there is a maximum increase in coverage of 23.56% in comparison to 20.67% in the single slot mode when the simulation environment population is 200 nodes. It is evident from the comparison of Figure 7.2 and Figure 7.3 that when an MS node is allowed to share more resources, the number of nodes that can be covered will rise at a faster rate as the population size within the cell increases. In this case the average coverage distance is almost the same with a peak increase of 14.54% in the coverage radius because, like the previous case, a one hop strategy is employed.

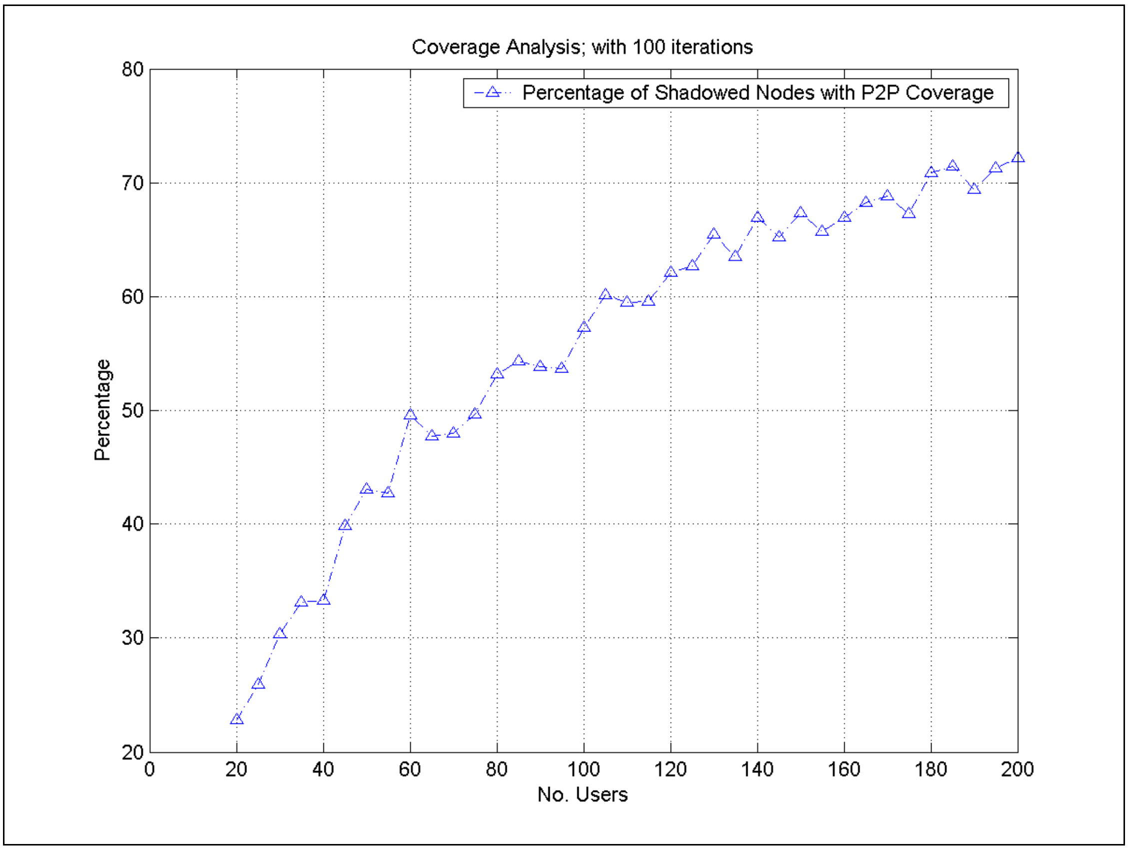

Figure 7.4: Ratio of Shadowed Nodes with Peer-to-peer Coverage to Shadowed Nodes for 1 Slot

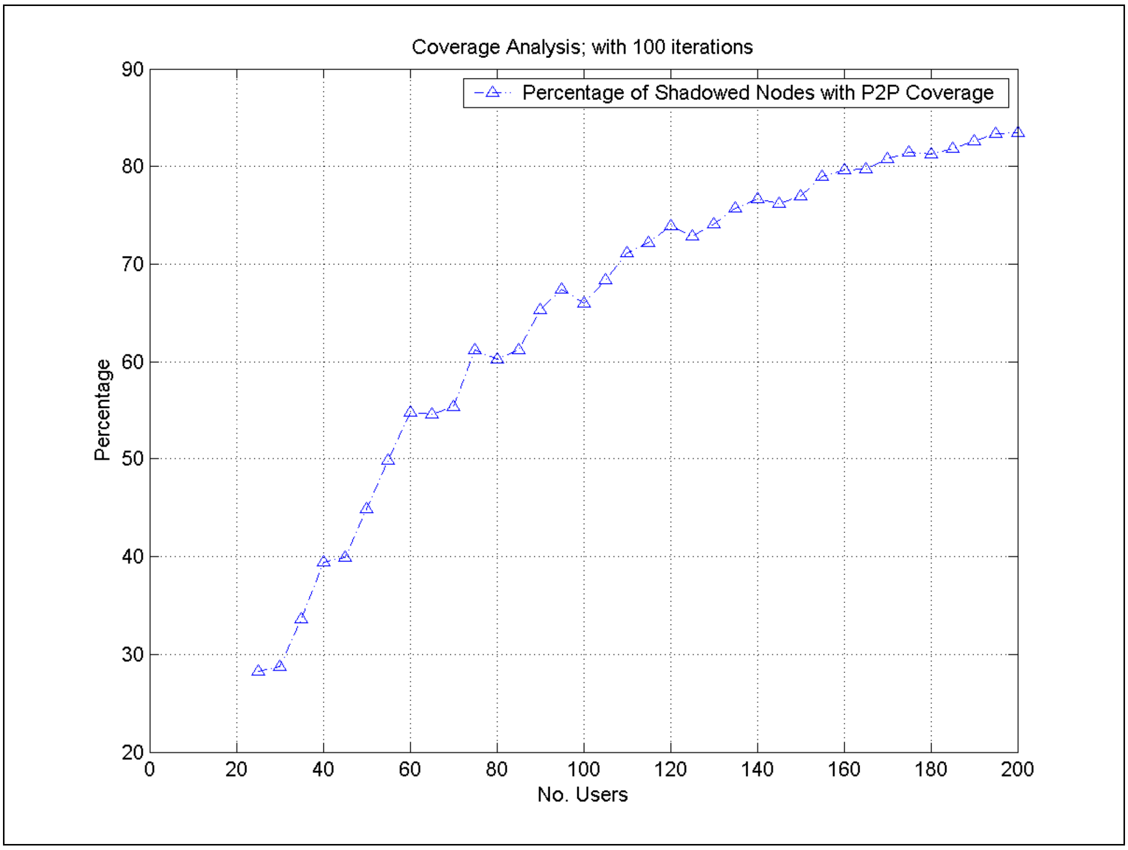

Figure 7.5: Ratio of Shadowed Nodes with Peer-to-peer Coverage to Shadowed Nodes for 2 Slots

Figure 7.4 and Figure 7.5 show the ratio of shadowed nodes that have received coverage via peer-to-peer access to the total number of shadowed nodes. Both figures indicate that nodes that are in radio shadows have a chance of getting coverage and that as the population increases, the amount of shadowed nodes that get access to the network via peer-to-peer connections will increase. This implies that the probability of getting service while in a radio shadow will increase with the increasing population size within a cell. In the case where an MS node will share two slots, the rate of increase of the ratio is faster than in the one slot scenario. This is because one MS unit can now serve two other orphaned[1] MS units whereas before, an MS was only capable of serving one other orphaned MS node.

[1] An orphaned node is a node that has no access to the network via a direct connection the base station.

7.3 Capacity Analysis Simulation

After peer-to-peer rules are applied, the SNI ratio improves relative to the standard infrastructure system, as seen in Figure 7.6. This is due to the autonomous power control that each node employs on each other. The nodes in close proximity to each other will reduce there EIRP to transmit to their neighbour. As a consequence, the interference from the MS units received at the BS reduces. Considering the relationship between path loss and distance, the reduction of transmit powers by the MS units will reflect a significant decrease in interference. The performance difference in the standard topology and the rule 1 of the hybrid topology is 0.3 dB on average. The small improvement is attributed to the fact that nodes that are unable to form links with base station will be searching at their maximum transmit EIRP until they find a peer or are in range of the base station. In this case they are provided coverage by a peer node that is in their range. At this point, the node will reduce its transmission power to communicate with the peer providing the hop to base station.

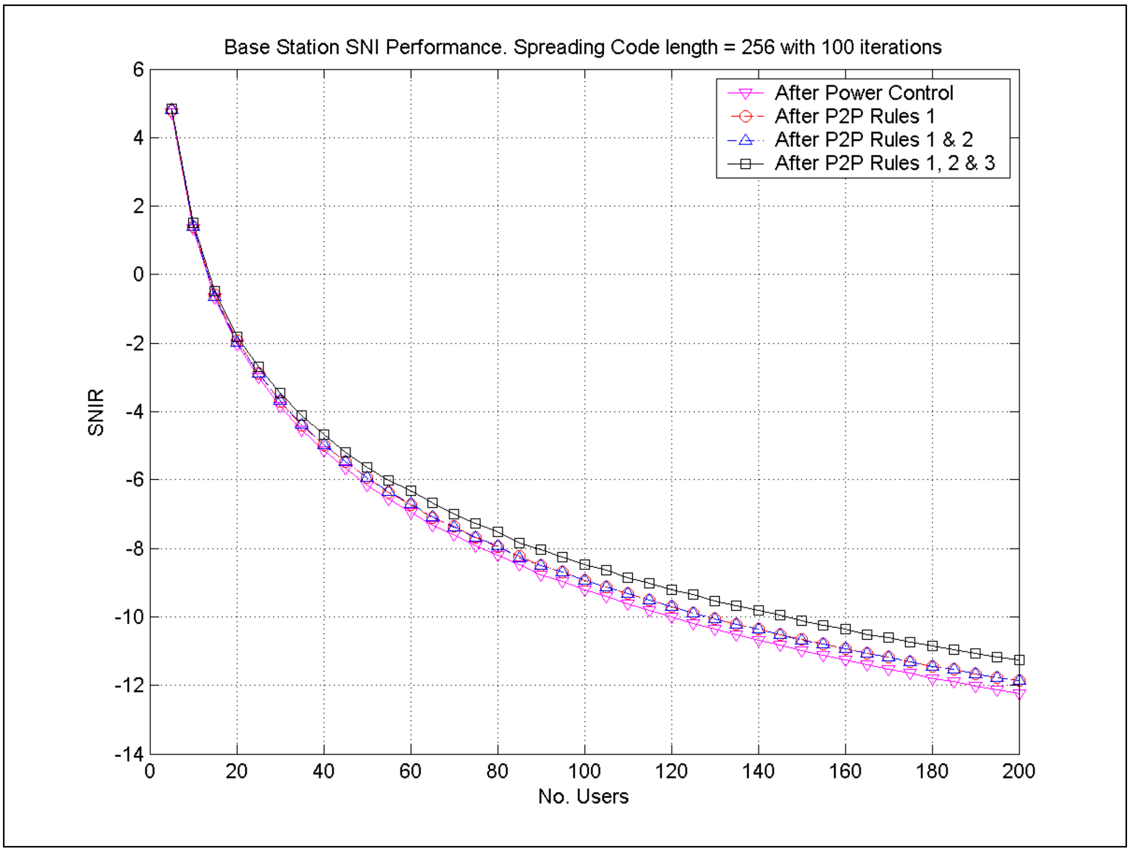

Figure 7.6: SNI Performance at the Base Station

When both rule 1 and rule 2 are applied the performance is almost identical. Rule 2 allows nodes to initiate inter cell calls. The probability of an inter cell call occurring is small and thus there is no significant difference in the performance of rule 1 and the performance of rule 1 along with rule 2. It must be highlighted that though the performance of rule 1 and 2 versus the standard remain very close, the number of the shadowed nodes that get serviced by peer-to-peer links has increased. This suggests that there are more users overall for (approximately) the same amount of interference at the base station. To put this in to context, at 40 MS nodes in the environment the base station experiences an SNI of -4.9812 with rule 1 and experiences and SNI of -5.1277 in the standard mode, a mere difference of 0.1465 dB, but at this point (referring to Figure 7.2 and Figure 7.3) there is a 9.21% increase in the number of nodes serviced for a 1 slot strategy and an 11.927% increase in the 2 slot mode. When rule 3 is applied, the amount of interference at the base station reduces further. This can be attributed to the peer-to-peer connection being made to alleviate the base station of traffic.

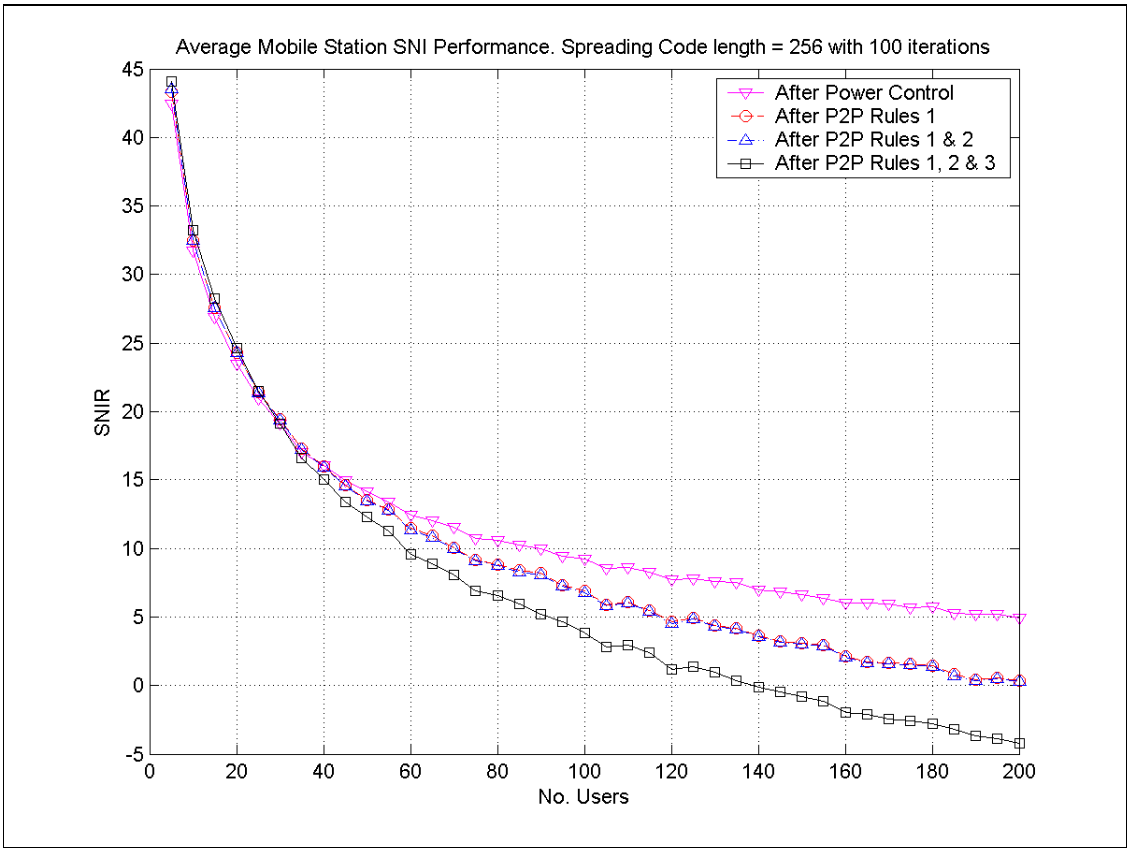

Figure 7.7: Average Mobile SNI Performance for a Single Slot Strategy

In Figure 7.7, the expected trend of deteriorating performance with the increasing number of nodes in the environment is observed. The average SNI for the MS units, in comparison to the base station, reduces when peer-to-peer rules are applied. As the peer-to-peer rules are applied there is a faster rate of deterioration in the signal-to-noise ratio versus the population. In other words, increased peer-to-peer activity will have a negative effect on the performance of the average performance of the MS units. A possible factor that reduces the systems performance is the localised interference due to node positions which was also mentioned in [25]. However, it is interesting to note that below 35 nodes in the environment, peer-to-peer activity contributed a small but noticeable increase in performance over the standard infrastructure topology.

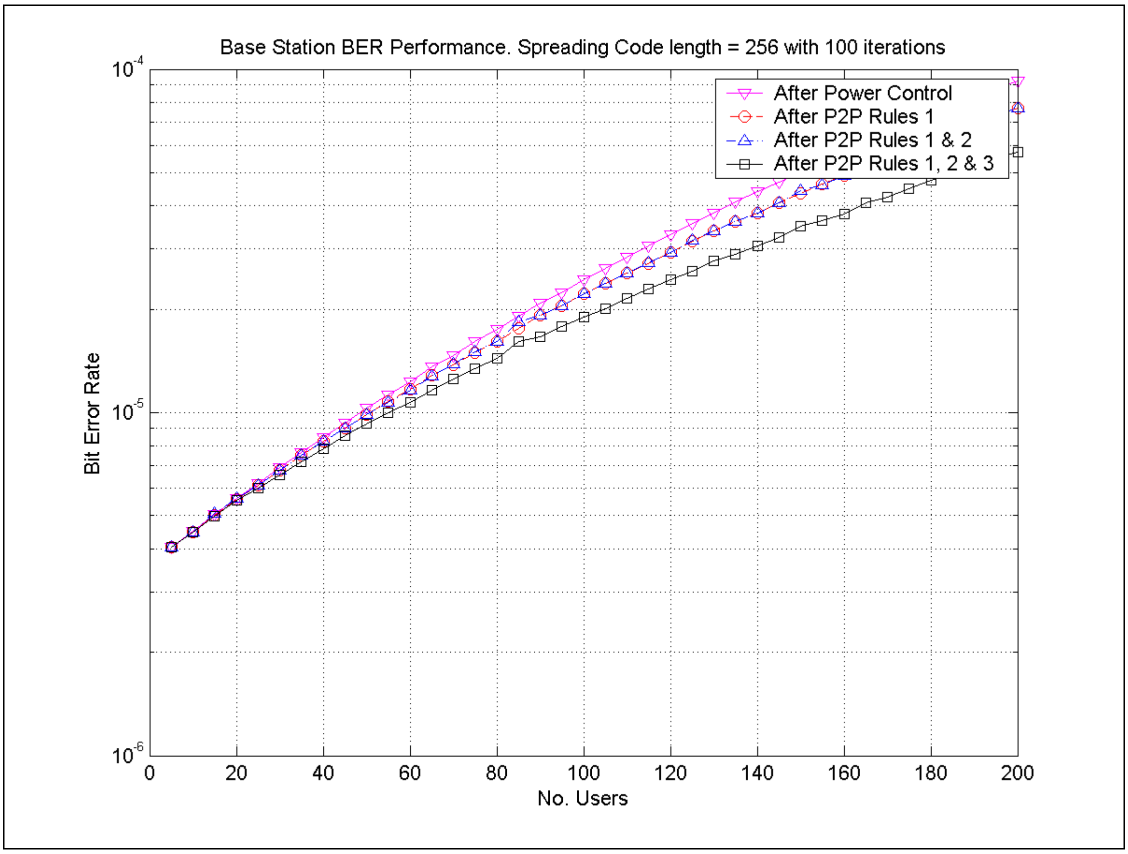

Figure 7.8: Base Station BER Performance

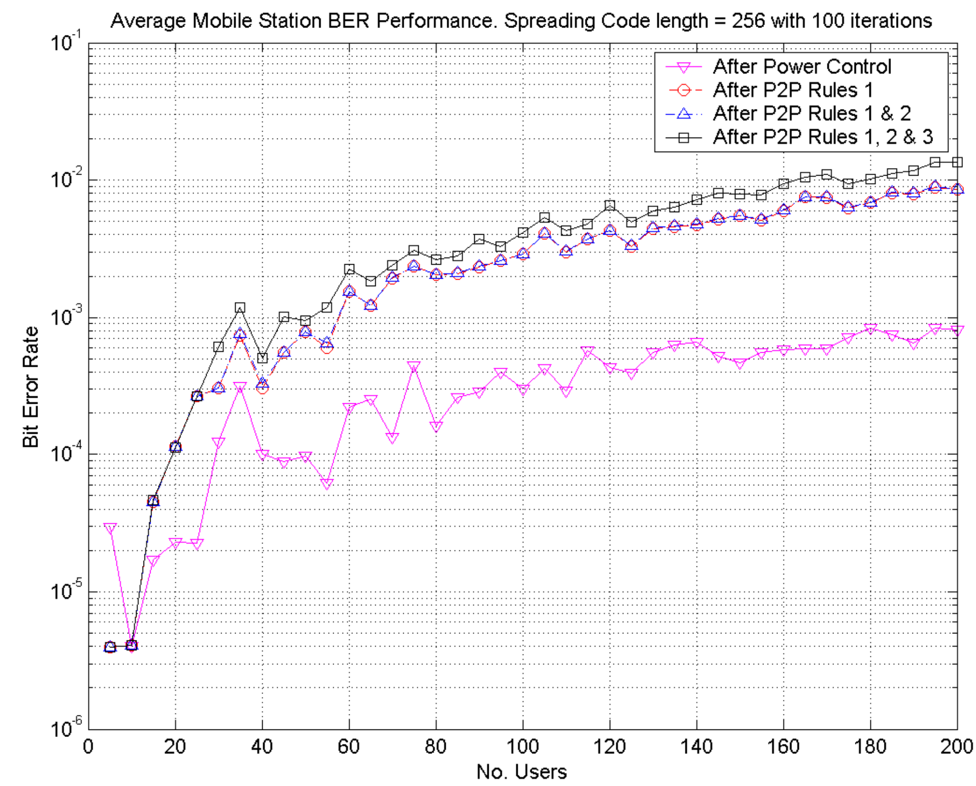

Figure 7.9: MS BER Performance

In Figure 7.8, the bit-error performance for a single user at the base station is best when all three peer-to-peer rules are applied. This is expected seeing that in Figure 7.6 there the SNI performance was improved with peer-to-peer activity. However, it can be seen in Figure 7.9 that the peer-to-peer activity will become destructive in terms of the average BER performance of the MS units. It can then be concluded that, even though rule 3 alleviates the base station of some interference, it does not positively affect the MS nodes in addition to the fact that it does not contribute in the coverage increase at all. It only facilitates nodes that are already connected to the base station to find an alternate route for data. Although the performance of the MS units is diminished in comparison to the standard infrastructure mode, the number of nodes being serviced has increased at the cell edges. This is the reason why the average performance of the MS is lower because there is an increase in the amount of interfering users at the same rate represented in Figure 7.2 and Figure 7.3.

7.4 Summary

In summary, hybrid topology will increase the coverage in terms of the number of nodes given service when those nodes were originally not serviced; and in terms of the expected coverage radius of a cell with a one hop routing strategy. It has also been shown that as the population of the cell increases the probability of a shadowed node attaining access to the network via a peer-to-peer connection increases.

Rule 1 along with rule 2 are recommended to increase the performance of the hybrid topology seeing that rule 1 is bound to increase the capacity and the coverage of the network by allowing orphaned MS nodes to communicate; and seeing that rule 2 does not significantly change the performance of the network topology. Rule 3, on the other hand, only aims to alleviate the base station traffic but as a consequence degrades the performances of the MS units on average. Therefore in can be concluded that a full ODMA implementation (rule 1 along with rule 3) is not necessary, but rather coverage enhancing routing (rule 1 along with rule 2) is preferred.

8 AREAS OF FURTHER RESEARCH

The project provided insight into just one portion of this vast field of hybrid networks that utilise code division duplexing. In the process of completing this project many interesting questions have been raised. Examples of the questions that have arisen are as follows:

- What are the consequences of using MS nodes to provide explicit diversity gain?
- How must multi-user detectors adapt with the implementation of a CDD system?
- Will routing in a hybrid topology mobile cellular network be considerably different from a standard network?
- How will channel assignment affect the systems performance?
- What are the design considerations that must be made for mobility in a hybrid topology network?

8.1 Explicit Diversity Gain

Explicit diversity gain is the gain that is achieved by using diversity techniques. In an ad-hoc network it is possible to provide time and space diversity by allowing neighbourhood nodes to relay copies of the signal in time, frequency or space. In terms of time diversity, multiple neighbourhood nodes would have to receive transmission intended for the target mobile and re-transmit at a suitable delay period such that the target mobile is able to combine the signals into useful information. Multiple neighbourhood nodes can also provide frequency diversity by relaying duplicate transmissions on a different frequency. Also neighbourhood nodes can provide space diversity seeing that neighbourhood nodes can transmit duplicates from different spatial position.

8.2 Multi-user Detection

The level of multiple access interference is different when code division duplexing is implemented, in contrast to a standard direct spread spectrum system. As a consequence, there is a need for modified versions of multi-user detectors. Taking an example where MS units are communicating with the base station, it has been shown that the multiple access interference is higher than a DSSS system due to a gain that stems from the sharing of codes. In the project, two ways were suggested to mitigate the gain due to the sharing of codes. However, this does add some amount of overhead to either the network or the mobile unit. Ideally, it would be more suited to incorporate multi-user detection for CDD in present multi-user detectors. The mitigation of the MAI gain in CDD is a vast area of research to be explored.

8.3 Routing Strategies

As IP networks grow in size the efficiency of the network depends tremendously on the routing algorithms that are implemented. In ad-hoc IP based wireless networks, like 802.11b, the nature of the routing algorithmic becomes even more critical. As the perspective changes to a cellular mobile network where delay and jitter are not tolerable as they are in traditional IP networks. Likewise, as the number of users in a cellular mobile network increases, the complexity of the routing decisions increase exponentially. In traditional mobile cellular networks routing is centralised, on the other hand, a hybrid topology network would have distributed routing. To make matters more complex, routing algorithms would have to consider more cost parameters for example SNR, CIR, path loss, delay time, number hops, MS velocity, handover probability, and MS unit power. It is clear that routing tables have the potential to grow to incredible sizes, in addition, the radio environment changes so rapidly that routing tables would also be consistently changing. Other difficult challenges are born out of the sheer number of users in a cellular mobile network. Large networks with autonomous nodes suffer from congestion and deadlock. Situations like this must also be catered for in the CDD/TDD CDMA network.

Questions arise: If a cellular mobile network has autonomous node with the ability to relay data, what routing algorithms and what routing criteria are to be proposed? What kind of processing power will MS nodes require in order to find the most efficient path from a complex routing table? How often should nodes update routing tables?

8.4 Channel Assignment Schemes

In hybrid topology networks, nodes must automatically assign some of its own resources to neighbouring peers. Some interesting questions are brought up. How much of its resources can it allocate and what will be the overall response of the network? In this project it was assumed that once neighbourhood nodes met the path loss requirements a connection could be made. However, could there be another indicator that could possibly be kept track of by the base station and relayed to the nodes informing them if it is efficient to share resources?

8.5 Mobility in Hybrid Networks

Mobility in a hybrid network present many challenges that are not present in the standard cellular infrastructure. One of the questions that is brought to the foreground is the question of hand over considerations. For instance, if a MS is engaged in a voice call via a hop on another MS, how will the connection be handed over when the relaying MS is no longer in sufficient radio range? Consider the scenario where the relaying MS must be prepared to hand over to another MS or the base station. In this instance, speed and movement prediction algorithms will be needed to determine the best MS to select, where the likelihood of the MS remaining in the range of the MS engaged in voice call is longest. This pre-calculation must be either executed by the relaying MS node before it moves out of range, the network or both. However, there will be a consequence of increased overhead on the network overall.

In a pure ad-hoc network employing store and forward relaying, it was shown in [14] that the overall capacity of the network was increased due to node mobility. However, this case is limited to delay tolerant communication. Does mobility also increase the capacity of hybrid networks in which nodes have the ability to store and forward delay tolerant data?

9 CONCLUSION

The CDD/TDD-CDMA system facilitates MS-MS communication on the physical layer, as well as the BS-MS communication. This results in the overlaying of an ad-hoc like architecture on the standard cellular infrastructure. The advantage of separating transmission direction using the code division duplexing is that unwanted transmission can be rejected (and thus ignored) by the receiver.

The MS and BS transmitter/receiver structure of such a network is considered as well as the use of Orthogonal Gold codes as the coding format used to provide the code level differentiation. In addition, a billing oriented call setup protocol was proposed that facilitates the integration of MS-MS communications into the standard network infrastructure.

The proposed system architecture offers the following benefits: simple frequency reuse planning for network operators and radio spectrum regulator bodies; simple power control; increase in system capacity because of the inherent advantages of a hybrid topology; conservation of power by mobile devices; and an increase in coverage.

It has been shown that code sharing in CDD increases the multiple access interference. Thus, network managed code assignment and random index code shifting were recommended to alleviate the MAI and offer performance approaching the performance of a direct sequence spread spectrum system.

The coverage and capacity simulation has demonstrated that a hybrid topology can increase the coverage capabilities of a network cell by providing coverage to MS nodes that are in radio shadows. Also, it has demonstrated that the overall system will yield an increase in the SNI for the base station, due to the reduction of transmit EIRP in MS nodes. On the other hand, the simulation showed that there would be a reduction in the bit-error rate performance of the MS nodes, due to increased peer-to-peer activity.

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APPENDIX I: PUBLICATIONS

The following publications resulted from this study:

[1] M.-P. Powell and P. J. Chitamu. “Performance Analysis of a Hybrid Topology CDD/TDD-CDMA Network Architecture,” SATNAC 2004 Conference proceedings Sep. 2004.

[2] M.-P. Powell and P. J. Chitamu. “Designing a CDD/TDD-CDMA Network Architecture,” SATNAC 2005 Conference proceedings. Sep. 2005.

APPENDIX II: MATLAB FUNTIONS

The following functions were written in MATLAB for the purpose of this research.

Table 9.1: MATLAB script functions for CDD analysis

Function Name	Syntax	Description
best_ogold	[ogoldseq1,ogoldseq2] = best_ogold(factor,id1,id2)	Works out the best pair of orthogonal Gold codes based on the factor. factor = n, where 2^n = the length of the m-sequences that are used in cross-correlation and auto correlation. The pairs of m-sequences codes with the lowest cross-correlation maxima are selected to form Gold codes. The codes are then padded with zeros to form orthogonal Gold codes.
bi_uni	result = bi_uni(value)	This function converts unipolar arrays to bipolar arrays. Example, [1 0 0 1] = [1 -1 -1 1]
bin2quad	result = bin2quad(data)	Converts a stream of binary symbols to a stream of quaternary symbols.
bpsk_baseband_test	[actual_err, theoretical_err] = bpsk_baseband_test(snr,frames)	Simulates a Base band BPSK transmission between two peers. The purpose of this simulation is to verify the BPSK function by seeing how closely it resembles the theoretical error rates, while indirectly verifying the additive white Gaussian noise function. The theoretical error rates based on the erfc function associated with BPSK. 1

		frame is 1280 bits.
channel_load	channel_load(snr,factor,bits, load)	This Function tests the channel loading capabilities of a basic DSSS CDMA system and the channel loading capabilities of a CDD DSSS CDMA system that only involves peers. The function between produces a graph that that compares BER to the number of users at a specified signal-to-noise ratio. For the DSSS CDMA system, the input 'factor' produces an orthogonal Gold code of length 2^n; for the CDD system it produces a Walsh code and an orthogonal Gold code of length 2^n. The input 'frames' is the number of frames to be sent. The value of the input 'load' must be a value between 0 and 1, to represent the percentage load to calculate. Primary feature, noise is added to each BPSK user then added together.
corr_mseq	[result, def_array] = corr_mseq(factor)	This function is used to correlate m-sequences in order to find auto and cross-correlation properties. The factor input is the power of the highest polynomial of the m-sequence definition.
despread	result = despread(data_in, spreader)	This function de-spreads the data input (data_in) with the specified spreading code (spreader).
find_ogold	result = find_ogold(pn1,pn2)	This function finds pairs of orthogonal Gold codes, where pn1 and pn2 are the m-sequence definitions.
find_m_seq	result = find_m_seq(factor)	This function finds m-sequence definitions, since not all PN sequences at that factor are

		m sequences; where the factor is the power of the most significant polynomial.
gold_code	result = gold_code(pn1,pn2,shift)	This function takes two PN m-sequence definitions and a shift parameter the second PN sequence is the one that is shifted.
ogold	code = ogold(pref_pn1,pref_pn2,id1)	Pads a zero to a PN m sequence to make a Gold code.
ogold_array	result = ogold_array(factor)	Works out an array of orthogonal Gold codes using the best pair of m-sequences codes based on the cross-correlation values of the parent PN sequences. The function uses corr_mseq(factor) to determine the definition of PN m-sequences.
peer2peer	[p2p_bit_perf,star_bit_perf] = peer2peer(snr, factor, frames, ogold_set, walsh_code_set)	Calculates the BER of 6 transmission links for CDD DSSS and basic DSSS for a given SNR
peer2peer_perf	peer2peer_perf(min_snr,max_snr,min_fac,max_fac,frames)	This function tests the performances of the CDD mode communication setup versus the standard infrastructure mode setup. 1 frame is 1 bit long.
peer2peer_s	peer2peer_s(min_snr,max_snr)	Plots the performance of a CDD DSSS versus a basic DSSS system for a single user.
pn_seq	result = pn_seq(pn_def)	Generates a pseudo noise sequence based on the input definition (pn_def). The input definition must be in the form that indicates the presence of a polynomial factor. Example if the definition is x^3 + x + 1 then pn_def should equal [1 0 1 1]
quad2bin	result = quad2bin(data)	Converts a data sequence of quaternary

		symbols to data sequence of binary symbols.
split	result = split(value,size)	This function splits a double into an array of single digits. Example: if value is 123, then the size is 123 and the result is [1 2 3].
spread	result = spread(data_in, spreader)	This function spreads a data input with (data_in) a spreading code (spreader). Each bit of the data_in is represented by the spreader or the inverse of the spreader depending on the bit (1/0). The resulting sequence is the length of the data_in * the length of the spreader.
uni_bi	result = uni_bi(value)	This function converts unipolar arrays to bipolar arrays.
user_rx	[data_rx,rx] = user_rx(tx,spreader)	Demodulates (BPSK) and de-spreads incoming signal
user_rx_qam	[data_rx,rx] = user_rx_qam(tx,spreader)	Demodulates (QPSK) and de-spreads incoming signal
user_tx	[pre_chan_tx, data_tx, spr_data] = user_tx(spreader,data_length)	Generates random bits and modulates them using BPSK. pre_chan_tx is the output after the data has been spread and modulated. data_tx is the data sequence that was generated; and spr_data is data_tx after is has been spread by the spreader (spreading sequence).
user_tx_qam	[pre_chan_tx, data_tx, spr_data] = user_tx_qam(spreader,data_length)	Generates random bits and modulates them using QPSK. pre_chan_tx is the output after the data has been spread and modulated. data_tx is the data sequence that was generated; and spr_data is data_tx after is has been spread by the spreader (spreading sequence).

walsh	[base_matrix,result] = walsh(factor,index)	Generates a Walsh code sequence of length 2^factor. There are 2^factor codes created. The index selects the one particular code. The index refers to the same code each run time provided the factor remains the same.
walsh_user_rx	[data_rx,rx] = walsh_user_rx(walsh_index, factor,tx,spreader)	Demodulates (BPSK) and de-spreads incoming signal using a specified Walsh code
walsh_user_tx	[pre_chan_tx, data_tx, spr_data] = walsh_user_tx(walsh_index, factor,spreader,data_length)	Generates random bits and modulates them using BPSK and a Walsh code to spread it. pre_chan_tx is the output after the data has been spread and modulated. data_tx is the data sequence that was generated; and spr_data is data_tx after is has been spread by the spreader (spreading sequence).
xor_array	result = xor_array(to_xor)	Performs XOR on all the bits in an array (to_xor). If there are an odd number of bits then the result is one, else the result is zero.
prob_dis_compare	prob_dis_compare(process_gain,users)	This function plots the theoretical error rates of the Gaussian approximation of interference limited DSSS and interference limited CDD-DSSS, where is assumed that there are no other users for the other duplexing directions.
cs_test	result = cs_test(factor,shift_slots)	This script creates orthogonal Gold codes of length 2^factor, shifts %one code and tests for orthogonality.

Table 9.2: MATLAB script functions for Coverage and Capacity Analysis

pl_limit	result = pl_limit(tx_node,rx_node)	Calculates the maximum allowable path loss between two nodes of any type. The first

		parameter is the sending node and the second parameter is the receiving node.
search	[distance] = search(node_element,ms_array)	This function returns an array of the distances from node_element to every ms units in the ms_array.
path_distance	loss = path_distance('Lee',height,pathloss)	Inverse function to the path_loss function. Used to find a given distance for a specified path loss. (Lee model only)
bs_range_plot	bs_range_plot(base,model)	Plots a radio coverage area (currently based on a simplified Lee model).
coverage coverage2 coverage3	coverage(ms_num,max_p2p_links) coverage2(ms_num,max_p2p_links) perc_incr = coverage3(ms_num,max_p2p_links)	Creates a coverage analysis by randomly placing nodes and analysing the improvement in the number of links that can be made. In coverage2, also calculates the number of local interferers. coverage 3 does not plot results and returns the percentage increase.
coverage_loop	coverage_loop(iterations)	Iterates converage3 to find statistical convergence for output.
chance	result = chance(percent)	Chance of success. A result of 1 means successful and 0 means failure
capacity3	[bs_ber_result, bs_sni_result, ms_ber_result, ms_sni_result,coverage_increase] = capacity2(ms_num,max_p2p_links,rule_level,plot_enable)	Randomly distributes nodes in an area and estimates the average SNI and BER for ms's and the bs in a single cell as well as the estimating coverage increase estimate. Input ms_num is the number of nodes to add (One BS node is always added). Input max_p2p_links is the number of peers an ms can accommodate Input rule_level is the rule

		to use. Input plot_enable will trigger the plot if set to 1, and will ignore plot if set to 0.

Appendix III: CDD Performance Results

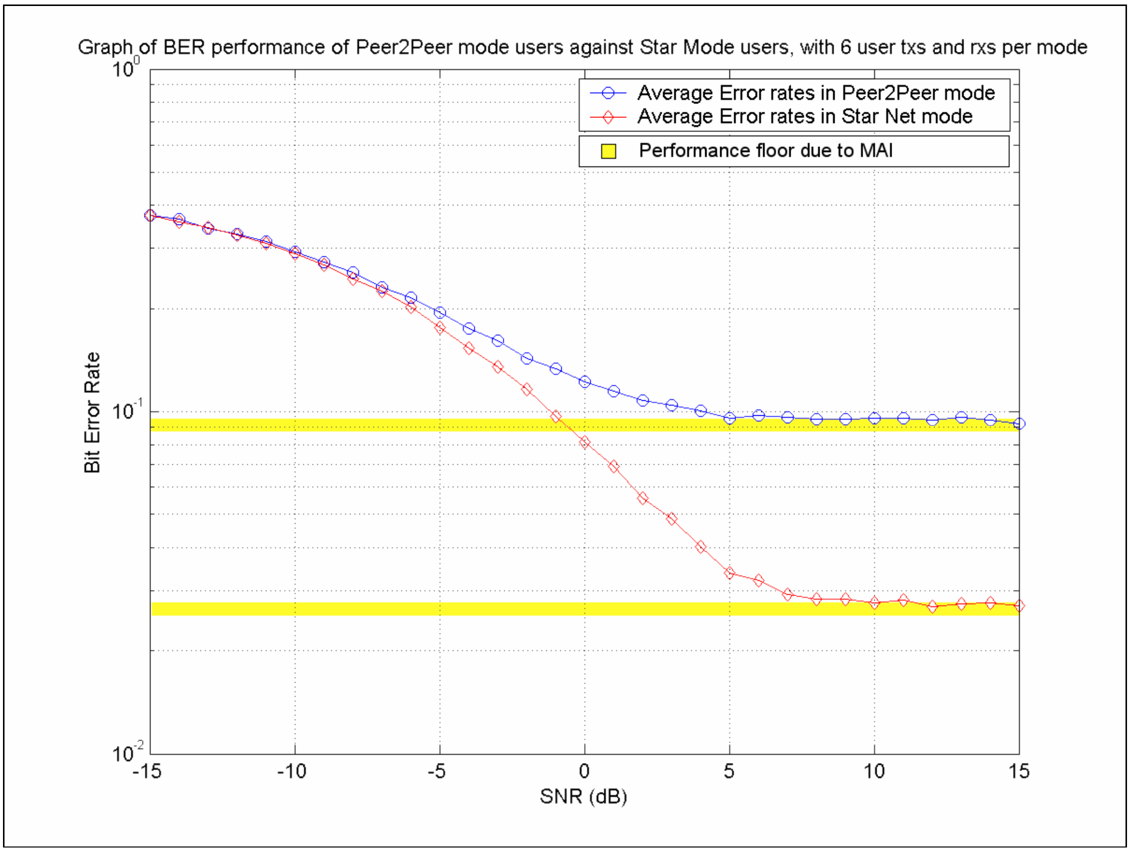

Figure 9.1: BER performance of 6 CDD users against 6 standard users with code length 16

APPENDIX IV: VERIFICATION OF BPSK FUNCTION

The BPSK signal is the generated from the binary data (generated randomly) and can be represented as followed:

$$\phi(t) = A\cos[\omega_c t + \Delta\theta p(t)] \tag{9.1}$$

Seeing that the additive white Gaussian noise that affects the can be represented as *n(t)*, it therefore follows that the received signal at the correlation detector is:

$$\phi_r(t) = \phi(t) + n(t) \tag{9.2}$$

The theoretical error rates for a BPSK signal (received with a conventional correlation detector) are given by the following functions:

$$2Q\left(\sqrt{\frac{2E_s}{\eta}}\right) \tag{9.3}$$

Or

$$\frac{1}{2}erfc\left(\frac{E_s}{\eta}\right) \tag{9.4}$$

The source of the BPSK function is from the MATLAB communication toolbox. Nonetheless, it is imperative that the function is still verified. The test was run and the results of the test were as followed:

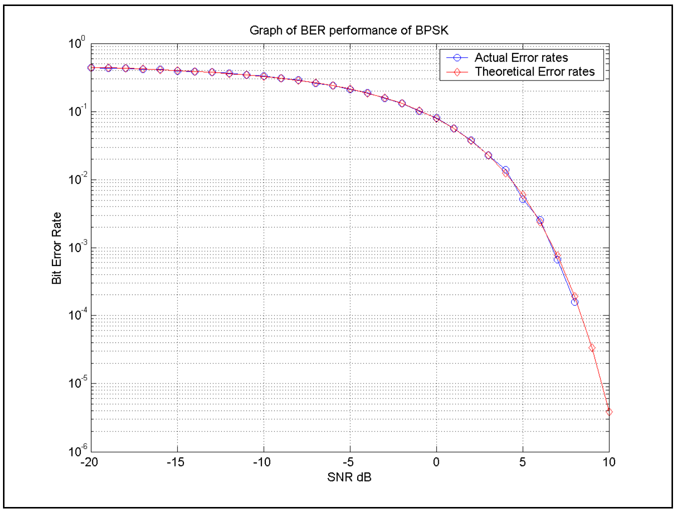

Figure 9.2: Plot of BPSK performance against theoretical error rates

As shown in Figure 9.2, the function performance of the function closely resembles the plot of the theoretical error rates. Based on these results, a simple DSSS system could be built to test channel loading.

As expected, the results (Figure 9.3) of the test setup showed that, as the number of user increase the error rates also increased. This is primarily due to the multiple access interference.

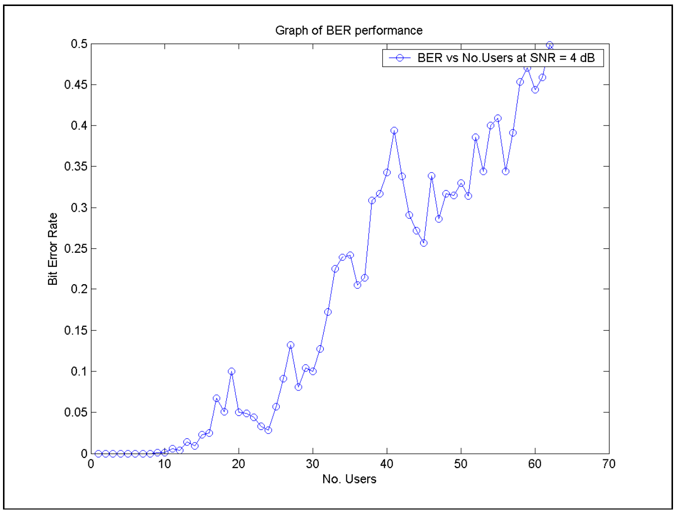

Figure 9.3: DSSS Channel loading test results

APPENDIX V: CONTENTS OF COMPACT DISC

The attached compact disc contains the electronic copy of this document along with the MATLAB script needed to run the simulation.

The requirements for the MATLAB simulation are as follows:

- MATLAB 6.5 R13
- Communication Toolbox

The compact disc associated with this project contains the following directory structure:

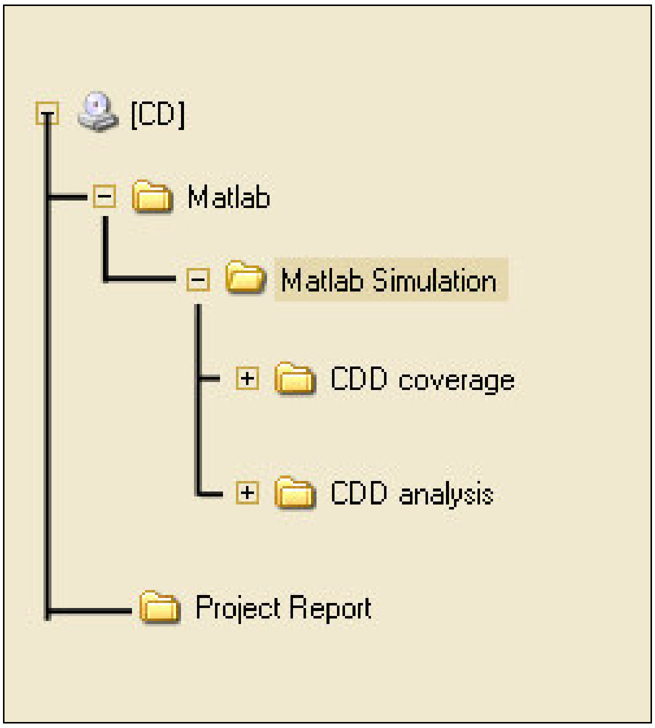

Figure 9.4: CD Contents