The suitability of OFDM as a modulation technique for wireless telecommunications, with a CDMA comparison.

Thesis submitted by

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in partial fulfilment of the requirements for the Degree of Bachelor of Engineering with Honours in Computer Systems Engineering at James Cook University.

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Finally, I would like to thank my computer for only crashing seriously once during the writing of this thesis.

Declaration

I declare that this thesis is my own work and has not been submitted in any form for another degree or diploma at any university or other institution for tertiary education. Information derived from published or unpublished work of others has been acknowledged in the text and a list of references is given.

Eric Lawrey
20th October 1997

Abstract

This thesis investigates the effectiveness of Orthogonal Frequency Division Multiplexing (OFDM) as a modulation technique for wireless radio applications. The main aim was to assess the suitability of OFDM as a modulation technique for a fixed wireless phone system for rural areas of Australia. However, its suitability for more general wireless applications is also assessed.

Several of the main factors affecting the performance of a OFDM system were measured, including multipath delay spread, channel noise, distortion (clipping), and timing requirements. The performance of OFDM was assessed by using computer simulations performed using Matlab, and practical measurements. These measurements were performed by recording a low bandwidth (audio) OFDM signal, generated using Matlab, on to a tape player. This recorded signal was then played back and recorded using the sound card of a PC. This was then decoded using a Matlab script.

Most third generation mobile phone systems are proposing to use Code Division Multiple Access (CDMA) as their modulation technique. For this reason, CDMA was also investigated so that the performance of CDMA could be compared with OFDM.

It was found that OFDM performs extremely well compared with CDMA, providing a very high tolerance to multipath delay spread, peak power clipping, and channel noise. In addition to this it provides a high spectral efficiency.

OFDM was found to have total immunity to multipath delay spread provided the reflection time is less than the guard period used in the OFDM signal. In fact, multipath signals lead to a strengthening of the received signal, improving the performance. In a typical system a delay spread of up to 100 μsec could be tolerated, corresponding to multipath reflections of 30 km. The only problem caused by multipath is frequency selective fading, which can result in carriers being heavily

attenuated due to destructive interference at the receiver. This can result in the carriers being lost in the noise.

For the modulation schemes investigated (BPSK, QPSK and 16 QAM), clipping of the OFDM signal was found to have little effect on the performance of the system, allowing the peak power of the signal to be clipped by up to 6 - 9dB before the symbol error rate became significant. This tolerance to clipping reduces the dynamic range overhead required in output stages of OFDM transmitters.

The noise performance of OFDM was found to depend solely on the modulation technique used for modulating each carrier of the signal. The performance of the OFDM signal was found to be the same as for a single carrier system, using the same modulation technique. The minimum signal to noise ratio (SNR) required for BPSK was ~7 dB, where as it was ~12 dB for QPSK and ~25 dB for 16PSK.

CDMA was found to perform poorly in a single cellular system, with each cell only allowing 7-16 simultaneous users in a cell, compared with 128 for OFDM. This was for a 1.25 MHz bandwidth and 19.5 kbps user data rate. This low cell capacity of CDMA was attributed to the use of non-orthogonal codes used in the reverse transmission link, leading to a high level of inter-user interference.

The only main weak point that was found with using OFDM, was that it is very sensitive to frequency, and phase errors between the transmitter and receiver. The main sources of these errors are frequency stability problems; phase noise of the transmitter; and any frequency offset errors between the transmitter and receiver. This problem can be mostly overcome by synchronizing the clocks between the transmitter and receiver, by designing the system appropriately, or by reducing the number of carriers used.

Table of Contents

Table of Figures

List of Tables

1. Introduction

The telecommunications industry faces the problem of providing telephone services to rural areas, where the customer base is small, but the cost of installing a wired phone network is very high. One method of reducing the high infrastructure cost of a wired system is to use a fixed wireless radio network. The problem with this is that for rural and urban areas, large cell sizes are required to obtain sufficient coverage. This results in problems cased by large signal path loss and long delay times in multipath signal propagation.

Currently Global System for Mobile telecommunications (GSM) technology is being applied to fixed wireless phone systems in rural areas or Australia. However, GSM uses Time Division Multiple Access (TDMA), which has a high symbol rate leading to problems with multipath causing inter-symbol interference.

Several techniques are under consideration for the next generation of digital phone systems, with the aim of improving cell capacity, multipath immunity, and flexibility. These include Code Division Multiple Access (CDMA) and Coded Orthogonal Frequency Division Multiplexing (COFDM). Both these techniques could be applied to providing a fixed wireless system for rural areas. However, each technique has different properties, making it more suited for specific applications.

COFDM is currently being used in several new radio broadcast systems including the proposal for high definition digital television, Digital Video Broadcasting (DVB) and Digital Audio Broadcasting (DAB). However, little research has been done into the use of COFDM as a transmission method for mobile telecommunications systems.

With CDMA systems, all users transmit in the same frequency band using specialized codes as a basis of channelization. The transmitted information is spread in bandwidth by multiplying it by a wide bandwidth pseudo random sequence. Both the base station and the mobile station know these random codes that are used to modulate the data sent, allowing it to de-scramble the received signal.

OFDM/COFDM allows many users to transmit in an allocated band, by subdividing the available bandwidth into many narrow bandwidth carriers. Each user is allocated several carriers in which to transmit their data. The transmission is generated in such a way that the carriers used are orthogonal to one another, thus allowing them to be packed together much closer than standard frequency division multiplexing (FDM). This leads to OFDM/COFDM providing a high spectral efficiency.

1.1 Third Generation Wireless Networks

The expansion of the use of digital networks has led to the need for the design of new higher capacity communications networks. The demand for cellular-type systems in Europe is predicted to be between 15 and 20 million users by the year 2000 [1], and is already over 30 million (1995) in the U.S. [2]. Wireless services have been growing at a rate greater than 50% per year [2], with the current second-generation European digital systems (GSM) being expected to be filled to capacity by the early 2000's[3]. The telecommunications industry is also changing, with a demand for a greater range of services such as video conferencing, Internet services, and data networks, and multimedia. This demand for higher capacity networks has led to the development of third generation telecommunications systems.

One of the proposed third generation telecommunications systems is the Universal Mobile Telecommunications System (UMTS), which aims to provide a more flexible data rate, a higher capacity, and a more tightly integrated service, than current second generation mobile systems. This section focuses on the services and aims of the UMTS. Other systems around the world are being developed, however many of these technologies are expected to be eventually combined into the UMTS.

The World Wide Web (WWW) has become an important communications media, as its use has increased dramatically over the last few years. This has resulted in an increased demand for computer networking services. In order to satisfy this, telecommunications systems are now being used for computer networking, Internet access and voice communications. A WWW survey revealed that more then 60% of users access the Internet from residential locations [10], where the bandwidth is often limited to 28.8kbps [8]. This restricts the use of the Internet, preventing the use of real time audio and video capabilities. Higher speed services are available, such as integrated-services digital network (ISDN). These provide data rates up to five times as fast, but at a much-increased access cost. This has led to the demand of a more integrated service, providing faster data rates, and a more universal interface for a variety of services. The emphasis has shifted away from providing a fixed voice service to providing a general data connection that allows for a wide variety of applications, such as voice, Internet access, computer networking, etc.

The increased reliance on computer networking and the Internet has resulted in demand for connectivity to be provided "any where, any time", leading to an increase in the demand for wireless systems. This demand has driven the need to develop new higher capacity, high reliability wireless telecommunications systems.

The development and deployment of third generation telecommunication systems aim to overcome some of the downfalls of current wireless systems by providing a high capacity, integrated wireless network. There are currently several third generation wireless standards, including UMTS, cdmaOne, IMT 2000, and IS-95 [10].

1.1.1 Evolution of Telecommunication Systems.

Many mobile radio standard have been developed for wireless systems throughout the world, with more standards likely to emerge.

Most first generations systems were introduced in the mid 1980's, and can be characterized by the use of analog transmission techniques, and the use of simple multiple access techniques such as Frequency Division Multiple Access (FDMA). First generation telecommunications systems such as Advanced Mobile Phone Service (AMPS) [4] only provided voice communications. They also suffered from a low user capacity, and security problems due to the simple radio interface used.

Second generation systems were introduced in the early 1990's, and all use digital technology. This provided an increase in the user capacity of around three times [6]. This was achieved by compressing the voice waveforms before transmission [7].

Third generation systems are an extension on the complexity of second-generation systems and will begin roll out of services sometime after the year 2001. The capacity of third generation systems is expected to be over ten times original first generation systems. This is going to be achieved by using complex multiple access techniques such as CDMA, or an extension of TDMA, and by improving flexibility of services available.

Table 1 and Table 2 show some of the major cellular mobile phone standards in North America and Europe.

Cellular System	**Year of Introduction**	**Transmission Type**	**Multiple Access Technique**	**Channel Bandwidth**	**System Generation**
Advanced Mobile Phone System (AMPS)	1983	Analog	FDMA	30kHz	First
Narrowband AMPS (NAMPS)	1992	Analog	FDMA	10kHz	First
U.S. Digital Cellular (USDC)	1991	Digital	TDMA	30kHz	Second
U.S Narrowband Spread Spectrum (IS-95)	1993	Digital	CDMA	1.25MHz	Second
Wideband cdmaOne	>2001	Digital	CDMA	-	Third

Table 1 Major Mobile Standards in North America[6]

Cellular System	Year of Introduction	Transmission Type	Multiple Access Technique	Channel Bandwidth	System Generation
E-TACS	1985	Analog	FDMA	25kHz	First
NMT-900	1986	Analog	FDMA	12.5kHz	First
Global System for Mobile (GSM)	1990	Digital	TDMA	200kHz	Second
Universal Mobile Tele-communications System (UMTS)	>2001	Digital	CDMA/ TDMA	-	Third

Table 2 Major Mobile Standards in Europe[6]

Figure 1 shows the evolution of current services and networks to the aim of combining them into a unified third generation network. Many currently separate systems and services such as radio paging, cordless telephony, satellite phones, private radio systems for companies etc, will be combined so that all these services will be provided by third generation telecommunications systems.

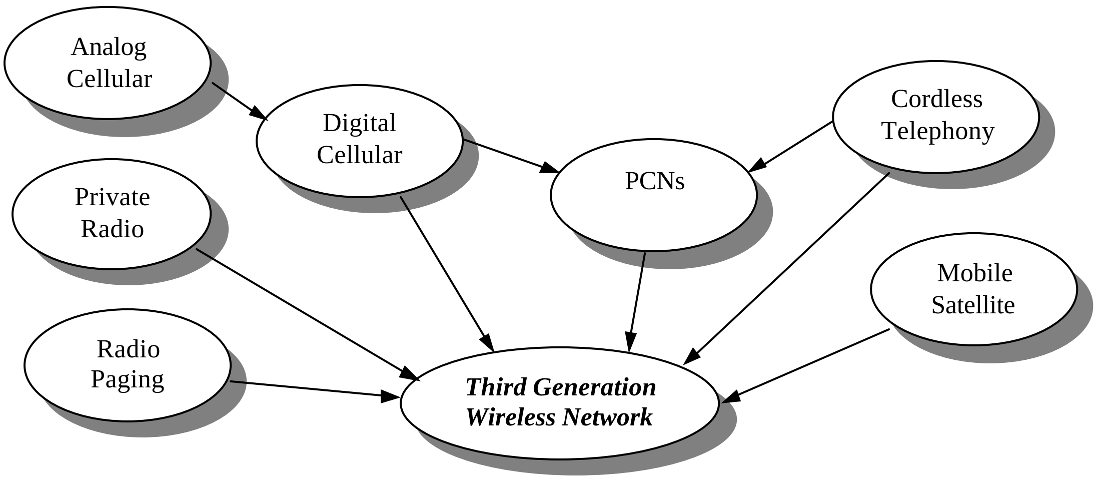


Figure 1 Evolution of current networks to the next generation of wireless networks (reproduced from [1])

1.1.2 Overall Aims of Universal Mobile Telecommunications System

The main aims of the Universal Mobile Telecommunications System is to provide a more unified high capacity network, in wireless and wired environments. UMTS will enable fixed and wireless services to converge. There are to be three main channel capacity connections: a mobile rate of 144kbps; a portable rate of 384kbps and an in-building rate of 2Mbps [10]. It will have the capacity to provide services and features requiring less then 2Mbps that would otherwise have been provided with a fixed network. UMTS must therefore provide on-demand, variable bandwidth allocation. It will also combine a range of applications including cordless phones, cellular phones, and mobile data networking for personal, business and residential use.

1.1.3 Tele-services

Many services have been identified for the UMTS, which can be categorized based on the data rate required, quality of service (reliability and allowable bit error rate (BER)), real time transfer rate. Each of the services has different characteristics in terms of delay tolerance and allowable bit error rates. Table 3 shows characteristics for some of the UMTS services.

Applications or Services	Data Rate Required	Quality of service required	Time critical data
Messaging (email, etc)	Low (1-10kbps)	High	No
Voice	Low (4-20kpbs)	Low (BER < 1e-3)	Yes
Web browsing	As high as possible (>10kbps-100kbps)	High (BER < 1e-9)	Depends on material. Generally not time critical.
Videoconferencing	High (100kbps-1Mbps)	Medium	Yes
Video Surveillance	Medium (50-300kbps)	Medium	No
High Quality Audio	High (100-300kbps)	Medium	Yes
Database access	High (>30kbps)	Very High	No

Table 3 UMTS Services, showing the data characteristics of each service

The data characteristics will determine the most suitable transmission methods. The type of data associated with each service determines the type of environment in which the service can be supported.

1.1.4 UMTS Environments

The aim of the UMTS systems is to provide an "any where, any time" service, thus the operating environment will vary depending on the user location. The environment in which the wireless system must operate affects the system capacity and type of services that can be provided. Table 4 lists some of the environments in which UMTS will be required to provide coverage.

Environment	Maximum supported Data Rate
Business (indoor)	384kbps
Suburban (indoor/outdoor)	144kbps
Urban vehicular (outdoor)	144kbps
Urban pedestrian (outdoor)	144kbps
Fixed (Outdoor)	144kbps / 384kbps
Local high bit rate (Indoor)	2Mbps

Table 4 Maximum supported data rates for UMTS, for various environments.

The maximum supported data rate for each environment is related to the cell size required to provide adequate coverage for the environment.

1.1.5 Cell types

A cellular network is required to ensure the UMTS can provide a high capacity network. As with any cellular system, the total capacity of the network is dependent on the size of the cells used. The smaller the cells are made, the larger the total capacity. However, the cell size is limited by the amount of infrastructure. The cell size also determines the maximum channel capacity for each cell, as propagation effects, such as multipath delay spread and high path loss, force large cells to have a lower data rate. Large cells also have to service a large number of users, and since the cell capacity is approximately fixed, each user can only have a reduced data rate, with respect to a smaller cell. In order to optimise the cellular network three cell types are used. These are the pico-cell, micro cell, and macro-cell. The three different cell types trade off cell size will total capacity and services. Table 5 shows the three cell types used in the UMTS system and some of the cell characteristics.

	Pico-cell	**Micro-cell**	**Macro-cell**
Cell radius	<100m	<1000m	<20km
Antenna	Ceiling/wall mounted	Below roof top height	Roof top mounting
Max. multipath delay spread	1usec	5usec	20usec
Applications and environments	Indoor/Outdoor Within buildings City centres Local high bit rate	High density outdoor Business (indoor) Fixed (Outdoor) Inner city areas	Low density areas Suburban areas Urban areas Fixed (outdoor)
Services and data rate supported	All services (up to 2Mbps)	Up to 384kbps	Limited sub-set (up to 144kpbs)

Table 5 Cell Types used in UTMS

The size and type of coverage of each cell type effects the radio propagation problems that will be encountered. This will determine the most suitable radio transmission technique to use.

1.1.6 Radio Interface

One of the aims identified for UMTS is to provide a wireless interface comparable to wired connections. The requirement to provide wide band services up to 2Mb/s, with flexible, on demand allocation of transmission capacity in a large range of radio environments, will call for a revolution in the radio access techniques used.

The radio interface is currently undergoing substantial research, with the relative performance of CDMA and TDMA being investigated [9]. Currently CDMA appears to be the most likely candidate for supporting the high data rate required. However, other techniques such as COFDM and hybrid solutions may also be appropriate for UMTS.

1.1.7 Satellite Networking

One of the aims of the UMTS is to provide access “any where, any time”. However, cellular networks can only cover a limited area due to the high infrastructure costs. For this reason, satellite systems will form an integral part of the UMTS network. Satellites will be able to provide an extended wireless coverage to remote areas and to aeronautical and maritime mobiles. The level of integration of the satellite systems with the terrestrial cellular networks is under investigation. A fully integrated solution will require mobiles to be dual mode terminals that would allow communications with orbiting satellites and terrestrial cellular networks. Low Earth Orbit (LEO) satellites are the most likely candidates for providing worldwide coverage.

Currently several low earth orbit satellite systems are being deployed for providing global telecommunications. These include the Teledesic System, which is scheduled to begin operation by the end of 2002 with 288 satellites [10], to provide high bandwidth two-way communications to virtually anywhere in the world. (Addendum, 10/2001: Current estimates put the release date of the Teledesic System sometime in 2005, see www.teledesic.com). However, the Teledesic System will not be able to meet even 20% of the demand [10], thus the need for broadband wireless networks.

1.1.8 Timetable for System Implementation

Across the globe, each region is moving to make third generation systems happen. Japan is looking at having a system up and running by year 2001. This is driven by the very high demand for mobile communications, which has been so great that their second-generation cellular networks are starting to run out of capacity [10]. It is expected that Europe will have a wide band CDMA system by the year 2005 [10]. The

U.S. is expected to implement a third generation system somewhere from 2000 to 2010 [10].

Manufacturers are creating several standards to meet requirements in each sector of the world. To date, the majority of systems are based on CDMA standards. Before infrastructure rolls out, third generations will be developed on a regional basis.

This process is being guided by the International Telecommunications Union's (ITU) effort to create the IMT 2000 standard. ITU will produce the IMT 2000 standard by the year 2000, with the aim of combining the regional systems into a unified standard [10].

1.1.9 Conclusion

Future communications will be driven by the need to provide a more integrated high capacity, wide coverage service. For the 21st century user there should ideally be no distinction in service capability between mobile or fixed network access. This will be achieved using a variety of technologies including satellite communications, advanced radio networking techniques, and high speed fixed networks.

1.2 Propagation Characteristics of mobile radio channels

In an ideal radio channel, the received signal would consist of only a single direct path signal, which would be a perfect reconstruction of the transmitted signal. However in a real channel, the signal is modified during transmission in the channel. The received signal consists of a combination of attenuated, reflected, refracted, and diffracted replicas of the transmitted signal. On top of all this, the channel adds noise to the signal and can cause a shift in the carrier frequency if the transmitter, or receiver is moving (Doppler effect). Understanding of these effects on the signal is important because the performance of a radio system is dependent on the radio channel characteristics.

1.2.1 Attenuation

Attenuation is the drop in the signal power when transmitting from one point to another. It can be caused by the transmission path length, obstructions in the signal path, and multipath effects. Figure 2 shows some of the radio propagation effects that cause attenuation. Any objects that obstruct the line of sight signal from the transmitter to the receiver can cause attenuation.

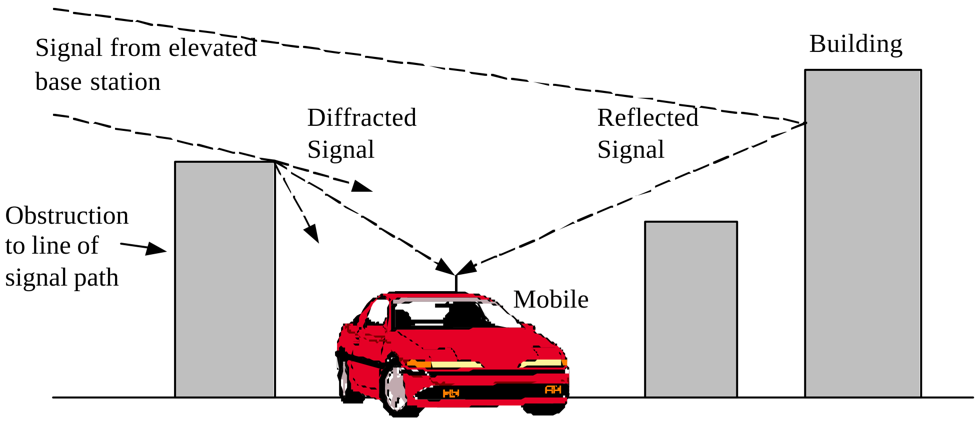


Figure 2 Radio Propagation Effects

Shadowing of the signal can occur whenever there is an obstruction between the transmitter and receiver. It is generally caused by buildings and hills, and is the most important environmental attenuation factor.

Shadowing is most severe in heavily built up areas, due to the shadowing from buildings. However, hills can cause a large problem due to the large shadow they produce. Radio signals diffract off the boundaries of obstructions, thus preventing total shadowing of the signals behind hills and buildings. However, the amount of diffraction is dependent on the radio frequency used, with low frequencies diffracting more then high frequency signals. Thus high frequency signals, especially, Ultra High Frequencies (UHF), and microwave signals require line of sight for adequate signal strength. To over come the problem of shadowing, transmitters are usually elevated as high as possible to minimise the number of obstructions. Typical amounts of variation in attenuation due to shadowing are shown in Table 6.

Description	Typical Attenuation due to Shadowing
Heavily built-up urban centre	20dB variation from street to street
Sub-urban area (fewer large buildings)	10dB greater signal power then built-up urban centre
Open rural area	20dB greater signal power then sub-urban areas
Terrain irregularities and tree foliage	3-12dB signal power variation

Table 6 Typical shadowing in a radio channel (Values from [11])

Shadowed areas tend to be large, resulting in the rate of change of the signal power being slow. For this reason, it is termed *slow-fading*, or *log-normal shadowing*.

1.2.2 Multipath Effects

1.2.2.1 Rayleigh fading

In a radio link, the RF signal from the transmitter may be reflected from objects such as hills, buildings, or vehicles. This gives rise to multiple transmission paths at the receiver. Figure 3 show some of the possible ways in which multipath signals can occur.

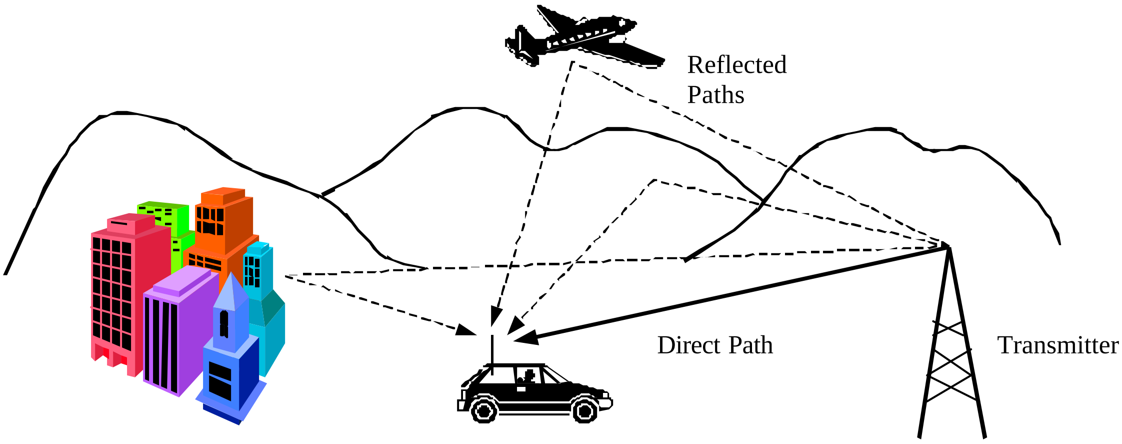


Figure 3 Multipath Signals

The relative phase of multiple reflected signals can cause constructive or destructive interference at the receiver. This is experienced over very short distances (typically at half wavelength distances), thus is given the term *fast fading*. These variations can vary from 10-30dB over a short distance. Figure 4 shows the level of attenuation that can occur due to the fading.

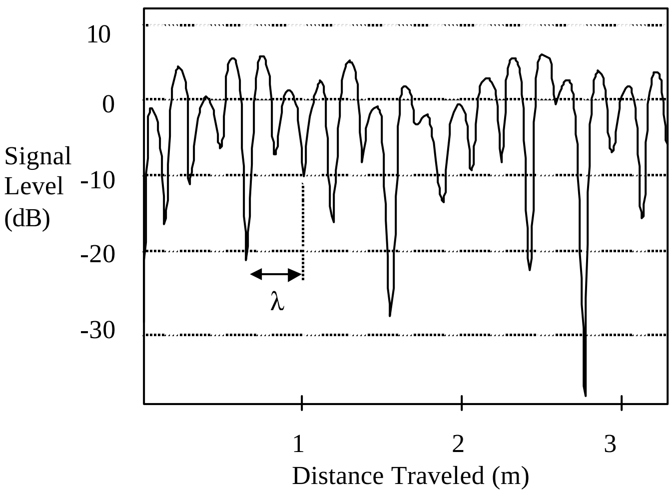


Figure 4 Typical Rayleigh fading while the Mobile Unit is moving (for at 900 MHz)[15]

The Rayleigh distribution is commonly used to describe the statistical time varying nature of the received signal power. It describes the probability of the signal level being received due to fading. Table 7 shows the probability of the signal level for the Rayleigh distribution.

Signal Level (dB about median)	% Probability of Signal Level being less then the value given
10	99
0	50
-10	5
-20	0.5
-30	0.05

Table 7 Cumulative distribution for Rayleigh distribution (Value from [15])

1.2.2.2 Frequency Selective Fading

In any radio transmission, the channel spectral response is not flat. It has dips or fades in the response due to reflections causing cancellation of certain frequencies at the receiver. Reflections off near-by objects (e.g. ground, buildings, trees, etc) can lead to multipath signals of similar signal power as the direct signal. This can result in deep nulls in the received signal power due to destructive interference.

For narrow bandwidth transmissions if the null in the frequency response occurs at the transmission frequency then the entire signal can be lost. This can be partly overcome in two ways.

By transmitting a wide bandwidth signal or spread spectrum as CDMA, any dips in the spectrum only result in a small loss of signal power, rather than a complete loss. Another method is to split the transmission up into many small bandwidth carriers, as is done in a COFDM/OFDM transmission. The original signal is spread over a wide bandwidth and so nulls in the spectrum are likely to only affect a small number of carriers rather than the entire signal. The information in the lost carriers can be recovered by using forward error correction techniques.

1.2.2.3 Delay Spread

The received radio signal from a transmitter consists of typically a direct signal, plus reflections off objects such as buildings, mountings, and other structures. The reflected signals arrive at a later time then the direct signal because of the extra path length, giving rise to a slightly different arrival times, spreading the received energy in time. Delay spread is the time spread between the arrival of the first and last significant multipath signal seen by the receiver.

In a digital system, the delay spread can lead to inter-symbol interference. This is due to the delayed multipath signal overlapping with the following symbols. This can cause significant errors in high bit rate systems, especially when using time division multiplexing (TDMA). Figure 5 shows the effect of inter-symbol interference due to delay spread on the received signal. As the transmitted bit rate is increased the amount of inter-symbol interference also increases. The effect starts to become very significant when the delay spread is greater then ~50% of the bit time.

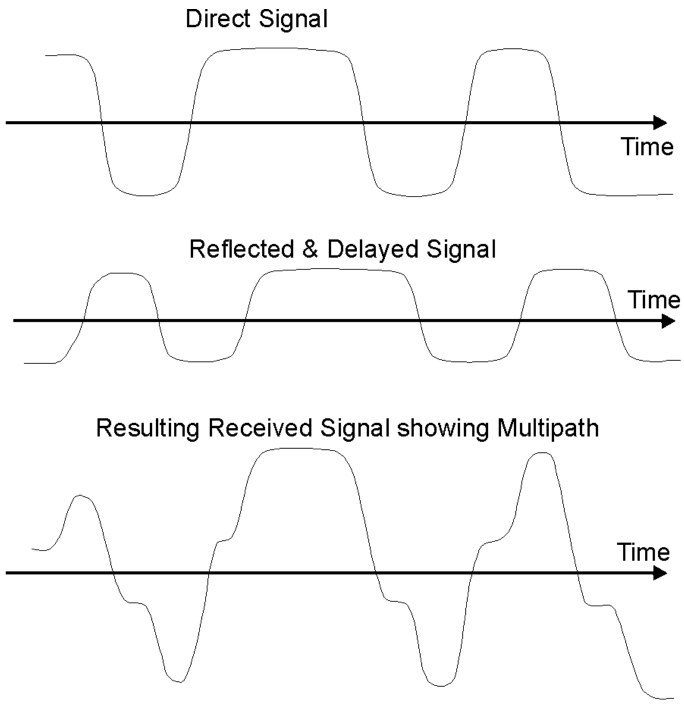


Figure 5 Multipath Delay Spread

Table 8 shows the typical delay spread for various environments. The maximum delay spread in an outdoor environment is approximately 20μs, thus significant inter-symbol interference can occur at bit rates as low as 25 kbps.

Environment or cause	Delay Spread	Maximum Path Length Difference
Indoor (room)	40ns – 200 ns	12 m – 60 m
Outdoor	1 μs – 20 μs	300 m – 6 km

Table 8 Typical Delay Spread

Inter-symbol interference can be minimized in several ways. One method is to reduce the symbol rate by reducing the data rate for each channel (i.e. split the bandwidth into more channels using frequency division multiplexing, or OFDM). Another is to use a coding scheme that is tolerant to inter-symbol interference such as CDMA.

1.2.3 Doppler Shift

When a wave source and a receiver are moving relative to one another the frequency of the received signal will not be the same as the source. When they are moving toward each other the frequency of the received signal is higher then the source, and when they are approaching each other the frequency decreases. This is called the Doppler effect. An example of this is the change of pitch in a car's horn as it approaches then passes by. This effect becomes important when developing mobile radio systems.

The amount the frequency changes due to the Doppler effect depends on the relative motion between the source and receiver and on the speed of propagation of the wave. The Doppler shift in frequency can be written:

$$\Delta f \approx \pm f_o \frac{v}{c}$$

(from [12])

where *Df* is the change in frequency of the source seen at the receiver, f_o is the frequency of the source, v is the speed difference between the source and transmitter, and c is the speed of light.

For example: Let f_o = 1GHz, and v = 60km/hr (16.7m/s) then the Doppler shift will be:

$$f_o = 10^9 \cdot \frac{16.67}{3 \times 10^8} = 55.5Hz$$

This shift of 55Hz in the carrier will generally not effect the transmission. However, Doppler shift can cause significant problems if the transmission technique is sensitive to carrier frequency offsets (for example OFDM) or the relative speed is very high as is the case for low earth orbiting satellites.

1.3 Multiple Access Techniques

Multiple access schemes are used to allow many simultaneous users to use the same fixed bandwidth radio spectrum. In any radio system, the bandwidth that is allocated to it is always limited. For mobile phone systems the total bandwidth is typically 50 MHz, which is split in half to provide the forward and reverse links of the system. Sharing of the spectrum is required in order increase the user capacity of any wireless network. FDMA, TDMA and CDMA are the three major methods of sharing the available bandwidth to multiple users in wireless system. There are many extensions, and hybrid techniques for these methods, such as OFDM, and hybrid TDMA and FDMA systems. However, an understanding of the three major methods is required for understanding of any extensions to these methods.

1.3.1 Frequency Division Multiple Access

For systems using Frequency Division Multiple Access (FDMA), the available bandwidth is subdivided into a number of narrower band channels. Each user is allocated a unique frequency band in which to transmit and receive on. During a call, no other user can use the same frequency band. Each user is allocated a forward link channel (from the base station to the mobile phone) and a reverse channel (back to the base station), each being a single way link. The transmitted signal on each of the channels is continuous allowing analog transmissions. The channel bandwidth used in most FDMA systems is typically low (30kHz) as each channel only needs to support a single user. FDMA is used as the primary subdivision of large allocated frequency bands and is used as part of most multi-channel systems.

Figure 6 and Figure 7 shows the allocation of the available bandwidth into several channels.

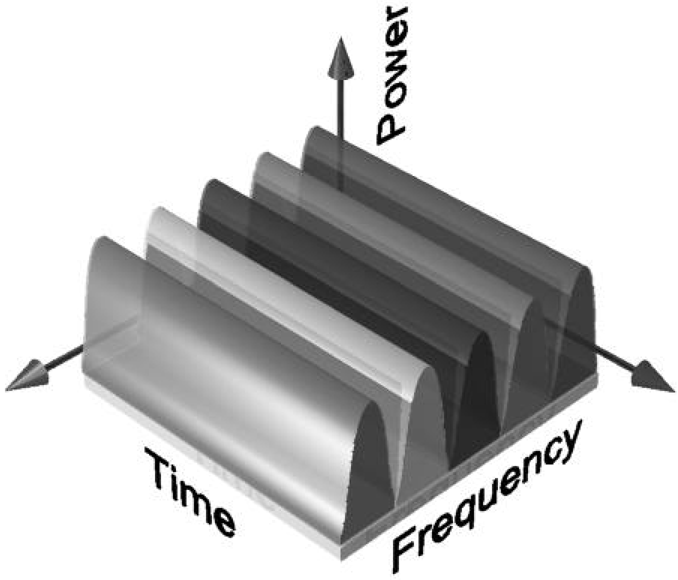


Figure 6 FDMA showing that the each narrow band channel is allocated to a single user

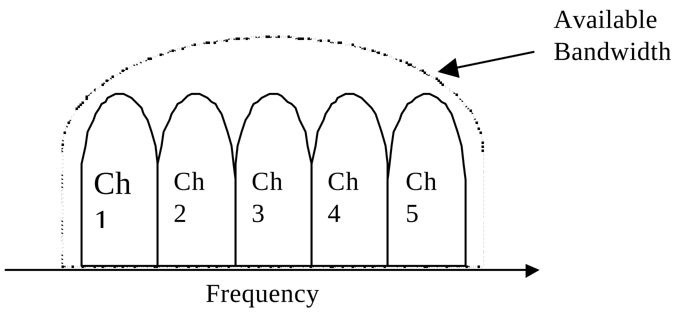


Figure 7 FDMA spectrum, where the available bandwidth is subdivided into narrower band channels

1.3.2 Time Division Multiple Access

Time Division Multiple Access (TDMA) divides the available spectrum into multiple time slots, by giving each user a time slot in which they can transmit or receive. Figure 8 shows how the time slots are provided to users in a round robin fashion, with each user being allotted one time slot per frame.

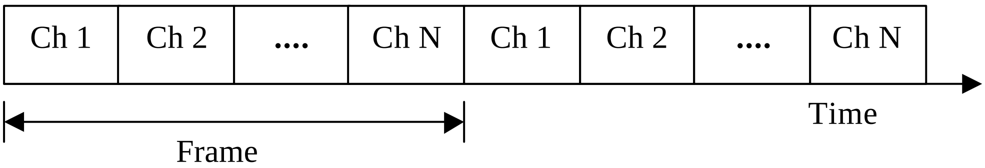


Figure 8 TDMA scheme where each user is allocated a small time slot

TDMA systems transmit data in a buffer and burst method, thus the transmission of each channel is non-continuous. The input data to be transmitted is buffered over the previous frame and burst transmitted at a higher rate during the time slot for the channel. TDMA can not send analog signals directly due to the buffering required, thus is only used for transmitting digital data. TDMA can suffer from multipath effects as the transmission rate is generally very high, resulting in significant inter-symbol interference.

TDMA is normally used in conjunction with FDMA to subdivide the total available bandwidth into several channels. This is done to reduce the number of users per channel allowing a lower data rate to be used. This helps reduce the effect of delay spread on the transmission. Figure 9 shows the use of TDMA with FDMA. Each channel based on FDMA, is further subdivided using TDMA, so that several users can transmit of the one channel. This type of transmission technique is used by most digital second generation mobile phone systems. For GSM, the total allocated bandwidth of 25MHz is divided into 125, 200kHz channels using FDMA. These channels are then subdivided further by using TDMA so that each 200kHz channel allows 8-16 users [13].

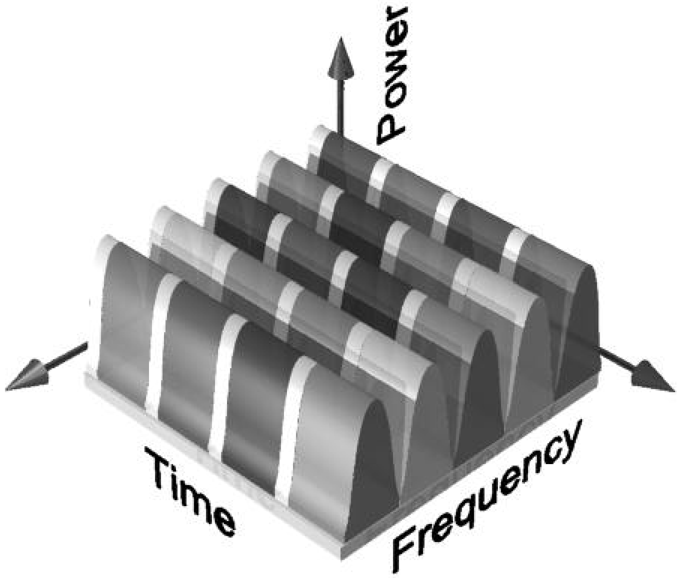


Figure 9 TDMA / FDMA hybrid, showing that the bandwidth is split into frequency channels and time slots

1.3.3 Code Division Multiple Access

Code Division Multiple Access (CDMA) is a spread spectrum technique that uses neither frequency channels nor time slots. With CDMA, the narrow band message (typically digitised voice data) is multiplied by a large bandwidth signal that is a pseudo random noise code (PN code). All users in a CDMA system use the same frequency band and transmit simultaneously. The transmitted signal is recovered by correlating the received signal with the PN code used by the transmitter. Figure 10 shows the general use of the spectrum using CDMA

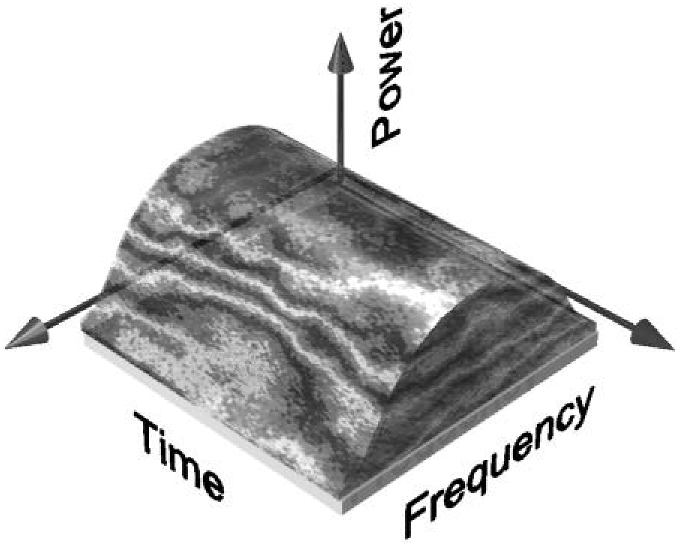


Figure 10 Code division multiple access (CDMA)

CDMA technology was originally developed by the military during World War II [14]. Researchers were spurred into looking at ways of communicating that would be secure and work in the presence of jamming. Some of the properties that have made CDMA useful are:

- Signal hiding and non-interference with existing systems.
- Anti-jam and interference rejection
- Information security
- Accurate Ranging
- Multiple User Access
- Multipath tolerance

For many years, spread spectrum technology was considered solely for military applications. However, with rapid developments in LSI and VLSI designs, commercial systems are starting to be used.

1.3.4 CDMA Process Gain

One of the most important concepts required in order to understand spread spectrum techniques is the idea of process gain. The process gain of a system indicates the gain or signal to noise improvement exhibited by a spread spectrum system by the nature of the spreading and despreading process. The process gain of a system is equal to the ratio of the spread spectrum bandwidth used, to the original information bandwidth. Thus, the process gain can be written as:

$$Gp = \frac{BW_{RF}}{BW_{\inf o}}$$

Where BW_{RF} is the transmitted bandwidth after the data is spread, and BW_{info} is the bandwidth of the information data being sent.

Figure 11 shows the process of a CDMA transmission. The data to be transmitted (a) is spread before transmission by modulating the data using a PN code. This broadens the spectrum as shown in (b). In this example the process gain is 125 as the spread spectrum bandwidth is 125 times greater the data bandwidth. Part (c) shows the received signal. This consists of the required signal, plus background noise, and any interference from other CDMA users or radio sources. The received signal is recovered by multiplying the signal by the original spreading code. This process causes the wanted received signal to be despread back to the original transmitted data. However, all other signals that are uncorrelated to the PN spreading code become more spread. The wanted signal in (d) is then filtered removing the wide spread interference and noise signals.

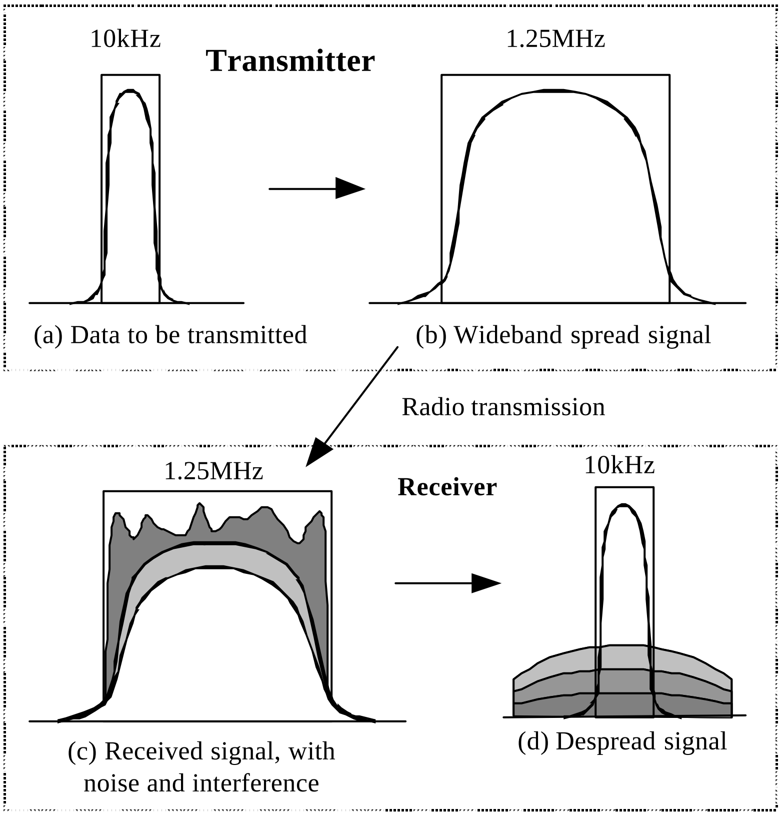


Figure 11 Basic CDMA transmission.

1.3.5 CDMA Generation

CDMA is achieved by modulating the data signal by a pseudo random noise sequence (PN code), which has a chip rate higher then the bit rate of the data. The PN code sequence is a sequence of ones and zeros (called chips), which alternate in a random fashion. Modulating the data with this PN sequence generates the CDMA signal. The CDMA signal is generated by modulating the data by the PN sequence. The modulation is performed by multiplying the data (XOR operator for binary signals) with the PN sequence. Figure 12 shows a basic CDMA transmitter.

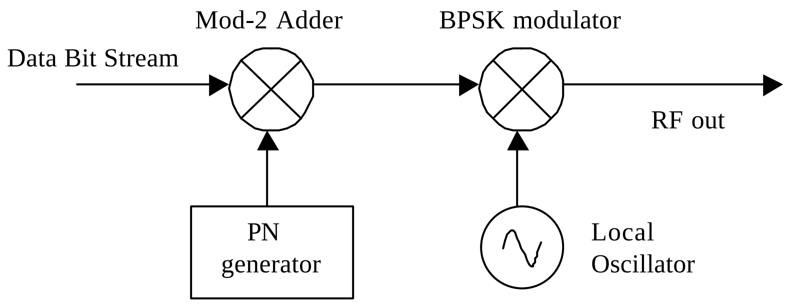


Figure 12 Simple direct sequence modulator

The PN code used to spread the data can be of two main types. A short PN code (typically 10-128 chips in length) can be used to modulate each data bit. The short PN code is then repeated for every data bit allowing for quick and simple synchronization of the receiver. Figure 13 shows the generation of a CDMA signal using a 10-chip length short code. Alternatively a long PN code can be used. Long codes are generally thousands to millions of chips in length, thus are only repeated infrequently. Because of this they are useful for added security as they are more difficult to decode.

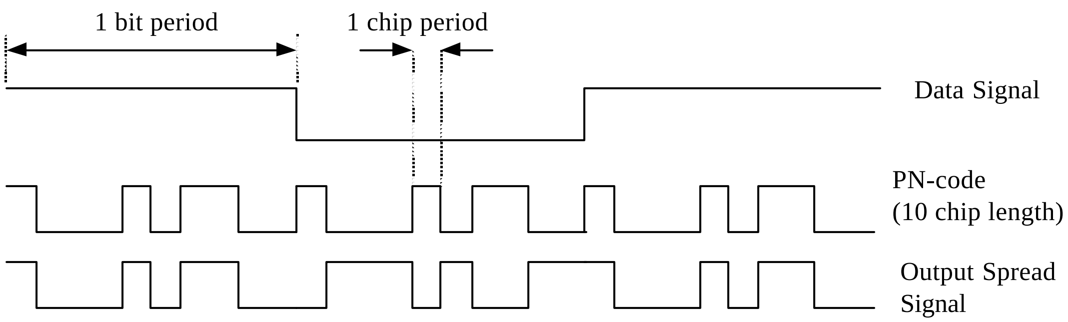


Figure 13 Direct sequence signals

1.3.6 CDMA Forward Link Encoding

The forward link, from the base station to the mobile, of a CDMA system can use special orthogonal PN codes, called Walsh codes, for separating the multiple users on the same channel. These are based on a Walsh matrix, which is a square matrix with binary elements and dimensions that are a power of two. It is generated from the basis that Walsh(1) = $W_1 = 0$ and that:

$$W_{2n} = \begin{bmatrix} W_n & W_n \\ W_n & \overline{W_n} \end{bmatrix}$$

Where W_n is the Walsh matrix of dimension n. For example:

$$W_2 = \begin{bmatrix} 0 & 0 \\ 0 & 1 \end{bmatrix}$$

$$W_4 = \begin{bmatrix} 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 1 \\ 0 & 0 & 1 & 1 \\ 0 & 1 & 1 & 0 \end{bmatrix}$$

Walsh codes are orthogonal, which means that the dot product of any two rows is zero. This is due to the fact that for any two rows exactly half the number of bits match and half do not.

Each row of a Walsh matrix can be used as the PN code of a user in a CDMA system. By doing this the signals from each user is orthogonal to every other user, resulting in no interference between the signals. However, in order for Walsh codes to work the transmitted chips from all users must be synchronized. If the Walsh code used by one user is shifted in time by more than about 1/10 of chip period, with respect to all the other Walsh codes, it looses its orthogonal nature resulting in inter-user interference. This is not a problem for the forward link as signals for all the users originate from the base station, ensuring that all the signal remain synchronized.

1.3.7 CDMA Reverse Link Encoding

The reverse link is different to the forward link because the signals from each user do not originate from a same source as in the forward link. The transmission from each user will arrive at a different time, due to propagation delay, and synchronization errors. Due to the unavoidable timing errors between the users, there is little point in using Walsh codes as they will no longer be orthogonal. For this reason, simple pseudo random sequences are typically used. These sequences are chosen to have a low cross correlation to minimise interference between users.

The capacity is different for the forward and the reverse links because of the differences in modulation. The reverse link is not orthogonal, resulting in significant inter-user interference. For this reason the reverse channel sets the capacity of the system.

1.3.8 Orthogonal Frequency Division Multiplexing

Orthogonal Frequency Division Multiplexing (OFDM) is a multicarrier transmission technique, which divides the available spectrum into many carriers, each one being modulated by a low rate data stream. OFDM is similar to FDMA in that the multiple user access is achieved by subdividing the available bandwidth into multiple channels, which are then allocated to users. However, OFDM uses the spectrum much more efficiently by spacing the channels much closer together. This is achieved by making all the carriers orthogonal to one another, preventing interference between the closely spaced carriers.

Coded Orthogonal Frequency Division Multiplexing (COFDM) is the same as OFDM except that forward error correction is applied to the signal before transmission. This is to overcome errors in the transmission due to lost carriers from frequency selective fading, channel noise and other propagation effects. For this discussion the terms OFDM and COFDM are used interchangeably, as the main focus of this thesis is on OFDM, but it is assumed that any practical system will use forward error correction, thus would be COFDM.

In FDMA each user is typically allocated a single channel, which is used to transmit all the user information. The bandwidth of each channel is typically 10kHz-30kHz for voice communications. However, the minimum required bandwidth for speech is only 3kHz. The allocated bandwidth is made wider then the minimum amount required to prevent channels from interfering with one another. This extra bandwidth is to allow for signals from neighbouring channels to be filtered out, and to allow for any drift in the centre frequency of the transmitter or receiver. In a typical system up to 50% of the total spectrum is wasted due to the extra spacing between channels. This problem becomes worse as the channel bandwidth becomes narrower, and the frequency band increases.

Most digital phone systems use vocoders to compress the digitised speech. This allows for an increased system capacity due to a reduction in the bandwidth required for each user. Current vocoders require a data rate somewhere between 4-13kbps [13], with depending on the quality of the sound and the type used. Thus each user only requires a minimum bandwidth of somewhere between 2-7kHz, using QPSK modulation. However, simple FDMA does not handle such narrow bandwidths very efficiently.

TDMA partly overcomes this problem by using wider bandwidth channels, which are used by several users. Multiple users access the same channel by transmitting in their data in time slots. Thus, many low data rate users can be combined together to transmit in a single channel that has a bandwidth sufficient so that the spectrum can be used efficiently.

There are however, two main problems with TDMA. There is an overhead associated with the change over between users due to time slotting on the channel. A change over time must be allocated to allow for any tolerance in the start time of each user, due to propagation delay variations and synchronization errors. This limits the number of users that can be sent efficiently in each channel. In addition, the symbol rate of each channel is high (as the channel handles the information from multiple users) resulting in problems with multipath delay spread.

OFDM overcomes most of the problems with both FDMA and TDMA. OFDM splits the available bandwidth into many narrow band channels (typically 100-8000). The carriers for each channel are made orthogonal to one another, allowing them to be spaced very close together, with no overhead as in the FDMA example. Because of this there is no great need for users to be time multiplex as in TDMA, thus there is no overhead associated with switching between users.

The orthogonality of the carriers means that each carrier has an integer number of cycles over a symbol period. Due to this, the spectrum of each carrier has a null at the centre frequency of each of the other carriers in the system. This results in no interference between the carriers, allowing then to be spaced as close as theoretically possible. This overcomes the problem of overhead carrier spacing required in FDMA.

Each carrier in an OFDM signal has a very narrow bandwidth (i.e. 1 kHz), thus the resulting symbol rate is low. This results in the signal having a high tolerance to multipath delay spread, as the delay spread must be very long to cause significant inter-symbol interference (e.g. > 100 μs).

1.3.9 OFDM generation

To generate OFDM successfully the relationship between all the carriers must be carefully controlled to maintain the orthogonality of the carriers. For this reason, OFDM is generated by firstly choosing the spectrum required, based on the input data, and modulation scheme used. Each carrier to be produced is assigned some data to transmit. The required amplitude and phase of the carrier is then calculated based on the modulation scheme (typically differential BPSK, QPSK, or QAM). The required spectrum is then converted back to its time domain signal using an Inverse Fourier Transform. In most applications, an Inverse Fast Fourier Transform (IFFT) is used. The IFFT performs the transformation very efficiently, and provides a simple way of ensuring the carrier signals produced are orthogonal.

The Fast Fourier Transform (FFT) transforms a cyclic time domain signal into its equivalent frequency spectrum. This is done by finding the equivalent waveform, generated by a sum of orthogonal sinusoidal components. The amplitude and phase of the sinusoidal components represent the frequency spectrum of the time domain signal. The IFFT performs the reverse process, transforming a spectrum (amplitude and phase of each component) into a time domain signal. An IFFT converts a number of complex data points, of length that is a power of 2, into the time domain signal of the same number of points. Each data point in frequency spectrum used for an FFT or IFFT is called a bin.

The orthogonal carriers required for the OFDM signal can be easily generated by setting the amplitude and phase of each frequency bin, then performing the IFFT. Since each bin of an IFFT corresponds to the amplitude and phase of a set of orthogonal sinusoids, the reverse process guarantees that the carriers generated are orthogonal.

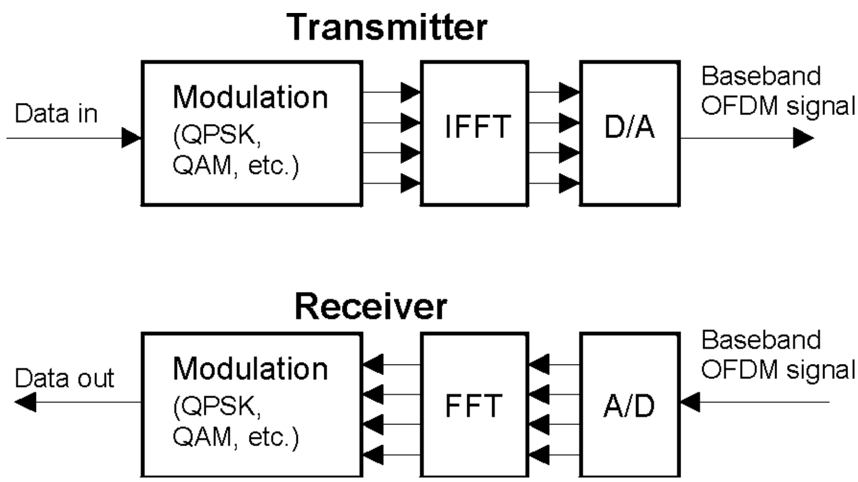


Figure 14 Basic FFT, OFDM transmitter and receiver

Figure 14 shows the configuration for a basic OFDM transmitter and receiver. The signal generated is at base-band and so to generate an RF signal the signal must be filtered and mixed to the desired transmission frequency.

1.3.10 Adding a Guard Period to OFDM

One of the most important properties of OFDM transmissions is its high level of robustness against multipath delay spread. This is a result of the long symbol period used, which minimises the inter-symbol interference. The level of multipath robustness can be further increased by the addition of a guard period between transmitted symbols. The guard period allows time for multipath signals from the pervious symbol to die away before the information from the current symbol is gathered. The most effective guard period to use is a cyclic extension of the symbol. If a mirror in time, of the end of the symbol waveform is put at the start of the symbol as the guard period, this effectively extends the length of the symbol, while maintaining the orthogonality of the waveform. Using this cyclic extended symbol the samples required for performing the FFT (to decode the symbol), can be taken anywhere over the length of the symbol. This provides multipath immunity as well as symbol time synchronization tolerance.

As long as the multipath delay echoes stay within the guard period duration, there is strictly no limitation regarding the signal level of the echoes: they may even exceed the signal level of the shorter path! The signal energy from all paths just add at the input to the receiver, and since the FFT is energy conservative, the whole available power feeds the decoder. If the delay spread is longer then the guard interval then they begin to cause inter-symbol interference. However, provided the echoes are sufficiently small they do not cause significant problems. This is true most of the time as multipath echoes delayed longer than the guard period will have been reflected of very distant objects.

Other variations of guard periods are possible. One possible variation is to have half the guard period a cyclic extension of the symbol, as above, an the other half a zero amplitude signal. This will result in a signal as shown in Figure 15. Using this method the symbols can be easily identified. This possibly allows for symbol timing to be recovered from the signal, simply by applying envelop detection. The disadvantage of using this guard period method is that the zero period does not give any multipath tolerance, thus the effective active guard period is halved in length. It is interesting to note that this guard period method has not been mentioned in any of the research papers read, and it is still not clear whether symbol timing needs to be recovered using this method.

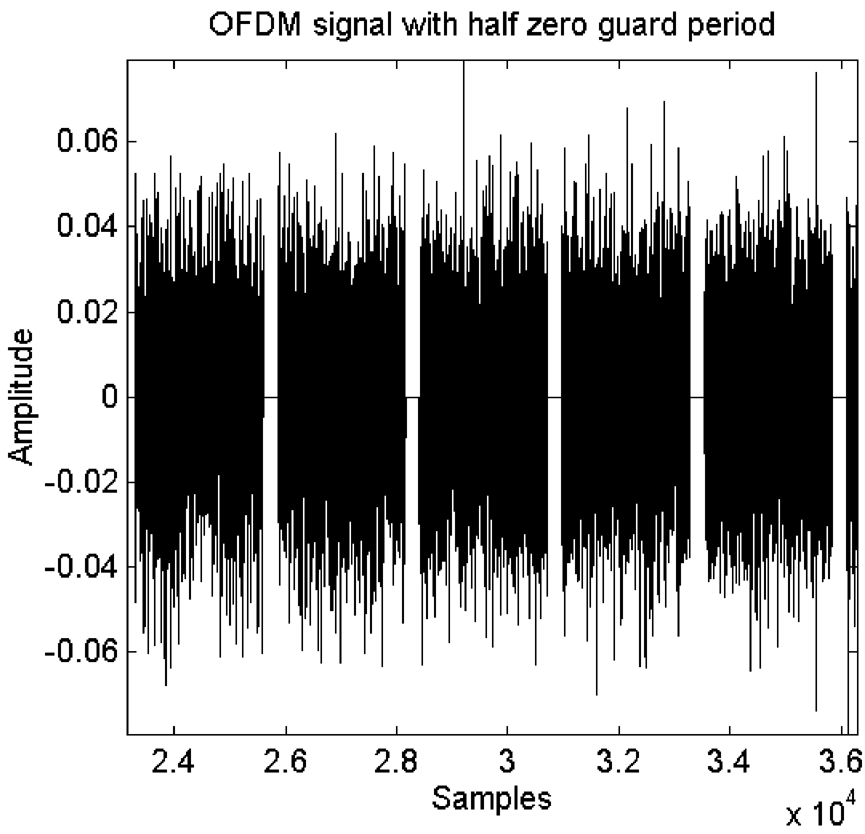


Figure 15 Section of an OFDM signal showing 5 symbols, using a guard period which is half a cyclic extension of the symbol, and half a zero amplitude signal. (For a signal using a 2048 point FFT and 512 sample total guard period)

(Errata Note 10/2001: The best method for guard period implementation to use a cyclic extension of the transmitted symbol over the entire guard period interval, rather than only half of the guard period as described above.)

2. OFDM Results

An OFDM system was modelled using Matlab to allow various parameters of the system to be varied and tested. The aim of doing the simulations was to measure the performance of OFDM under different channel conditions, and to allow for different OFDM configurations to be tested. Four main criteria were used to assess the performance of the OFDM system, which were its tolerance to multipath delay spread, peak power clipping, channel noise and time synchronization errors.

2.1 OFDM Model Used

The OFDM system was modelled using Matlab and is shown in Figure 16. A brief description of the model is provided below.

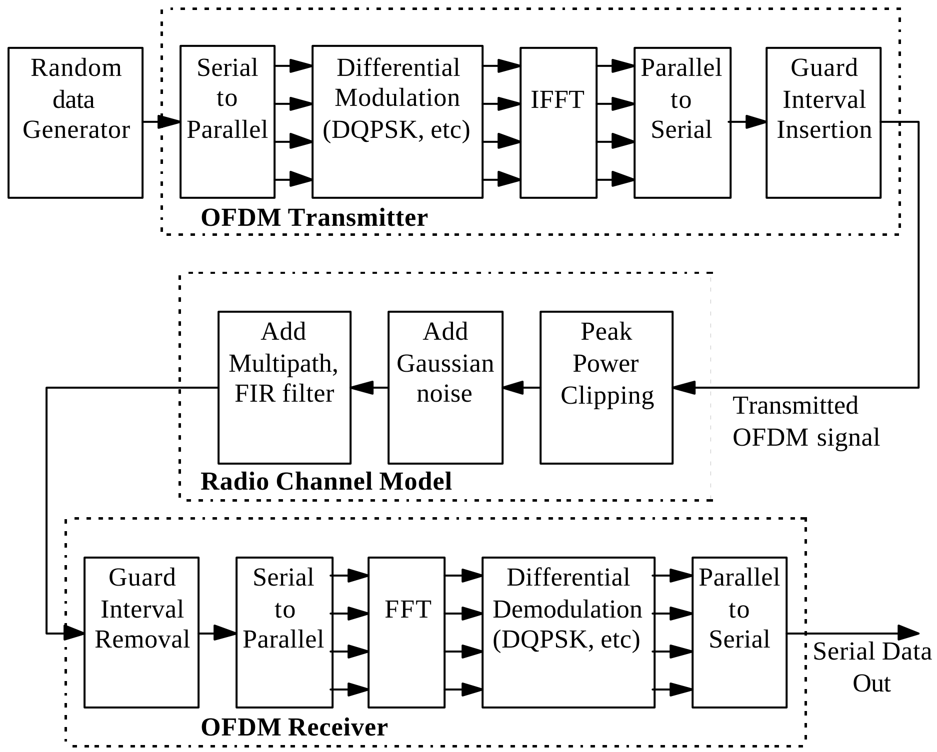


Figure 16 OFDM Model used for simulations

2.1.1 Serial to Parallel Conversion

The input serial data stream is formatted into the word size required for transmission, e.g. 2 bits/word for QPSK, and shifted into a parallel format. The data is then transmitted in parallel by assigning each data word to one carrier in the transmission.

2.1.2 Modulation of Data

The data to be transmitted on each carrier is then differential encoded with previous symbols, then mapped into a Phase Shift Keying (PSK) format. Since differential encoding requires an initial phase reference an extra symbol is added at the start for this purpose. The data on each symbol is then mapped to a phase angle based on the modulation method. For example, for QPSK the phase angles used are 0, 90, 180, and 270 degrees. The use of phase shift keying produces a constant amplitude signal and was chosen for its simplicity and to reduce problems with amplitude fluctuations due to fading.

2.1.3 Inverse Fourier Transform

After the required spectrum is worked out, an inverse fourier transform is used to find the corresponding time waveform. The guard period is then added to the start of each symbol.

2.1.4 Guard Period

The guard period used was made up of two sections. Half of the guard period time is a zero amplitude transmission. The other half of the guard period is a cyclic extension of the symbol to be transmitted. (As discussed in section 1.3.10). This was to allow for symbol timing to be easily recovered by envelope detection.

However it was found that it was not required in any of the simulations as the timing could be accurately determined position of the samples.

After the guard has been added, the symbols are then converted back to a serial time waveform. This is then the base band signal for the OFDM transmission.

2.1.5 Channel

A channel model is then applied to the transmitted signal. The model allows for the signal to noise ratio, multipath, and peak power clipping to be controlled. The signal to noise ratio is set by adding a known amount of white noise to the transmitted signal. Multipath delay spread then added by simulating the delay spread using an FIR filter. The length of the FIR filter represents the maximum delay spread, while the coefficient amplitude represents the reflected signal magnitude.

2.1.6 Receiver

The receiver basically does the reverse operation to the transmitter. The guard period is removed. The FFT of each symbol is then taken to find the original transmitted spectrum. The phase angle of each transmission carrier is then evaluated and converted back to the data word by demodulating the received phase. The data words are then combined back to the same word size as the original data.

2.1.7 OFDM simulation parameters

Table 9 shows the configuration used for most of the simulations performed on the OFDM signal. An 800-carrier system was used, as it would allow for up to 100 users if each were allocated 8 carriers. The aim was that each user has multiple carriers so that if several carriers are lost due to frequency selective fading that the remaining carriers will allow the lost data to be recovered using forward error correction. For this reason any less then 8 carriers per user would make this method unusable. Thus 400 carriers or less was considered too small. However more carriers were not used due to the sensitivity of OFDM to frequency stability errors. The greater the number of carriers a system uses, the greater it required frequency stability.

For most of the simulations the signals generated were not scaled to any particular sample rate, thus can be considered to be frequency normalized. Three carrier modulation methods were tested to compare their performances. This was to show a trade off between system capacity and system robustness. DBPSK gives 1 b/Hz spectral efficiency and is the most durable method, however system capacity can be increased using DQPSK (2 b/Hz) and D16PSK (4 b/Hz) but at the cost of a higher BER. The modulation method used is shown as BPSK, QPSK, and 16PSK on all of the simulation plots, because the differential encoding was considered to be an integral part of any OFDM transmission.

(Addendum 10/2001: Many OFDM systems now use coherent modulation instead of differential modulation as coherent modulation allows the use of Quadrature Amplitude Modulation (QAM) carrier modulation, which improves the spectral efficiency.)

Parameter	Value
Carrier Modulation used	DBPSK, DQPSK, D16PSK
FFT size	2048
Number of carrier used	800
Guard Time	512 samples (25%)
Guard Period Type	Half zero signal, half a cyclic extension of the symbol

Table 9 OFDM system parameters used for the simulations

(Errata Note: 10/2001. The simulation results presented here show the overall Symbol Error Rate (SER) rather than the Bit Error Rate (BER) as indicated on the graph results and discussions. For BPSK the SER equals the BER, however for QPSK the BER will be approximately half the SER. This is because two bits of information are transferred for each QPSK symbol and typically only single bit errors occur when suitable mapping is used (gray coding) and the noise level is low. This approximation is valid for a SER below approximately $1x10^{-2}$. For 16PSK the same argument applies, thus the true BER is approximately one quarter the SER.)

2.2 OFDM Simulated Results

2.2.1 Multipath Delay Spread Immunity

For this simulation the OFDM signal was tested with a multipath signal containing a single reflected echo. The reflected signal was made 3 dB weaker then the direct signal as weaker reflections then this did not cause measurable errors, especially for BPSK. Figure 17 shows the simulation results.

It can be seen from Figure 17 that the BER is very low for a delay spread of less then approximately 256 samples. In a practical system (i.e. one with a 1.25 MHz bandwidth) this delay spread would correspond to ~80 µsec. This delay spread would be for a reflection with 24 km extra path length. It is very unlikely that any reflection, which has travelled an extra 24 km, would only be attenuated by 3 dB as used in the simulation, thus these results show extreme multipath conditions. The guard period used for the simulations consisted of 256 samples of zero amplitude, and 256 samples of a cyclic extension of the symbol. The results show that the tolerable delay spread matches the time of the cyclic extension of the guard period. It was verified that the tolerance is due to the cyclic extension not the zeroed period with other simulations. These test however are not shown to conserve space.

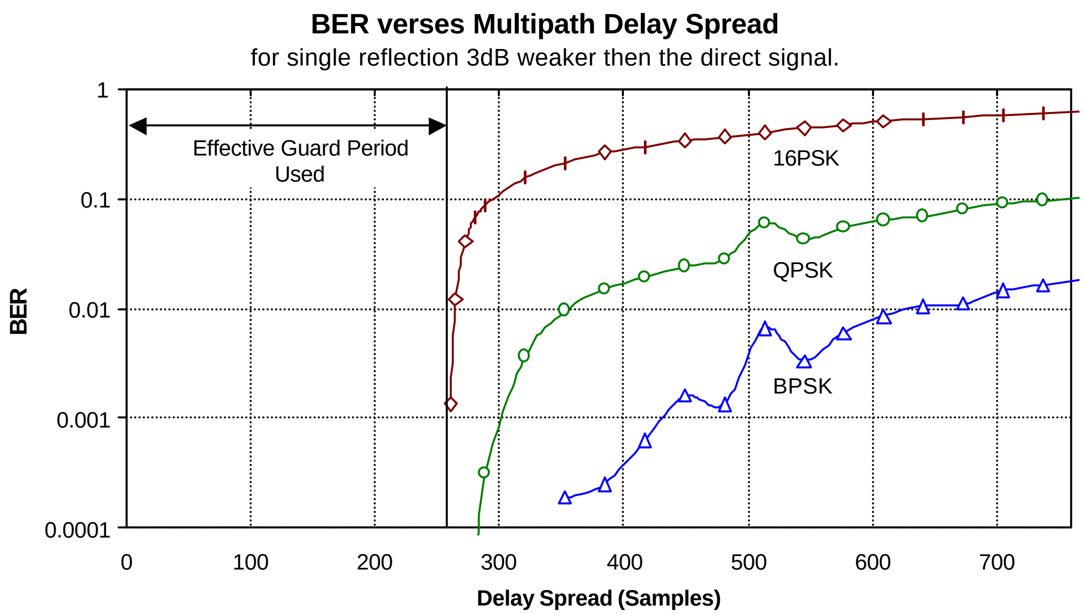


Figure 17 Delay Spread tolerance of ODFM

For a delay spread that is longer than the effective guard period, the BER rises rapidly due to the inter-symbol interference. The maximum BER that will occur is when the delay spread is very long (greater then the symbol time) as this will result in strong inter-symbol interference.

In a practical system the length of the guard period can be chosen depending on the required multipath delay spread immunity required.

2.2.2 Peak Power Clipping

It was found that the transmitted OFDM signal could be heavily clipped with little effect on the received BER. In fact, the signal could the clipped by up to 9 dB without a significant increase in the BER. This means that the signal is highly resistant to clipping distortions caused by the power amplifier used in transmitting the signal. It also means that the signal can be purposely clipped by up to 6 dB so that the peak to RMS ratio can be reduced allowing an increased transmitted power.

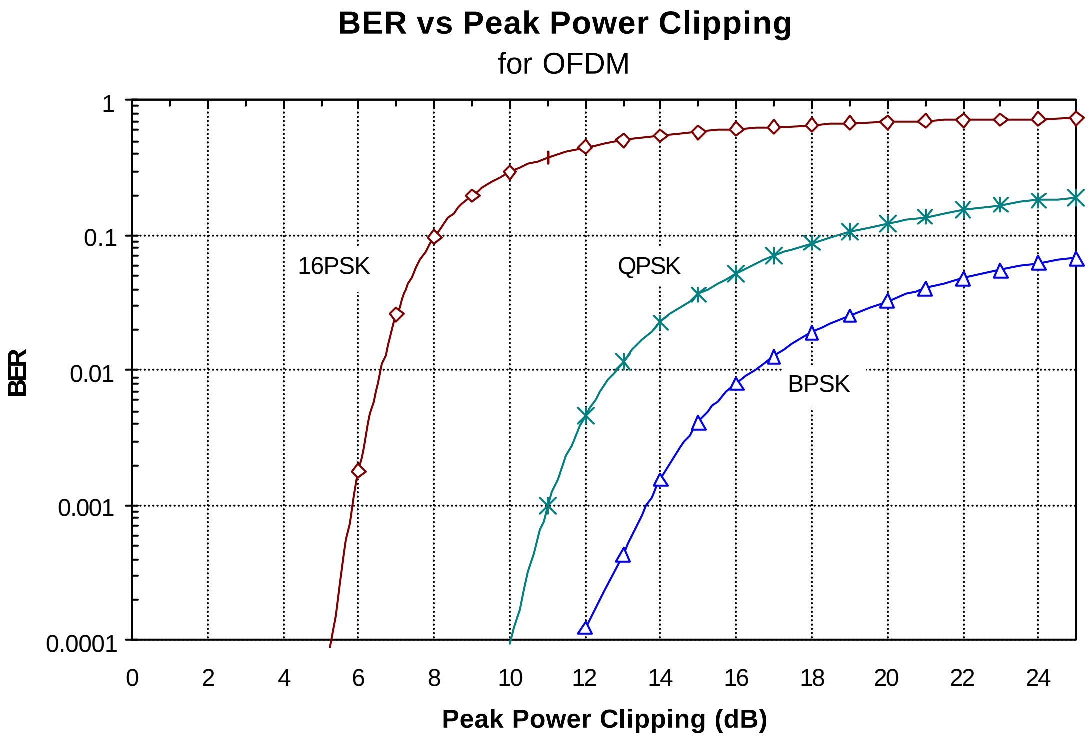


Figure 18 Effect of peak power clipping for OFDM

2.2.3 Gaussian Noise Tolerance of OFDM

It was found that the SNR performance of OFDM is similar to a standard single carrier digital transmission. This is to be expected, as the transmitted signal is similar to a standard Frequency Division Multiplexing (FDM) system. Figure 1 shows the results from the simulations. The results show that using QPSK the transmission can tolerate a SNR of >10-12 dB. The bit error rate BER gets rapidly worse as the SNR drops below 6 dB. However, using BPSK allows the BER to be improved in a noisy channel, at the expense of transmission data capacity. Using BPSK the OFDM transmission can tolerate a SNR of >6-8 dB. In a low noise link, using 16PSK can increase the capacity. If the SNR is >25 dB 16PSK can be used, doubling the data capacity compared with QPSK.

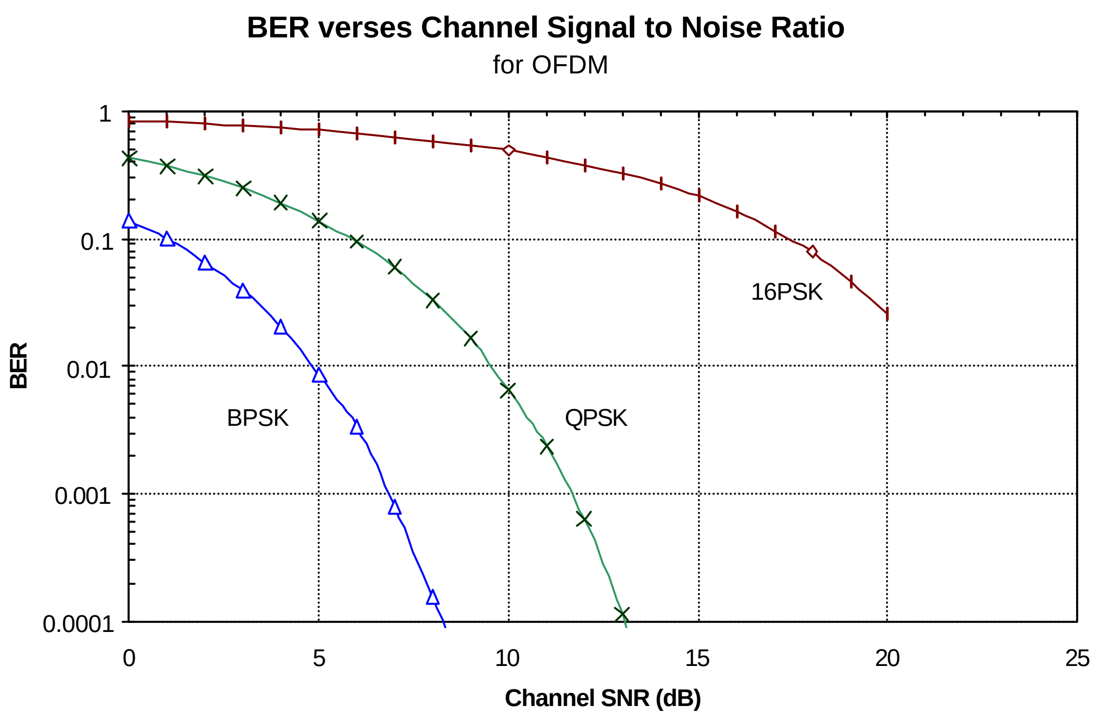


Figure 19 BER verse SNR for OFDM using BSPK, QPSK and 16PSK

(Errata 10/2001: The simulation results shown in Figure 19 are slightly incorrect due to the calculation of the noise level in the simulation. The simulations shown above calculated the signal to noise ratio based on the power of the time domain signal waveform and the power of the time domain noise waveform, with no consideration of the signal bandwidth. At the receiver the signal is filtered by the FFT stage, thus making the receiver only see noise within the signal bandwidth. The simulations were performed using 800 carriers and generated using a 2048-point IFFT. The nyquist bandwidth is half the transmission sample rate as the signal is real (i.e. no imaginary components) and so the nyquist bandwidth corresponds to 1024 carriers. The signal bandwidth is thus 800/1024 = 0.781 or 78.1% of the nyquist bandwidth. Since the receiver was only seeing 78.1% of the total noise the error rate is lower than it should be. The correct SNR values can be found by adding 1.07 dB ($10.\log_{10}(0.781)$) to the scale in Figure 19. Also note the simulation results actually show the Symbol Error Rate rather than the Bit Error Rate.)

2.2.4 Timing Requirements

One of the big questions at the start of the thesis was how tolerant OFDM would be to a starting time error. The problem was that when an OFDM receiver is initially switched on it will not be synchronized with the transmitted signal. So a synchronization method was required. The proposed method was that the OFDM signal could be broken up into frames, where each frame transmits a number of symbols (somewhere between 10-1000). At the start of each frame a null symbol is transmitted, thus allowing the start of the frame to be detected using envelope detection. However using envelope detection only allows the start to be detected to within a couple of sample, depending on the noise in the system. It was not known whether this timing accuracy was sufficient. This method was used for the synchronization in the practical tests performed. (See section 2.3)

Figure 20 shows the effect of start time error on the received BER. This shows that the starting time can have an error of up to 256 samples before there is any effect of the BER. This length matches the cyclic extension period of the guard interval, and is due to the guard period maintaining the orthogonality of the signal.

In any practical system, the timing error made be either early or late, thus any receiver would aim for the middle of the expected starting time to allow for an error of ± 128 samples. In addition, if the signal is subject to any multipath delay spread, this will reduce the effective stable time of the guard period, thus reducing the starting time error tolerance.

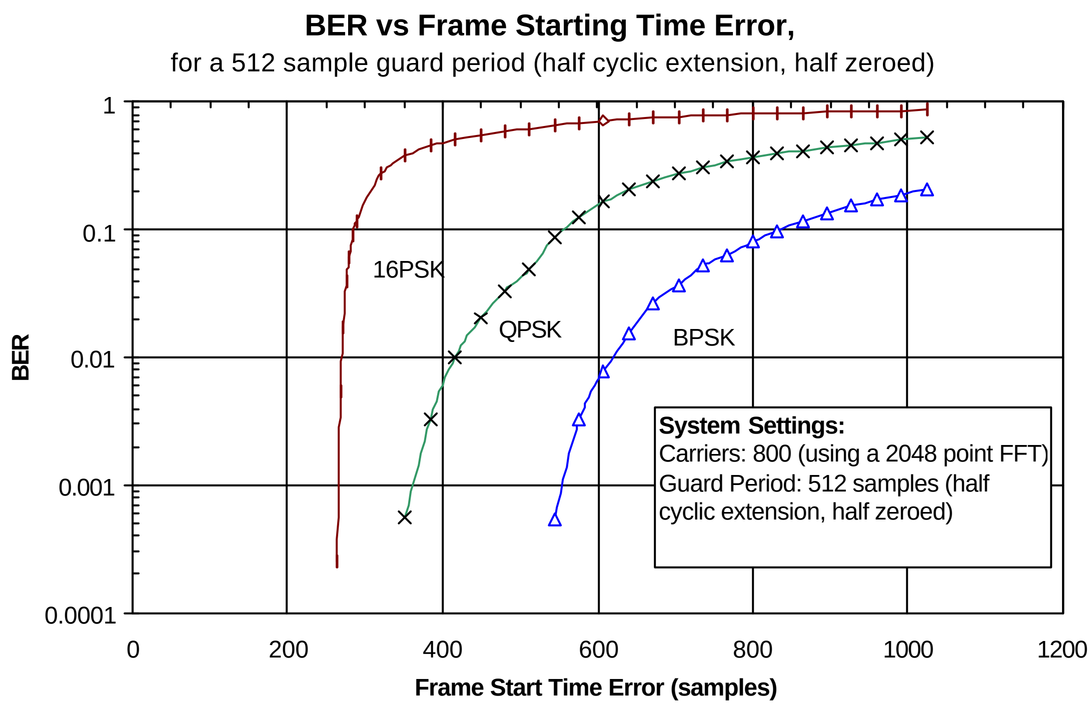


Figure 20 Effect of frame synchronization error on the received OFDM signal.

2.3 Practical Measurements

A set of practical measurements was made on the OFDM system. This was done so that the simulated results could be partly verified and so that difficulties in implementing a practical system could be tackled, and to measure some effects which were hard to simulate.

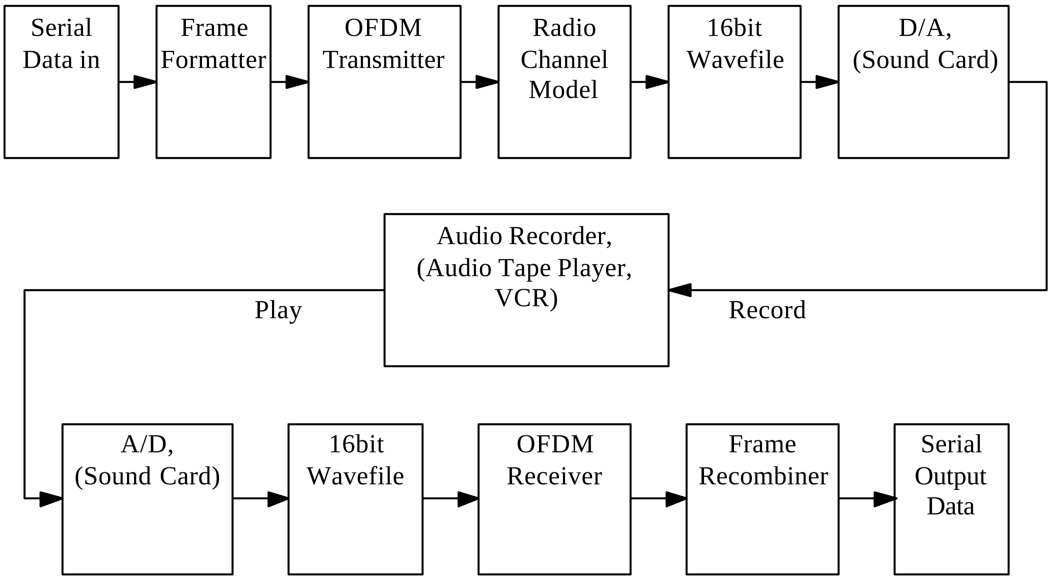
The practical measurements were performed using a Personal Computer (PC) as an OFDM generator and receiver. A Matlab program was developed to process the input and output signals and a Sound Blaster 16 card was used to play the transmitted OFDM signal, and used to record the received signal. Only one PC was available with a sound card and so the transmission was performed in two steps. The transmitted signal was generated using Matlab then played out the sound card and recorded onto an audio recorder. This signal was then played back and re-recorded by the sound card on the computer. The received signal was then processed using a Matlab script. Two different audio recorders were tested, as each gave a different performance.

The first used was the HiFi stereo audio track on a high quality VCR (Panasonic Super-VHS FS90). This gave a high quality audio channel (SNR >90dB, 20Hz - 20kHz range, and crystal accuracy stability of frequency (0.005%)), which was a good model for a near perfect radio channel.

The system was also tested using an audio tape player. The tape player had a much poorer performance, in noise (~50dB SNR), audio bandwidth (20Hz-15kHz), and particularly frequency stability (2%). This provided a good test of the performance of OFDM in a channel with very poor frequency stability.

2.3.1 Extended Model

The basic OFDM model used for the simulations (see section 2.1) was extended to allow for the received signal to be automatically be synchronized to the OFDM frame structure, and for large data files to be able to be transmitted.



2.3.2 Transmission Protocol

A basic frame structure was used to allow the receiver to synchronize with the transmitted signal. The frames were marked out by having a null symbol (zero amplitude) between frames (see Figure 21), allowing the start of each frame to be detected using an envelope detector in software. The transmitted signal consisted of a number of frames (typically 1- 100), with a preamble at the start and a post signal at the end (see Figure 22). The preamble was used to provide the start time of the first frame and consisted on a mixture of tones. This was required so that the envelope detector had a signal to initialise the filtering required.

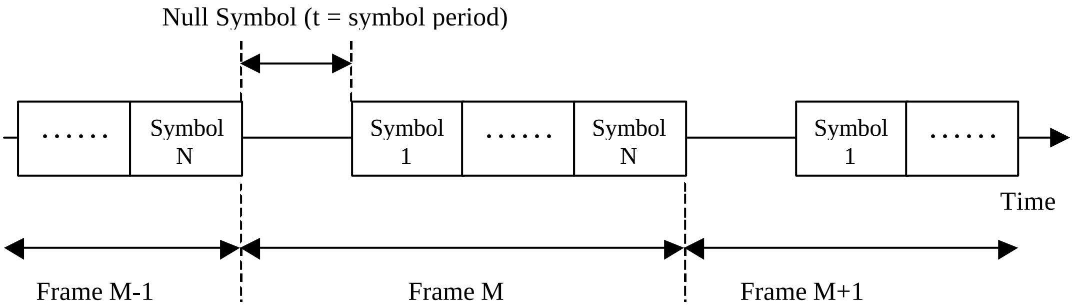


Figure 21 Frame Structure, showing the null symbol between frames

The envelope detector was implemented by rectifying the input signal then applying a moving average filter to the signal. The length of the filter was made exactly the same length as the null symbol. This results in the filtered signal having a minimum amplitude at the start of a frame. This minimum was used to find the starting location in which to decode the entire frame. Each frame consists of a number of symbols (typically 5-40) that contain the actual data.

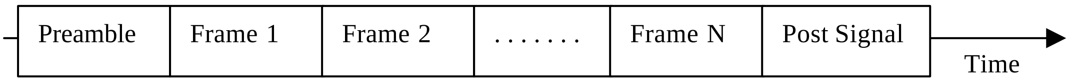


Figure 22 Frame Structure used for the OFDM transmission

2.3.3 Video Recorder

The first simulated channel used was the audio track of a VCR.

2.3.3.1 Number of Carriers Used

For an OFDM system, the number of carriers used in the transmission determines several parameters about the system. These include the processing speed required, the symbol time (thus the maximum delay spread that can be tolerated), the number of users the available bandwidth can be split over (i.e. one carrier per user), and the frequency stability required.

Since there was no easy way of simulating this effect with the Matlab model developed, it was decided to measure it in a practical way. This was done by varying the number of carriers used when transmitting the OFDM over the link simulated by the VCR. The frequency stability of the VCR's recording and playing was considered to be approximately the same for each of the tests performed. This allowed the relative effect of varying the number of carriers for a fixed frequency stability.

The data used in the transmission was that of a grey-scale bit-map image of a fish. The original of the image is show below in Figure 23.

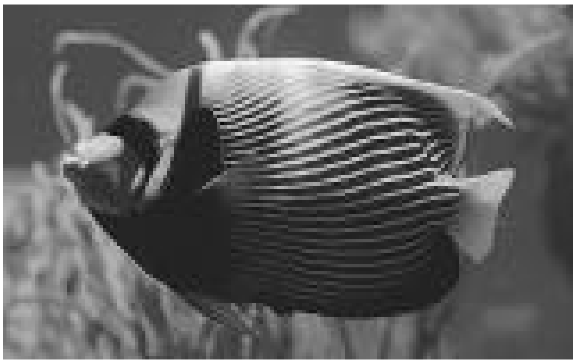


Figure 23 Image used in transmission tests

The transmission was sent using 256PSK. This was chosen because it gave the highest transmission efficiency (~8 bits/Hz), thus resulting in the smallest transmission data size. By sending each 8 bit grey scale pixels as one carrier per symbols, any phase errors in the transmission directly correspond to a change in the intensity of the received signal. This allows the phase angle errors to be judged from the image quality. Since the phase error due to a frequency offset or phase noise has the same effect on the received data as gaussian noise, it can be effectively converted to an equivalent signal to noise ratio of the received image.

The image quality and phase error was measured at different number of carriers used to transmit the signal.

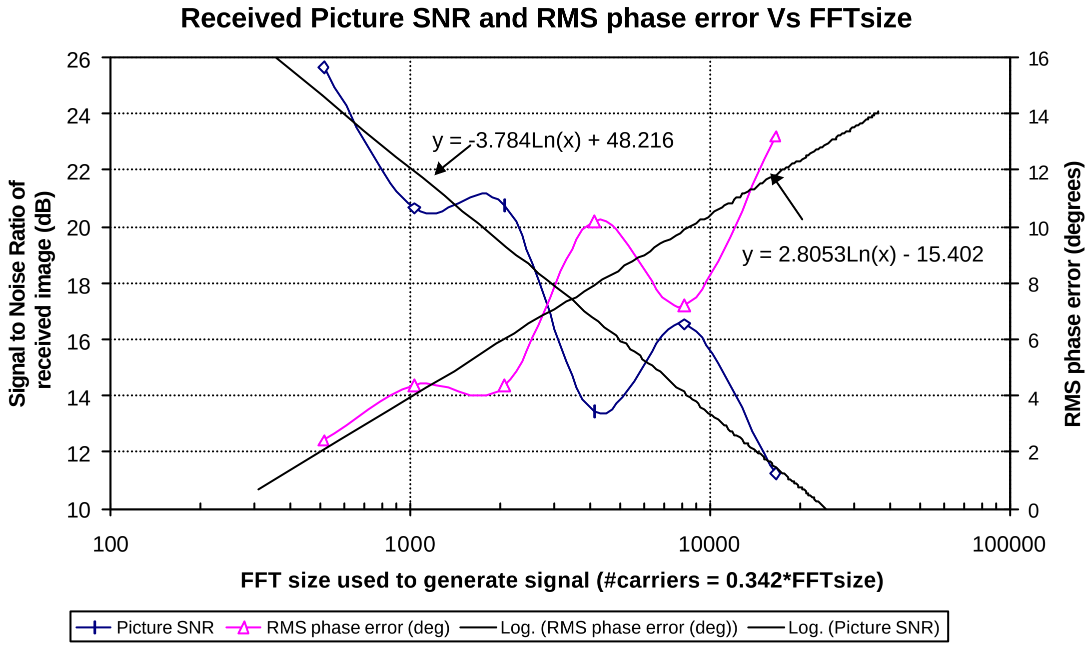


Figure 24 Performance of the OFDM using the VCR as a channel, as a function of the number of carriers used

Figure 24 show the effect that increasing the number of carriers had on the received signal. It can be seen that the larger the FFT size (and number of carriers used), the worse the performance of the system. This is due to the VCR having a fixed frequency stability thus the closer the carriers are in the transmission, the worse the effect of the frequency error. The SNR of the received image for an FFT size of 2048 was only 20dB. This is much worse then the SNR ratio that would be expected from the channel noise. The VCR has excellent noise performance (>70-90dB SNR for gaussian noise) however this limit is no where near reached due to frequency stability problems causing an effective SNR of between 10-30dB. This indicates that the performance of the system is not limited by the gaussian noise of the system, but the frequency stability.

Since the frequency stability of the is such a problem in an practical radio OFDM system the receiver would have to be frequency locked to the transmitter in order achieve the maximum performance.

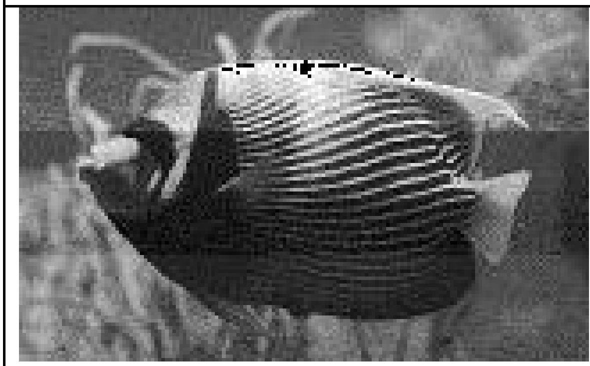
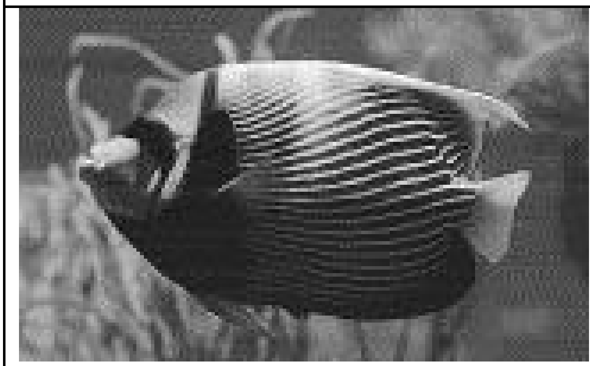
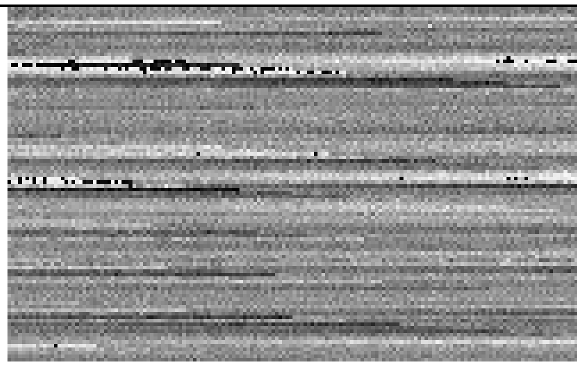
Received Image from VCR audio channel	Error in received image (contrast enhanced to show the errors)
 16384 point FFT, 5600 carriers	 16384 point FFT, 5600 carriers
 256 point FFT, 75 carriers	 256 point FFT, 75 carriers

Table 10 Received OFDM images using the audio channel of a VCR

Table 10 shows some of the received images that were used to generate Figure 24. It can be seen that the image transmitted using 5600 carriers has bands in the image, due to the phase errors (and thus pixel intensity errors) in the transmission. Also some of the pixels that are white (on the top of the fish) have wrapped round to show white. This is due to the received phase error causing a wrap around from 255 to 0 in intensity.

2.3.4 Peak OFDM Performance for the VCR link

After trying out different OFDM system parameters such as the number of carriers used, system bandwidth and guard period length, it was found that very high spectral efficiency could be achieved. Figure 25 shows the maximum performance that could be achieved on the VCR audio track.

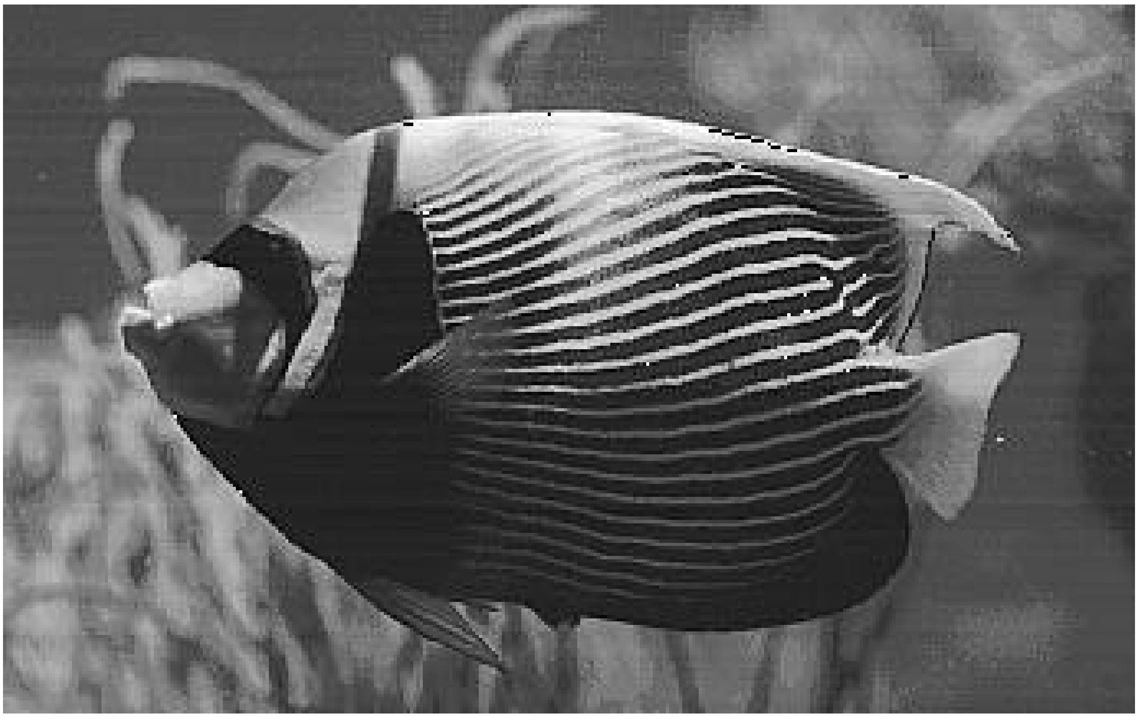


Figure 25 Image transferred at 134kbps in an 18.2kHz bandwidth on the VCR audio channel, using 210 carriers.

The image of the fish was transferred using 256PSK. The total transmission time was 4.54 seconds for 76246 bytes of data, with only 18.2kHz bandwidth. This gives a spectral efficiency of 7.4 bits/Hz. This is just under the theoretical limit of 8 bits/Hz for 256PSK and is due to overhead in the guard period and frame symbols. The signal was generated using a 512-point FFT, using 210 carriers, and a guard period of 32 samples. The carriers used were based on the frequency response of the VCR link measured in section 2.3.4.2, so that the maximum bandwidth could be used.

The received image in Figure 25 has slight phase errors, which are just noticeable as bands in the image. Also some pixel on the top fin of the fish have wrapped around from white to black.

2.3.4.1 Peak Power Clipping

The clipping tolerance of the OFDM signal was tested to verify that OFDM can handle a large amount of peak power clipping before any significant increase in the bit error rate (BER) occurs. The simulations indicated that OFDM could handle up to 9dB of clipping (for QPSK) before the BER became detectable. This result was slightly surprising as any non-linearity in the system lead to intermodulation distortion. Thus the initial expectation was that OFDM would be susceptible to any clipping of the signal.

The clipping of the signal was performed during the recording from the VCR back to the computer for decoding. The clipping of the signal was achieved by using back-to-back germanium diodes across at the output of the VCR with a resistor in series with the VCR to limit the current flow. The signal was observed on a CRO and increased in amplitude until clipping occurred. The peak power clipping was measured by finding the ratio of the peak signal level before clipping, to the peak signal level after clipping.

Peak Signal before clipping (Vp-p)	Peak Signal after clipping (Vp-p)	Calculated Peak Power Clipping (dB)	Measured BER of received signal	Predicted BER from simulation
1.45	0.72	6.08	<0.00006	
1.88	0.80	7.45	<0.00006	
2.01	0.805	7.95	<0.00006	
2.65	0.853	9.85	0.0004	<0.00009
3.55	0.917	11.8	0.0036	0.0038
4.6	0.935	13.84	0.0125	0.0208

Table 11 Results of clipping the OFDM signal, showing the resulting BER

Table 11 shows the measured error rates when the signal was clipped, and the expected BER based on the simulations. The BER was found to be too small (<0.00006) for peak power clipping up to 8 dB. The BER was only detectable for peak power clipping of >8-10 dB, matching the expected result of 9 dB measured from the simulations (see section 2.2.2). For high levels of clipping, from 12- 14 dB, the measured BER was actually lower then the simulated results. This is probably due to the fact that the germanian diodes used for clipping of the signal were not clipping as abruptly as in the simulation resulting in lower intermodulation distortion and a lower BER.

2.3.4.2 VCR performance

The audio performance of the VCR link was measured so that the quality of the channel used for the practical measurements could be assessed. The loop back frequency response of the channel was measured by using the sound card to generate white noise. This noise was then recorded on the VCR, then played back and recorded back on to the computer. The power spectral density of the noise was then measured giving an estimate frequency response of the system. Since the signal was generated from the sound card the measured frequency response is the combined performance of the Sound Blaster 16 card and the VCR.

Figure 26 shows the frequency response of the VCR link used.

The frequency stability of the VCR is quoted by the manufacturer as 0.005%. This was verified by recording a 10kHz sine wave on to the VCR using the sound card to generate the signal. The frequency of the played back signal was measured using a frequency meter. It was found that the played back signal to be stable to better then the resolution of the frequency meter used (i.e. 10.000kHz). Thus the sound card, VCR combination was stable to better then 0.01%.

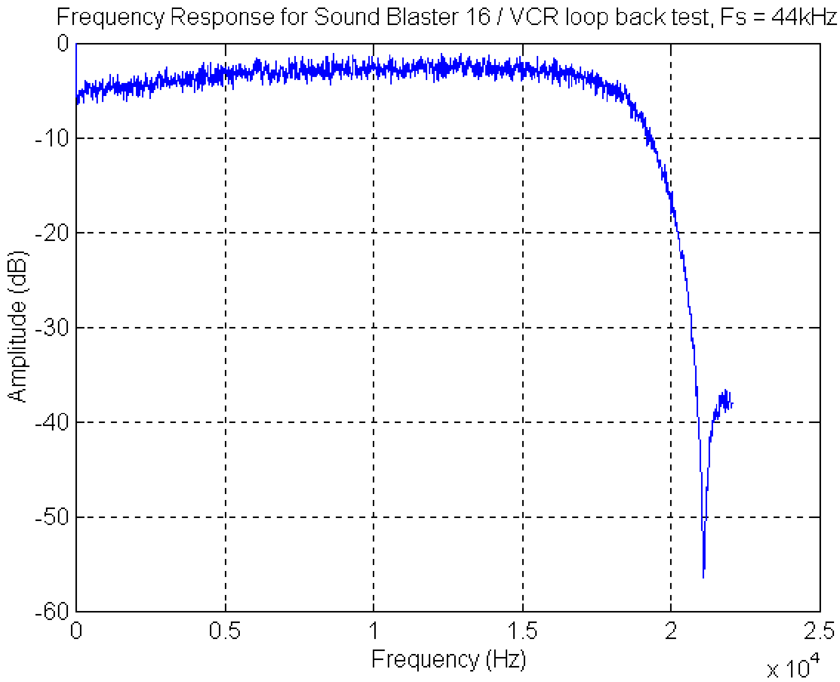


Figure 26 Record / Play back frequency response of the Panasonic FS90 VCR / Sound Blaster 16 combination using a sample rate of 44.1kHz

2.3.5 Audio Tape Player

After much success in using the VCR as a link, the OFDM system was also tested using an audio tape player as the channel. The audio tape player used is not frequency locked, as with the VCR, thus has a very poor frequency stability. This can lead to a large frequency offset error in the recording. Since OFDM is very susceptible to frequency offset errors, the performance of the system was poor on the tape player.

2.3.5.1 Carrier Number

Several test OFDM transmissions were tried using the tape player as a channel. It was found that the only way to successfully store an image on the tape player was to greatly reduce the number of carriers used. Table 12 shows two images that were recorded on the audiotape play then recorded and decoded back onto the computer. It can be seen that the image quality when using only 5 carriers was much better than using 21 carriers. The image transmitted using 21 carriers, has lines and smears through it due to phase errors. This is due to the frequency instability of the tape player.

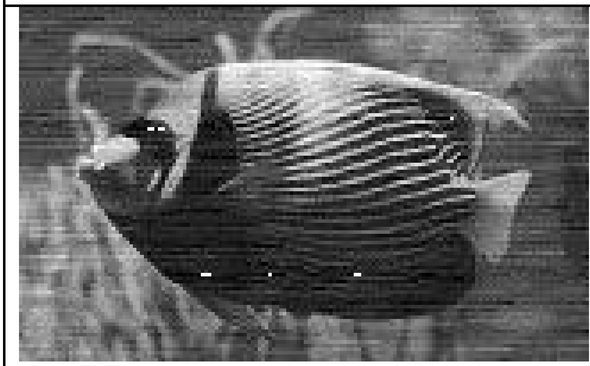
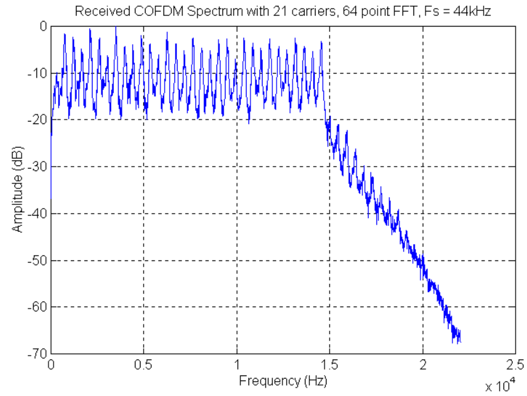
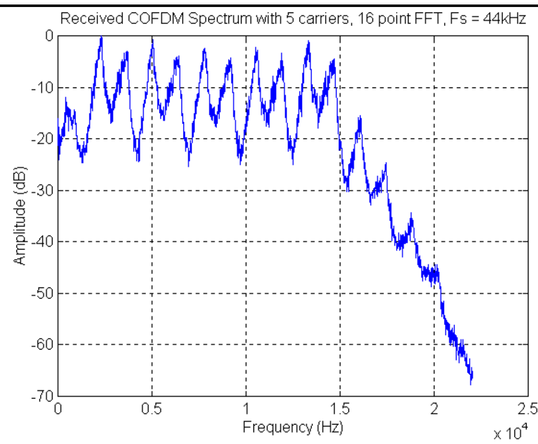
Received Image	Spectrum of the received signal
 64 point FFT, 21 carriers	
 16 point FFT, 5 carriers	

Table 12 Received images from the audio tape player, using OFDM with 5 and 21 carriers.

2.3.5.2 Tape Player Performance

The frequency stability and frequency response of the tape player were measured to assess the effective quality of the channel. All tests were performed using the audio tape player in conjunction with the Sound Blaster 16 card, thus the measurements are the combination of their performances.

The measurements were performed using the same technique used for performance measurement of the VCR (see section 2.3.4.2).

Frequency Stability

A 10kHz tone was generated using the sound card by playing a windows 3.1 WAV file generated by a Matlab script. This tone was then recorded on the tape player. The frequency and amplitude of the played back signal was then measured over a 10 second period. This time length was used because it was approximately the same length of time that is taken in transmitting the fish image. The results are shown below in Table 13.

	Frequency (kHz)	Amplitude Variation (dB)
Max	10.000	0.21
Mean	9.987	0.01
Min	9.823	-0.25

Table 13 Frequency and amplitude fluctuations in a 10second, 10kHz tone played back from the audio tape player

Thus the frequency offset is -0.13% and the variation is 1.77%. The level of frequency instability is very high as even for only 21 carriers this represents the carrier frequencies shifting by up to 40% of a bin. Also the frequency instability would also cause large phase errors to occur.

Frequency Response

The tape player / Sound Blaster 16 frequency response is shown below in Figure 27. The bandwidth of the tape player is about 15kHz, plus the response is reasonably flat (±3dB). The frequency response of the tape player should not cause any significant problems for the OFDM signal.

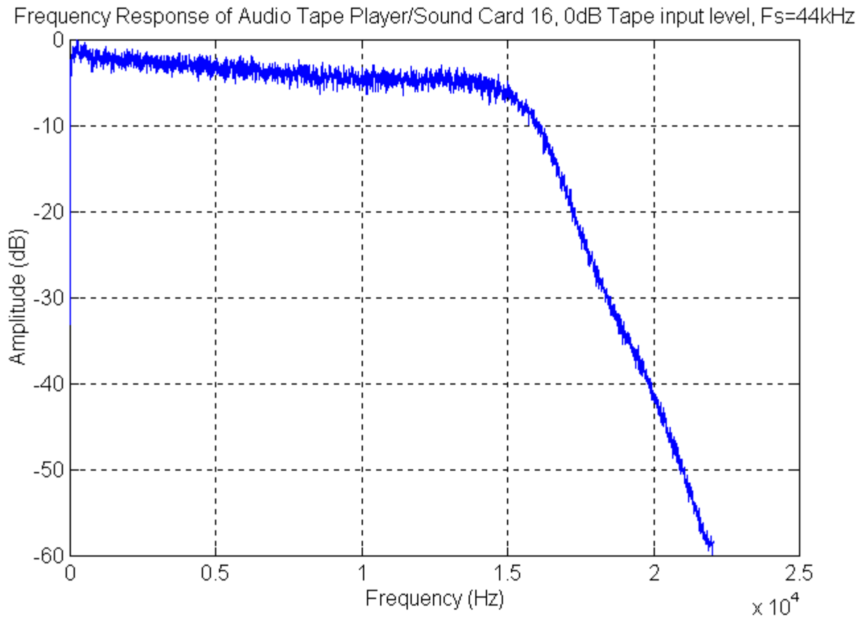


Figure 27 Record / Play back frequency response of the JVC TD-W444 / Sound Blaster 16 with a sample rate of 44.1kHz

2.4 Picture quality verse signal to noise ratio

This section describes some work that was done relating to improving the degradation performance of OFDM when the channel noise becomes very high. One problem with many digital communication systems is that the performance of the system is fine, up to some critical channel noise level, above which the system fails very quickly. This can cause potential problems for wireless telecommunications where the received signal quality can vary greatly depending of the location of the mobile station. This problem leads to drop outs in the signal, decreasing the reliability of the system. It would be far better if the system simply gave a worse voice or image quality under high noise conditions, rather then completely loosing the signal.

It was noticed when doing the practical measurements on OFDM (see section 2.3) that the received quality of the image was very good using 256PSK even though the measured error rate was very high. Using 256PSK to transmit the image is similar to sending an analog transmission as channel noise simply appears as noise in the image. The main question was, does sending the image using 256PSK result in better performance then QPSK in a high noise environment? To answer this question, a simple comparison was set up.

2.4.1 Experimental comparison between QPSK and power averaged 256PSK

The same Matlab scripts were used for this test as in the previous practical measurements, however the signal wasn't recorded to any external audio channel as in the practical tests. The channel noise was simulated using the same model as used in the OFDM simulations (see section 2.1). The signal to noise ratio (SNR) of the channel was varied from 0 dB up to 15 dB, with the image quality measured at 3 dB increments. No forward error correction was used for either modulation technique.

Using 256PSK allows 8 bits to be sent on each carrier per symbol, compared to only 2 bits for QPSK. Thus, 256PSK on its own allows for up to 4 times the transmission data rate. Since the aim of the experiment was to compare the relative performance differences between QPSK and 256PSK under the same transmission bandwidth and data rate, the 256PSK transmission was slowed down by 4 times to have the same transmission rate. This was achieved by repeating the 256PSK symbols four times, thus reducing the data rate to the same as using QPSK. The four repetitions were then combined at the receiver by averaging the received demodulated phase angle. This averaging reduces the received phase noise by up to 6 dB.

Transmitting the signal using 256PSK with linear mapping of the pixel intensity is very similar to transmitting the imaging in an analog manner. This modulation technique is essentially linear Phase Modulation. Transmitting the signal in an analog manner result in a direct mapping between phase errors caused by the channel, and amplitude noise (pixel intensity) in the received image. Using a linear mapping tends to reduce the chance of the large catastrophic errors that can occur when sending digital data that is not mapped in a linear manner.

However when transmitting using a standard digital modulation technique such as QPSK, the SNR of the demodulated image can be much higher then the channel SNR, provided the channel SNR is greater then ~10dB. This is because small phase errors are not big enough to cause a bit error and thus no noise is seen on the received signal. However when the SNR of the channel becomes worse (< 10 dB), phase errors cause the received signal to be misread thus causing bit errors. Since the mapping of the bits to the amplitude of the transmitted signal are not linearly related, a single bit error can cause a significant error in received amplitude. This results in a rapid drop in performance of the received signal as the SNR of the channel drops below ~10dB for QPSK.

2.4.2 Results

Figure 28 shows how the signal to noise of the received image decreases as the signal to noise ratio of the transmission channel also worsens. It can be seen that the 256PSK signal performs better then the QPSK transmission below a channel SNR of 9dB.

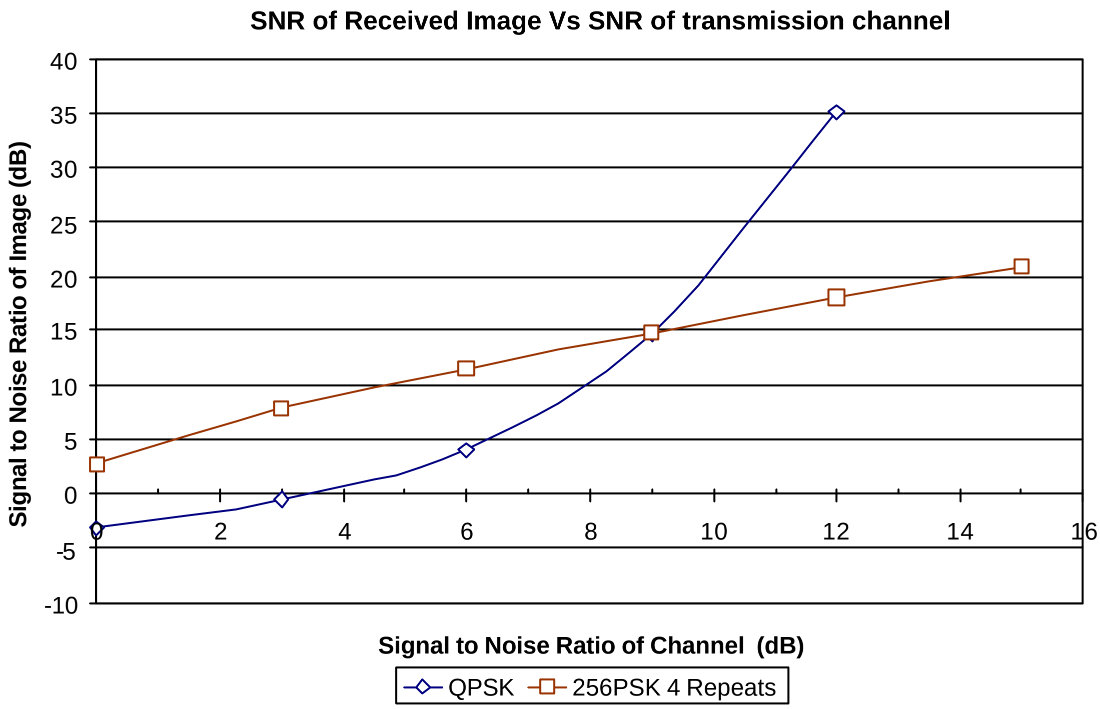


Figure 28 Comparison between the received image SNR using QPSK and 256PSK verses the SNR.

For a channel SNR of 0 – 6 dB the 256PSK signal averaged approximately 7 - 9 dB better picture quality then the QPSK signal. This gain is to be expected as there are two reasons for the improvement. Firstly, the signal has been averaged (thus reducing the phase noise), and that the phase angle mapping used for 256PSK, as the 256PSK signal is averaged over 4 repeats, giving approximately a 6 dB improvement compared with the channel SNR.

Table 14 shows some of the received images. It is easy to see that the image quality of the signal transmitted using 256PSK is much better then using QPSK.

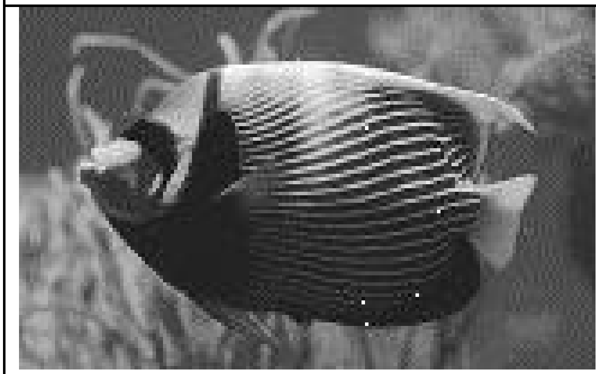
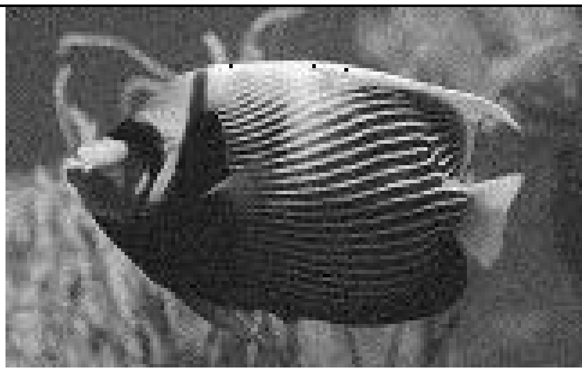
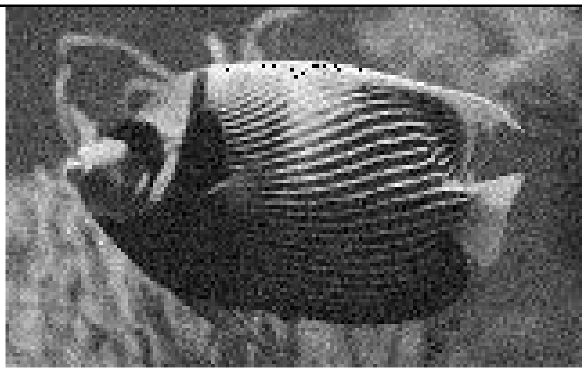
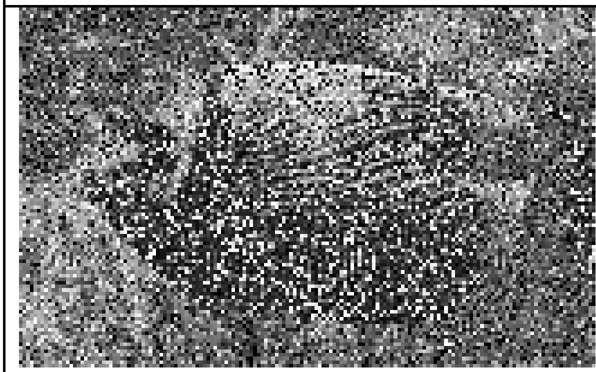
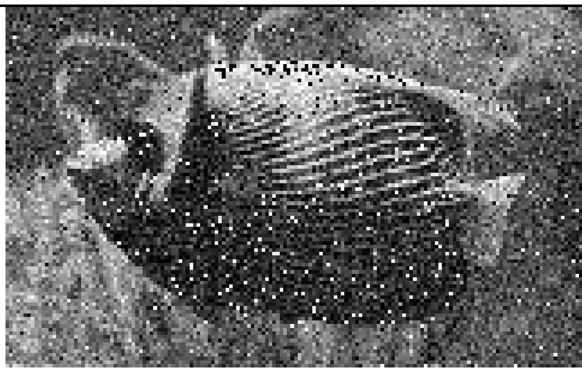
QPSK	256 PSK with 4 times data averaging
 SNR = 12 dB	 SNR = 12 dB
 SNR = 6 dB	 SNR = 6 dB
 SNR = 0 dB	 SNR = 0 dB

Table 14 Comparison between QPSK and 256PSK for transmitting an image under noisy conditions.

It was found that the picture sent using QPSK had a much better quality then the 256PSK signal in a low noise channel, where as the 256PSK performed better then QPSK under high noise conditions with a SNR < 9 dB.

Note: the performance of the QPSK signal can be significantly improved using advanced forward error correction techniques. This however cannot be applied to 256PSK using these same techniques as the matching between the phase mapping and the pixel intensity will be lost and the error is too high (typically 0.8-0.95 for SNR <12 dB) for most error correction to work successfully. The use of forward error correction techniques may result in the performance of QPSK and 256PSK being the same or QPSK performing better.

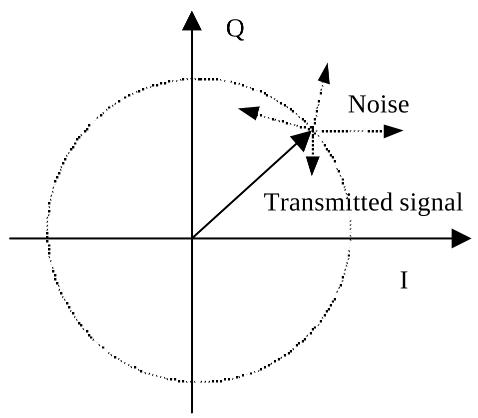
2.5 Mathematical Model for OFDM performance

The aim was to develop a mathematical model of the performance (BER) of OFDM verses the channel noise. This was so that the simulated results could be verified, and to get a more in depth understanding of the transmission mechanism.

The model developed is based on the transmission modulation technique being phase shift keying, and that the channel noise is gaussian noise (i.e. white noise).

2.5.1 RMS Demodulated Phase Error

If we assume that the transmission modulation method used is phase shift keyed then any noise added to the transmitted signal will result in a phase error. If we look at the IQ diagram of the transmitted signal then the transmitted signal will be a phasor of fixed magnitude, and of phase corresponding to the data to be transmitted. The noise can then be considered as the random vector added to the transmitted signal. The magnitude of the phase error depends on two things, the relative phase angle of the noise vector, and the magnitude of the noise vector.



The received vector will be the vector sum of the transmitted signal and the noise. If we assume that the noise is a constant magnitude vector equal to its RMS magnitude, and that it has a random phase angle then the problem of working out the received angle would be as follows.

2.5.2 BER verses Channel Noise

Figure 29 show the effect of noise on the received phase angle. If we let the amplitude of the transmitted signal be 1, and the length of the noise vector be A with angle ϕ, then the received phase error is θ_{err}.

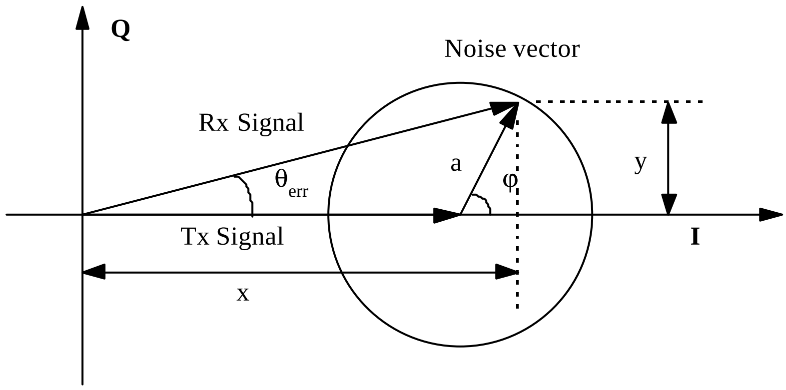


Figure 29 Received Phasor, showing effect of noise on the received phase angle.

Using trigonometry,

$$x = 1 + a\cos\varphi$$

$$y = a\sin\varphi$$

Since,

$$\theta_{err} = \tan^{-1}\left(\frac{y}{x}\right)$$

Therefore,

$$\theta_{err} = \tan^{-1}\left(\frac{a\sin\varphi}{1 + a\cos\varphi}\right)$$

The signal to ratio determines the relative amplitude of the received signal and the noise level. Since the signal is scaled to an amplitude of 1, the amplitude of the noise is:

$$a = \frac{1}{S_{NR}}$$

Note: The SNR is base on the amplitudes of the signals thus must be scaled correctly when converting it to dB.

If we substitute this in we get,

$$\theta_{err} = \tan^{-1}\left(\left(\frac{1}{S_{NR}}\right)\frac{\sin\varphi}{1+\left(\frac{1}{S_{NR}}\right)\cos\varphi}\right)$$

$$\theta_{err} = \tan^{-1}\left(\frac{\sin\varphi}{SNR+\cos\varphi}\right)$$

The noise signal can be of any phase angle. What we need is to find is the RMS phase error, so if we find the average phase error (assuming the noise phase angle is always positive) the this can be scaled to find the RMS error. The average phase angle can be found by integrating θ_{err} over a half circle, i.e. φ varies from 0 to π.

$$Av\theta_{err} = \frac{1}{\pi}\int_0^{\pi} \tan^{-1}\left(\frac{\sin\varphi}{SNR+\cos\varphi}\right)d\varphi$$

The RMS phase error will be greater by $\sqrt{2}$, thus

Equation 1

$$\boxed{RMS\theta_{err} = \frac{\sqrt{2}}{\pi}\int_0^{\pi} \tan^{-1}\left(\frac{\sin\varphi}{SNR+\cos\varphi}\right)d\varphi}$$

This equation was used to predict the RMS phase error for different channel SNRs. This was compared with the results obtained using the simulation of OFDM. Figure 30 shows that the predicted results based on the above mathematical derivation match the simulated results very well.

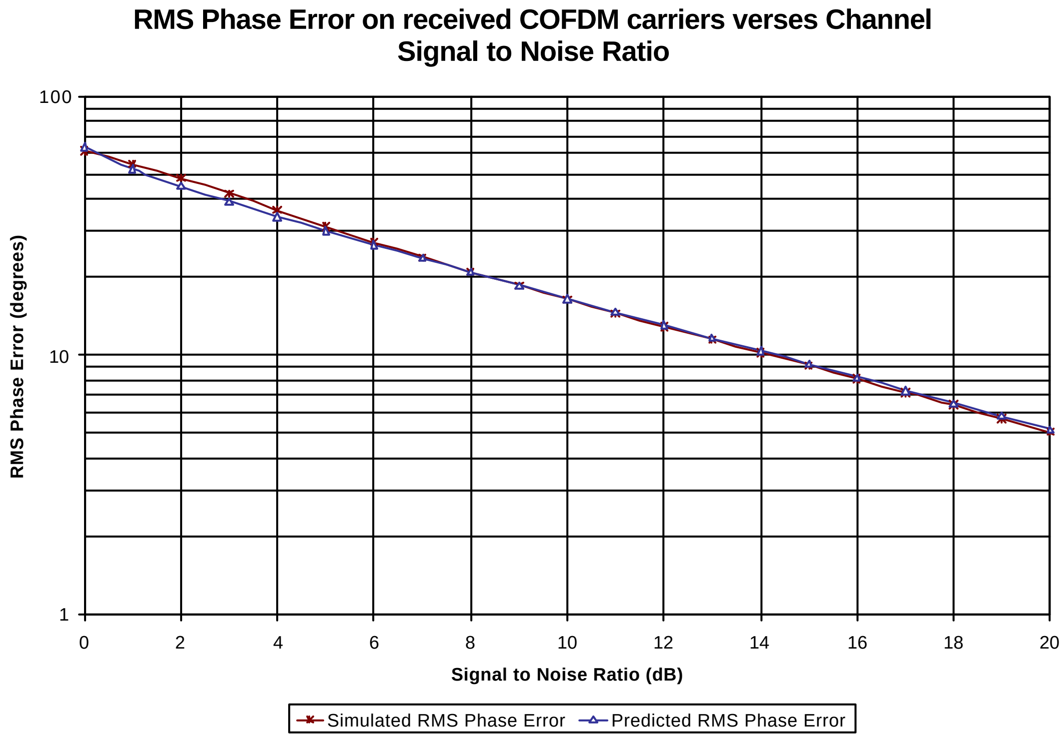


Figure 30 Comparison between the measured RMS phase error using the simulations and the predicted result. (Also shown in Table 22)

Once the RMS phase error has been calculated the BER can be easily calculated using simple statistics. The RMS phase error is the standard deviation of the phase error. An error will occur if the phase error gets bigger then the maximum allowed for the modulation method used. Thus the BER can be determined by finding the probability of the phase error being greater the plus, minus the maximum phase error for a standard deviation equal to the RMS phase error. This is outlined in the following example.

The Bit Error Rate (BER) of an OFDM link can be predicted based on the channel signal to noise ratio (SNR) and phase modulation used (e.g. BPSK, QPSK, etc). This is done by finding out what the expected RMS phase error ($\theta_{error(rms)}$) there will be on the signal (due to the channel noise). The bit error rate can then be found by comparing the magnitude of the RMS phase error to that of the maximum phase allowed for the particular phase modulation used (θ_{max}).

2.5.2.1 Maximum Allowable Phase Angle

θ_{max} is the maximum phase error allowed on the received symbol, before an error will occur on the received word.

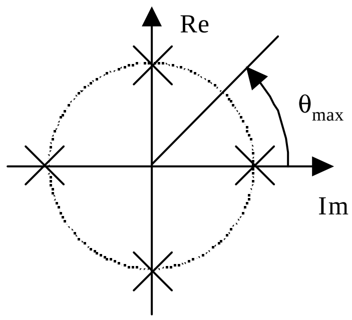


Figure 31 IQ diagram for QPSK, showing the phase locations for data (crosses) and that q_{max} is 45 degrees

Modulation Technique	Maximum Phase Error Allowed (q_{max}) in degrees
BPSK	90
QPSK	45
16PSK	11.25
256PSK	0.70313

Once θ_{max} and $\theta_{error(rms})$ have been established, Z can be calculated, and the BER found from Table 22.

$$Z = \frac{q_{max}}{q_{error(rms)}}$$

2.5.2.2 Example

For a QPSK transmission if the signal to noise ratio (SNR) of the channel is 10dB, find the BER:

Note: Table 21 shows are summary calculated from Equation 1

From Table 21, $\theta_{error(rms)}$ = 16.5 degrees,

For QPSK, θ_{max} = 45 degrees,

Therefore,

$$Z = \frac{45}{16.5} = 2.727$$

From Table 22 the BER is between 0.0053 and 0.0091, with a result of 0.0077 if the results are interpolated.

BER = 0.0077

2.6 OFDM system implementation

The proposed final application for OFDM is to use it for wireless communications systems such as cellular mobile phone systems, fixed wireless phone systems, wireless data links and wireless computer local area networks. If OFDM is to be used in any of these applications then the bandwidth used must be sufficiently high to compete with other radio technologies. This section discusses the processing power required to implement a practical OFDM system.

An OFDM system mainly involves digital signal processing, thus the main focus of the performance of the system depends on the availability of high performance signal processing. There are two main ways in which the OFDM signal can be processed, which are using a general purpose DSP, or by implementing the processing in hardware using customized IC's.

2.6.1 Using general purpose DSP's

There are several processing stages required to generate and receive an OFDM signal. However most of the processing is required in performing the Fast Fourier Transform (FFT).

The complexity of performing an FFT is dependent on the size of the FFT. The larger the FFT the greater the number of calculations required, however since as the symbol period is longer the increased processing required is less then the straight increase in processing to perform a single FFT. Table 15 shows the number of calculations required for an FFT (radix-2) of size N, and also the relative processing for various FFT sizes. It can be seen that because the symbol period increases with a larger FFT that the extra processing required is minimal.

Size FFT (N)	Total number of complex calculations (values from [21])	Relative processing required for OFDM generation (normalized to 1024 point FFT)
32	240	0.5
64	576	0.6
128	1344	0.7
256	3072	0.8
512	6912	0.9
1024	15360	1.0
2048	33792	1.1
4096	73728	1.2

Table 15 Processing complexity for FFT

The processing efficiency of a DSP processor depends on the architecture of the processor, however for most single instruction DSP's the number of cycles required to calculate an FFT is twice the total number of calculations shown in Table 15. This is due to complex calculations requiring two operations per calculation.

Required Processing Power

To get an estimate of the processing power required to implement a practical phone system, lets consider an example.

Basic System Parameter	Value
Total Bandwidth	1.25MHz
User Capacity	64 users
Modulation Used	QPSK
FFT size	2048
Guard period	512 samples

Table 16 Example OFDM system

Table 16 shows a example system. From these basic system parameters the required number of carriers, user data rate and symbol rate can be calculated. These are shown in

Derived System Parameter	Value
No. of active carriers	832
Data Rate of each User	39kbps
Useful symbol time	666µsec
Total symbol time	833µsec

Table 17 Derived system parameters for the example OFDM system

From Table 15 the number of complex calculations required for a 2048 point FFT is 33792. The maximum time that can be taken in performing the calculation is once every symbol thus once every 833 µsec. If we assume that the processor used requires 2 instructions to perform a single complex calculation, and that there is an overhead of 30% for scheduling of tasks and other processing. The minimum processing power required for this is then:

$$MIPS = \frac{33792 \times 2}{833 \times 10^{-6}} \times 1.3 \times 10^{-6} = 105$$

Thus the transmitter requires >105 MIPS in order to implement to OFDM transmitter. The receiver will require just as much as the transmitter, thus a full OFDM transceiver will require >210 MIPS. This is a lot of processing required. Most current cheap DSP's are only 25-50 MIPS (i.e. AD2181 is 33 MIPS). Currently the fastest general purpose DSP is produced by Texas Instruments. The TMS320C62xx is capable of up to 1600 MIPS, making it sufficiently fast enough for an OFDM transceiver. However, the price of the TMS320C62xx is not known and is expected to be very expensive.

(Addendum 10/2001: Texas Instruments are now producing the TMS320C64xx, which is the next generation of fixed point DSP's. These processors are capable of up to 4800 MIPS)

OFDM clearly requires a large amount of processing power, however since computer technology is advancing so fast this may not become a problem in the future.

2.6.2 Future DSP Processing Power

Computer technology, and signal processing technology is improving at a rapid rate. The IC industry is on a self-prescribed course of development of ever-smaller devices and faster circuits. By the year 2000, logic IC will employ up to 60 million transistors and operate at speeds of over 1GHz.

Figure 32 shows the expected performance increase in general purpose microprocessors until the year 2010. If this is considered as an indicator for DSP improvement in speed, this shows that for the next 5 years we can expect a speed improvement in processing of greater than 5 times. Thus an OFDM system will be easily achievable using general purpose DSPs in 5 years time.
(Addendum 10/2001: Texas Instruments announced during 1999 that they predict that by 2010, Digital Signal Processors will be capable of more than 1 Trillion Operations Per Second, www.ti.com/sc/docs/news/1999/99086.htm)

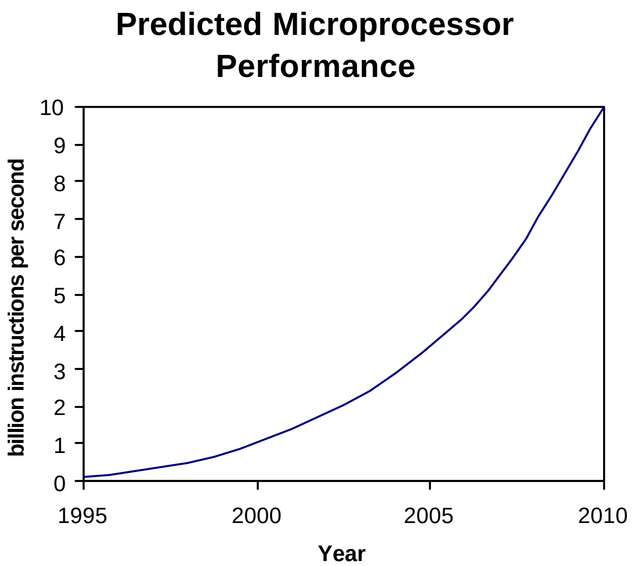


Figure 32 The performance of general-purpose microprocessors will climb from 100 million operations per second in 1995 to more than 10 billion by 2010 (Source [20])

2.6.3 Hardware FFT Implementation

Another way of implementing the FFT processing required for generating and receiving the OFDM signal is by doing the FFT's in hardware using FFT Integrated Circuits (IC). Several 1024-point FFT chips are shown in Table 18. For the previous example a 2048 point FFT needed to be performed in less then 830μsec. This is much greater then the execution time required by any of the FFT IC's in Table 18. Although these IC's only perform a 1024-point FFT, clearly the processing can be easily achieved using hardware implementation of the FFT processing.

Processor	Year	Datapath Width (bits)	Execution Time (1024-pt FFT)	Number of Chips Required	Pins	Power (mW)
Cobra (Colorada State)	1994	23	9.5μsec	>16	391	7700
PDSP16510A (Plessey)	1989	16	98μsec	1	84	3000
DSP-24 (DSP Architectures)	1997	24	21μsec	7	308	3500+
Spiffe ULP (Stanford)	1995	20	61μsec	1	70	8

Table 18 1024 point FFT Chip Comparison

3. CDMA Results

3.1 Simulated Model

3.1.1 Forward Link

The forward link of the CDMA system modelled uses orthogonal Walsh codes to separate the users. Each user is randomly allocated a Walsh code to spread the data to be transmitted.

The transmitted signals from all the users are combined together, then passed through a radio channel model. This allows for clipping of the signal, adding multipath interference, and adding white gaussian noise to the signal.

The receiver uses the same Walsh code that was used by the transmitter to demodulate the signal and recover the data. After the received signal has been despread using the Walsh code, it is sub-sampled back down to the original data rate. This is done by using an integrate-and-dump filter, followed by a comparator to decide whether the data was a 1 or a 0.

The received data is then compared with the original data transmitted to calculate the bit error rate (BER).

The RMS amplitude error is also worked out. The signal level after it has been demodulated and filtered is compared with the expected amplitude of the signal based on the transmitted data. The RMS amplitude error directly relates to the bit error rate, so is a useful measurement to make.

Figure 33 shows the model used for the simulations of the CDMA forward link.

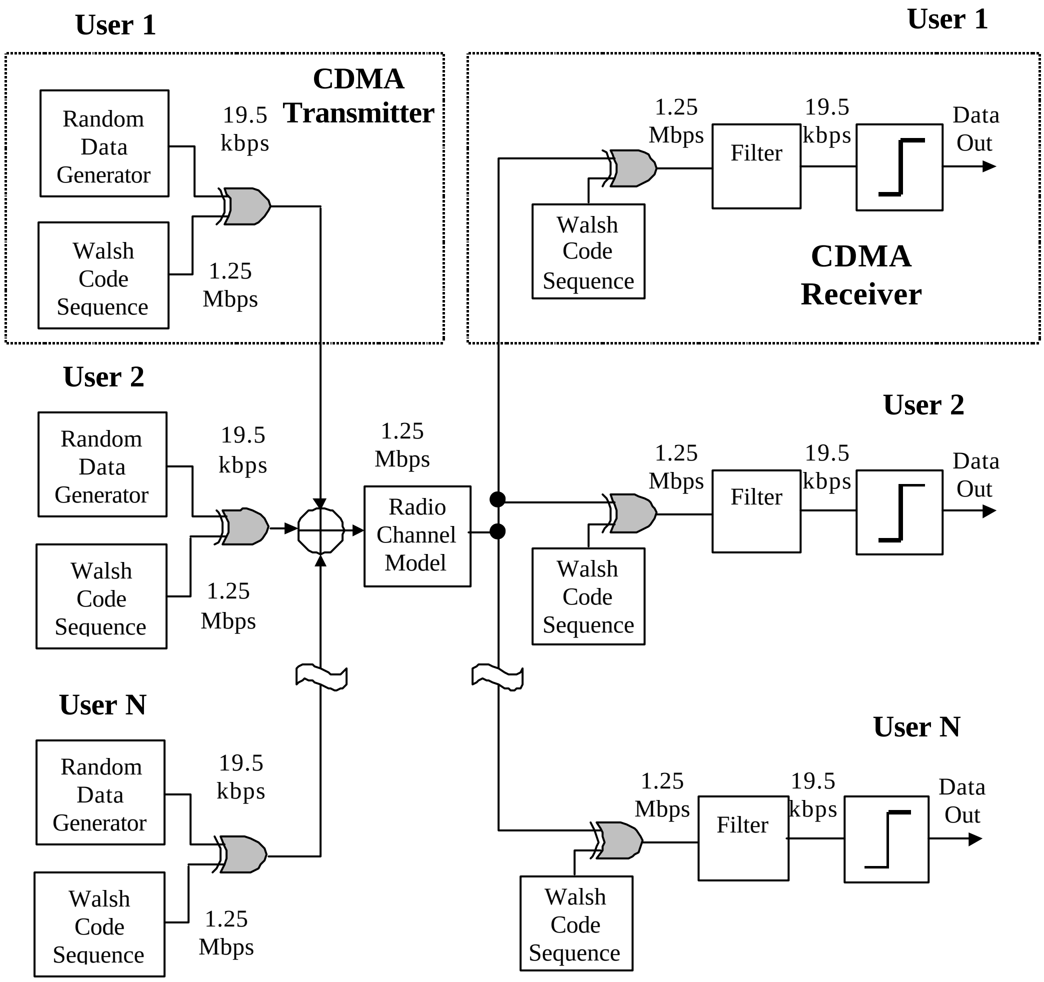


Figure 33 Model used for the CDMA forward link.

3.1.2 Reverse Path

The reverse link of the CDMA link was simulated in a very similar way to the forward link except that orthogonal Walsh codes are not used. As shown earlier it is extremely difficult to use orthogonal codes effectively in a reverse link from the mobiles to the base station, because of the difficulty in synchronizing the system accurately enough. Because of this simply long pseudo random codes were used instead of the Walsh codes.

3.2 Simulation Results

3.2.1 BER verses the number of users in a cell

The reverse links of a CDMA system, from the mobiles to the base station, use non-orthogonal codes, which are pseudo random noise codes (PN codes). This leads to the signals from each user interfering with each other. The signals transmitted by each user are uncorrelated with each other as each user uses a unique pseudo random sequence code, resulting in the signal appearing a noise to other users.

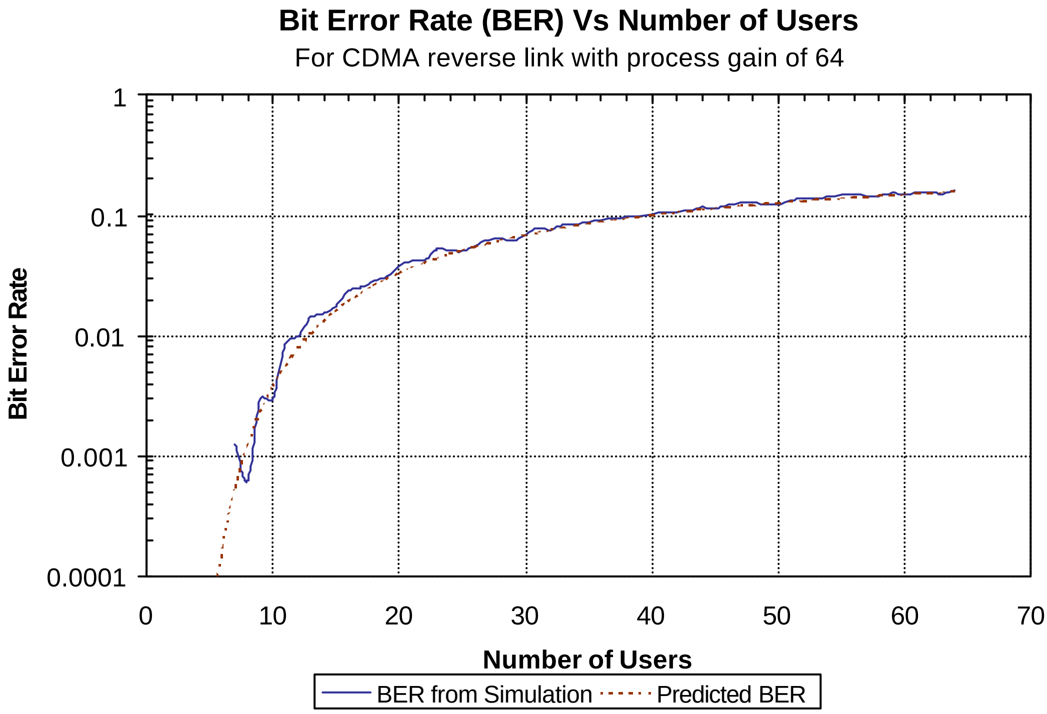


Figure 34 BER verse the number of users in a cell, for the reverse link in a CDMA system.

The BER for the reverse link of a CDMA system, increase as more users use the same cell. Figure 34 shows the BER expected base on the number of users in a cell. This result, is for an isolated cell with no interference from neighbouring cells, no multipath effects, and no channel noise. Any of these effects would worsen the BER. From Figure 34 it can be seen that the BER becomes significantly large if the number of users is greater then 8 users. This represents only 12.5% of the total user capacity of 64 users. The maximum number of users in the cell can be increased by using, advanced forward error correction, voice activity detection, and cell sectorization.

For computer transfer applications such as wireless local area networks, the data load is generally in bursts. This allows a reduction in the duty cycle of each user which is similar to voice activity detection. This can lead to a vast increase in the number of users possible as the interference from each user is reduced. However, it is at the cost of total data throughput of each user.

It is clear from the result obtained from Figure 34 that the inter-user interference in the reverse link is the weak point in the CDMA system. It is this interference that limits the cell capacity to approximately 8-12 users.

3.2.1.1 Multipath Immunity

CDMA is inherently tolerant to multipath delay spread signals as any signal that is delayed by more than one chip time becomes uncorrelated to the PN code used to decode the signal. This results in the multipath simply appearing as noise. This noise leads to an increase in the amount of interference seen by each user subjected to the multipath and thus increases the received BER.

Figure 35 shows the effect of delay spread on the reverse link of a CDMA system. It can be seen that the BER is essentially flat for delay spreads of greater than one chip time (0.8usec), which is to be expected as the reflected signal becomes uncorrelated.

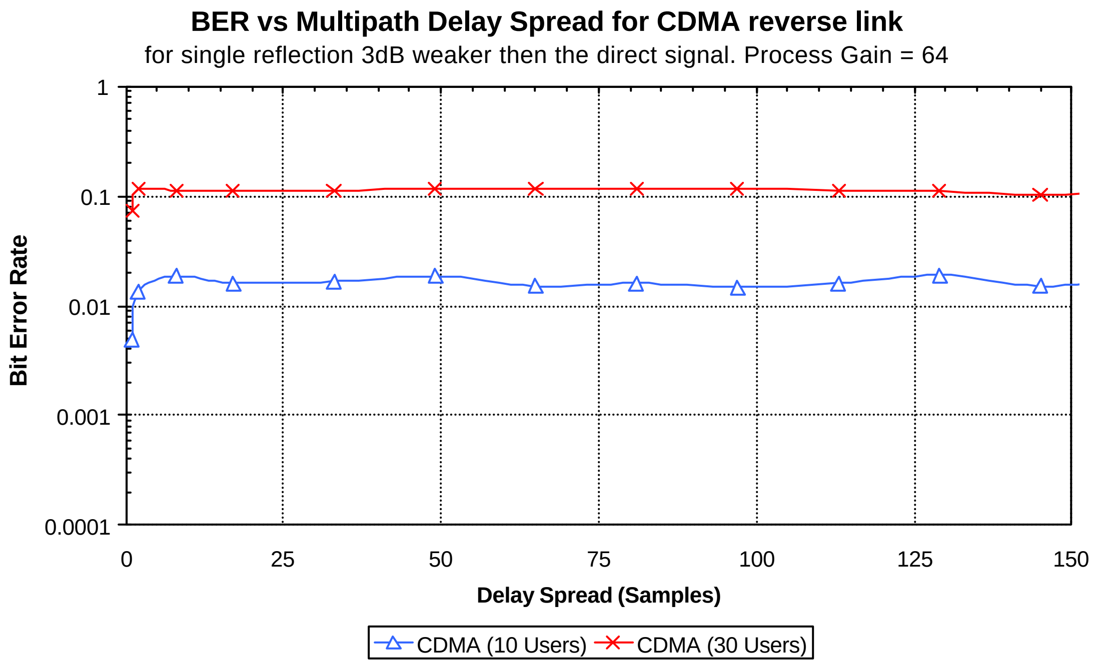


Figure 35 Effect of multipath delay spread on the reverse link of a CDMA system.

The multipath delay spread leads to an increase in the equivalent number of users in the cell, as it increases the amount of interference seen by the receiver.

Figure 36 shows a how the multipath power leads to an increase in the effective number of users in the cell. This simulation was performed using a fixed number of users in the CDMA link. A multipath signal of 10 samples in delay (to ensure that it is uncorrelated) was then added. It was found that as the amplitude of the reflected signal was increased, so did the bit error rate (BER). This BER was compared with Figure 34 to find out the equivalent number of users that result in the same BER.

The addition of multipath to the signal increases the total interference in the cell. The level of this increase is proportional to the number of users in the cell and the multipath signal strength. Figure 36 also shows the predicted result based on the increase interference power.

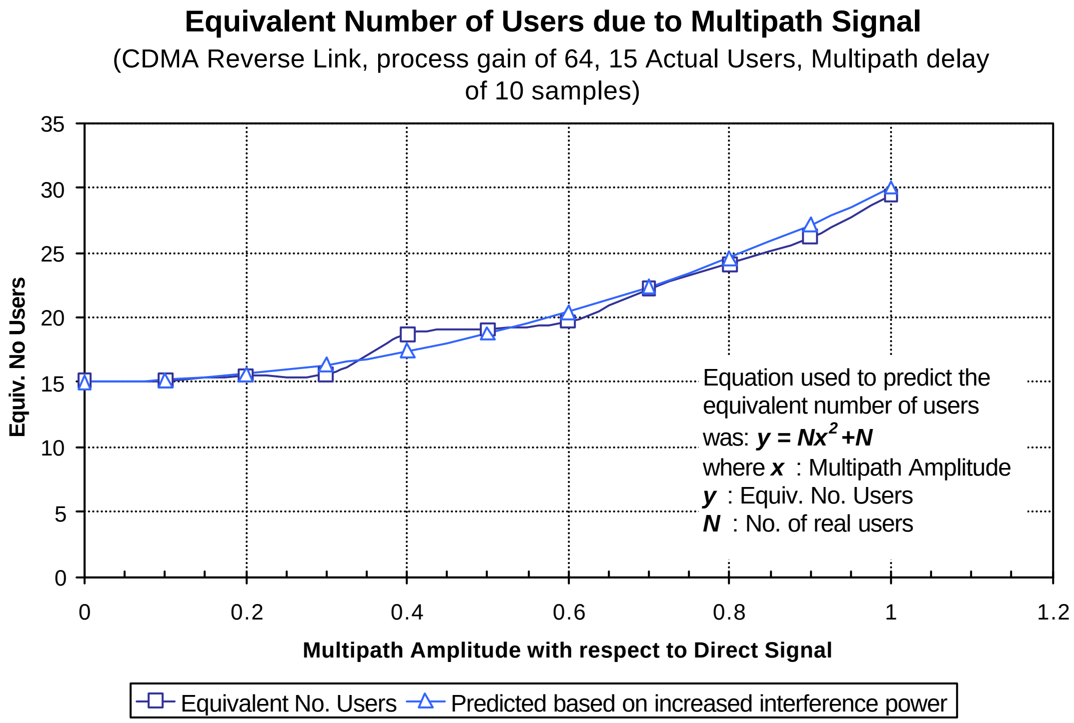


Figure 36 Interference increase seen by the receiver due to multipath delay spread.

3.2.1.2 Peak Power Clipping

The distortion tolerance of any transmission technique is very important, as it determines what type of power amplifier can be used, and how much dynamic range overhead is required. If a transmission technique is tolerant to peak power clipping, then it allows the signal to be clipped. This clipping of the signal reduces the peak to RMS signal power ratio thus allowing the signal power to be increased for the same sized transmitter. Figure 37 shows the effect of peak power clipping on both the reverse and forward links for CDMA.

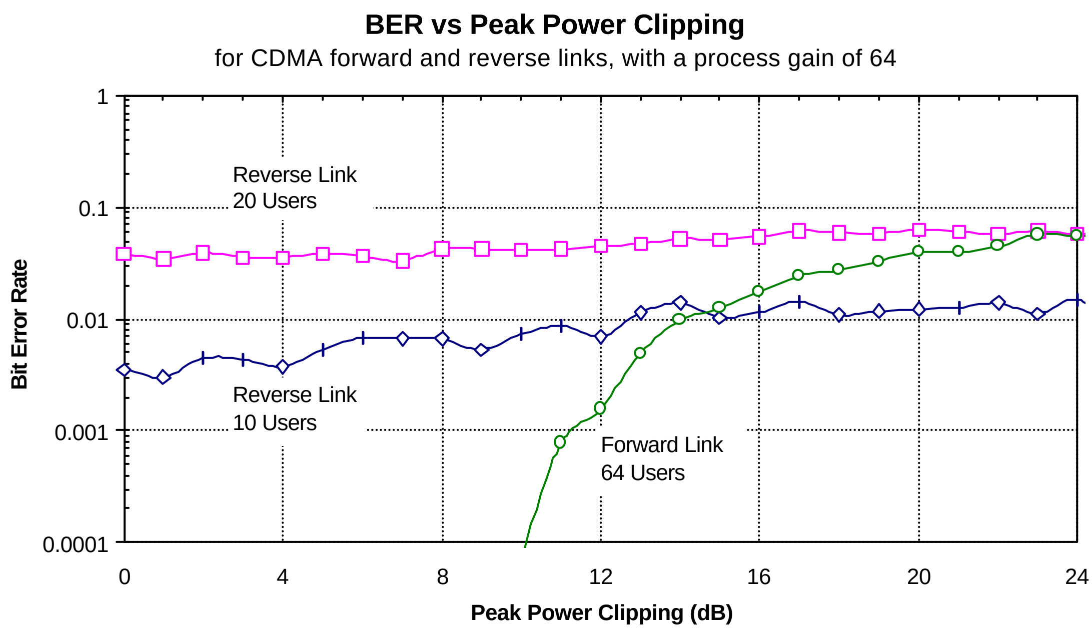


Figure 37 Effect of peak power clipping on the BER for the forward and reverse links of CDMA.

For the reverse link the BER starts high initially due to inter-user interference. The peak power clipping of the signal has little effect on the reverse link because the extra noise due to the distortion is not very high compared with the inter-user interference, plus any added noise is reduced by the process gain of the system. Peak power clipping for the reverse link is also likely to be small as clipping would only ever occur due to distortion in the base station receiver, as this is the only point where all the signals are combined. A well-designed receiver is unlikely to cause significant clipping of the signal and thus the result shown in Figure 37 is not very important.

(Addendum 10/2001: CDMA mobile phones require that the CDMA signal is band-pass filtered before being transmitted. This is to prevent interference between neighbouring channels. This band pass filtering changes the transmitted signal from a simple binary phase transmission (i.e. constant amplitude) to one that varies continuously in amplitude. This variable amplitude requires linear amplification, and is thus affected by distortion. This means that some distortion is likely to result in each of the mobiles, as well as the base station.)

The forward link result is more important as significant clipping of the transmitted signal could occur at the base station transmitter. The result for the forward link is completely different to the reverse link. The peak power clipping tolerance of the forward link is very similar to the result obtained for OFDM (see Figure 18 in section 2.2.2). The BER is low for a peak power clipping of less then 10dB, above which the orthogonal nature of the Walsh codes used begins to collapse.

3.2.1.3 Channel Noise

The noise performance of the CDMA reverse link is shown in Figure 38. This shows that the BER rises as the SNR of the channel worsen. Due to the high level of inter-user interference the addition of channel noise leads to only a gradual rise in the BER. The BER of each of the lines (10 user, 20 users and 30 users) approaches approximately the same BER at a SNR of 0dB. At 0dB the effect noise of the channel is the same as adding an additional 64 users to the cell, thus the difference between 10, 20 and 30 users becomes insignificant. The BER is very bad for more than 10 users regardless of the channel SNR, thus making 20 or 30 users unusable. However, for 10 users the BER becomes greater the 0.01 at approximately a SNR of 14dB, which is the maximum BER that can be normally tolerated for voice communications.

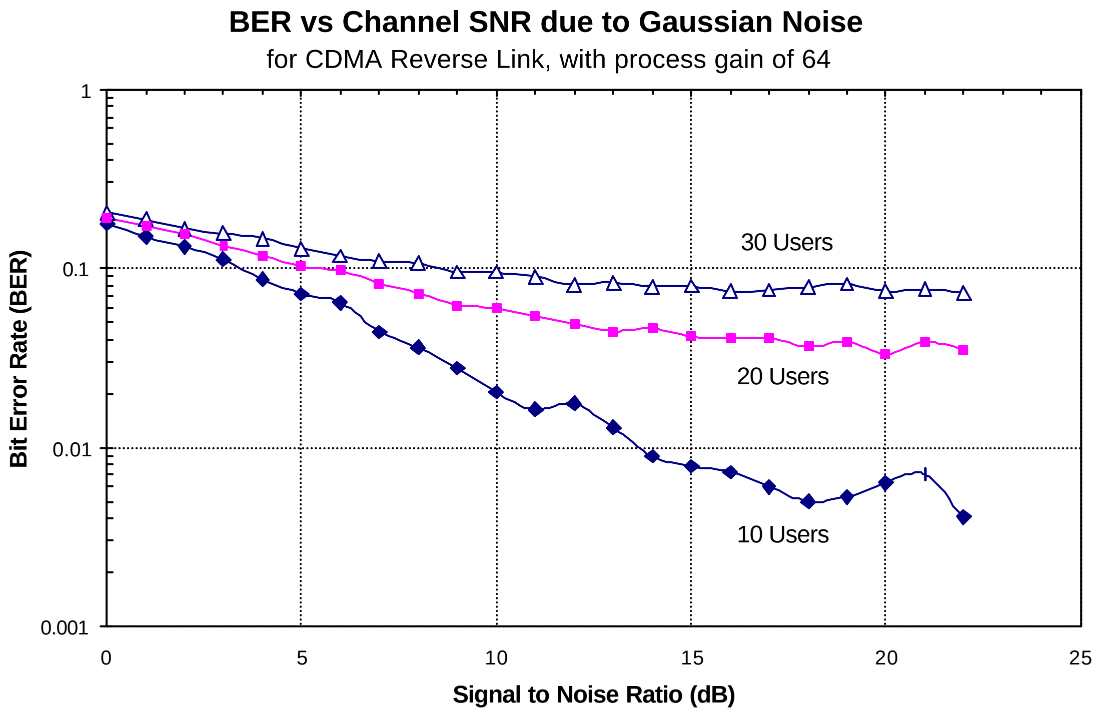


Figure 38 BER verses the radio channel SNR for the reverse link of a CDMA system.

3.3 Mathematical Model for Reverse Link

3.3.1 Cell Capacity for a CDMA system

The capacity of a CDMA system is limited by the reverse link. The reverse link uses uncorrelated, non-orthogonal PN codes, which makes it limited by interference from other users. Each other user appears as noise as additional noise to the cell. If we initially assume a single cell then the noise in the system will be determined by the number of users in the cell. If we let the number of users be N, and the transmitted power from each user to be S, the received signal will consists of the received signal power for the desired user (S) and the interference from N-1 other users, thus the signal to noise ratio will be.

$$SNR = \frac{S}{(N-1)S} = \frac{1}{N-1}$$

Since the noise in the channel is reduced by the process gain during demodulation, the noise on each data bit seen after demodulation will be less. The process gain is the ratio of the total bandwidth (W) to the base band information bit rate (R). Thus the received energy per bit to noise ratio (E_b/N_o) is

$$\frac{E_b}{N_o} = \frac{W}{R}\frac{1}{(N-1)}$$

The above equation does not take into account thermal noise. The thermal noise simply increased the effective amount of noise. Let the thermal noise be ***n***. Thus, the E_b/N_o becomes

$$\frac{E_b}{N_o} = \frac{W}{R}\frac{1}{(N-1)+n/S}$$

In order to achieve an increased capacity, the interference from users needs to be reduced. This can be achieved by monitoring the voice activity so that the transmitter is switched off during periods of no voice activity. This reduces the effective interference level by the reduced duty cycle of the transmitted signal. Using antenna sectorization can also reduce the interference. If for example the cell was sub divided using three antennas, each having a beam width of 120°, then the interference seen by each antenna is one third that of an omni-direction antenna. If we let ***d*** be the duty cycle of the voice activity, and ***G*** be the cell sectorization then equation becomes

$$\frac{E_b}{N_o} = \frac{W}{R\left[(N-1)\frac{d}{G}+n/S\right]}$$

Thus the capacity of a single cell CDMA system would be

Equation 2

$$N = \frac{G}{d}\left[\frac{N_o}{E_b}\frac{W}{R} - \frac{n}{S}\right] + 1$$

Where:

G is the antenna sectorization,

d is the voice duty cycle,

E_b/N_o is the energy per bit to noise ratio,

W is the total transmission bandwidth,

R is the base band bit rate,

n/S is the ratio of received thermal noise to user signal power.

3.3.2 Capacity of a single CDMA cell

The cell capacity of a CDMA system is dependent on the bandwidth used the process gain and the allowable error rate. For this discussion we will consider the a system with the same bandwidth and user data rate the same as the OFDM example system in section 2.6.

The OFDM example used a bandwidth of 1.25MHz. The OFDM system could handle 64 users each at 39kbps, or 128 users at 19.5kbps depending on the spectrum allocation. For CDMA if we use a process gain of 64, this will give each user a data rate capacity of 19.5kbps, making it comparable to the 128 user OFMD system. Since the capacity of a CDMA system is dependent on the noise tolerance of data if we assume an E_b/N_o of 8dB this will give a BER of ~0.006 which is acceptable for voice communications (see Table 23 in Appendix III for more detail). For a the CDMA link that has no voice detection activity and no cell sectorization then cell capacity can be calculated using Equation 2 as follows:

G = 1, **d** = 1, **E_b/N_o** = 8dB = 6.31, **W** = 1.25MHz, **R** = 19.5kHz and **n/S** = 0 (Assume no thermal noise)

From Equation 2

$$N = 1\left[\frac{1.25\times10^{6}/19530}{6.31}\right]+1 = 10.1+1 = 11.1$$

This gives a spectral efficiency of only:

$$= \frac{11.1 \times 19530}{1.25 \times 10^6} = 0.173 bits / Hz$$

This result is pretty poor as the cell capacity is more then 10 times lower then it was for OFDM. However the efficiency of CDMA can be improved by using voice detection to reduce the duty cycle of each user, and by using cell sectorization. Note however, that voice activity detection can only be used for voice communications and not for general data transfer. Thus all it is effectively doing is reducing the data throughput allowed for each user.

Appling both voice duty cycle detection and cell sectorization the effective capacity is increased. If we assume that the cell is split three ways then the ideal cell sectorization factor will be 3. However, side-lobes of the antennas used will always reduce this, there reducing the factor to about 2.55.

Using **G** = 2.55, **d** = 0.4 (i.e. 40%) the cell capacity becomes:

$$N = \frac{2.55}{0.4}\left[\frac{1.25 \times 10^6 / 19530}{6.31}\right] + 1 = 65.7$$

The spectral efficiency is thus

$$= \frac{65.7 \times 19.53 \times 10^3}{1.25 \times 10^6} = 1.026 bit / Hz$$

This is still half the capacity of the OFDM system, and it comes at the cost of reduced total data throughput. Table 19 shows how the over capacity of the CDMA system changes depending on what BER is allowed. This is different to OFDM as the BER is ideally 0 for the 128 user as in the example.

E_b/N_o (dB)	Expected Bit Error Rate (BER)	Max. No. Users for single cell (no voice detection, no cell sectorization)	Spectral Efficiency (bits/Hz)	Max. No. Users for single cell (voice detection, cell sectorization)	Spectral Efficiency (bits/Hz)
6	0.023007	17.1	0.267	103.6	1.62
8	0.006004	11.1	0.173	65.7	1.03
10	0.000783	7.4	0.116	41.8	0.65
12	3.43E-05	5.0	0.078	26.5	0.41

Table 19 Predicted cell capacity for a single CDMA cell with process gain of 64, depending on the tolerable E_b/N_o

3.3.3 Capacity of CDMA and OFDM with Multiple Cells

With any cellular system, interference from neighbouring cells lowers the overall capacity of each cell. For conventional FDMA and TDMA systems, each cell must have a different operating frequency from its immediate neighbouring cells. This is to reduce the amount of interference to an acceptable level. The frequencies are reused in a pattern, with the spacing between cells using the same frequency determined by the reuse factor. The required frequency reuse factor depends on the interference tolerance of the transmission system. Analog systems typically require a carrier to interference ratio (C/I) of greater then 18dB [18], which requires reuse factor of 1/7 (see Figure 39 (b)). Most digital systems only require a C/I of 12dB, allowing a reuse factor of 1/3-1/4 (see Figure 39 (a)). CDMA however, uses the same frequency in all cells, thus ideally allowing a reuse factor of 1 (see Figure 39).

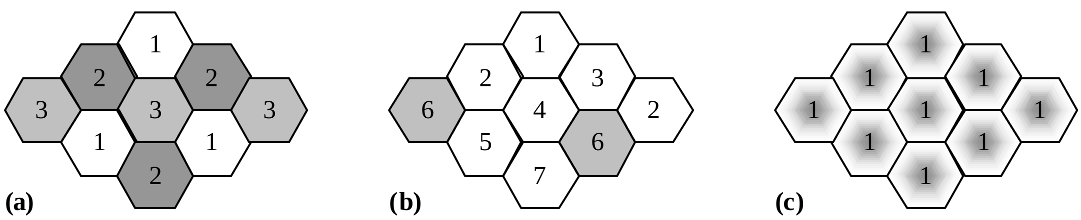


Figure 39 Frequency reuse patterns for (a) 3 frequencies (Digital systems), (b) 7 frequencies (Analog FDMA), (c) CDMA

In practice, the frequency reuse efficiency of CDMA is somewhat lower 1, as neighbouring cells cause interference, thus reducing the user capacity of both systems. The frequency reuse factor for a CDMA system is about 0.65 [16]. Figure 40 shows the interference from neighbouring cells. Note that most of the neighbouring interference is from the immediate neighbours of the cell.

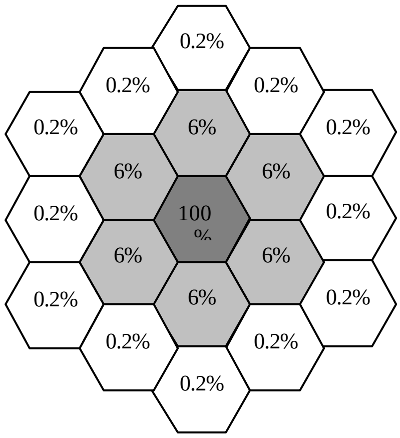


Figure 40 Interference contributions from neighbouring cells in a CDMA system (source [16]).

The cell capacity for a multi-cellular CDMA system equal to the single cell capacity reduced by the frequency reuse factor. Table 20 shows the effect of this on the CDMA capacity. The cell capacity for CDMA is very low if voice activity detection and cell sectorization is used. A straight CDMA system can only have somewhere between 5-11 users/cell/1.25MHz. Using cell sectorization and voice activity detection allows the capacity to be increased by up to 6.4 time, allowing somewhere between 30-70 user/cell/1.25MHz.

OFDM would require a frequency reuse pattern to be used in a multi-cellular environment to reduce the level of inter-cellular interference. The C/I required would need to be greater then ~12dB (see Gaussian Noise Tolerance of OFDM, section 2.2.3). This could be done with a frequency reuse factor of ~3. This should easily be able to be achieved as cell sectorization could also be used to reduce the level of interference. This would result in the cell capacity for an OFDM system of approximately 128/3 = 42.7 users/cell/1.25MHz in a multicellular environment. The matches the same user capacity as CDMA for an E_b/N_o of 8dB and using cell sectorization and voice detection.

E_b/N_o (dB)	Expected Bit Error Rate (BER)	Max. No. Users for single cell (no voice detection, no cell sectorization)	Max. No. Users for single cell (voice detection, cell sectorization)
6	0.023007	11.1	67.3
8	0.006004	7.2	42.7
10	0.000783	4.8	27.2
12	3.43E-05	3.3	17.2

Table 20 Predicted cell capacity for a CDMA cell in a multi-cellular environment, for a process gain of 64.

4. Conclusion

The current status of the research is that OFDM appears to be a suitable technique as a modulation technique for high performance wireless telecommunications. An OFDM link has been confirmed to work by using computer simulations, and some practical tests performed on a low bandwidth base-band signal. So far only four main performance criteria have been tested, which are OFDM's tolerance to multipath delay spread, channel noise, peak power clipping and start time error. Several other important factors affecting the performance of OFDM have only been partly measured. These include the effect of frequency stability errors on OFDM and impulse noise effects.

OFDM was found to perform very well compared with CDMA, with it out-performing CDMA in many areas for a single and multicell environment. OFDM was found to allow up to 2-10 times more users than CDMA in a single cell environment and from 0.7 - 4 times more users in a multi-cellular environment. The difference in user capacity between OFDM and CDMA was dependent on whether cell sectorization and voice activity detection is used.

It was found that CDMA only performs well in a multi-cellular environment where a single frequency is used in all cells. This increases the comparative performance against other systems that require a cellular pattern of frequencies to reduce inter-cellular interference.

One important major area, which hasn't been investigated, is the problems that may be encountered when OFDM is used in a multiuser environment. One possible problem is that the receiver may require a very large dynamic range in order to handle the large signal strength variation between users.

This thesis has concentrated on OFDM, however most practical system would use forward error correction to improve the system performance. Thus more work needs to be done on studying forward error correction schemes that would be suitable for telephony applications, and data transmission.

Several modulation techniques for OFDM were investigated in this thesis including BPSK, QPSK, 16PSK and 256PSK, however possible system performance gains may be possible by dynamically choosing the modulation technique based on the type of data being transmitted. More work could be done on investigating suitable techniques for doing this.

OFDM promises to be a suitable modulation technique for high capacity wireless communications and will become increasing important in the future as wireless networks become more relied on.

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Appendix I. Acronyms

AMPS	Advanced Mobile Phone System
BER	Bit Error Rate. Probability of a data word being transmitted being in error.
BPSK	Binary Phase Shift Keying
CDMA	Code Division Multiple Access
COFDM	Coded Orthogonal Frequency Division Multiplexing
DAB	Digital Audio Broadcasting system
DS-CDMA	Direct Sequence Code Division Multiple Access
E_bN_o	Energy per bit to noise energy ratio (similar to SNR)
FDM	Frequency Division Multiplexing
FDMA	Frequency Division Multiple Access
FFT	Fast Fourier Transform
FH-CDMA	Frequency Hopping Code Division Multiple Access
FIR	Finite Impulse Response Filter
GSM	Global System for Mobile telecommunications
IFFT	Inverse Fast Fourier Transform
IS-95	International Standard for the CDMA phone system developed by Qualcomm Inc.
LEO	Low Earth Orbit satellite
OFDM	Orthogonal Frequency Division Multiplexing
QAM	Quadrature Amplitude Modulation
QPSK	Quadrate Phase Shift Keying
RMS	Root Mean Square value
SNR	Signal to Noise Ratio
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Access
UMTS	Universal Mobile Telecommunications System
VCR	Video Cassette Recorder

Appendix II. OFDM Guassian Noise Performance Prediction

The performance of any OFDM system using phase shift keying can be worked out using the Table 21 and Table 22.

SNR (dB)	**RMS Phase Error (degrees)** $q_{error(rms)}$
0	63.63
2	44.85
4	34.25
6	26.65
8	20.92
10	16.5
12	13.05
14	10.34
16	8.198
18	6.505
20	5.164
22	4.1
24	3.256
26	2.586
28	2.054
30	1.631
32	1.296
34	1.029
36	0.8175
38	0.6494
40	0.5158
42	0.4097
44	0.3254
46	0.2585
48	0.2053
50	0.1631

Table 21 Expected Phase Error on a OFDM carrier at difference SNR levels

Z (number of standard deviations)	BER
0	1
0.2	0.841481
0.4	0.689157
0.6	0.548506
0.8	0.423711
1	0.317311
1.2	0.230139
1.4	0.161513
1.6	0.109599
1.8	0.071861
2	0.0455
2.2	0.027807
2.4	0.016395
2.6	0.009322
2.8	0.00511
3	0.0027
3.2	0.001374
3.4	0.000674
3.6	0.000318
3.8	0.000145
4	6.34E-05
4.2	2.67E-05
4.4	1.08E-05
4.6	4.23E-06
4.8	1.59E-06
5	5.74E-07

Table 22 Expected Bit Error Rate for various noise levels. Z is the ratio of the maximum allowable phase angle / RMS phase error.

Appendix III. BER verses Eb/No for a CDMA system

Table 23 shows the bit error rate (BER) that would occur for a CDMA system that does not use forward error correction. The energy per bit to noise ratio (Eb/No), is the energy in the demodulated data bit, to the noise energy in the same bit. It is similar to the signal to noise ratio. The Eb/No is the effective signal to noise ratio of the demodulated, despread CDMA signal. Any noise, or interference in the radio channel is reduced by a factor equal to the process gain during despreading. The minimum allowable Eb/No that can be used for a particular system depends on the forward error correction scheme used, and the type of data being sent. Voice communications typically requires a BER better then ~1/100 or 0.01. This is assuming some forward error correction is used.

Eb/No (dB)	BER
0	0.158655
2	0.104029
4	0.056495
6	0.023007
8	0.006004
10	0.000783
12	3.43E-05
14	2.7E-07

Table 23 Shows the Expected BER verses the energy per bit to noise ratio for a CDMA system