Thesis on

Evaluation Performance between OFDM and CDMA in perspective of wireless communication.

Supervised By
Munawwar Mahmud Sohul

Thesis Submitted By

Md. Waheed
Md. Lutful Karim
Md. Faisal Amin

Spring 2007.

Statement on Access to Thesis

We, the undersigned, the authors of this thesis, understand the BRAC University
will make it available for use within the University Library. All users consulting this thesis will have to sign the following statement:

"*In consulting this thesis I agree not to copy or closely paraphrase it in whole or part without the written consent of the author; and to make proper written acknowledgment for any assistance which I obtained from it.*"

Beyond this, we do not wish to place any restriction on access to this thesis.

Acknowledgements

We would like to thank our supervisor Monnuwar Mahmood Shahul for the endless hours of help, suggestions, ideas and advice during the development of this thesis. We would also like to thank Aneeta Quader for her help in MATLAB simulation.

Declaration

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any form for another degree or diploma at any university or other institution for Information derived from published or unpublished work of others has been acknowledged in the text and a list of references is given.

Abstract

In recent years orthogonal frequency division multiplexing (OFDM) and code division multiple access (CDMA) systems have gained considerable attention due to their use in high speed wireless communication. Both OFDM and CDMA have distinguishing features, for example, the former is almost completely immune to multipath fading effects, and the later has multi-user capability. Orthogonal frequency division multiplexing-code division multiple access (OFDM-CDMA) attempts to combine these features, so that we can achieve higher data rates for multiple users simultaneously.

The OFDM technique is an interesting approach in mobile communications in order to achieve a high spectral efficiency and to combat the frequency selectivity of the channel. OFDM effectively squeezes multiple modulated carriers tightly together, reducing the required bandwidth but keeping the modulated signals orthogonal so they do not interfere with each other.

CDMA is different than traditional ways like FDMA, TDMA in that it does not allocate frequency or time in user slots but gives the right to use both to all users simultaneously. To do this, it uses a technique known as Spread Spectrum. In effect, each user is assigned a code which spreads its signal bandwidth in such a way that only the same code can recover it at the receiver end.

OFDM performs extremely well compared with CDMA, providing a very high tolerance to multipath delay spread, peak power clipping, and channel noise.

In addition it provides a high spectral efficiency and resistance to multipath make it an extremely suitable technology to meet the demands of wireless data traffic. This has made it not only ideal for such new technology like WiMAX and Wi-Fi but also currently one of the prime technologies being considered for use in future Next Generation Networks (NGN).

TABLE OF CONTENTS

Chapter 1

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 result 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 canalizations.

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 Evolution of Mobile Communication

First generation (1G)

1G analog system for mobile communications saw two key improvements during the 1970s: the invention of the microprocessor and the digitization of the control link between the mobile phone and the cell site. AMPS(Advance mobile phone system) was first launched by US which is 1G mobile system. It is best on FDMA technology which allows users to make voice calls within one country.

Second generation (2G)

2G digital cellular systems were first developed at the end of the 1980s. These systems digitized not only the control link but also the voice signal. The new system provided better quality and higher capacity at lower cost to consumers. GSM (Global system for mobile communication) was the first commercially operated digital cellular system which is based on TDMA.

Third generation (3G)

All 2G wireless systems are voice-centric. GSM includes short message service (SMS), enabling text messages of up to 160 characters to be sent, received and viewed on the handset. Most 2G systems also support some data over their voice paths, but at painfully slow speeds usually 9.6 Kb/s or 14.4 Kb/s. So in the world of 2G, voice remains king while data is already dominant in wireline communications. And, fixed or wireless, all are affected by the rapid growth of the Internet. Planning for 3G started in the 1980s. Initial plans focused on multimedia applications such as videoconferencing for mobile phones. When it became clear that the real killer application was the Internet, 3G thinking had to evolve. As personal wireless handsets become more common than fixed telephones, it is clear that personal wireless Internet access will follow and users will want broadband Internet access wherever they go.
Today's 3G specifications call for 144 Kb/s while the user is on the move in an automobile or train, 384 Kb/s for pedestrians, and ups to 2 Mb/s for stationary users.

That is a big step up from 2G bandwidth using 8 to 13 Kb/s per channel to transport speech signals.

The second key issue for 3G wireless is that users will want to roam worldwide and stay connected. Today, GSM leads in global roaming. Because of the pervasiveness of GSM, users can get comprehensive coverage in Europe, parts of Asia and some U.S. coverage. A key goal of 3G is to make this roaming capacity universal.

A third issue for 3G systems is capacity. As wireless usage continues to expand, existing systems are reaching limits. Cells can be made smaller, permitting frequency reuse, but only to a point. The next step is new technology and new bandwidth. International Mobile Telecommunications-2000 (IMT-2000) is the official International Telecommunication Union name for 3G and is an initiative intended to provide wireless access to global telecommunication infrastructure through both satellite and terrestrial systems, serving fixed and mobile phone users via both public and private telephone networks. GSM proponents put forward the universal mobile telecommunications system (UMTS), an evolution of GSM, as the road to IMT-2000. Alternate schemes have come from the U.S., Japan and Korea. Each scheme typically involves multiple radio transmission techniques in order to handle evolution from 2G. Agreeing on frequency bands for IMT-2000 has been more difficult and the consensus included five different radio standards and three widely different frequency bands. They are now all part of IMT-2000. To roam anywhere in this "unified" 3G system, users will likely need a quintuple-mode phone able to operate in an 800/900 MHz band, a 1.7 to 1.9 GHz band and a 2.5 to 2.69 GHz band.

Third-generation wireless also requires new infrastructure. There are two mobility infrastructures in wide use. GSM has the mobile access protocol, GSM-MAP. The North American infrastructure uses the IS-41 mobility protocol. These protocol sets define the messages passed between home location registers and visitor location registers when locating a subscriber and the messages needed to deal with hand-offs as a subscriber moves from cell to cell. 3G proponents have agreed on an evolution path so that existing operators, running on either a GSM-MAP or an IS-41 infrastructure, can interoperate. But the rest of the landline infrastructure to support IMT-2000 will be in flux in the near future.

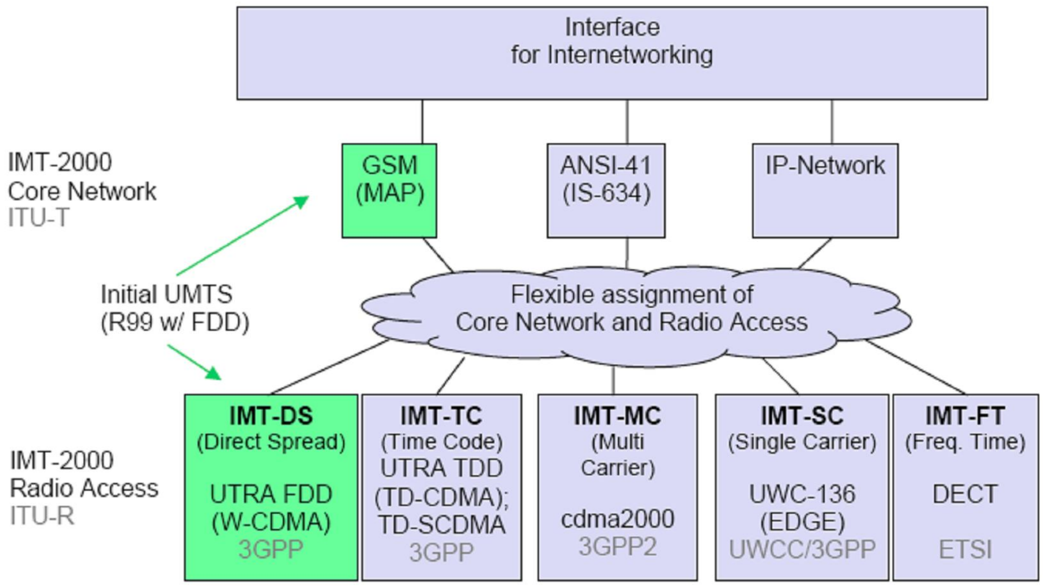


Fig1.1: IMT-2000 family

Transport Technology	Description	Typical Use / Data Transmission Speed	Pros/cons
TDMA	Time Division Multiple Access is 2G technology	Voice and data Up to 9.6kbps	Low battery consumption, but transmission is one-way, and its speed pales next to 3G technologies
GSM	Global System for Mobile Communications is a 2G digital cell phone technology	Voice and data. This European system uses the 900MHz and 1.8GHz frequencies. In the United States it operates in the 1.9GHz PCS band up to 9.6kbps	Popular around the globe. Worldwide roaming in about 180 countries, but GSM's short messaging service (GSM-SMS) only transmits one-way, and can only deliver messages up to 160 characters long
GPRS	General Packet Radio Service is a 2.5G network that supports data packets	Data Up to 115kbps; the AT&T Wireless GPRS network will transmit data at 40kbps to 60kbps	Messages not limited to 160 characters, like GSM SMS
EDGE	Enhanced Data GSM Environment is a 3G digital network	Data Up to 384kbps	May be temporary solution for operators unable to get W-CDMA licenses
CDMA	Code Division Multiple Access is a 2G technology developed by Qualcomm that is transitioning to 3G		Although behind TDMA in number of subscribers, this fast-growing technology has more capacity than TDMA
W-CDMA (UMTS)	Wideband CDMA (also known as Universal Mobile Telecommunications System-UMTS) is 3G technology. On November 6, 2002, NTT DoCoMo, Ericsson, Nokia, and Siemens agreed on licensing arrangements for W-CDMA, which should set a benchmark for royalty rates	Voice and data. UMTS is being designed to offer speeds of at least 144kbps to users in fast-moving vehicles Up to 2Mbps initially. Up to 10Mbps by 2005, according to designers	Likely to be dominant outside the United States, and therefore good for roaming globally. Commitments from U.S. operators are currently lacking, though AT&T Wireless performed UMTS tests in 2002. Primarily to be implemented in Asia-Pacific region
CDMA2000 1xRTT	A 3G technology, 1xRTT is the first phase of CDMA2000	Voice and data Up to 144kbps	Proponents say migration from TDMA is simpler with CDMA2000 than W-CDMA, and that spectrum use is more efficient. But W-CDMA will likely be more common in Europe
CDMA2000 1xEV-DO	Delivers data on a separate channel	Data Up to 2.4Mbps	(see CDMA2000 1xRTT above)
CDMA2000 1xEV-DV	Integrates voice and data on the same channel	Voice and data Up to 2.4Mbps	(see CDMA2000 1xRTT above)

Table1.1: Evolution of mobile communication from 1G to 3G.

Next Generation Network (4G)

With technological advancement and social changes, a proliferation of access networks (e.g. 2G, 3G, WLAN, HIPERLAN) with diverse data rates and quality- of-service (QoS) requirements have emerged. The operators have invested heavily into these networks and are working hard to maintain (even expand) their market share by improving the services (such as throughput and data rate). It is difficult to predict which of these technologies will ultimately come on top, reinforcing the likelihood of their coexistence in the future. Next Generation Mobile Network (NGMN) (at times also referred to by the networking jargon: 4G or beyond 3G (B3G)) is expected to offer ubiquitous roaming across these networks by inter-connecting these and emerging technologies through a common Internet protocol (IP) based platform, thereby providing end-to-end IP connectivity between peer end terminals. Inter-connectivity through a common platform will enable individual networks to evolve independently (through the adoption/modification of new/current system) while at the same time allow newer technologies to seamlessly integrate with the NGMN framework. IP is selected as he underlying transport technology to streamline wireless networks towards global Internet, and to support the growing number of wireless users, new applications and addressing requirements. In terms of the radio network, whether a common radio interface will be adopted across the entire NGMN to support high data rates or multiple interfaces (e.g. FDMA, TDMA, Narrowband CDMA, Wideband CDMA, OFDM) in their current form will continue to coexist is an open issue. This is influenced as much by the operator’s willingness to invest into the new interface (in line with the market trend) as by the need to safeguard their present investment.

Deploying a common multi-access radio system (such as MIMO-OFDM, TD-CDMA, CDMA-HSDPA, etc. offering high speed options) across NGMN would involve considerable capital investment which will ultimately lead to a concerted effort by the operators to resist changes in the existing radio interfaces. So far no compelling winner has emerged to dictate the transition since it is difficult to reach a consensus regarding the priority of one radio system over another.

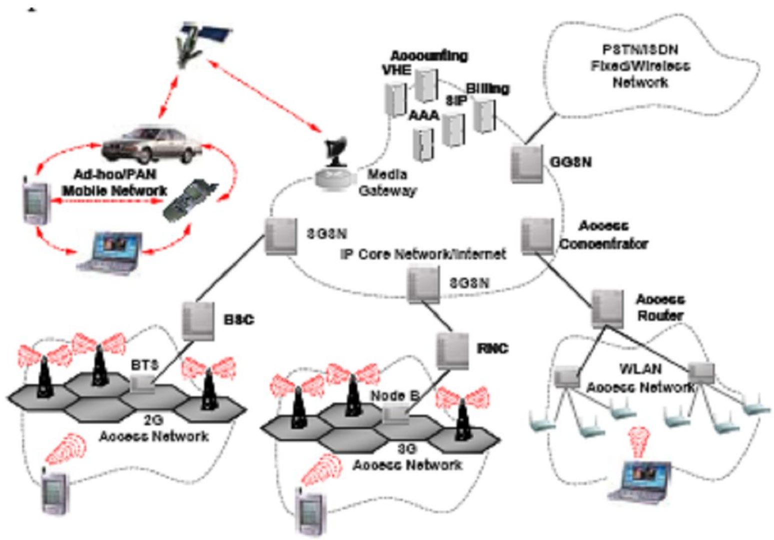


Fig1.2: The generic 4G mobile network architecture.

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.

1.3 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.

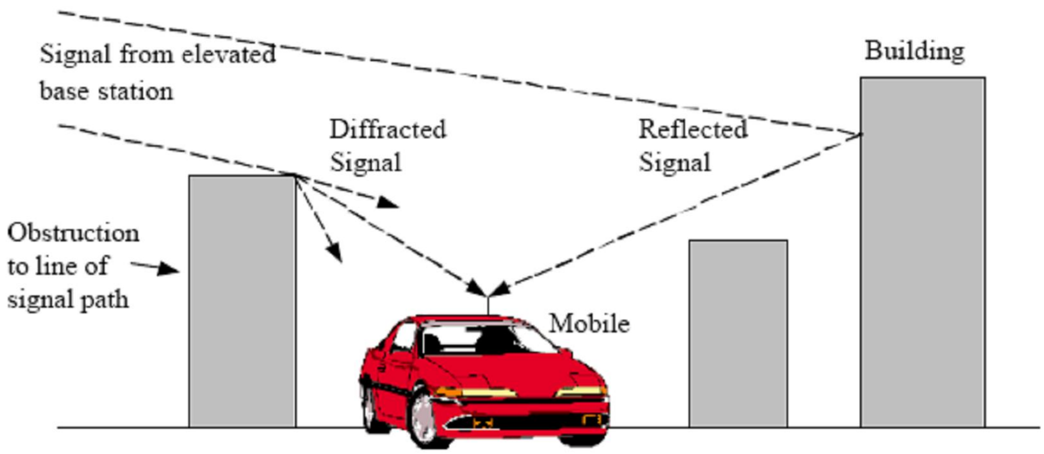


Fig1.3 Radio Propagation Effects

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 1.2.

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 1.2

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

1.4 Multipath Effects

1.4.1 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.

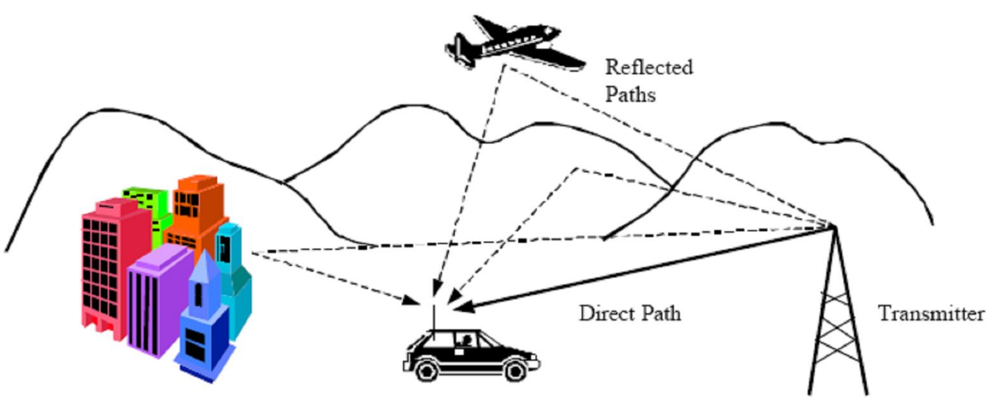


Fig: 1.4 Multipath Signals

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.

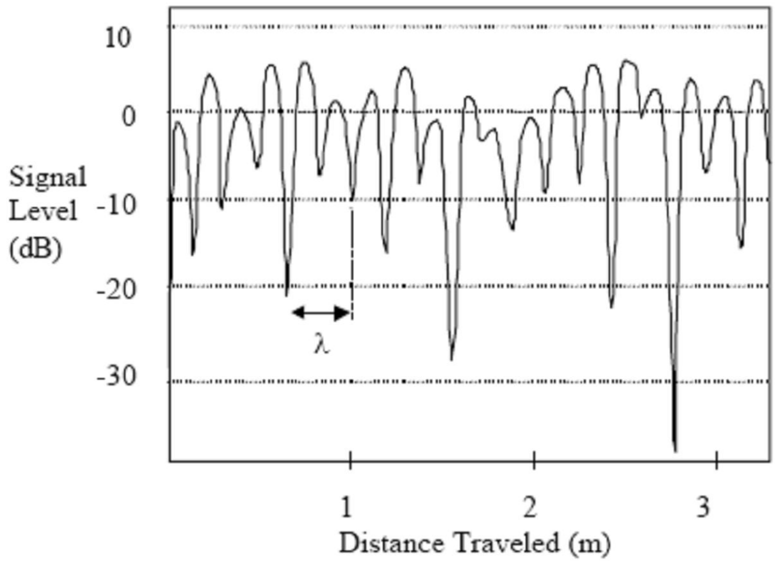


Fig 1.5 Typical Rayleigh fading while the Mobile Unit is moving

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 1.3 Cumulative distributions for Rayleigh distribution

1.4.2 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.

1.4.3 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.

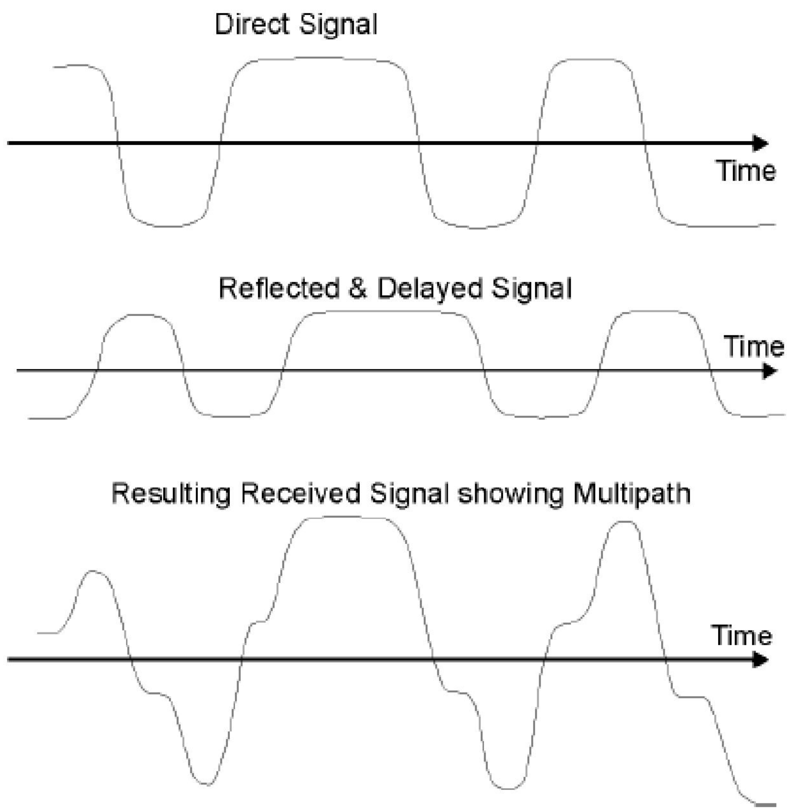


Fig 1.5 Multipath Delay Spread

Table 1.4 shows the typical delay spread for various environments. The maximum delay spread in an outdoor environment is approximately 20s, thus significant intersymbol 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 1.4 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.

1.4.4 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:

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

where Δf 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:

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

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.

Chapter 2

2 Code Division Multiple Access

2.1 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.

2.2 Code Division Multiple Access (CDMA)

Let's take a stright forward binary signal of symbol rate 2.

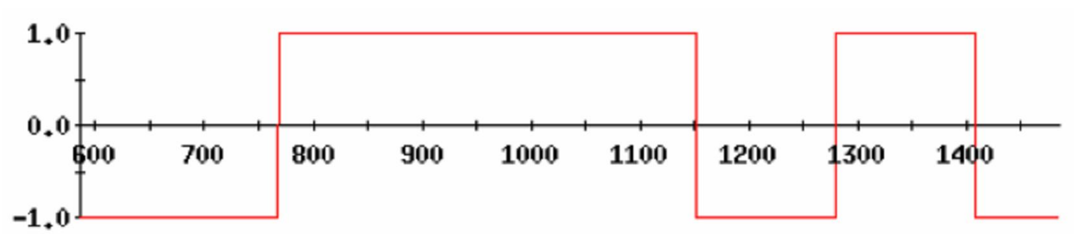


Fig: 1.6

To modulate this signal, we would multiply this sequence with a sinusoid and its spectrum would look like as In figure 1.6. The main lobe of its spectrum is 2 Hz wide. The larger the symbol rate the larger the bandwidth of the signal.

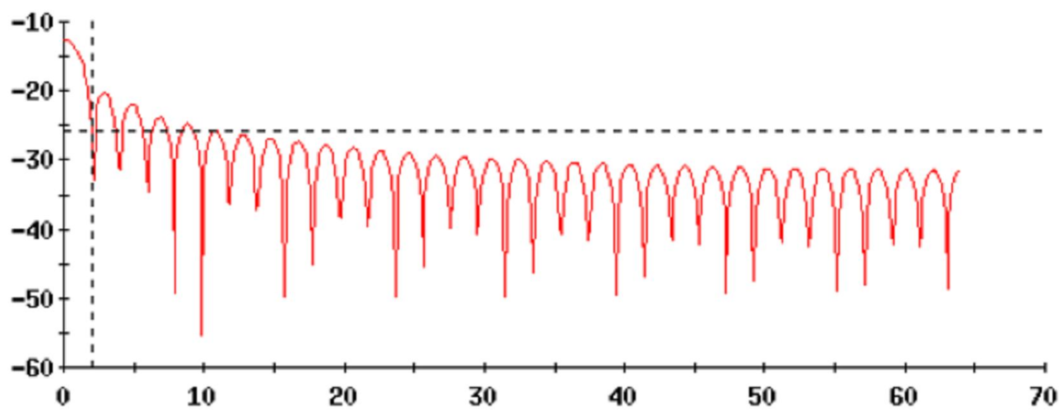


Fig: 1.7

Now we take an another binary sequence of data rate 8 times larger than of sequence

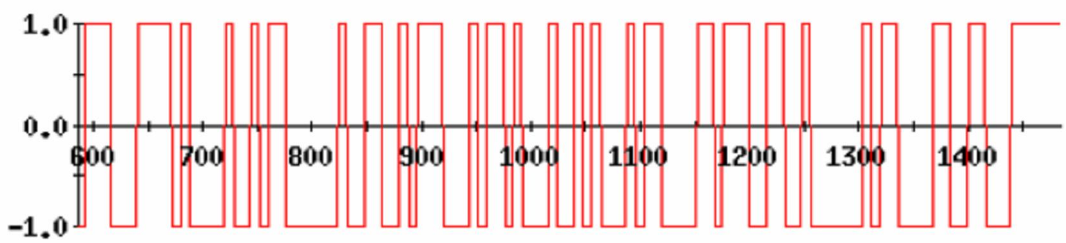


Fig1.8 – A new binary sequence which will be used to modulate the information sequence.

Instead of modulating with a sinusoid, we will modulate the sequence 1 with this new binary sequence which we will call the code sequence for sequence 1. The resulting signal looks like Fig. 1.9. Since the bit rate is larger now, we can guess that the spectrum of this sequence will have a larger main lobe.

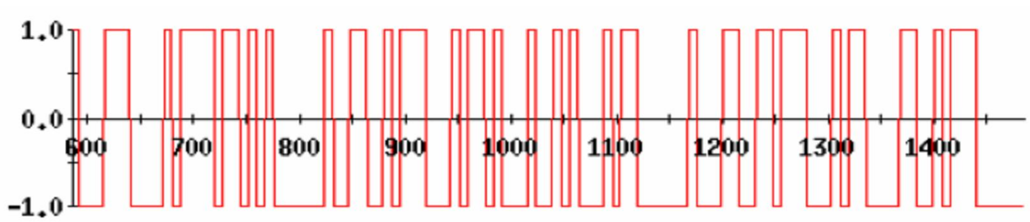


Fig 1.9 – Each bit of sequence 1 is replaced by the code sequence

The spectrum of this signal has now spread over a larger bandwidth. The main lobe bandwidth is 16 Hz instead of 2 Hz it was before spreading. The process of multiplying the information sequence with the code sequence has caused the information sequence to inherit the spectrum of the code sequence (also called the spreading sequence).

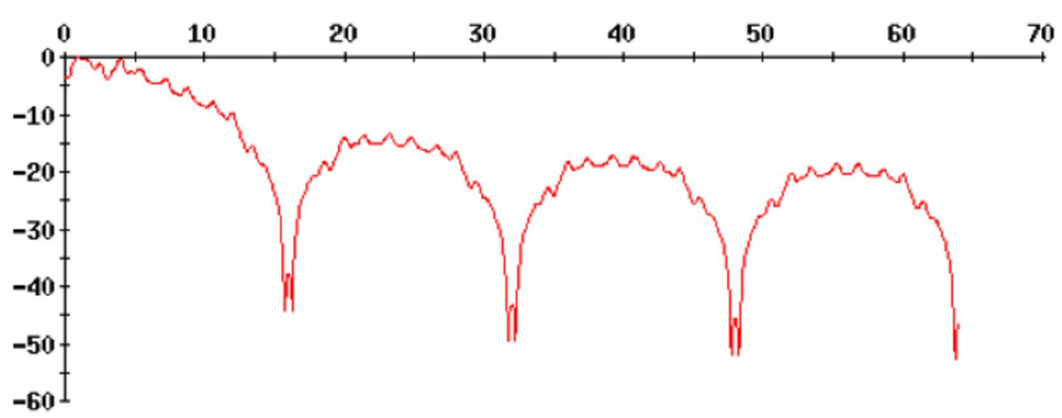


Fig 2.0 – The spectrum of the spread signal is as wide as the code sequence

The spectrum has spread from 2 Hz to 16 Hz, by a factor of 8. This number is called the spreading factor or the processing gain (in dBs) of the system. This process can also be called a form of binary modulation. Both the Data signal and the modulating sequence in this case are binary signals. If original signal is x(t) of power P_s, and the code sequence is given by g(t), the resultant modulated signal is

$$s(t) = \sqrt{2P_s}\,\underline{d(t)g(t)}$$

The multiplication of the data sequence with the spreading sequence is the first modulation. Then the signal is multiplied by the carrier which is the second modulation.

The carrier here is analog.

$$s(t) = \sqrt{2P_s}\,\underline{\underline{d(t)g(t)}\sin(2\pi f_c t)}$$

On the receive side, we multiply this signal again with the carrier. What we get is this.

$$rcv(t) = \sqrt{2P_s}\, d(t) g(t) \sin^2(2\pi f_c t)$$

By the trigonometric identity

$$\sin^2(2\pi f_c t) = 1 - \underline{\cos(4\pi f_c t)}$$

we get

$$rcv(t) = \sqrt{2P_s}\, d(t) g(t) (1 - \underline{\cos(4\pi f_c t)})$$

Where the underlined part is the double frequency extraneous term, which we filter out
and we are left with just the signal.

$$rcv(t) = \sqrt{2P_s}\, d(t) g(t)$$

Now we multiply this remaining signal with g(t), the code sequence and we get

$$rcv(t) = \sqrt{2P_s}\, d(t) g(t) g(t)$$

Now from having used a very special kind of sequence, we say that correlatation of g(t)
with itself (only when perfectly aligned) is a certain scalar number which can be removed, and we get the original signal back.

$$rcv(t) = \sqrt{2P_s}\, d(t)$$

In CDMA we do modulation twice. First with a binary sequence g(t), the properties of which we will discuss below and then by a carrier. The binary sequence modulation ahead of the carrier modulation accomplishes two functions, 1. It spread the signal and 2.
It introduces a form of encryption because the same sequence is needed at the receiver to demodulate the signal.

In IS-95 and CDMA 2000 we do this three times, once with a code called Walsh, then with a code called Short Code and then with one called Long code.

2.2.1 Properties of spreading codes

Multiplication with the code sequence which is of a higher bit rate, results in a much wider spectrum. The ratio of the code rate to the information bit rate is called both the spreading factor and the processing gain of the CDMA system. In IS-95, the chipping rate is 1.2288 and the spreading factor is 64. Processing gain is usually given in dBs. To distinguish the information bit rate from the code rate, we call the code rate, chipping rate. In effect, we take each data bit and convert it into k chips, which is the code sequence. We call it the chipping rate because the code sequence applied to each bit is as you can imagine it chipping the original bit into many smaller bits.

For CDMA spreading code, we need a random sequence that passes certain "quality" criterion for randomness. These criterion are

1. The number of runs of 0's and 1's is equal. We want equal number of two 0's and 1's, a length of three 0's and 1's and four 0's and 1's etc. This property gives us a perfectly random sequence.
2. There are equal number of runs of 0's and 1's. This ensures that the sequence is balanced.
3. The periodic autocorrelation function (ACF) is nearly two valued with peaks at 0 shift and is zero elsewhere. This allows us to encrypt the signal effectively and using the ACF peak to demodulate quicklt. Binary sequences that can meet these properties are called optimal binary sequences, or pseudo-random sequences. There are many classes of sequences that mostly meet these requirements, with m-sequences the only ones that meet all three requirements strictly. These sequences can be created using a shift-registers with feedback-taps. By using a single shift-register, maximum length sequences can be created and called often by their shorter name of m-sequence, where m stands for maximum.

2.2.2 m-sequences and the Linear Feed Shift-Register

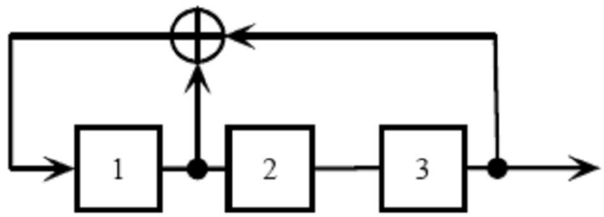


Fig 2.1

3 stage LFSR generating m-sequence of period 7., using taps 1 and 3.

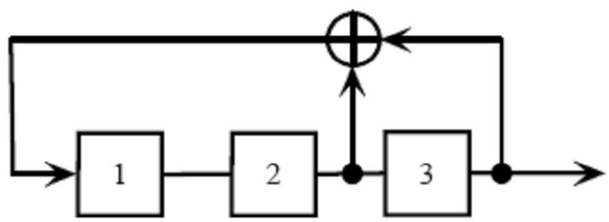


Fig: 2.2

Another 3 stage LFSR generating m-sequence of period 7, using taps 2 and 3 Figure 2.1 – The structure of linear feedback registers (LFSR) from which m-sequences can be created

m–sequences are created using linear feedback registers (LFSR). Figure 2.1 shows a three register LFSR with two different tap connection arrangements. The tap connections are based on primitive polynomials on the order of the number of registers and unless the polynomial is irreducible, the sequence will not be a m-sequence and will not have the desired properties.

Each configuration of N registers produces one sequence of length 2^N. If taps are changed, a new sequence is produced of the same length. There are only a limited number of m-sequences of a particular size. The cross correlation between an m-sequences and noise is low which is very useful in filtering out noise at the receiver. The cross correlation between any two different msequences is also low and is useful in providing both encryption and spreading. The low amount of cross-correlation is used by the receiver to discriminate among user signals generated by different m-sequences. Think of m-sequence as a code applied to each message. Each letter (bit) of the message is changed by the code sequence. The spreading quality of the sequence is an added dimensionality and benefit in CDMA systems.

2.2.3 Gold sequences

Combining two m-sequences creates **Gold codes**. These codes are used in asynchronous

CDMA systems. Gold sequences are an important class of sequences that allow construction of long sequences with three valued Auto Correlation Function ACFs. Gold sequences are constructed from pairs of preferred m-sequences by modulo-2 addition of two maximal sequences of the same length. Gold sequences are in useful in non-orthogonal CDMA. (CDMA 2000 is mostly an orthogonal CDMA system) Gold sequences have only three cross-correlation peaks, which tend to get less important as the length of the code increases. They also have a single auto-correlation peak at zero, just like ordinary PN sequences. The use of Gold sequences permits the transmission to be asynchronous. The receiver can synchronize using the auto-correlation property of the Gold sequence.

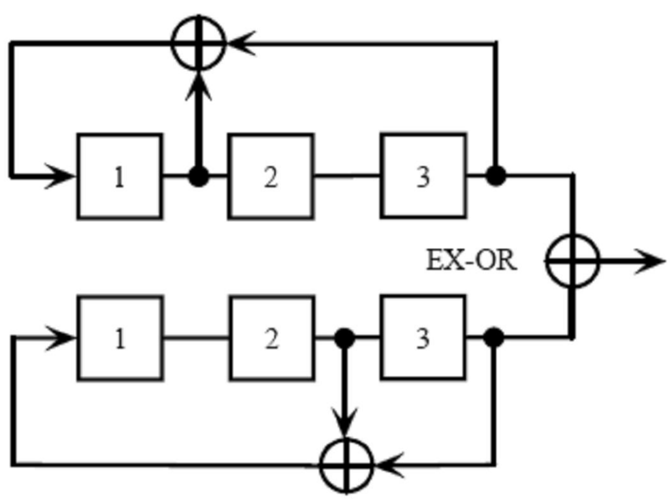


Fig 2.3 Generating Gold codes by combining two preferred pairs of m-sequences

2.2.4 More codes

IS-95 and IS-2000 use two particular codes that are really m-sequences but have special names and uses. These are called long codes and short codes.

2.2.5 Long code

The Long Codes are 2^{42} bits (created from a LFSR of 42 registers) long and run at 1.2288 Mb/s. The time it takes to recycle this length of code at this speed is 41.2 days. It is used to both spread the signal and to encrypt it. A cyclically shifted version of the long code is generated by the cell phone during call setup. The shift is called the Long Code Mask and is unique to each phone call. CDMA networks have a security protocol called CAVE that requires a 64-bit authentication key, called A-key and the unique ESN (Electronic Serial Number, assigned to mobile based on the phone number). The network uses both of these to create a random number that is then used to create a mask for the long code used to encrypt and spread each phone call. This number, the long code mask is not fixed but changes each time a connection is created.

There is a Public long code and a Private long code. The Public long code is used by the mobile to communicate with the base during the call setup phase. The private long code is one generated for each call then abandoned after the call is completed.

2.2.6 Short code

The short code used in CDMA system is based on a m-sequence (created from a LFSR of 15 registers) of length $2^{15} – 1 = 32,767$ codes. These codes are used for synchronization in the forward and reverse links and for cell/base station identification in the forward link The short code repeats every 26.666 milliseconds. The sequences repeat exactly 75 times in every 2 seconds. We want this sequence to be fairly short because during call setup, the mobile is looking for a short code and needs to be able find it fairly quickly. Two seconds is the maximum time that a mobile will need to find a base station, if one is present because in 2 seconds the mobile has checked each of the allowed base stations in its database against the network signal it is receiving. Each base station is assigned one of these codes. Since short code is only one sequence, how do we assign it to all the stations? We cyclically shift it. Each station gets the same sequence but it is shifted. From properties of the m-sequences, the shifted version of a m-sequences has a very small cross correlation and so each shifted code is an independent code. For CDMA this shift is 512 chips for each adjacent station. Different cells and cell sectors all use the same short code, but use different phases or shifts, which is how the mobile differentiates one base station from another. The phase shift is known as the PN Offset.

The moment when the Short code wraps around and begins again is called a PN Roll. If I call the word “please” a short code, then I can assign, “leasep” to one user, “easepl” to another and so on. The shift by one letter would be my PN Offset. So if I say your ID is 3, then you would use the code “aseple”. A mobile is assigned a short code PN offset by the base station to which it is transmitting. The mobile adds the short code at the specified PN offset to its traffic message, so that the base station in the region knows that the particular message is meant for it and not to the adjacent base station. This is essentially the way the primary base station is identified in a phone call. The base station maintains a list of nearby base stations and during handoff, the mobile is notified of the change in the short code.

There are actually two short codes per base station. One for each I and Q channels to be used in the quadrature spreading and despreading of CDMA signals.

2.2.7 Walsh codes

In addition to the above two codes, another special code, called Walsh is also used in CDMA. Walsh codes do not have the properties of m-sequences regarding cross correlation.. IS-95 uses 64 Walsh codes and these allow the creation of 64 channels from the base station. In other words, a base station can talk to a maximum of 64 (this number is actually only 54 because some codes are used for pilot and synch channels) mobiles at the same time. CDMA 2000 used 256 of these codes. Walsh codes are created out of Haddamard matrices and Transform. Haddamard is the matrix type from which Walsh created these codes. Walsh codes have just one outstanding quality. In a family of Walsh codes, all codes are orthogonal to each other and are used to create channelization within the 1.25 MHz band.

Here are first four Hadamard matrices. The code length is the size of the matrix. Each row is one Walsh code of size N. The first matrix gives us two codes; 00, 01. The second matrix gives: 0000, 0101, 0011, 0110 and so on.

$$H1 = \begin{pmatrix} 0 & 0 \\ 0 & 1 \end{pmatrix}$$

$$H2 = \begin{pmatrix} 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 1 \\ 0 & 0 & 1 & 1 \\ 0 & 1 & 1 & 0 \end{pmatrix}$$

$$H3 = \begin{pmatrix} 0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 1 & 0 & 1 & 0 & 1 \\ 0 & 0 & 1 & 1 & 0 & 0 & 1 & 1 \\ 0 & 1 & 1 & 0 & 0 & 1 & 1 & 0 \\ 0 & 0 & 0 & 0 & 1 & 1 & 1 & 1 \\ 0 & 1 & 0 & 1 & 1 & 0 & 1 & 0 \\ 0 & 0 & 1 & 1 & 1 & 1 & 0 & 0 \\ 0 & 1 & 1 & 0 & 1 & 0 & 0 & 1 \end{pmatrix}$$

In general each higher level of Hadamard matrix is generated from the previous by the

2.2.8 Hadamard transform

$$H_{N+1} = \begin{pmatrix} H_N & H_N \\ H_N & \overline{H_N} \end{pmatrix}$$

Where $\overline{H_N}$ is the inverse of H_N.

Their main purpose of Walsh codes in CDMA is to provide orthogonality among all the users in a cell. Each user traffic channel is assigned a different Walsh code by the base station. IS-95 has capability to use 64 codes, whereas CDMA 2000 can use up to 256 such codes. Walsh code 0 (which is itself all 0s) is reserved for pilot channels, 1 to 7 for synch and paging channels and rest for traffic channels. They are also used to create an orthogonal modulation on the forward link and are used for modulation and spreading on the reverse channel.

Orthogonal means that cross correlation between Walsh codes is zero when aligned. However, the auto-correlation of Walsh-Hadamard codewords does not have good characteristics. It can have more than one peak and this makes it difficult for the receiver to detect the beginning of the codeword without an external synchronization. The partial sequence cross correlation can also be non-zero and un-synchronized users can interfere with each other particularly as the multipath environment will differentially delay the sequences. This is why Walsh-Hadamard codes are only used in synchronous CDMA and only by the base station which can maintain orthogonality between signals for its users.

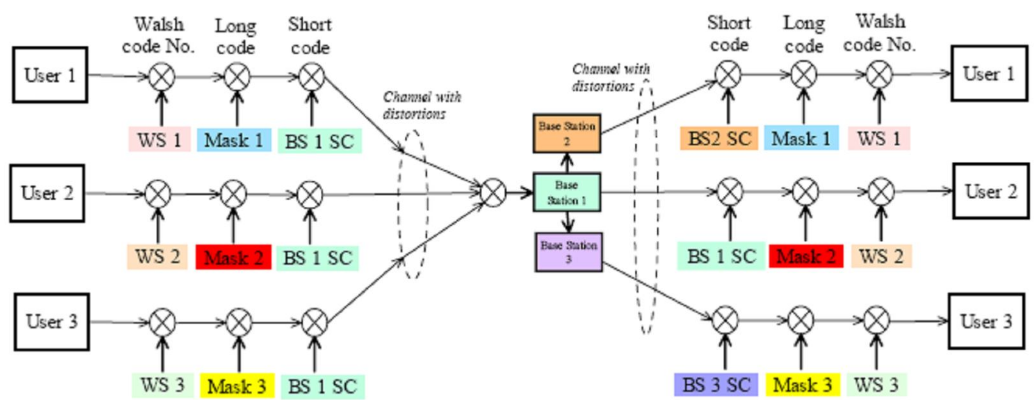


Fig 2.4 Relationship codes used in CDMA

The above is simplified look at the use of these codes. Assume there are three users in one cell. Each is trying to talk to someone else. User 1 wants to talk to someone who is outside its cell and is in cell 2. User 3 wants to talk to someone in cell 3. Let's take User 1. Its data is first covered by a channel Wash code, which is any Walsh code from 8 to 63. It is assigned to the user by the base station 1 in whose cell the mobile is located. The Base Station has also assigned different Walsh codes to users 2 and 3. All three of these are different are assigned by base station 1 and are orthogonal to each other. This keeps the data apart at the base station. Now based on the random number assigned by the BS, the mobile generates a long code mask (which is just the starting point of the long code sequence and is a scalar number). It now multiplies the signal by this long code starting at the mask ID. Now it multiplies it by the short code of the base station to whom it is directing the signal. When the base station receives this signal, it can read the long code and see that the message needs to be routed to base station 2. So it strips off 1st short code and adds on the short code of base station

2 which is then broadcast by the BS 1 to BS 2 or sent by landlines. BS2 then broadcasts this signal along to all mobiles in its cell. The users who is located in this cell, now does the reverse. It multiplies the signal by the BS 2 short code (it knows nothing about BS 1 where the message generated) then it multiplies the signal by the same long code as the generating mobile. How? During the call paging, the mobile was given the same random number from which it creates the same long code mask. After that it multiplies it by the Walsh code sequence (also relayed during call setup). So that's about it with some additional bells and whistles, which we shall get to shortly.

2.2.9 Channel Waveform Properties

The communications between the mobile and the base station takes place using specific channels. Figure below shows the architecture of these channels. The forward channel (from base station to mobile) is made up of the following channels: Pilot channel (always uses Walsh code W0) (Beacon Signals) Paging channel(s) (use Walsh codes W1-W7)

Sync channel (always uses Walsh code W32) Traffic channels (use Walsh codes W8-W31 and W33-W63) The reverse channel (from mobile to base station) is made up of the following channels:

Access channel

Traffic channel

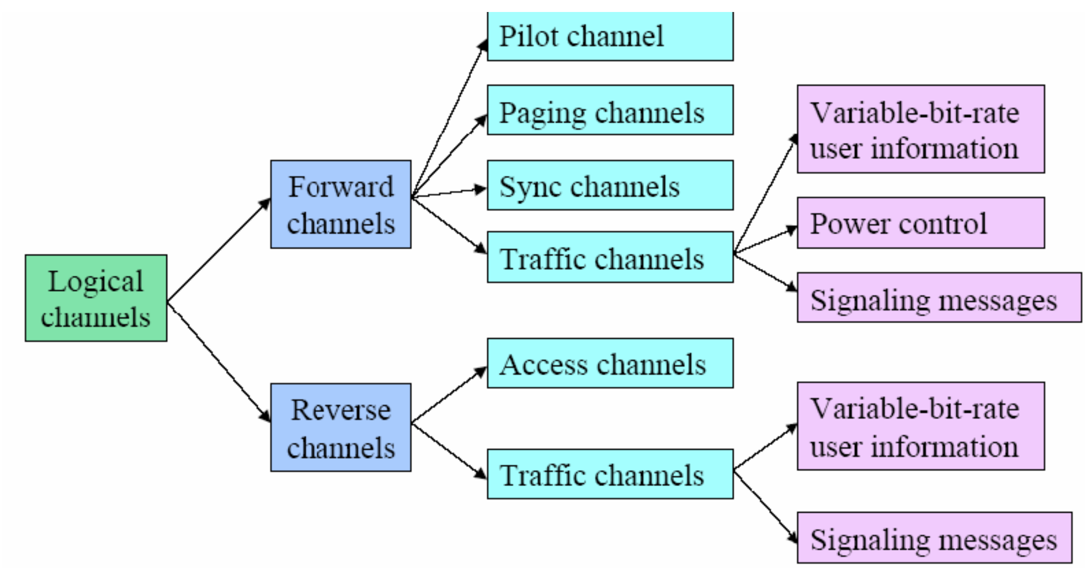


Fig 2.5 Forward channel

Forward Channel description

A base station can communicate on up to 64 channels. It has one pilot signal, one synch channel and 8 paging channels. The remaining are used for traffic with individual mobiles.

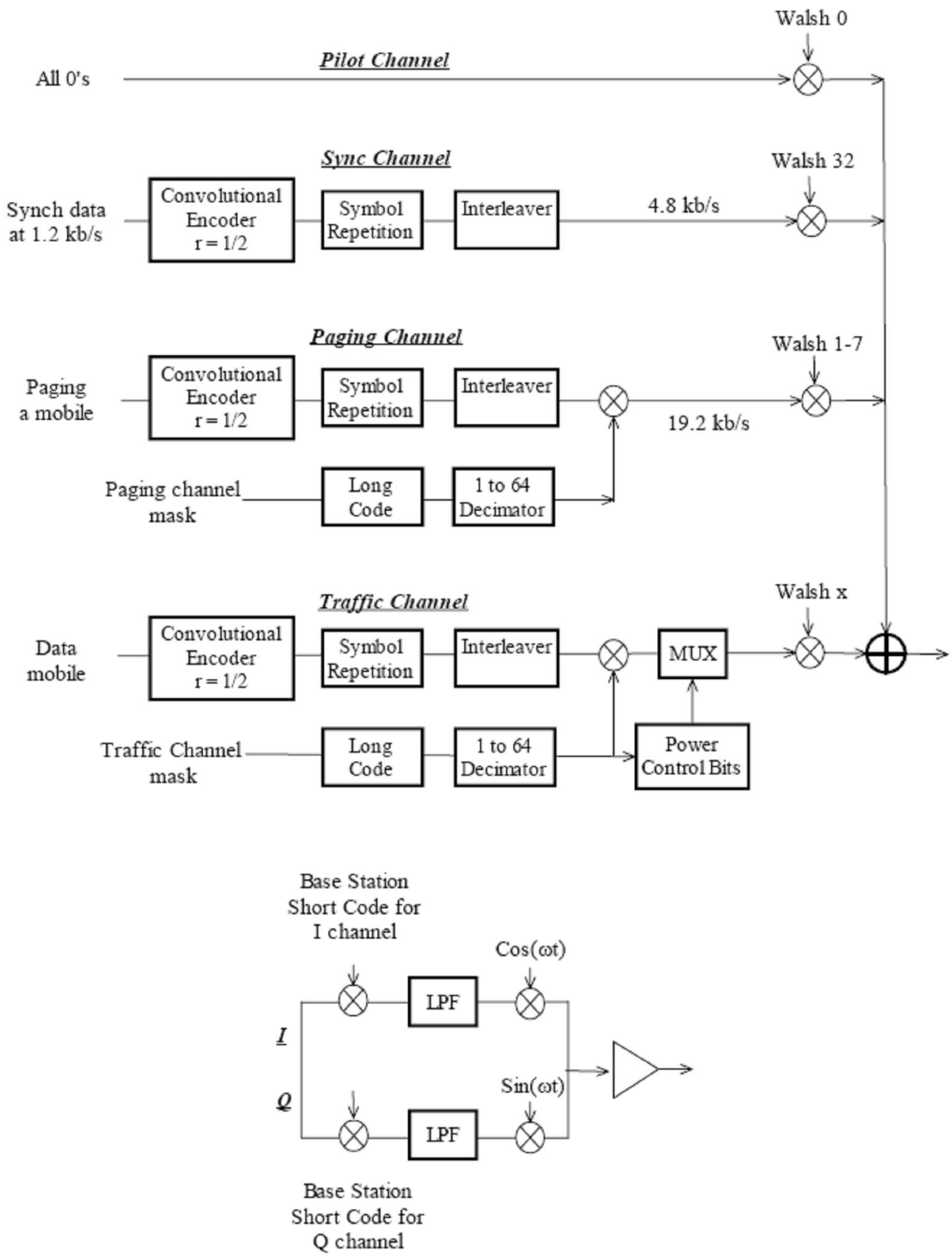


Fig 2.6 – Forward channel is the transmission of all traffic from the base station within its cell. All data is sent simultaneously.

2.2.10 Pilot Channel

Let's start with how the base station establishes contact with the mobiles within its cell. It is continually transmitting an all zero signal, which is covered by a Walsh code 0, a all 0's code. So what we have here is a one very long bit of all zeros. For this reason, the pilot channel has very good SNR making it easy for mobiles to find it. This all zero signal is then multiplied by the base stations' short code, which if you recall is the same short code that all base station use, but each with different PN offset. Pilot PN Offsets are always assigned to stations in multiples of 64 chips, giving a total of 512 possible assignments. The 9-bit number that identifies the pilot phase assignment is called the Pilot Offset.

This signal is real so it only goes out on the I channel, and is up-converted to the carrier frequency which in the US is 845 MHz. On the receive side, the mobile picks up this signal and notes the base station that is transmitting it. Here is a question, if the short code is cyclical, how does the receiver know what the phase offset is. Do not all the signals from all the other nearby base stations look the same? Yes, and the mobile at this point does not know which base station it is talking to, only that it has found the network. To determine of all the possible base station and there can 256 of them, each using a 512 chip shifted short code, the network uses the GPS signal and timing.

The zero offset base station aligns its pilot transmission with every even second time tick of GPS. So let's say that your mobile is in the cell belonging to a base station with PN offset ID of 10. That means that is will start its transmission 10 x 512 chip = 5120 chips after every even second time tick. So when the mobile wakes up and looks at it time, it knows exactly where each base station short code should be. Then all it has to do is to do a correlation of the bits it is seeing with each of the 256 possible sequences. Of course, it tries the base station where it was last but if it has been moved then theoretical it will have to go through all 256 correlations to figure out where it is. But it does do it and at the end of the process, it knows exactly which of the base stations it is hearing.

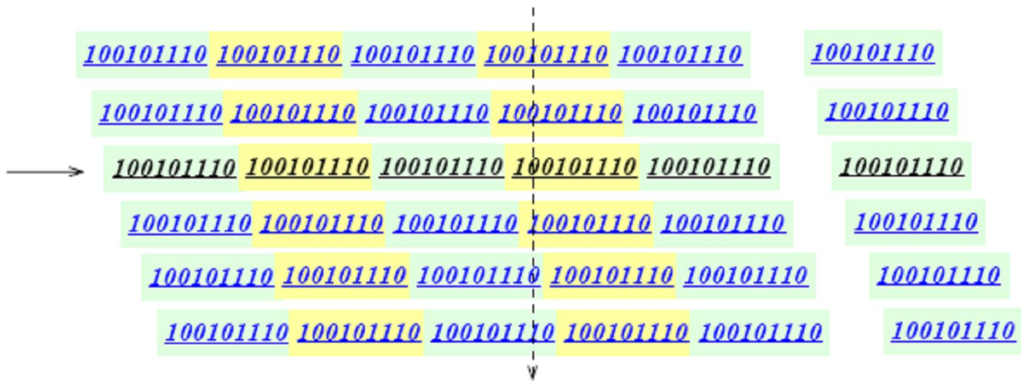


Fig 2.7 The mobile looks for the code that aligns with GPS timing. It picks off the code received at this time, does a correlation with stored data and knows which base station it has found.

2.2.11 Synch Channel

The Synch channel information includes the pilot offset of the pilot the mobile has acquired. This information allows the mobile to know where to search for the pilots in the neighbor list. It also includes system time, the time of day, based on Global Positioning Satellite (GPS) time. The system time is used to synchronize system functions. For instance, the PN generators on the reverse link use zero offset relative to the even numbered seconds in GPS time. However, the mobiles only know system time at the base stations plus an uncertainty due to the propagation delay from its base station to the mobile's location. The state of the long code generator at system time is also sent to the mobile in the Synchronization message. This allows the mobile to initialize and run its long code generator very closely in time synchronism with the long code generators in the base stations. The Synchronization message also notifies the mobile of the paging channel data rate, which may be either 4800 or 9600 bits/sec. The data rate of this channel is always 1200 bps.

2.2.12 Paging Channel

Now the mobile flashes the name of the network on its screen and is ready to receive and make calls. Your paging channel may now be full of data. It may include a ring tone or a “voicemail received” message. The data on the paging channel sent by the base station, includes mobile Electronic Serial Identification Number (ESIN), and is covered by a long code. How does the mobile figure out what this long code is? At the paging level, the system uses a public long code. This is because it is not talking to a specific mobile, it is paging and needs to reach all mobiles. When the correct mobile responds, a new private long code will be assigned at that time before the call will be connected. The mobile while scanning the paging channel recognizes its phone number and responds by ringing. When you pick up the call, an access message goes back to the base station. The mobile using Qualcomm CDMA generator is a 18-bit code. The mobile sends this authentication sequence to the base station during the sync part of the messaging protocol. The base station checks the authentication code before allowing call setup. It then issues a random number to the mobile, which the mobile uses in the CAVE algorithm to generate a call specific long code mask. At the same time, the base station will also do exactly that. The two now have the same long code with which to cover the messages.

2.2.13 Traffic Channel

The base station can transmit traffic data to as many as 54 mobiles at the same time. It keeps these channel separate by using Walsh codes. This is a code division multiplexing rather than a frequency based channelization. Walsh codes are used only by the base station and in this fashion, it is a **synchronous CDMA** on the forward link, whereas on the return link it is **asynchronous CDMA**, because there is no attempted separation between the various users. But the use of m-sequences for spreading, the quality of orthogonality although not perfect is very very good. The traffic channel construct starts with baseband data at 4.8 kbps. It is then convolutionally encoded at rate of ½, so the data rate now doubles to 9.6 kbps. Symbol repetition is used to get the data rate up to 19.2 kbps. All information rates are submultiples of this rate. Data is then interleaved. The interleaving does not change the data rate, only that the bits are reordered to provide protection against burst errors. Now at this point, we multiply the resulting data sequences with the long code, which starts at the point determined by the private random number generated by both the base station and the mobile jointly. This start point is call-based and changes every time. Mobiles do not have a fixed long code assigned to them. Reverse CDMA Channel can have up to 2^{42}-1 logical channels or the total number of calls that can be served are 17179869184. Now the data is multiplied by a specific Walsh codes which is the nth call that the base station is involved in. Mobile already knows this number from the paging channel. The base station then combines all its traffic channels (each covered by a different Walsh code) and all paging channels (just 8) and the one pilot channel and one synch channel adds them up, does serial to parallel conversion to I and Q channels. Each is then covered by a I and a Q short code and is QPSK modulated up to carrier frequencies and then transmitted in the cell.

2.2.14 Reverse Channels

In IS-95, there are just two channels on which the mobile transmits, and even that never simultaneously. It is either on the access channel or it is transmitting traffic. The channel structure is similar but simpler to the forward channel, with the addition of 64-ary modulation.

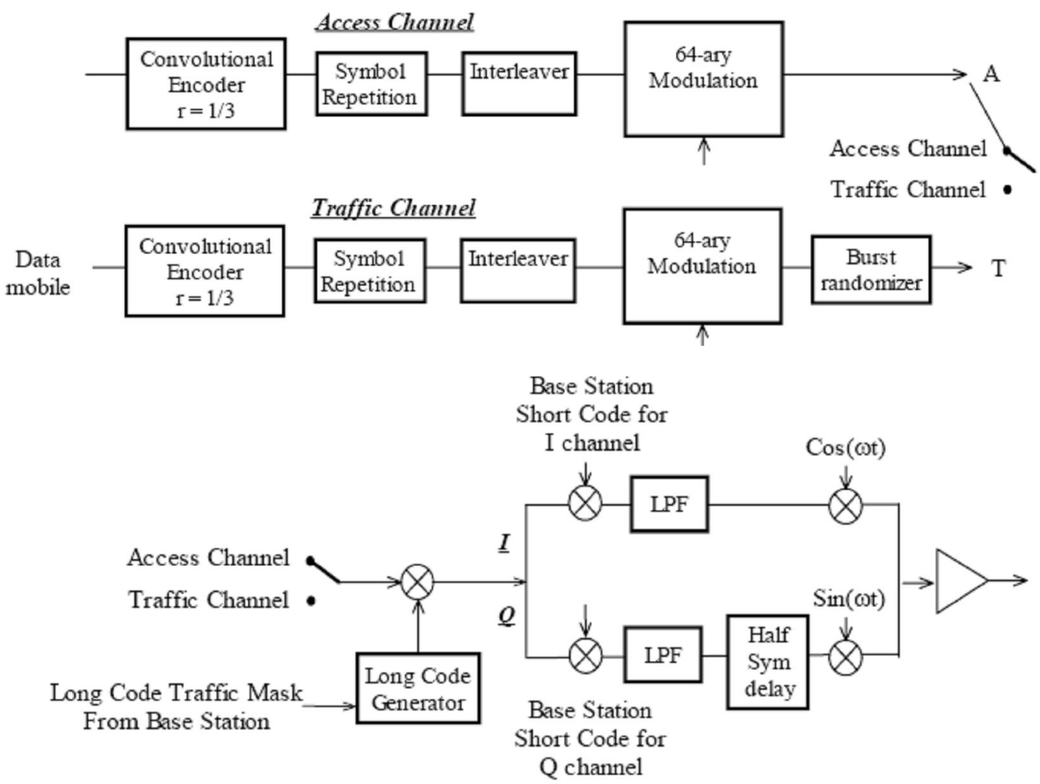


Fig 2.8 Reverse Channel - from mobile to base station communication

2.2.15 64-ary modulation

This block takes a group of six incoming bits (which makes $2^6 = 64$ different bit sequences of 6 bits) and assigns a particular Walsh code to each. We know that each Walsh code sequence is orthogonal to all the others so in this way, a form of spreading has been forced on the arbitrarily created symbols of 6 bits. And this spreading also forces the symbols to be orthogonal. It is not really a modulation but is more of a spreading function because we still have not up converted this signal to the carrier frequency. After this, a randomization function is employed to make sure we do not get too many 0's or 1's in a row. This is because certain Walsh codes have a lot of consecutive 0's.

Next comes multiplication with the long code starting at a particular private start point. Then comes serial to parallel conversion, and application of baseband filtering which can be a Gaussian or a root cosine shaping. Then the Q channel (or I, it makes no difference) is delayed by half a symbol, as shown below. The reason this is done is to turn this into n offset QPSK modulated signal. The offset modulated signal has a lower non-linearity susceptibility and is better suitable to being transmitted by a class C amplifier such as may be used in a CDMA cell phone. From there, each I and Q channel is multiplied by the rf carrier, (a sine and a cosine of frequency f_c) and off the signal goes to the base station.

On the demodulation side, the most notable item is the Rake receiver. Due to the presence of multipath, Rake receivers which allow maximal combining of delayed and attenuated signal, make the whole thing work within reasonable power requirements. Without Rake receivers, your cell phone would not be as small as it is.

2.2.16 Power control

Assume that there is only one user of the system. The carrier power

$$C = SNR = E_b/T_b = R\,E_b$$

If we define the transmit power equal to W and signal bandwidth equal to B, then the Interference power at the receiver is equal to

$$I = W\,N_0$$

Now we can write

$$\frac{C}{I} = \frac{R\,E_b}{W\,N_0} = \frac{E_b/N_0}{W/R}$$

The quantity W/R is the processing gain of the system. Now let's call M the number of users in this system. The total interference power is equal to

$$I = C\ (M-1)$$

Substituting this in the above equation, we get,

$$\frac{C}{I} = \frac{C}{C(M-1)} = \frac{1}{M-1}$$

and with one more substitution we get

$$\frac{C}{I} = \frac{E_b / N_0}{W / R} = \frac{1}{M-1}$$

$$M \approx M-1 = \frac{W}{R}\frac{1}{E_b / N_0}$$

So we conclude that the system capacity is a direct function of the processing gain for a given Eb/N0. What you may not have noticed is that we made an assumption that all users have similar power level so the interferences are additive. No one user overwhelms all the others. If the power levels of all users are not equal then the system capacity is compromised and the C/I expression above is not valid. The CDMA systems manage the power levels of all mobiles so that the power level of each mobile is below a certain required level and is about the same whether the mobile is very close to the base station or far at the edge of the cell. Multipath and fading also attenuate power levels so the system maintains a power control loop. IS-95 has a open-loop and a closed loop power management system. The open loop is a quicker way to manage power levels. The forward and reverse links are at different frequencies so they fade differently and open loop power control allows the mobile to adjust its power without consulting with the base station. In closed loop power control the base station measures the power level of the access channel signal sent by the mobile and then commands with 1 in the synch channel if the power needs to be raised and with 0 if it is to be reduced by 1 dB at a time. The closed loop power control also uses an outer loop power control. This method measures the Frame Error Rate (FER) both by the mobile and the base station and then adjusts the power according to whether the FER is acceptable.

Chapter 3

3 Orthogonal Frequency Division Multiplexing (OFDM)

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 3 kHz. The allocated bandwidth is made wider then the minimum amount required preventing channels from interfering with one another. This extra bandwidth is to allow for signals from neighboring 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, 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.

3.1 OFDM History

The concept of using parallel data transmission by means of frequency division multiplexing (FDM) was published in mid 60s. Some early development can be traced back in the 50s. A U.S. patent was filled and issued in January, 1970. The idea was to use parallel data streams and FDM with overlapping subchannels to avoid the use of high speed equalization and to combat impulsive noise, and multipath distortion as well as to fully use the available bandwidth. The initial applications were in the military communications. In the telecommunications field, the terms of discrete multi-tone (DMT), multichannel modulation and multicarrier modulation (MCM) are widely used and sometimes they are interchangeable with OFDM. In OFDM, each carrier is orthogonal to all other carriers. However, this condition is not always maintained in MCM. OFDM is an optimal version of multicarrier transmission schemes.

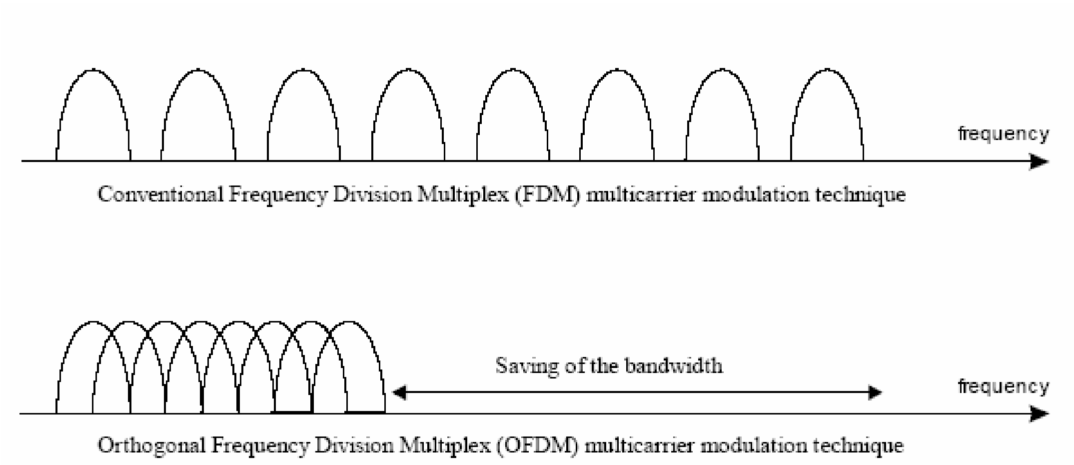


Figure 3.1

For a large number of subchannels, the arrays of sinusoidal generators and coherent demodulators required in a parallel system become unreasonably expensive and complex. The receiver needs precise phasing of the demodulating carriers and sampling times in order to keep crosstalk between subchannels acceptable. Weinstein and Ebert applied the discrete Fourier transform (DFT) to parallel data transmission system as part of the modulation and demodulation process. In addition to eliminating the banks of subcarrier oscillators and coherent demodulators required by FDM, a completely digital implementation could be built around special-purpose hardware performing the fast Fourier transform (FFT). Recent advances in VLSI technology enable making of high-speed chips that can perform large size FFT at affordable price.

In the 1980s, OFDM has been studied for high-speed modems, digital mobile communications and high-density recording. One of the systems used a pilot tone for stabilizing carrier and clock frequency control and trellis coding was implemented. Various fast modems were developed for telephone networks. In 1990s, OFDM has been exploited for wideband data communications over mobile radio FM channels, high-bit-rate digital subscriber lines (HDSL, 1.6 Mb/s), asymmetric digital subscriber lines (ADSL, 1,536 Mb/s), very high-speed digital subscriber lines (VHDSL, 100 Mb/s), digital audio broadcasting (DAB) and HDTV terrestrial broadcasting.

3.2 Qualitative Description of OFDM

In multimedia communication, a demand emerges for high-speed, high-quality digital mobile portable reception and transmission. A receiver has to cope with a signal that is often weaker than desirable and that contains many echoes. Simple digital systems do not work well in the multipath environment.

In a conventional serial data system, the symbols are transmitted sequentially, with the frequency spectrum of each data symbol allowed to occupy the entire available bandwidth. In a parallel data transmission system several symbols are transmitted at the same time, what offers possibilities for alleviating many of the problems encountered with serial systems. In OFDM, the data is divided among large number of closely spaced carriers. This accounts for the “frequency division multiplex” part of the name. This is *not* a multiple access technique, since there is no common medium to be shared. The entire bandwidth is filled from a single source of data. Instead of transmitting in serial way, data is transferred in a parallel way. Only a small amount of the data is carried on each carrier, and by this lowering of the bitrate per carrier (not the total bitrate), the influence of intersymbol interference is significantly reduced. In principle, many modulation schemes could be used to modulate the data at a low bit rate onto each carrier.

It is an important part of the OFDM system design that the bandwidth occupied is greater than the correlation bandwidth of the fading channel. A good understanding of the propagation statistics is needed to ensure that this condition is met. Then, although some of the carriers are degraded by multipath fading, the majority of the carriers should still be adequately received. OFDM can effectively randomize burst errors caused by Rayleigh fading, which comes from interleaving due to paralellisation. So, instead of several adjacent symbols being completely destroyed, many symbols are only slightly distorted. Because of dividing an entire channel bandwidth into many narrow subbands, the frequency response over each individual subband is relatively flat. Since each subchannel covers only a small fraction of the original bandwidth, equalization is potentially simpler than in a serial data system. A simple equalization algorithm can minimize mean-square distortion on each subchannel, and the implementation of differential encoding may make it possible to avoid equalization altogether. This allows the precise reconstruction of majority of them, even without forward error correction (FEC).

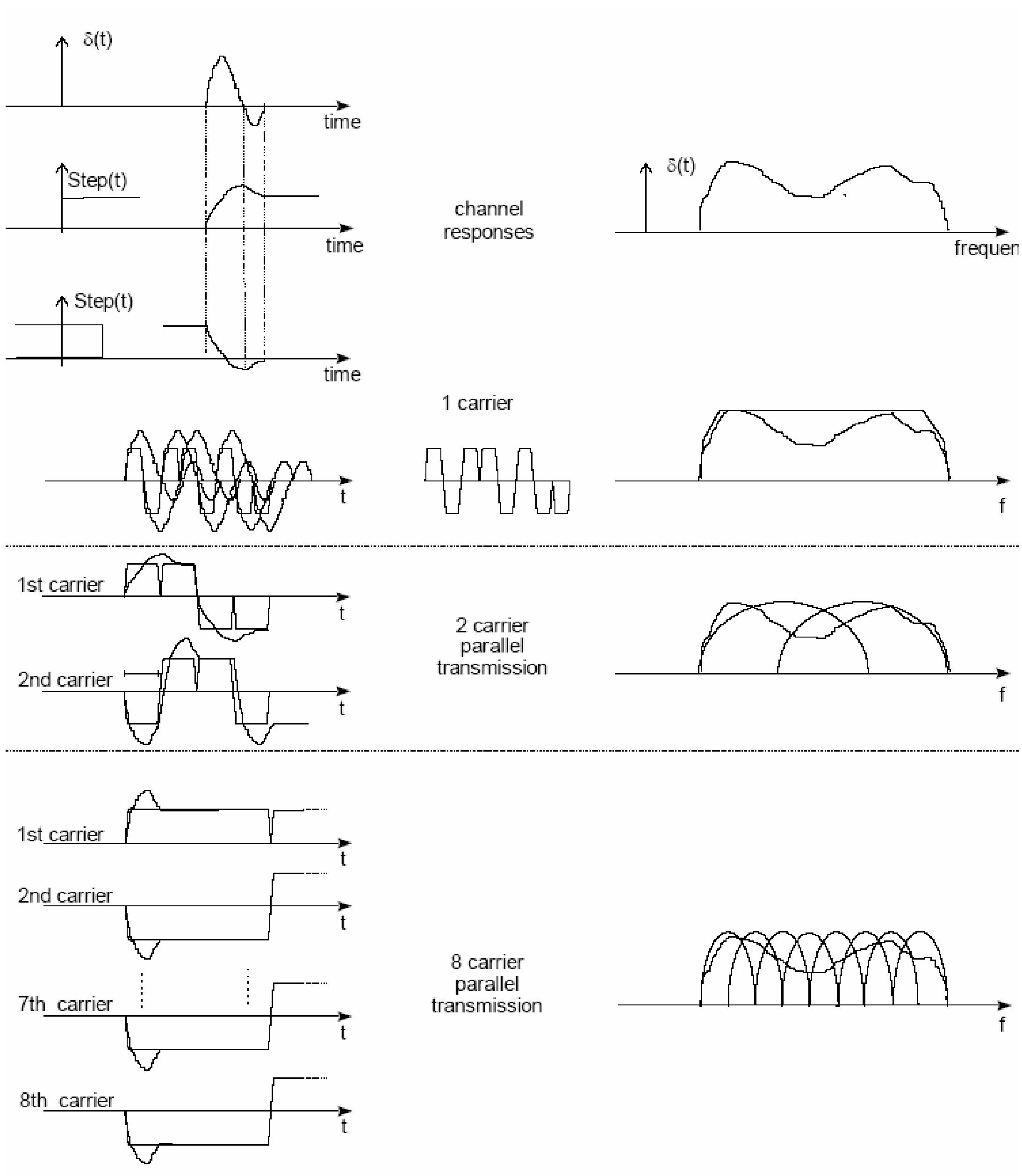


Figure 3.2

The effect of adopting a multicarrier system. For a given overall data rate, increasing the number of carriers reduces the data rate that each individual carrier must convey, and hence (for a given modulation system) lengthens the symbol period. This means that the intersymbol interference affects a smaller percentage of each symbol as the number of carriers and hence the symbol period increases (after [10[DM1]]). For example, on the picture is shown a 8 bit long part of a data sequence. For a single carrier system, the responses of individual bits are overlapping, thus creating ISI. Multicarrier system is robust against these physical effects.

In addition, by using a guard interval the sensitivity of the system to delay spread can be reduced. In a classical parallel data system, the total signal frequency band is divided into N nonoverlapping frequency subchannels. Each subchannel is modulated with a separate symbol and, then, the N subchannels are frequency multiplexed. There are three schemes that can be used to separate the subbands:

1. Use filters to completely separate the subbands. This method was borrowed from the conventional FDM technology. The limitation of filter implementation forces the bandwidth of each subband to be equal to (1+□)f_m, where □ is the roll-off factor and f_m is the Nyquist bandwidth. Another disadvantage is that it is difficult to assemble a set of matched filter when the number of carriers is large.

2. Use staggered QAM to increase the efficiency of band usage. In this way the individual spectra of the modulated carriers still use an excess bandwidth, but the are overlapped at the 3 dB frequency. The advantage is that the composite spectrum is flat. The separability or orthogonality is achieved by staggering the data (offset the data by half a symbol). The requirement for filter design is less critical than that for the first scheme.

3. Use discrete Fourier transform (DFT) to modulate and demodulate parallel data. The individual spectra are now *sinc* functions and are not band limited. The FDM is achieved, not by bandpass filtering, but by baseband processing. Using this method, both transmitter and receiver can be implemented using efficient FFT techniques that reduce the number of operations from N^2 in DFT, down to NlogN. OFDM can be simply defined as a form of multicarrier modulation where its carrier spacing is carefully selected so that each subcarrier is orthogonal to the other subcarriers. As is well known, orthogonal signals can be separated at the receiver by correlation techniques; hence, intersymbol interference among channels can be eliminated.

Orthogonality can be achieved by carefully selecting carrier spacing, such as letting the carrier spacing be equal to the reciprocal of the useful symbol period. Mathematical deduction of the orthogonal carrier frequencies is given in. In order to occupy sufficient bandwidth to gain advantages of the OFDM system, it would be good to group a number of users together to form a wideband system, in order to interleave data in time and frequency (depends how broad is one user signal).

3.3 The Importance of Orthogonality

The “orthogonal” part of the OFDM name indicates that there is a precise mathematical relationship between the frequencies of the carriers in the system. In a normal FDM system, the many carriers are spaced apart in such way that the signals can be received using conventional filters and demodulators. In such receivers, guard bands have to be introduced between the different carriers (Figure. 1.), and the introduction of these guard bands in the frequency domain results in a lowering of the spectrum efficiency. It is possible, however, to arrange the carriers in an OFDM signal so that the sidebands of the individual carriers overlap and the signals can still be received without adjacent carrier interference. In order to do this the carriers must be mathematically orthogonal. The receiver acts as a bank of demodulators, translating each carrier down to DC, the resulting signal then being integrated over a symbol period to recover the raw data. If the other carriers all beat down to frequencies which, in the time domain, have a whole number of cycles in the symbol period (□), then the integration process results in zero contribution from all these carriers. Thus the carriers are linearly independent (i.e. orthogonal) if the carrier spacing is a multiple of 1/□.

Mathematically, suppose we have a set of signals □ , where □p is the *p*-th element in the set. The signals are orthogonal if

$$\int_a^b \Psi_p(t)\Psi_q^*(t)dt = \begin{cases} K & for\ p = q \\ 0 & for\ p \neq q \end{cases}$$

where the * indicates the complex conjugate and interval [a,b] is a symbol period. A fairly simple mathematical proof exists, that the series *sin*(*mx*). Much of transform theory makes the use of orthogonal series, although they are by no means the only example.

3.4 Mathematical Description of OFDM

After the qualitative description of the system, it is valuable to discuss the mathematical definition of the modulation system. This allows us to see how the signal is generated and how receiver must operate, and it gives us a tool to understand the effects of imperfections in the transmission channel. As noted above, OFDM transmits a large number of narrowband carriers, closely spaced in the frequency domain. In order to avoid a large number of modulators and filters at the transmitter and complementary filters and demodulators at the receiver, it is desirable to be able to use modern digital signal processing techniques, such as fast Fourier transform (FFT).

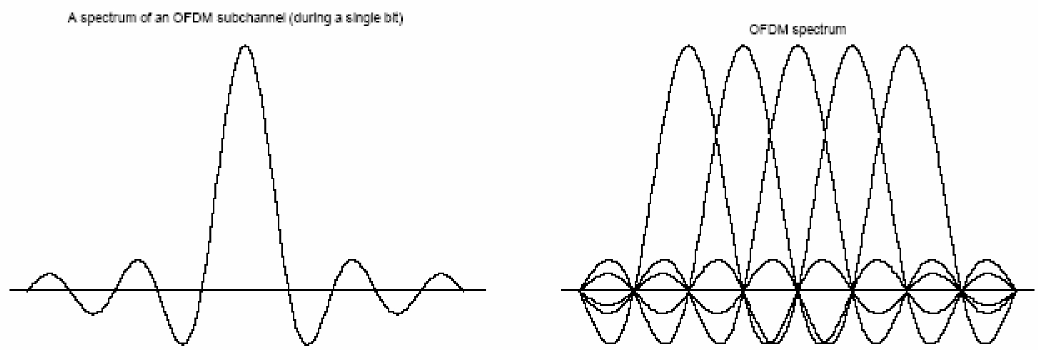


Figure. 3.3

Examples of OFDM spectrum (a) a single subchannel, (b) 5 carriers

At the central frequency of each subchannel, there is no crosstalk from other subchannels

Mathematically, each carrier can be described as a complex wave:

$$s_c(t) = A_c(t) e^{j[\omega_c t + \phi_c(t)]} \qquad (1)$$

The real signal is the real part of *sc(t)*. Both *Ac(t)* and *fc(t),* the amplitude and phase of the carrier, can vary on a symbol by symbol basis. The values of the parameters are constant over the symbol duration period □.

OFDM consists of many carriers. Thus the complex signals *ss(t)* (Fig. 4) is represented by:

$$s_s(t) = \frac{1}{N}\sum_{n=0}^{N-1} A_N(t)\, e^{j\left[\omega_n t + \phi_n(t)\right]} \qquad (2)$$

where,

$$\omega_n = \omega_0 + n\Delta\omega$$

This is of course a continuous signal. If we consider the waveforms of each component of the signal over one symbol period, then the variables *Ac(t)* and *fc(t)* take on fixed values, which depend on the frequency of that particular carrier, and so can be rewritten:

$$\phi_n(t) \Rightarrow \phi_n$$
$$A_n(t) \Rightarrow A_n$$

If the signal is sampled using a sampling frequency of 1/T, then the resulting signal is represented by:

$$s_s(kT) = \frac{1}{N}\sum_{n=0}^{N-1} A_n\, e^{j\left[(\omega_0 + n\Delta\omega)kT + \phi_n\right]} \qquad (3)$$

At this point, we have restricted the time over which we analyse the signal to N samples. It is convenient to sample over the period of one data symbol.

If we now simplify eqn. 3, then the signal becomes:

$$s_s(kT) = \frac{1}{N}\sum_{n=0}^{N-1} A_n e^{j\phi_n} e^{j(n\Delta\omega)kT} \quad (4)$$

Now Eq. 4 can be compared with the general form of the inverse Fourier transform:

$$g(kT) = \frac{1}{N}\sum_{n=0}^{N-1} G\left(\frac{n}{NT}\right) e^{j2\pi nk/N} \quad (5)$$

In eq. 4, the function $A_n e^{j\phi_n}$ is no more than a definition of the signal in the sampled frequency domain, and *s(kT)* is the time domain representation. Eqns. 4 and 5 are equivalent if:

$$\Delta f = \frac{\Delta\omega}{2\pi} = \frac{1}{NT} = \frac{1}{\tau} \quad (6)$$

This is the same condition that was required for orthogonality (see *Importance of orthogonality*). Thus, one consequence of maintaining orthogonality is that the OFDM signal can be defined by using Fourier transform procedures.

3.5 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.

3.6 The Fourier Transform

The Fourier transform allows us to relate events in time domain to events in frequency domain. There are several version of the Fourier transform, and the choice of which one to use depends on the particular circumstances of the work.

The conventional transform relates to continuous signals which are not limited to in either time or frequency domains. However, signal processing is made easier if the signals are sampled. Sampling of signals with an infinite spectrum leads to aliasing, and the processing of signals which are not time limited can lead to problems with storage space. To avoid this, the majority of signal processing uses a version of the discrete Fourier transform (DFT). The DFT is a variant on the normal transform in which the signals are sampled in both time and the frequency domains. By definition, the time waveform must repeat continually, and this leads to a frequency spectrum that repeats continually in the frequency domain.

The fast Fourier transform (FFT) is merely a rapid mathematical method for computer applications of DFT. It is the availability of this technique, and the technology that allows it to be implemented on integrated circuits at a reasonable price, that has permitted OFDM to be developed as far as it has. The process of transforming from the time domain representation to the frequency domain representation uses the Fourier transform itself, whereas the reverse process uses the inverse Fourier transform.

3.7 The Use of FFT in OFDM

The main reason that the OFDM technique has taken a long time to become a prominence has been practical. It has been difficult to generate such a signal, and even harder to receive and demodulate the signal. The hardware solution, which makes use of multiple modulators and demodulators, was somewhat impractical for use in the civil systems.

The ability to define the signal in the frequency domain, in software on VLSI processors, and to generate the signal using the inverse Fourier transform is the key to its current popularity. The use of the reverse process in the receiver is essential if cheap and reliable receivers are to be readily available. Although the original proposals were made a long time ago, it has taken some time for technology to catch up. At the transmitter, the signal is defined in the frequency domain. It is a sampled digital signal, and it is defined such that the discrete Fourier spectrum exists only at discrete frequencies. Each OFDM carrier corresponds to one element of this discrete Fourier spectrum. The amplitudes and phases of the carriers depend on the data to be transmitted. The data transitions are synchronized at the carriers, and can be processed together, symbol by symbol (Fig. 4).

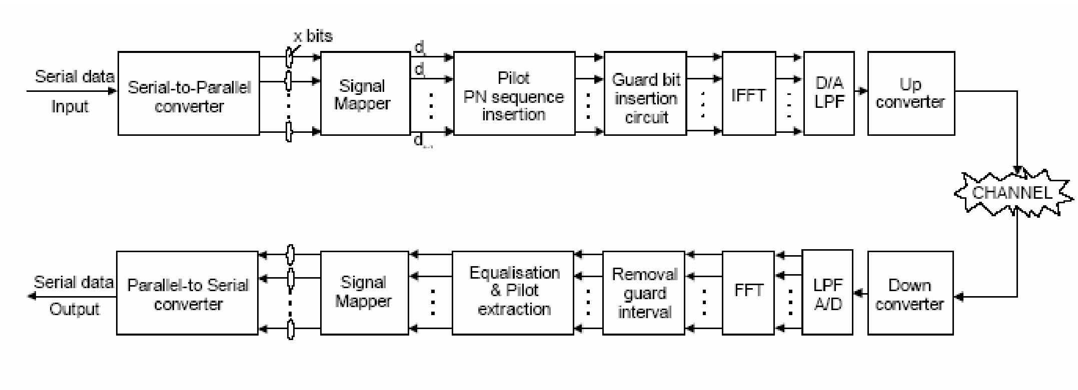


Figure 3.4: Block diagram of OFDM system

The definition of the (N-point) discrete Fourier transform (DFT) is:

$$X_p[k] = \sum_{n=0}^{N-1} x_p[n] e^{-j(2\pi/N)kn} \quad \text{(DFT)} \qquad (7)$$

and the (N-point) inverse discrete Fourier transform (IDFT):

$$x_p[n] = \frac{1}{N} \sum_{k=0}^{N-1} X_p[k] e^{j(2\pi/N)kn} \quad \text{(IDFT)} \qquad (8)$$

A natural consequence of this method is that it allows us to generate carriers that are orthogonal. The members of an orthogonal set are linearly independent. Consider a data sequence (d0, d1, d2, … , dN-1), where each dn is a complex number dn=an+jbn.

$$D_m = \sum_{n=0}^{N-1} d_n e^{-j(2\pi nm/N)} = \sum_{n==0}^{N-1} d_n e^{-j2\pi f_n t_m} \quad k=0,1,2,\ \ldots,\ N\text{-}1 \qquad (9)$$

The real part of the vector *D* has components

$$Y_m = \mathrm{Re}\{D_m\} = \sum_{n=0}^{N-1}\left[(a_n \cos(2\pi f_n t_m) + b_n \sin(2\pi f_n t_m))\right], \qquad k=0,1,\ldots,N-1 \tag{10}$$

If these components are applied to a low-pass filter at time intervals □t, a signal is obtained that closely approximates the frequency division multiplexed signal

$$y(t) = \sum_{n=0}^{N-1}\left[(a_n \cos(2\pi f_n t_m) + b_n \sin(2\pi f_n t_m))\right], \qquad 0 \le t \le N\Delta t \tag{11}$$

Figure. 3.5 illustrates the process of a typical FFT-based OFDM system. The incoming serial data is first converted form serial to parallel and grouped into *x* bits each to form a complex number. The number *x* determines the signal constellation of the corresponding subcarrier, such as 16 QAM or 32QAM. The complex numbers are modulated in the baseband by the inverse FFT (IFFT) and converted back to serial data for transmission. A guard interval is inserted between symbols to avoid intersymbol interference (ISI) caused by multipath distortion. The discrete symbols are converted to analog and low-pass filtered for RF up conversion. The receiver performs the inverse process of the transmitter. One-tap equalizer is used to correct channel distortion. The tap-coefficients of the filter are calculated based on the channel information.

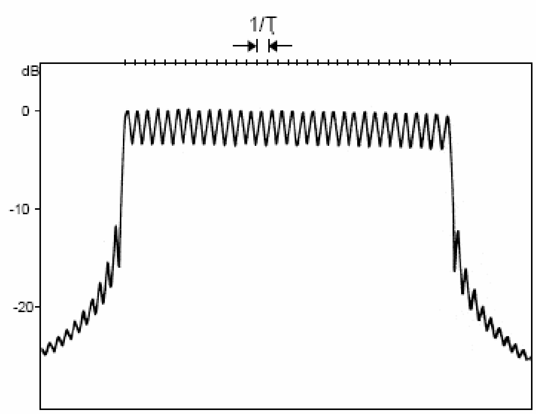


Figure: 3.5

Example of the power spectral density of the OFDM signal with a guard interval **D** = TS/4 (number of carriers N=32)]

3.8 Guard Interval and its Implementation

The orthogonality of subchannels in OFDM can be maintained and individual subchannels can be completely separated by the FFT at the receiver when there are no intersymbol interference (ISI) and intercarrier interference (ICI) introduced by transmission channel distortion. In practice these conditions can not be obtained. Since the spectra of an OFDM signal is not strictly band limited (*sinc(f)* function), linear distortion such as multipath cause each subchannel to spread energy into the adjacent channels and consequently cause ISI. A simple solution is to increase symbol duration or the number of carriers so that distortion becomes insignificant. However, this method may be difficult to implement in terms of carrier stability, Doppler shift, FFT size and latency.

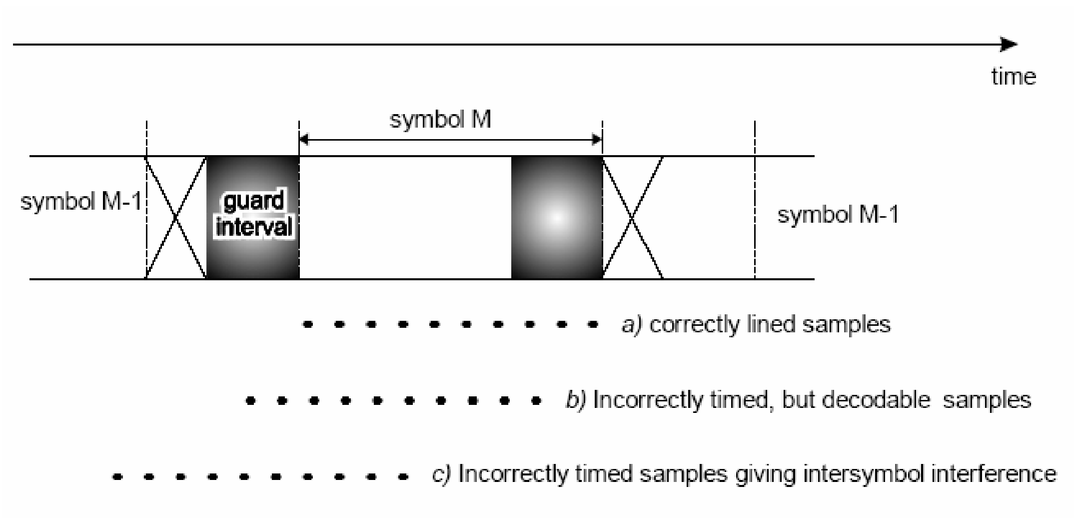


Figure: 3.6
The effect on the timing tolerance of adding a guard interval. With a guard interval included in the signal, the tolerance on timing the samples is considerably more relaxed.

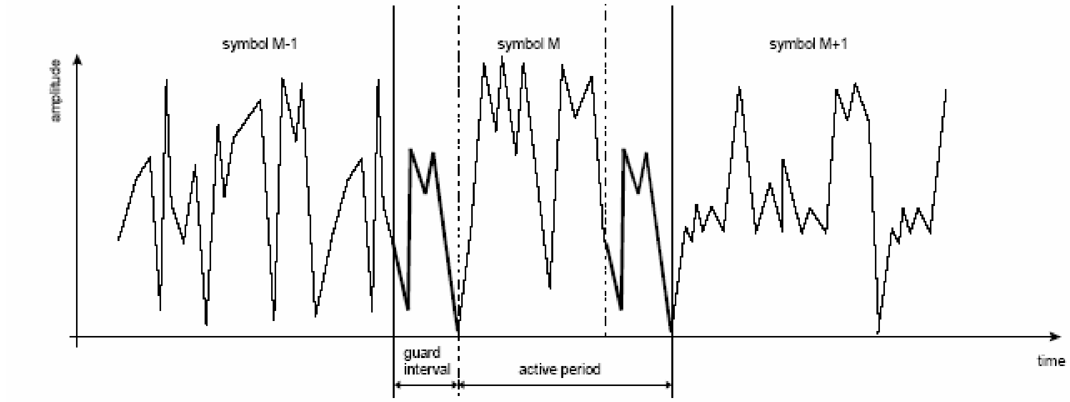


Figure: 3.7

Example of the guard interval. Each symbol is made up of two parts. The whole signal is contained in the
active symbol (shown highlighted for the symbol M) The last part of which (shown in bold) is also repeated at thestart of the symbol and is called the guard interval

One way to prevent ISI is to create a cyclically extended guard interval (Fig. 6, 7), where each OFDM symbol is preceded by a periodic extension of the signal itself. The total symbol duration is $T_{total}=T_g+T$, where T_g is the guard interval and T is the useful symbol duration. When the guard interval is longer than the channel impulse response (Figure. 2), or the multipath delay, the ISI can be eliminated.

However, the ICI, or in-band fading, still exists. The ratio of the guard interval to useful symbol duration is application-dependent. Since the insertion of guard interval will reduce data throughput, T_g is usually less than $T/4$.

The reasons to use a cyclic prefix for the guard interval are:

- To maintain the receiver carrier synchronization ; some signals instead of a long silence must always be transmitted;

- Cyclic convolution can still be applied between the OFDM signal and the channel response to model the transmission system.

3.9 Choice of the key elements

3.9.1 Useful symbol duration

The useful symbol duration *T* affects the carrier spacing and coding latency. To maintain the datathroughput, longer useful symbol duration results in increase of the number of carriers and the size of FFT (assuming the constellation is fixed). In practice, carrier offset and phase stability may affect how close two carriers can be placed. If the application is for the mobile reception, the carrier spacing must be large enough to make the Doppler shift negligible. Generally, the useful symbol duration should be chosen so that the channel is stable for the duration of a symbol.

3.10 Number of carriers

The number of subcarriers can be determined based on the channel bandwidth, data throughput and useful symbol duration.

$$N = \frac{1}{T}$$

The carriers are spaced by the reciprocal of the useful symbol duration. The number of carriers corresponds to the number of complex points being processed in FFT. For HDTV applications, the numbers of subcarriers are in the range of several thousands, so as to accommodate the data rate and guard interval requirement.

Modulation scheme in an OFDM system can be selected based on the requirement of power or spectrum efficiency. The type of modulation can be specified by the complex number $d_n=a_n+jb_n$, defined in section *The use of FFT in OFDM*. The symbols a_n and b_n can be selected for 16QAM and for QPSK. In general, the selection of the modulation scheme applying to each subchannel depends solely on the compromise between the data rate requirement and transmission robustness. Another advantage of OFDM is that different modulation schemes can be used on different subchannels for layered services.

3.11 OFDM Results

An OFDM system was modeled 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.

3.12 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.

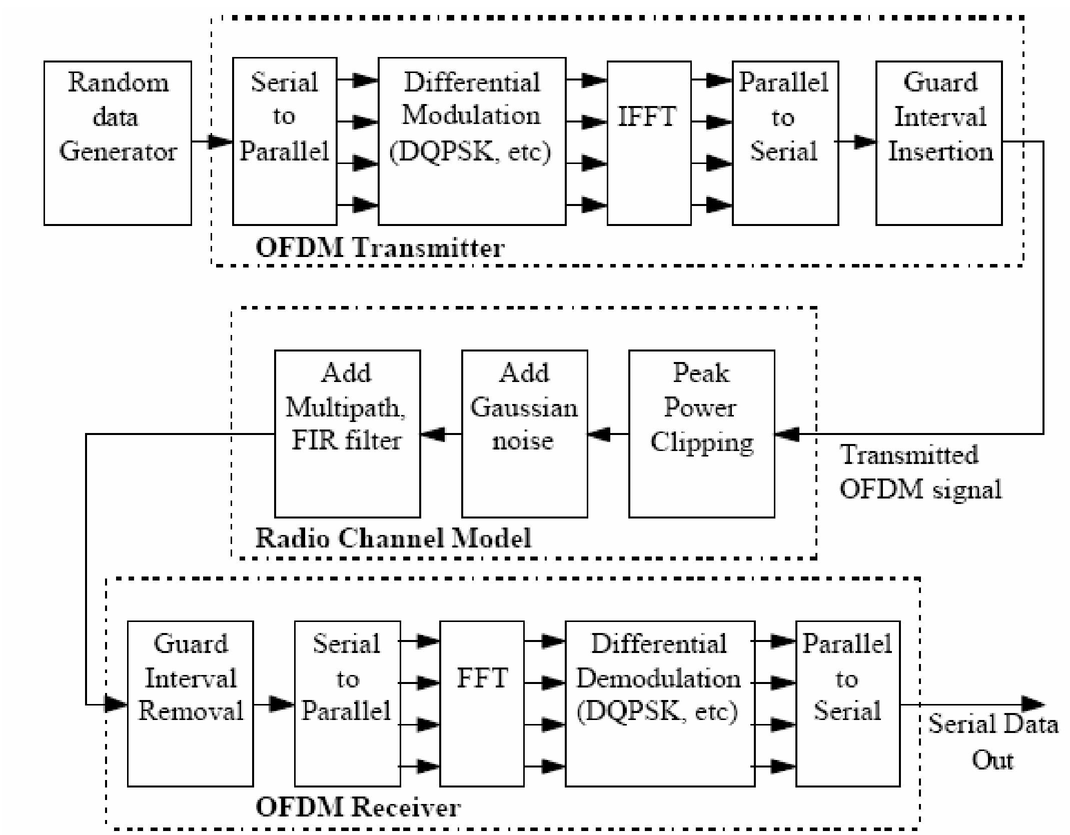


Figure: 3.8 OFDM Model used for simulations

3.13 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.

3.14 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.

3.15 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.

3.16 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. 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.

3.17 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.

3.18 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.

3.19 OFDM Simulation Parameters

Table 1 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.

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

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.

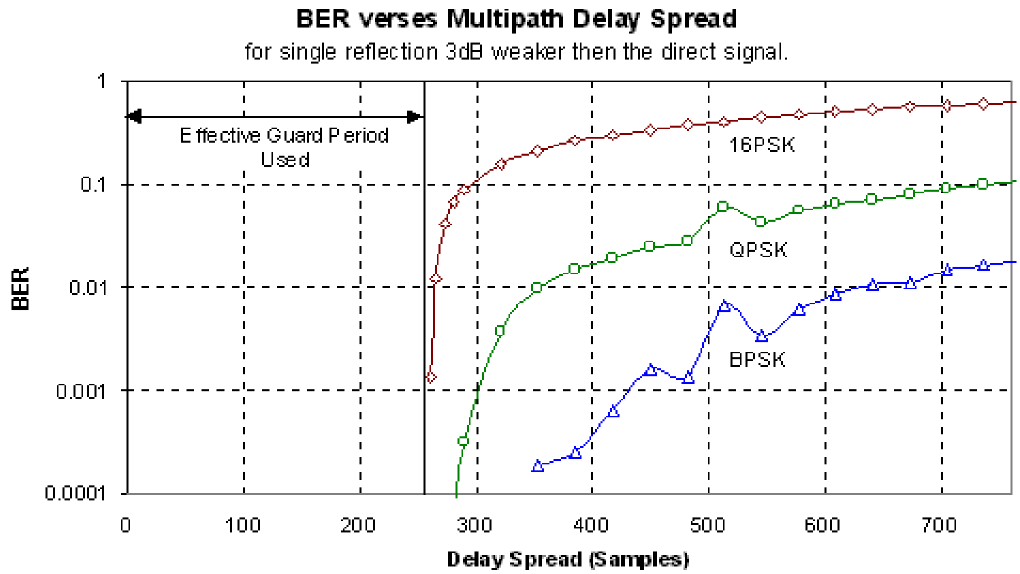
Chapter 4

4 Simulation Result

4.1 OFDM Simulation Result

4.1.1 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 below figure 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 msec. 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 tests however are not shown to conserve space.

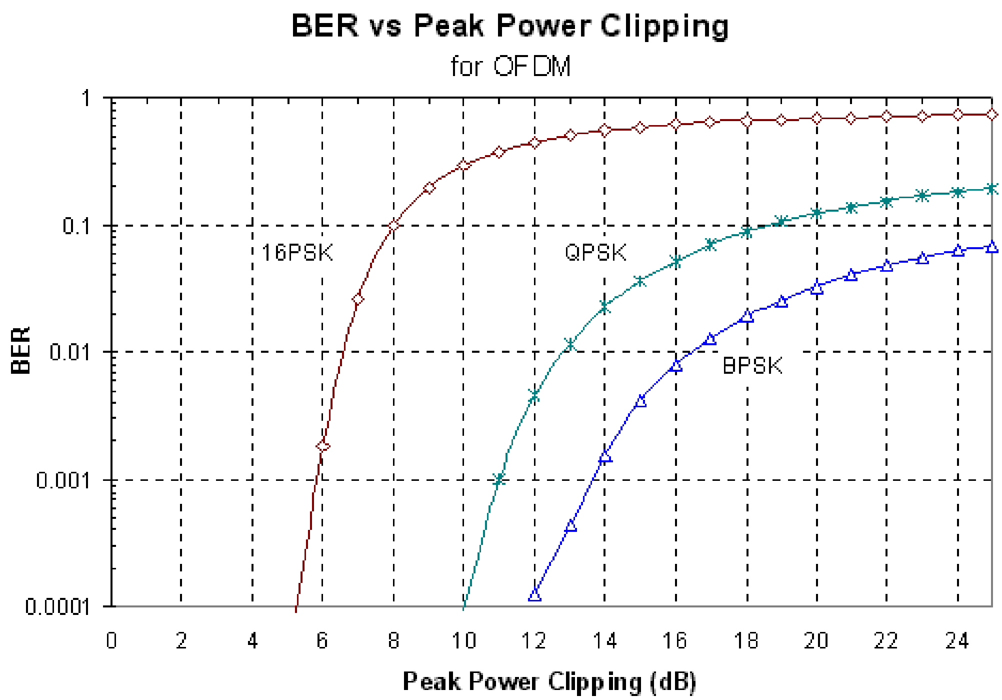


For a delay spread that is longer than the effective guard period, the BER rises rapidly due to the inter-symbol interference. The maximum BER that will occur is when the delay spread is very long (greater then the symbol time) as this will result in strong inter-symbol interference.

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

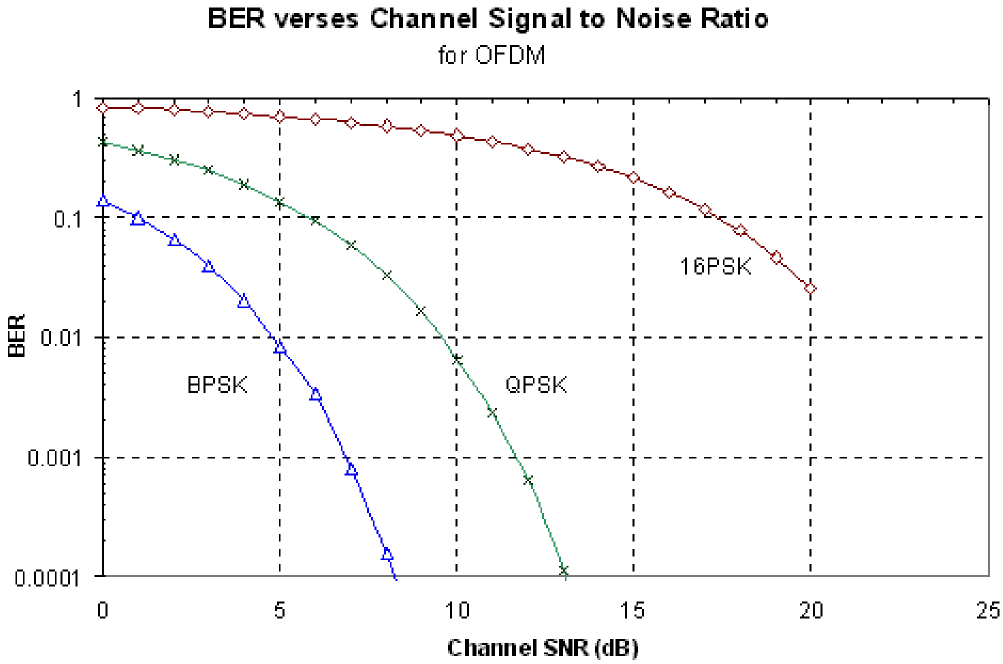
4.1.2 Peak Power Clipping

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.



4.1.3 Gaussian Noise Tolerance of OFDM

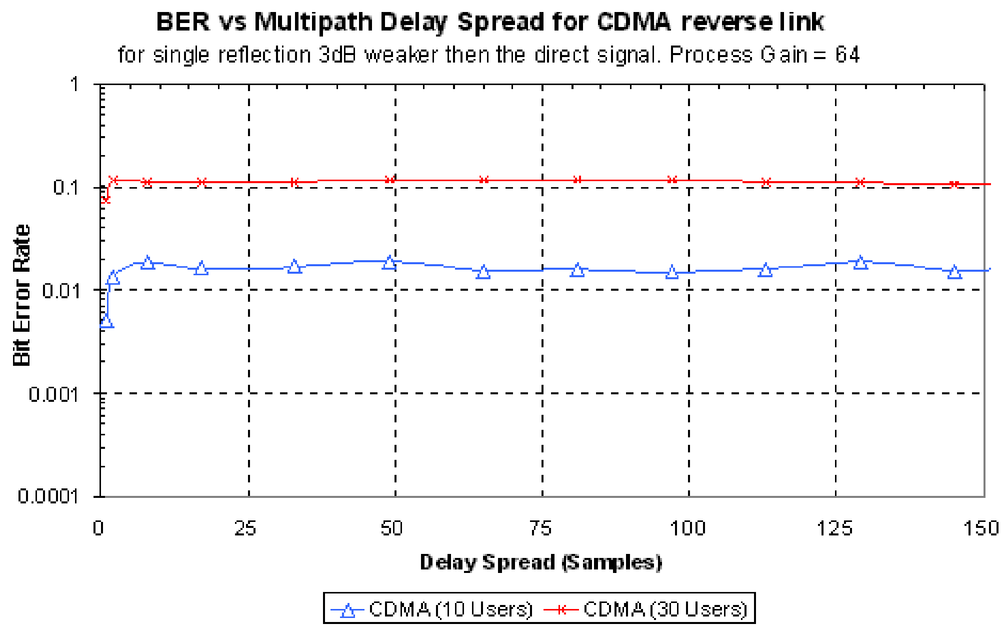
It was found that the SNR performance of OFDM is similar to a standard single carrier digital transmission. This is to be expected, as the transmitted signal is similar to a standard Frequency Division Multiplexing (FDM) system. Figure 1 shows the results from the simulations. The results show that using 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.



4.2 CDMA Simulation Results

4.2.1 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.



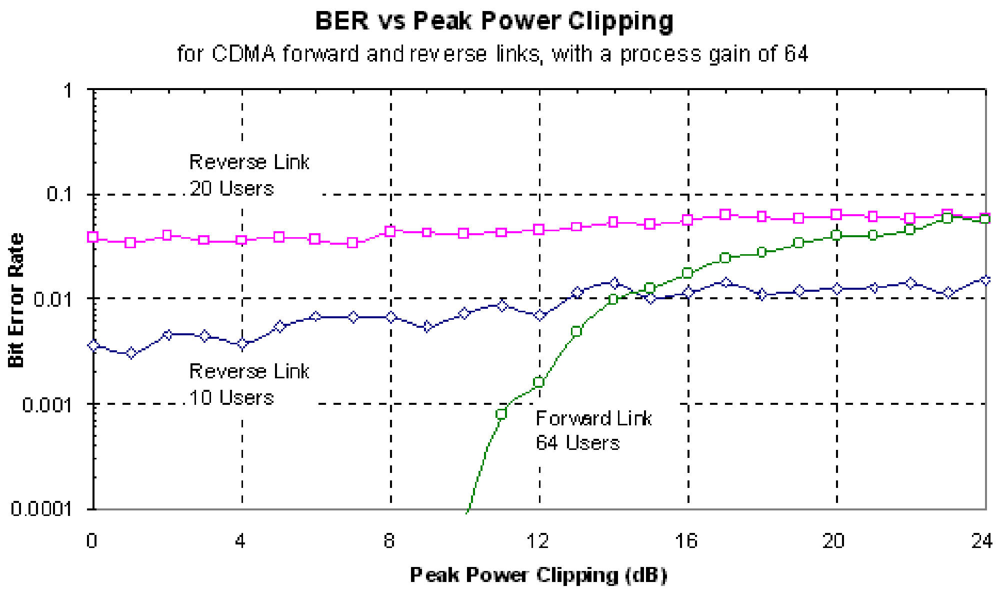
The multipath delay spread leads to an increase in the equivalent number of users in the cell, as it increases the amount of interference seen by the receiver.

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.

4.2.2 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.



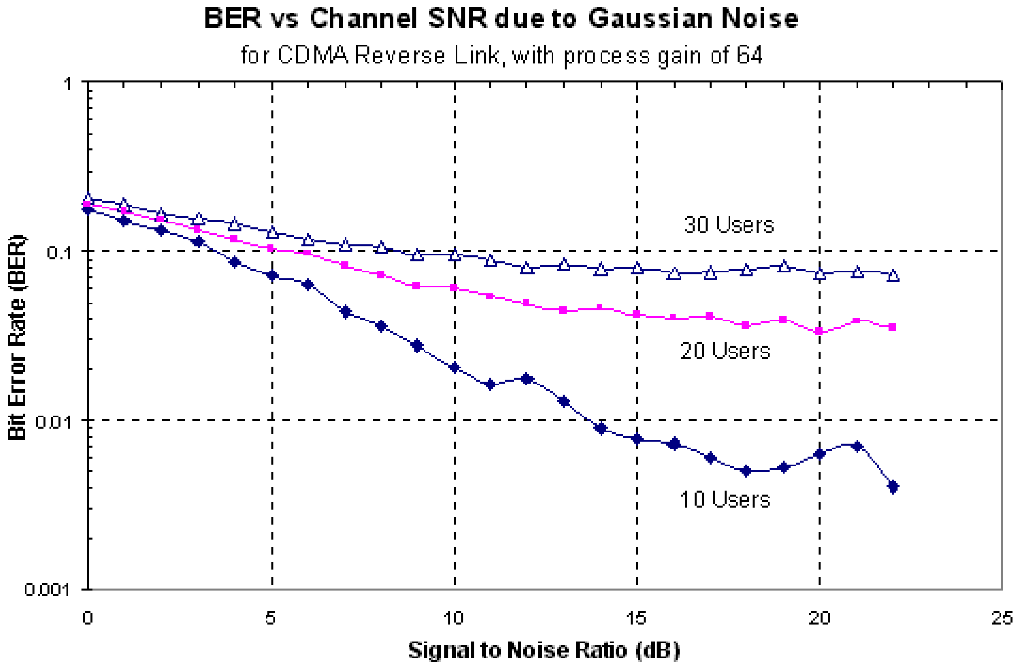
For the reverse link the BER starts high initially due to inter-user interference. The peak power clipping of the signal has little effect on the reverse link because the extra noise due to the distortion is not very high compared with the inter-user interference, plus any added noise is reduced by the 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.

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. 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.

4.2.3 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 interuser 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.



5. 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 were 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.

BIBLIOGRAPHY

1. T. Rappaport, "Wireless Communications, Principle & Practice", IEEE Press, Prentice Hall, pp. 3, 1996

2. www.skydsp.com/publications/4thyrthesis

3. www.webopedia.com/TERM/O/OFDM

4. www.eg3.com/WebID/mobile/ofdm/blank/paper/a-z.htm

5. http://bwrc.eecs.berkeley.edu/People/Grad_Students/dejan/ee225c/ofdm.pdf

6. http://www.eg3.com/mobile/index.htm

7. http://www.iec.org/online/tutorials/gsm/

8. http://www.intel.com/netcomms/technologies/wimax/303787.pdf

9. http://bwrc.eecs.berkeley.edu/People/Grad_Students/dejan/ee225c/ofdm.pdf

10. http://www.comsoc.org/livepubs/pci/public/2006/dec/index.html

11. http://rfdesign.com/images/archive/0101Puegel30.pdf

12. http://cegt201.bradley.edu/projects/proj2001/ofdmabsh/OFDM_Final_Report.pdf

13. http://cegt201.bradley.edu/projects/proj2001/ofdmabsh/OFDM_Final_Report.pdf

14. http://www.zdnet.com.au/membership/register.htm

15. www.ubicom.tudelft.nl/MMC/Docs/introOFDM.pdf

16. http://wirelessman.org/tutorial/80216t-00_02.pdf

17. http://www.wave-report.com/tutorials/OFDM.htm

18. www.ece.mtu.edu/faculty/ztian/ee5560/mud_project03.htm

19. http://www.zdnet.com.au/whitepaper/0,2000063328,22104691p-16001478q,00.htm

20. http://www.mathworks.com

21. http://www.educypedia.be/electronics/rfdigmod.htm

22. http://www.complextoreal.com/tutorial.htm

23. http://en.wikipedia.org/wiki/GSM_frequency_ranges