NEAR EAST UNIVERSITY

Faculty of Engineering

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ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING (OFDM)

Graduation Project
EE- 400

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Nicosia – 2006

ACKNOWLEDGMENTS

First of all I would like to thanks ALLAH for guiding me through my study.

More over I want to pay special regards to my parents who are enduring these all expenses and supporting me in all events. I am nothing without their prayers. They also encouraged me in crises. I shall never forget their sacrifices for my education so that I can enjoy my successful life as they are expecting.

Also, I feel proud to pay my special regards to my project adviser "Dr. Ali Serener". He never disappointed me in any affair. He delivered me too much information and did his best of efforts to make me able to complete my project. He has Devine place in my heart and I am less than the half without his help. I am really thankful to my teacher.

The best of acknowledge, I want to honor those all persons who have supported me or helped me in my project. I also pay my special thanks to my all friends who have helped me in my project and gave me their precious time to complete my project. Also my especial thanks go to my friends, Murad elsmade, Samer Hattab, Omar Yasin, Abo Maloh, Sameer ghannem, Fehmi, Waled oteala.

At the end I am again thankful to the doctors and professors encouraged me to complete my project.

ABSTRACT

Nowadays technology is growing quickly, and communication theory one of the fields that is developing frequently.

This report discusses orthogonal frequency division multiplexing (OFDM), which is used for high speed rate communication. During transmission in the channel intersymbol interference (ISI) and fading distorts the signal.

The main goal of this report is that to show OFDM is robust against intersymbol interference and fading.

MATLAB simulations are performed to show that OFDM improves the performance compared to single carrier communication systems operating in slow flat fading channels.

It is also proved using simulations that no performance improvement is obtained. When OFDM is used in additive white Gaussian noise (AWGN) channel. The results are compared to Shannon limit, which is the bound for error free communication.

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INTRODUCTION

Communication channels suffer from the effects of multipath, and intersymbol interference (ISI). Multipath occurs from the reflection in difference phases, and it creates small scale fading. Intersymbol interference occurs when transmitted pulses are distorted in the channel, when they interfere with each other.

This report investigates single carrier and multiple carrier communications over various types of channels. It particularly investigates the additive white Gaussian noise (AWGN) channel and slow flat fading channel. Using MATLAB simulations, the performances of a orthogonal frequency division multiplexing (OFDM) modulated multi-carrier communication system and a single carrier communication system over the AWGN and slow, flat fading channels are analyzed. The results confirm the superiority of OFDM over single carrier system.

The first chapter describes the elements of a digital communication system and the noise within the system.

Chapter two describes fading and intersymbol interference in detail and shows how they affect the transmitted signal.

Chapter three gives the details of OFDM.

Finally, chapter four includes the results obtained through simulation. The MATLAB code is included in the appendix section.

CHAPTER ONE

COMMUNICATION SYSTEM OVERVIEW

1.1 Communication system

Electrical communication systems are designed to send messages or information from a source that generates the messages to one or more destinations. In general, a communication system can be represented by the functional block diagram shown in Figure 1.1. The information generated by the source may be of the form of voice (speech source), a picture (image source), or plain text in some particular language, such as English, Japanese, German, French, etc. An essential feature of any source that generates information is that its output is described in probabilistic terms; i.e., the output of a source is not deterministic. Otherwise, there would be no need to transmit the message.

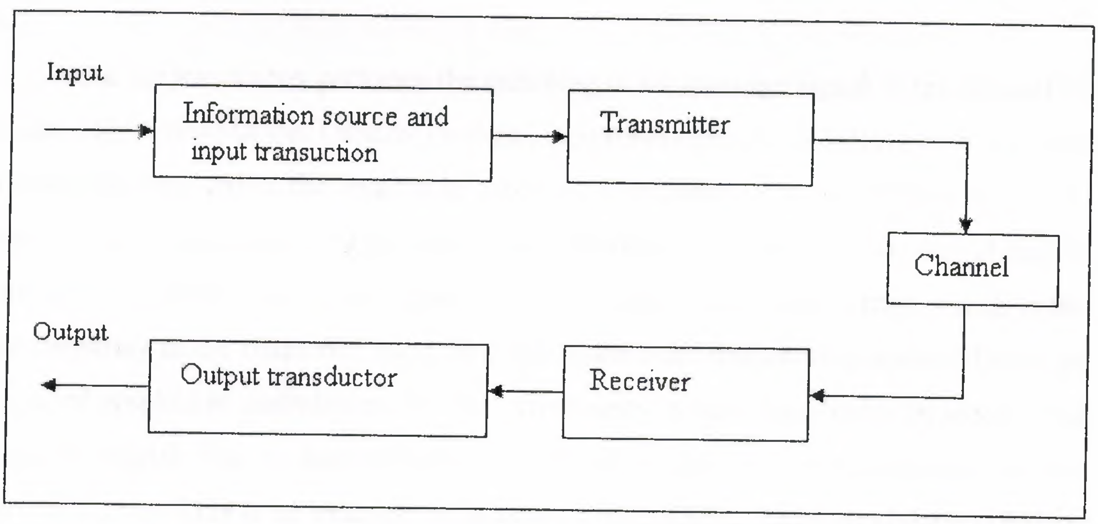


Figure 1.1 Functional block diagram of communication system [1].

A transducer is usually required to convert the output of a source into an electrical signal that is suitable for transmission. For example, a microphone serves as the transducer that converts an acoustic speech signal into an electrical signal, and a video camera converts an image into an electrical signal. At the destination, a similar transducer is required to convert the electrical signals that are received into a form that is suitable for the user: e.g. Acoustic signals, images, etc.

The heart of the communication system consists of three basic parts, namely, the transmitter, the channel and the receiver. The functions performed by these three elements are described next.

1.1.1 The Transmitter

The transmitter converts the electrical signal into a form that is suitable for transmission through the physical channel or transmission medium. In radio and TV broadcast, the Federal Communications Commission (FCC) specifies the frequency range for each transmitting station. Hence the transmitter must translate the information signal in be transmitted into the appropriate frequency range mat matches the frequency allocation assigned to the transmitter, thus signals transmitted by multiple radio stations do not interfere with one another. Similar functions are performed in telephone communication systems where the electrical speech signals from many users are transmitted over the same wire.

In general the transmitter performs the matching of the message signal to the channel by a process called modulation. Usually modulation involves the use of information signal to systematically vary either the amplitude, frequency, or phase of a sinusoidal carrier. For example in AM (Amplitude modulation) radio broadcast, the information signal that is transmitted is contained in the amplitude variations of the sinusoidal carrier, which is the center frequency in the frequency band allocated to the radio transmitting station, This is an example of amplitude modulation. In FM (frequency modulation) radio broadcast, the information signal that is transmitted is contained in the frequency variations of the sinusoidal carrier. This is an example of frequency modulation. Phase modulation (PM) is yet a third method for impressing the information signal on a sinusoidal carrier.

In general carrier modulation such as AM, FM, and PM is performed at the trans-mitter, as indicated above to convert the information signal to a form that matches the characteristics of the channel. Thus through the process of modulation, the information signal is translated in frequency to match the allocation of the channel. The choice of the type of modulation is based on several factors, such as the amount of bandwidth allocated, the types of noise and interference that the signal encounters m transmission over the channel, and the electronic devices that are available for signal amplification prior to transmission. In AM case, the modulation process makes it possible for us to accommodate the transmission of multiple messages from many users over the same physical channel.

In addition to modulation, other functions that are usually performed at the transmitter are filtering of the information bearing signal, amplification of the modulated signal, and in the case of wireless transmission, radiation of the signal by means of a transmitting antenna.

1.1.2 The Channel

The communications channel is the physical medium that is used to send the signal from the transmitter to the receiver. In wireless transmission the channel is usually the atmosphere (free space). On the other hand telephone channels usually employ a variety of physical media, including wireless optical fiber cables and wireless (microwave radio), whatever the physical medium for signal transmission. The essential feature is that the transmitted signal is corrupted in a random manner by a variety of possible mechanisms.

The most common form of signal degradation comes in the form of additive noise, which is generated at the front end of the receiver, where signal amplification is performed, This noise is often called thermal noise. In wireless transmission additional additive disturbances are man made noise, and atmospheric noise picked up by a receiving antenna. Automobile ignition noise is an example of man made noise, and electrical lightning discharges from thunderstorms is an example of atmospheric noise. Interference from other users of the channel is another form of additive noise that often arises in both wireless and wireless communication systems.

In some radio communication channels, such as the ionospheric channel that is used for long range, short-wave radio transmission another form of signal degradation is multi-path propagation.

Such signal distortion is characterized as a non-additive signal disturbance which manifests itself as time variations in the signal amplitude, usually called fading. This phenomenon is described in more detail in the following section.

Both additive and non-additive signal distortions are usually characterized as random phenomena and described in statistical terms. The effect of these signal distortions must be taken into account on the design of the communication system.

In the design of a communication system, the system designer works with mathematical models that statistically characterize the signal distortion encountered on physical channels. Often the statistical description that is used in a mathematical model is a result of actual empirical measurements obtained from experiments involving signal transmission over such channels, In such cases, there is a physical justification for the mathematical model used in the design of communication systems. On the other hand in some communication system designs, the statistical characteristics of the channel may vary significantly with time. In such cases the system designer may design a communication system that is robust to the variety of signal distortions. This can be accomplished by having the system adapt some of its parameters to the channel distortion encountered.

1.1.3 The Receiver

The function of the receiver is to recover the message signal contained in the received signal. If the message signal is transmitted by carrier modulation, the receiver performs carrier demodulation in order to extract the message from the sinusoidal carrier. Since the signal demodulation is performed in the presence of additive noise and possibly other signal distortion, the demodulated message signal is generally degraded to some extent by the presence of these distortions in the received signal. The fidelity of the received message signal is a function of the type of modulation the strength of the additive noise, the type and strength of any other additive interference, and the type of any non-additive interference.

Besides performing the primary function of signal demodulation, the receiver also performs a number of peripheral functions, including signal filtering and noise suppression.

1.2 Digital Communication System

Up to this point, we have described an electrical communication system in rather broad terms based on the implicit assumption that the message signal is a continuous time varying waveform. We refer to such continuous time signal waveforms as analog signals and to the corresponding information sources that produce such signals as analog sources. Analog signals can be transmitted directly via carrier modulation over the communication channel and demodulated accordingly at the receiver we call such a communication system an analog communication system.

Alternatively, an analog source output may be converted into a digital form and the message can be transmitted via digital modulation and demodulated as a digital signal at the receiver. There are some potential advantages to transmitting an analog signal by means of digital modulation. The most important reason is that signal fidelity is better controlled through digital transmission than analog transmission in particular. Digital transmission allows us to regenerate the digital signal in long distance transmission, thus eliminating effects of noise at each regeneration point. In contrast, the noise added in analog transmission is amplified along with the signal when amplifiers are used periodically to boost the signal level in long distance transmission.

Another reason for choosing digital transmission over analog is that the analog message signal may be highly redundant with digital processing, Redundancy may be removed prior to modulation thus conserving channel bandwidth, yet a third reason may be that digital communication systems are often cheaper to implement.

In some applications, the information to be transmitted is inherently digital (in the form of English text, computer data, etc.). In such cases the information source that generates the data is called a discrete (digital) source.

In a digital communication system, the functional operations performed at the transmitter and receiver must be expanded to include message signal discretization at the transmitter and message signal synthesis or interpolation at the receiver. Additional functions include redundancy removal, and channel codding and decoding.

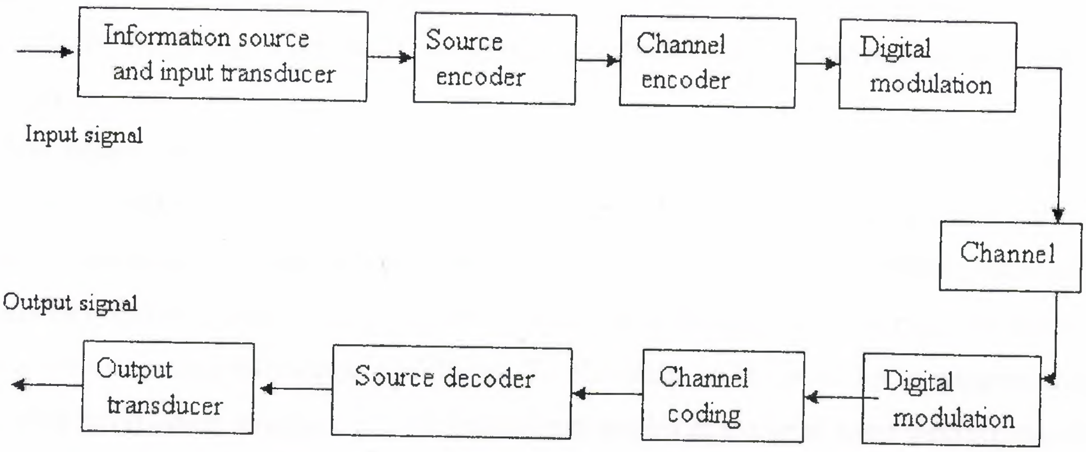


Figure 1.2 Basic elements of a digital communication system [1].

Figure 1.2 illustrates the functional diagram and the basic elements of digital communication system, the source output may be either an analog signal such as audio of video signal or a digital signal such as the output of a computer, which as discrete in time and has a finite number of output characters. In a digital communication system the messages produced by the source are usually converted into a sequence of binary digits as possible. In other words we seek an efficiently converting the output of either an analog or a digital source into a sequence of binary digits is called source encoding or data compression.

The sequence of binary digits from the source encoder, which we call the information sequence is passed to the channel encoder. The purpose of the channel encoder is to introduce in a controlled manner. Some redundancy in the binary information sequence, which can be used at the receiver to overcome the effects of noise and interference encountered in the transmission of the signal through the channel. Thus the added redundancy serves to increase the reliability of the received data and improves the fidelity of the received signal. In effect, redundancy in the information sequence aids the receiver in decoding the desired information sequence. A (trivial) form of encoding of the binary information sequence is simply to repeat each binary digit m times where m is some positive integer.

More sophisticated (non-trivial) encoding involves taking k information bits at a time and mapping each k bit sequence into a unique n-bit sequence called a code word. The amount of redundancy introduced by encoding the data in this manner is measured by the ratio n/k, the reciprocal of this ratio namely k/n. is called the rate of the code or simply the code rate.

The binary sequence at the output of the channel encoder is passed to the digital modulator, which serves as the interface to the communications channel. Since nearly all of the communication channels encountered in practice are capable of transmitting electrical signals (waveforms), the primary purpose of the digital modulator is to map the binary information sequence into signal waveforms. To elaborate on the point let us suppose that the coded information sequence is to be transmitted one bit at a time at some uniform rate R bits/s. The digital modulator may simply map the binary digit 0 into a waveform $S_0(t)$ and the binary digit 1 into a waveform $S_1(t)$, In this manner each bit from the channel encoder is transmitted separately.

We call this binary modulator alternatively, she modulator may transmit b coded information bits at a time by using $M = 2^B$ distinct waveforms $S_i(t)$ where i=0,1,...M-1 one waveform for each of the 2^B possible B bit sequences. We call this M-ary modulation (M > 2). Note that a new B bit sequence enters the modulator every B/R seconds, Hence when the channel bit rate R is fixed the amount of time available to transmit one of the M waveforms corresponding to B bit sequence is B times the time period in a system that uses binary modulation.

At the receiving end of a digital communications system, the digital demodulator processes the channel corrupted transmitted waveform and reduces each waveform to a single number that represents an estimate of the transmitted data symbol (binary or M-ary). When binary modulation is used the demodulator may process the received waveform and decide on whether the transmitted bit is a 0 or a 1. In such a case, we say the demodulator has made a binary decision.

As one alternative, the demodulator may make a ternary decision, it decides that the transmitted bit is either a 0 or 1 or it makes no decision at all, depending on the apparent quality of the received signal when no is made on a particular bit, we say mat the demodulator has inserted an erasure in the demodulated data.

Using the redundancy in the transmitted data the decoder attempts to fill in the positions where ensures occurred. Viewing the decision process performed by the demodulator as a form of quantization we observe there binary and ternary decisions are special cases of a demodulator that quantizes to Q levels where $Q \geq 2$, In general of the digital communications system employs Mary modulation where m=0,1,....,M-1 represent the M possible transmitted symbol each corresponding to $b = \log_2 M$ bits. The demodulator may make a Q-ary decision, where $Q \geq M$ in the extreme case where no quantization is performed $Q = \infty$.

When there is no redundancy in the transmitted information, the demodulator must decide which of the M waveforms was transmitted in any given time interval. Consequently Q=M and since there is no redundancy in the transmitted information no discrete decoder is used following the demodulator. On the other hand when there is redundancy introduced by a discrete channel encoder at the transmitter the Q-ary output from the demodulator occurring every b/R seconds is fed to the decoder which attempts to reconstruct the original information sequence from knowledge of the code used by the channel encoder and the redundancy contained in the receiver data.

A measure of how well the demodulator and encoder perform is the frequency with which errors occur in the decoded sequence. More precisely the average probability of a bit error at the output of the decoder is a measure of the performance of the demodulation decoder combination.

In general, the probability of error is a function of the code characteristic, the types of waveforms is used to transmit the over the channel, the nature of noise, the nature of interference, and the of demodulation and decoding.

When an analog output is desired, the source decoder accepts the output sequence from the channel decoder and, from knowledge of the source encoding method used, attempts to reconstruct the original signal from the source. Due to channel decoding errors and possible distortion introduced by the source encoder and perhaps the source decoder, the signal at the output of the source decoder is an approximation to the original source output. The difference or some function of the difference between the original signal and the reconstructed signal, is a measure of the distortion introduced by the digital communications system.

1.3 Characteristic of communication channels

As indicated in our preceding discussion the communication channel provides the connection between the transmitter and the receiver. The physical channel may be a pair of wires that carry the electrical signal, or an optical fiber that carries the information on a modulated light beam, or an underwater ocean channel in which the information is transmitted acoustically, or free space over which the information bearing signal is radiated by use of an antenna. Other media that can be characterized as communication channels are data storage media, such as magnetic tape, magnetic disks, and optical disks.

One common problem in signal transmission through any channel is additive noise. In general, additive noise is generated internally by components such as resistors and solid-state devices used to implement the communication system. This is sometimes called thermal noise. Other sources of noise and interference may arise externally to the system, such as interference from other users of the channel. When such noise and interference occupy the same frequency band as the desired signal, its effect can be minimized by proper design of the transmitted signal and its demodulator at the receiver. Other types of signal degradations that may be encountered in transmission over the channel are signal attenuation, amplitude and phase distortion, and multi-path distortion.

The effects of noise may be minimized by increasing the power in the transmitted signal. However equipment and other practical constraints limit the power level in the transmitted signal. Another basic limitation is the available channel bandwidth. A bandwidth constraint is usually due to the physical limitations of the medium and the electronic components used to implement the transmitter and the receiver. These two limitations result in constraining the amount of data that can he transmitted reliably over any communications channel. Shannon's basic results relate the channel capacity to the available transmitted power and channel bandwidth.

1.3.1 Wireline Channels

The telephone network makes extensive use of wire lines for voice signal transmission, as well as data and video transmission.

Twisted-pair wire lines and coaxial cable are basically guided electromagnetic channels which provide relatively modest bandwidths, telephone wire generally used to connect a customer to a central office has a bandwidth of several hundred kilohertz (KHz). On the other hand coaxial cable has a usable bandwidth of several megahertz (MHz).

Signals transmitted through such channels are distorted in both amplitude and phase, and further corrupted by additive noise. Twisted-pair wire-line channels are also prone to crosstalk interference from physically adjacent channels. Because wire-line channels carry a large percentage of our daily communications around the country and the world much research has been performed on the characterization of their transmission properties and on methods for mitigating the amplitude and phase distortion encountered in signal transmission.

1.3.2 Fiber optic channels

Optical fibers offer the communication system designer a channel bandwidth that is several orders of magnitude larger than coaxial cable channels. During the past decade optical fiber cables have been developed which have a relatively low signal attenuation and high reliable photonic devices have been developed for signal generation and signal detection. These technological advances have resulted in a rapid deployment of optical fiber channels both in domestic telecommunication systems as well as for transatlantic and transpacific communication. With the large bandwidth available on fiber optic channels it is possible for the telephone companies to offer subscribers a wide array of telecommunication services, including voice, data, facsimile, and video.

The transmitter or modulator in a fiber optic communication system is a light source either a light-emitting diode (LED) or a laser. Information is transmitted by varying {modulating) the intensity of the light source with the message signal. The signal propagates through the fiber as a light wave and is amplified periodically (in the case of digital transmission, it is detected and regenerated by repeaters) along the transmission path to compensate for signal attenuation.

At the receiver, the light intensity is detected by a photodiode, whose output is an electrical signal that varies in direct proportion to the power of the light impinging on the photodiode.

It is envisioned that optical fiber channels will replace nearly all wire-line channels in the telephone network in the next few years.

1.3.3 Wireless electromagnetic channels

In radio communication systems, electromagnetic energy is coupled to the propagation medium by an antenna, which serves as the radiator, the physical size and the configuration of the antenna depend primarily on the frequency of operation.

To obtain efficient radiation of electromagnetic energy, the antenna must be longer than 1/10 of the wavelength. Consequently, a radio station transmitting in the AM frequency band, say at 1 MHz (corresponding to a wavelength of a $\lambda = \frac{c}{f_c}$ 300 m) requires an antenna of at least 30 meters.

The mode of propagation of electromagnetic waves in the atmosphere and in free space may be subdivided into three categories, namely, ground-wave propagation, sky-wave propagation, and line-of-sight (LOS) propagation. In the VLF and ELF frequency bands, where the wavelengths exceed 10 km, the earth and the ionosphere act as a waveguide for electromagnetic wave propagation. In these frequency ranges, communication signals practically propagate around the globe. For this reason, these frequency bands are primarily used to provide navigational aids from shore to ships around the world. The channel band widths available in these frequency bands are relatively small (usually from 1-10% of the center frequency), and hence, the information that is transmitted through these channels is relatively 4ow speed and, generally, confined to digital transmission.

A dominant type of noise at these frequencies is generated from thunderstorm activity around the globe, especially in tropical regions interference results from the many users of these frequency bands.

Ground-wave propagation, is the dominant mode of propagation for frequencies in the MF band (0.3-3 MHz). This is the frequency band used for AM broadcasting and maritime radio broadcasting. In AM broadcast the range with ground-Wave propagation of even the more powerful radio station is limited to about 100 miles. Atmospheric noise man made noise, and thermal noise from electronic components at the receiver are dominant disturbances for signal transmission of MF(Medium frequency).

Sky-wave propagation results from transmitted signals being reflected (bent or refracted) from the ionosphere, which consists of several layers of charged particles ranging in altitude from 30-250 miles above the surface of the earth. During the daytime hours the heating of the Slower atmosphere by the sun causes the formation of the lower layers at altitudes below 75 miles.

These lower layers, especially the D-layer serve to absorb frequencies below 2 MHz, thus, severely limiting sky-wave propagation of AM radio broadcast. However, during the night-time hours, the electron density in the lower layers of the ionosphere drops sharply and the frequency absorption that occurs during the day time is significantly reduced. As a consequence, powerful AM radio broadcast stations can propagate over large distances via sky-wave over the F-layer of the ionosphere, which ranges from 90-250 miles above the surface of the earth.

A frequently occurring problem with electromagnetic wave propagation via skywave in the HF frequency range is signal multi-path. Signal multi-path occurs when the transmitted signal arrives at the receiver via multiple propagation paths at different delays. Signal multipath generally results in inter-symbol interference in a digital communication system.

Moreover the signal components arriving via different propagation paths may add destructively, resulting in a phenomenon called signal fading, which most people have experienced when listening to a distant radio station at night, when skywave is the dominant propagation mode, additive noise at HF is a combination of atmospheric noise and thermal voice.

Skywave ionosphere propagation ceases to exist at frequencies above approximately 30 MHz, which is the end of the HF band. However, it is possible to have ionospheric scatter propagation at frequencies in the range of 30-60 MHz, resulting from signal scattering from the lower ionosphere. It is also possible to communicate over distances of several hundred miles by use of tropospheric scattering at frequencies in the range of 40-300 MHz. Troposcatter results from signal scattering due to particles in the atmosphere at altitudes of 10 miles or less. Generally, ionospheric scatter and tropospheric scatter involve large signal propagation losses and require a large amount of transmitter power and relatively large antennas.

Frequencies above 30 MHz propagate through the ionosphere with relatively little loss and make satellite and extraterrestrial communications possible. Hence, at frequencies in the VHF band and higher, the dominant mode of electromagnetic propagation is line of signal (LOS) propagation. For terrestrial communication systems, this means that the transmitter and receiver antennas must be in direct LOS with relatively little or no obstruction. For this reason, television stations transmitting in the VHF and UHF frequency bands mount their antennas on high towers in order to achieve a broad coverage area.

In general, the coverage area for LOS propagation is limited by the curvature of the earth. If the transmitting antenna is mounted at a height h feel above the surface of the earth, the distance to the radio horizon, assuming no physical obstructions such a mountains is approximately $d = \sqrt{2h}$ miles. For example, a TV antenna mounted on a tower of 1000 ft in height provides a coverage of approximately 50 miles. As example, microwave radio relay systems used extensively for telephone and video transmission at frequencies above 1 GHz have antennas mounted on tall towers or on the top of tall buildings.

1.3.4 Underwater acoustic channel

Over the past few decades ocean exploration activity has been steadily increasing, Coupled with that increase in ocean exploration is the need to transmit data, collected by sensors placed underwater to the surface of the ocean. From there it is possible to relay the data via a satellite to a data collection center.

Electromagnetic waves do not propagate over long distances underwater, except at extremely low frequencies. However the transmission of signals at such low frequencies is prohibitively expensive because of the large and powerful transmitters required. The attenuation of electromagnetic waves in water can be expressed in terms of the skin depth, which is the distance a signal is attenuated by 1/e. For sea water the skin depth $\delta = 250/\sqrt{f}$ where f is expressed in Hz and δ is in meters, For example at 10 kHz, the skin depth is 2.5 m, In contrast acoustic signals propagate over distances of tens and even hundreds of kilometers,

A shallow water acoustic channel is characterized as a multi-path channel due to signal reflections from the surface and the bottom of the sea. Due to wave motion the signal multipath components undergo time varying propagation delays which result in signal fading. In addition there is frequency dependent attenuation, which is approximately proportional to the square of the signal frequency,

Ambient ocean acoustic noise is caused by shrimp, fish and various mammals, Near harbors, there is also man made acoustic noise in addition to the ambient noise.

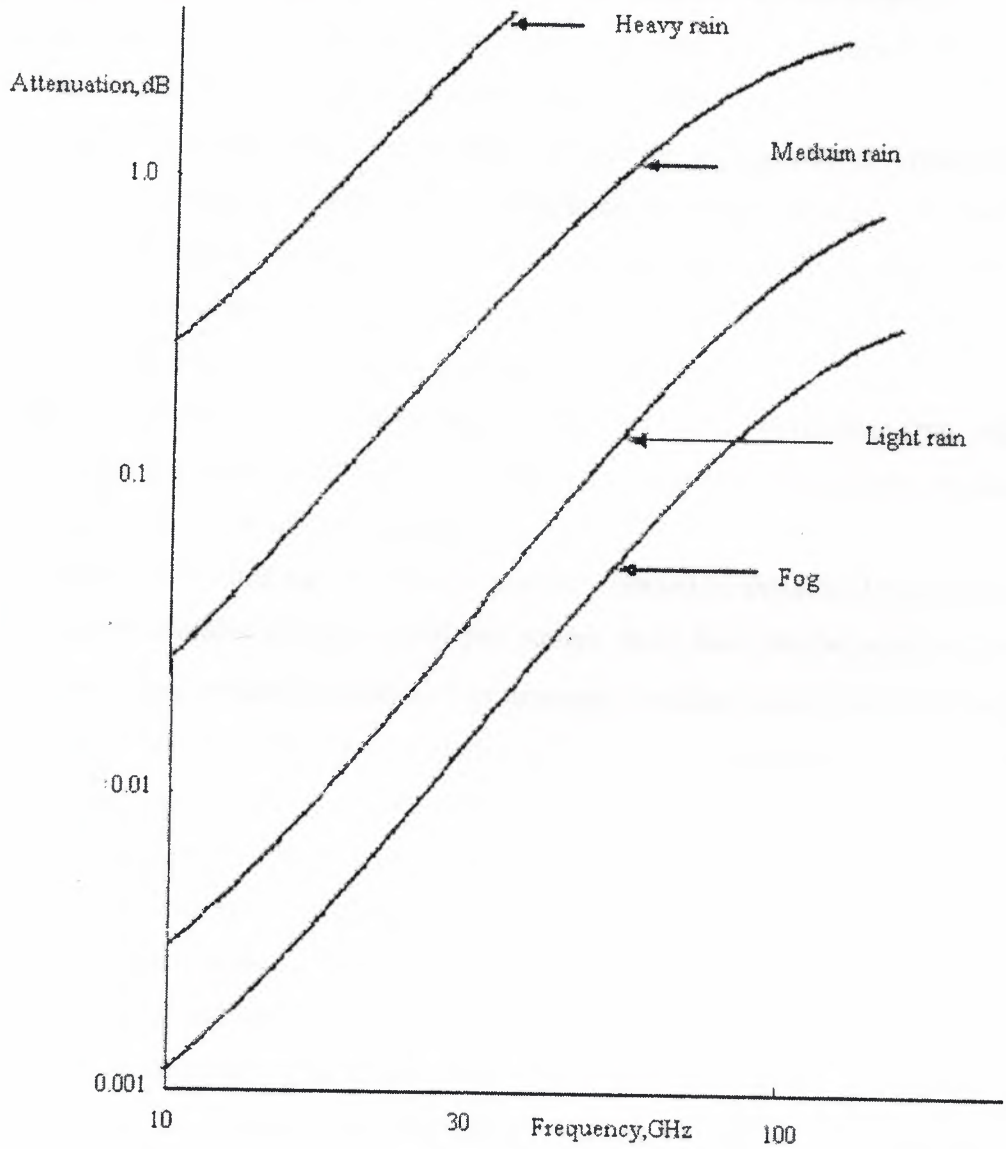


Figure 1.3 Signal attenuation due to precipitation [1].

In spite of this hostile environment, it is possible to design and implement efficient and highly reliable underwater acoustic communication systems for transmitting digital signals over large distances.

1.3.4 Storage channels

Information storage and retrieval systems constitute a very significant part of our data handling activities on a daily basis. Magnetic tape including digital audio tape and video tape magnetic disks used for storing large amounts of computer data, and optical disks used for computer data storage, music (compact disks) and video are examples of data storage systems that can be characterized as communication channels.

The process of storing data on a magnetic tape or a magnetic or optical disk is equivalent to transmitting a signal over a telephone or a radio channel. The readback process and the signal processing involved in storage systems to recover the stored information is equivalent to the functions performed by a receiver in a telephone or radio communication system to recover the transmitted information.

Additive noise generated by the electronic components and interference from adjacent tracks is generally present in the readback signal of a storage system, just as is the case in a telephone or a radio communication system.

The amount of data that can be stored is generally limited by the size of the disk or tape and the density (number of bits stored per square inch) that can be achieved by the write/read electronic systems and heads. For example a packing density of 10^9 bits/sq, in has been recently demonstrated in an experimental magnetic disk storage system, (Current commercial magnetic storage products achieve a much lower density), the speed at which data can be written on a disk or tape and the speed at which it can be read back is also limited by the associated mechanical and electrical subsystems that constitute an information storage system.

Channel coding and modulation are essential components of a well designed digital magnetic or optical storage system. In the readback process, the signal is demodulated and the added redundancy introduced K the channel encoder is used to correct errors in the readback signal.

1.4 Mathematical models for communication channels

In the design of communication systems for transmitting information through physical channels, we find it convenient to construct mathematical models that reflect the most important characteristics of the transmission medium. Then the mathematical model for the channel is used in the design of the channel encoder and modulator at the transmitter and the demodulator and channel decoder at the receiver.

Next we provide a brief description of the channel models that are frequently used to characterize many of the physical channels that we encounter in practice.

1.4.1 The additive noise channel

The simplest mathematical model for a communication channel is the additive noise channel, illustrated in Figure 1.4. In this model the transmitted signal s{t) is corrupted by an additive random noise process n(t).

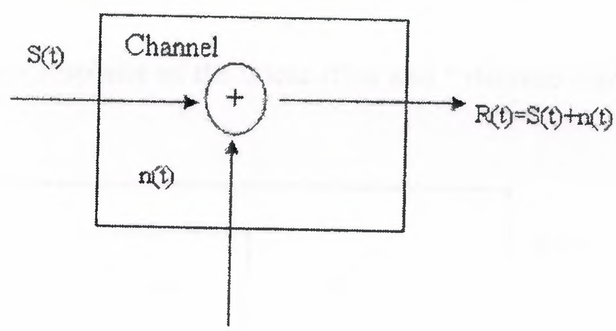


Figure 1.4 The additive noise channel [1].

Physically the additive noise process may arise from electronic components and amplifiers at the receiver of the communication system, or from interference encountered in transmission, as in case of radio signal transmission.

If the noise is introduced primarily by electronic components and amplifiers at the receiver, it may be characterized as thermal noise. This type of noise is characterized statistically as Gaussian noise process. Hence, the resulting mathematical model for the channel is usually called the additive Gaussian noise channel. Because this channel model applies to a board class of physical communication channels and because of its mathematical tractability, this is the predominant channel model used in our communication system analysis and design. Channel attenuation is easily incorporated into the model. When the signal undergoes attenuation in transmission through the channel , the received signal is

$$R(t) = as(t) + n(t) \tag{1.1}$$

where a represents the attenuation factor.

1.4.2 The Linear filter channel

In some physical channels such as wire-line telephone channels, filters are used to ensure that the transmitted signals do not exceed specified bandwidth limitations and, thus, do not interfere with one another. Such channels are generally characterized mathematically as linear filter channels with additive noise, as illustrated in Figure 1.5. Hence if the channel input is the signal S(t) the channel output is the signal

$$R(t) = S(t) * h(t) + n(t) = \int_{-\infty}^{+\infty} h(\tau)S(t-\tau)d\tau + n(t) \tag{1.2}$$

where h(t) is the impulse response of the linear filter and * denotes convolution.

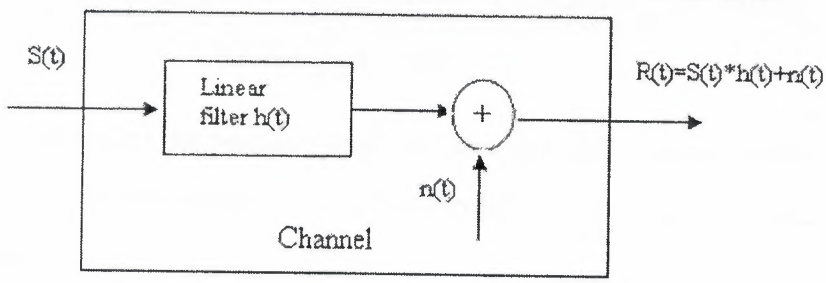


Figure 1.5 The linear filter channel with additive noise [1].

1.4.3 The Linear Time-Variant Filter Channel

In some physical channels such as underwater acoustic channels and ionospheric radio channels which result in time variant multi-path propagation of the transmitted signal may be characterized mathematically as time-variant linear filters. Such linear filters are characterized by time-variant channel impulse response $h(\tau,t)$ where $h(\tau,t)$ is the response of the channel at time t, due to an impulse applied at time t- τ . Thus τ represents the "age" (elapsed time) variable. The linear time variant filter channel with additive noise is illustrated Figure 1.6 for an input signal S(t) the channel output signal is

$$R(t) = S(t) * h(\tau;t) + n(t) = \int_{-\infty}^{\infty} h(\tau;t) S(t-\tau) d\tau + n(t) \tag{1.3}$$

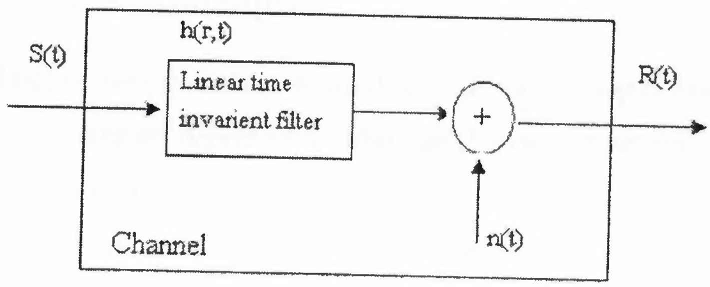


Figure 1.6 Linear time variant filter channel with additive noise [1].

A good model for multi-path signal propagation through physical channels, such as the ionosphere (at frequencies bellow 30 MHz) and mobile cellular radio channels, is a special case of Equation (1.3) is which the time-variant impulse response has the form

$$h(\tau;t) = \sum_{K=1}^{L} a_K(t) \delta(\tau - \tau_K) \tag{1.4}$$

Where the a_K represent the possibly time invariant attenuation factors for the L multi-path propagation paths, if equation (1.4) is substituted into equation (1.3) the received signal has the form

$$R(t) = \sum_{K=1}^{L} a_K(t) s(t - \tau_K) + n(t) \tag{1.5}$$

Hence the received signal consists of L multi-path components, where each component is attenuated by $\{a_K\}$ and delayed by $\{\tau_K\}$.

The three mathematical models described above adequately characterize a large majority of physical channels encountered in practice, these three channel models are used in this next for the analysis and design of communication systems.

CHAPTER TWO

FADING AND INTERFERENCE

2.1 Small scale fading and multipath

Small-scale fading, or simply fading, is used to describe the rapid fluctuations of the amplitudes, phases, or multipath delays of a radio signal over a short period or travel distance, so that large-scale path loss effects may be ignored.

Fading is caused by interference between two or more versions of the transmitted signal, which arrive at the receiver at slightly different times. These waves called multipath waves combine at the receiver antenna to give a resultant signal which can vary widely in amplitude and phase, depending on the distribution of the intensity and relative propagation time of the waves and the bandwidth of the transmitted signal.

2.1.1 Small-scale multipath propagation

Multipath in the radio channel creates small-scale fading effects. The three most important effects are:

- Rapid changes in signal strength over a small travel distance or time interval.
- Random frequency modulation due to varying Doppler shifts on different multipath signal.
- Time dispersion (echoes) caused by multipath propagation delays.

In built up urban areas, fading occurs because the height of the mobile antennas are well below the height of surrounding structures, so there is no single line of sight even when a line of sight exists. Multipath still occurs due to reflections from the ground and surrounding structures. The incoming radio waves arrive different direction with different propagation delays. The signal received by the mobile at any point in space may consist of a large number of plane waves having randomly distributed amplitudes, phases, and angle of arrival.

These multipath components combine vectorially at the receiver antenna, and can cause the signal received by the mobile to distort or fade, even when a mobile receiver is stationary. The received signal may fade due to movement of surrounding objects in the radio channel.

If objects in the radio channel are static, and motion is considered to be only due to that of the mobile, then fading is purely a spatial phenomenon. The spatial variations of the resulting signal are seen as temporal variations by the receiver as it moves through the multipath field. Due to the constructive and destructive effects of multipath waves summing at various points in space, a receiver moving at high speed can pass through several fades in a small period of time.

In a more serious case a receiver may stop at a particular location at which the received signal is in a deep fade. Maintaining good communications can then become very difficult, although passing vehicles or people walking in the vicinity of the mobile can often disturb the field pattern, thereby diminishing the likelihood of the received signal remaining in a deep null for along period of time.

Due to the relative motion between the mobile and the base station, each multipath wave experiences an apparent shift in frequency. The shift in received signal frequency due to motion is called the Doppler shift, and is directly proportional to the velocity and direction of motion of the mobile with respect to the direction of arrival of the received multipath wave.

2.1.1.1 Factors influencing small scale fading

Many physical factors in the radio propagation channel influence small scale fading. These include the following:

- Multipath propagation

The presence of reflecting objects and scatterers in the channel creates a constantly changing environment that dissipates the signal energy in amplitude, phase, and time. These effects result in multiple versions of the transmitted signal that arrive at the receiving antenna, displaced with respect to one another in time and spatial orientation.

The random phase and amplitudes of the different multipath components cause fluctuations in signal strength, thereby inducing small-scale fading, signal distortion, or both. Multipath propagation often lengthens the time required for the base band portion of the signal to reach the receiver which can cause signal smearing due to inter symbol interference.

- Speed of the mobile

The relative motion between the base station and the mobile results in random frequency modulation due to different Doppler shifts on each of the multipath components. Doppler shift will be positive or negative depending on whether the mobile receiver is moving toward or away from the base station.

- Speed of surrounding objects

If objects in the radio channel are in motion, they induce a time varying Doppler shift on multipath components. If the surrounding objects move at a greater rate than the mobile, then this effect dominates the small-scale fading.

Otherwise, motion of surrounding objects may be ignored, and only the speed of rhe mobile need be considered. The coherence time defines the "staticsness" of the channel, and is directly impacted by the Doppler shift.

- The transmission bandwidth of the signal

If the transmitted radio signal bandwidth is greater than the "bandwidth" of the multipath channel, the received signal will be distorted, but the received signal strength will not fade much over a local area (the small scale signal fading will not be significant). As will be shown the bandwidth of the channel can be quantified by the coherence bandwidth which is related to the specific multipath structure of the channel.

The coherence bandwidth is a measure of the maximum frequency difference for which signals are still strongly correlated in amplitude. If the transmitted signal has a narrow bandwidth as compared to the channel, the amplitude of the signal will change rapidly, but the signal will not be distorted in time. Thus the statistics of small-scale signal strength and the likelihood of signal smearing appearing over small-scale distances are very much related to the specific amplitudes and delays of the multipath channel, as well as the bandwidth of the transmitted signal.

2.1.1.2 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, 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

$$\Delta f \cong \pm f_o \frac{c}{c} \qquad (2.1)$$

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.

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.

2.1.2 Impulse response model of a multipath channel

The small-scale variations of a mobile radio signal can be directly related to the impulse response of the mobile radio channel. The impulse response is a wideband channel characterization and contains all information necessary to simulate or analyze any type of radio transmission through the channel.

This stems from the fact that a mobile radio channel may be modeled as a linear filter with a time varying impulse response, where the time variation is due to receiver motion in space. The filtering nature of the channel is caused by the summation of amplitudes and delays of the multiple arriving waves at any instant of time.

The impulse response is a useful characterization of the channel, since it may be used to predict and compare the performance of many different mobile communication systems and transmission bandwidths for a particular mobile channel condition. In a radio link, the radio frequency (RF) signal from the transmitter may be reflected off objects such as hills, buildings, vehicles, walls etc. Some of these reflections will arrive at the receiver effectively creating multiple transmission paths, commonly referred to as a multipath environment. The radio signal travels over a different distance for each of these paths, and thus takes a different amount of propagation time.

If we were to transmit an RF pulse in a multipath environment, we would receive a signal like the one shown in Figure 2.1. (a) Each impulse corresponds to one path, with the strength of each impulse dependent on the path loss for that path, for a fixed frequency signal, (a sine wave) the propagation delay results in a phase rotation of the signal, the amount of phase rotation corresponds to 360° for each wavelength of path length traveled, each of the multipath signals will have a different propagation distance and thus a different phase rotation, these signals add at the receiver resulting in constructive or destructive interference.

Each of the multipath signals can be represented as a phasor, which has vector length corresponding to signal power and the angle corresponding to the phase, the received signal corresponds to the vector sum of the multipath phasors (see Figure 2.1 (b)). Destructive interference occurs when the vector sum adds to zero, this is also referred to as a 'null'. Constructive interference occurs when all the signals have a similar phase, reinforcing each other.

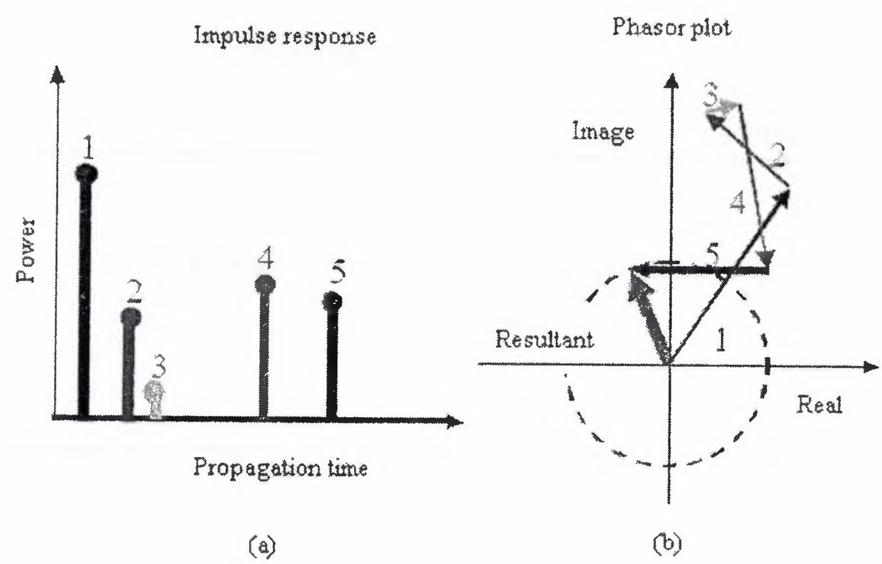


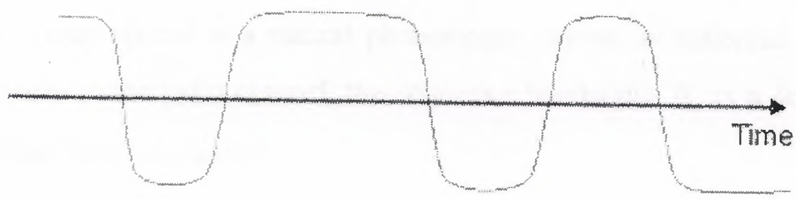
Figure 2.1 Impulse response and phasor plot for multipath channel [2].

2.1.3 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). 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.

1. Direct signal.



2. Reflected and delayed signal.

3. Resulting received signal showing multipath.

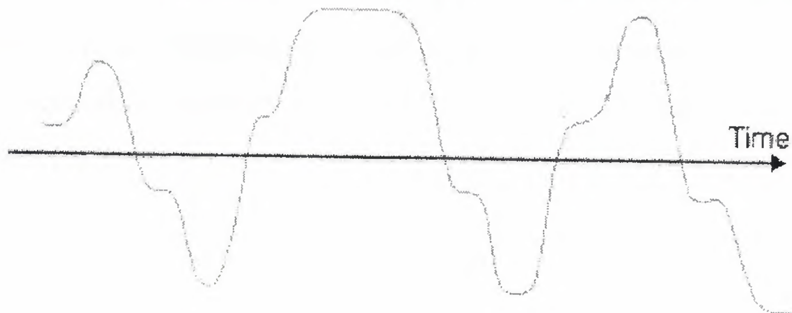


Figure 2.2 Multipath delay spread [2].

Table 2.1 Delay spread for various environments.

Environment	Delay spread	Maximum path length difference
Indoor (room)	40ns-200 ns	300m-6km
Outdoor	1μ - 20 μs	12m - 60m

Table 2.1 shows the typical delay spread for various environments, the maximum delay spread in an outdoor environment is approximately 20ms, thus significant intersymbol interference can occur at bit rates as low as 25 kbps.

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 (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.

2.1.4 Coherence bandwidth

While the delay spread is a natural phenomenon caused by reflected and scattered propagation paths in the radio channel, the coherence bandwidth B_c is a defined relation derived from the rms delay spread.

Coherence bandwidth is a statistical measure of the range of frequencies over which the channel can be considered "flat" (a channel which passes all spectra! components with approximately equal gain and linear phase). In other words, coherence bandwidth is the range of frequencies over which two frequency components have a strong potential for amplitude correlation.

Two sinusoids with frequency separation greater than B_c, are affected quite differently by the channel. If the coherence bandwidth is defined as the bandwidth, over which the frequency correlation function is above 0.9. Then the coherence bandwidth is approximately [3]

$$B_c \cong \frac{1}{50\,\sigma_\Gamma} \tag{2.2}$$

If the definition is relaxed s that the frequency correlation functions above 0.5, then the coherence bandwidth is approximately

$$B_c \cong \frac{1}{5\sigma_\tau} \qquad (2.3)$$

It is important to note that an exact relationship between coherence bandwidth and rms delay spread is a function of specific channel impulse responses and applied signals, and equations (2.2) and (2.3) are "ball park estimates".

In general, spectral analysis techniques and simulation are required to determine the exact impact that time varying multipath has on a particular transmitted signal [4]. For this reason accurate multipath channel models must be used in the design of specific modems for wireless applications.

2.1.5 Doppler spread and coherence time

Delay spread and coherence bandwidths are parameters, which describe the time dispersive nature of the channel in a local area. However, they do not offer information about the time varying nature of the channel caused by either relative motion between the mobile and base station, or by movement of objects in the channel. Doppler spread and coherence time are parameters, which describe the time varying nature of the channel in a small-scale region.

Doppler spread B_D is a measure of the spectral broadening caused by the time rate of change of the mobile radio channel and is defined as the range of frequencies over which the received Doppler spectrum is essentially non-zero.

When a pure sinusoidal tone of frequency f_C is transmitted, the received signal spectrum called the Doppler spectrum, will have components in the range f_C - f_d to $f_C + f_d$, where f_d is the Doppler shift. The amount of spectral broadening depends on f_d which is a function of the relative velocity of the mobile, and the angle θ between the direction of motion of the mobile and direction of arrival of the scattered waves.

If the baseband signal bandwidth is much greater than B_D, the effects of Doppler spread are negligible at the receiver. This is a slow fading channel.

Coherence time T_C is the time domain dual of Doppler spread and is used to characterize the time varying nature of the frequency dispersiveness of the channel in the time domain, the Doppler spread and coherence time are inversely proportional to one another That is

$$T_C \cong \frac{1}{f_M} \tag{2.4}$$

Coherence time is actually a statistical measure of the time duration over which the channel impulse response is essentially invariant, and quantifies the similarity of the channel response at different times.

In other words coherence time is the time duration over which two received signals have a strong potential for amplitude correlation. If the reciprocal bandwidth of the base band signal is greater than the coherence time of the channel, then the channel will change during the transmission of the base band message, thus causing distortion at the receiver. If the coherence time is defined as the time over which the time correlation function is above 0.5 then the coherence time is approximately [5]

$$T_C = \frac{9}{\pi 16\, f_M} \tag{2.5}$$

Where f_M is the maximum Doppler shift given by $f_M = \frac{v}{\lambda}$, In practice equation (2.4) suggests a time duration during which a Rayleigh fading signal may fluctuate wildly, and equation (2.5) is often too restrictive.

A popular rule of thumb for modern digital communications is to define the coherence time as the geometric mean of Equations (2.4) and (2.5). That is

$$T_C = \sqrt{\frac{9}{\pi 16 f_M{}^2}} \tag{2.6}$$

The definition of coherence time implies that two signals arriving with a time separation greater than T_C are affected differently by the channel. For example for a vehicle traveling 60 mph using a 900 MHz carrier, a conservative value of T_C can be shown to be 2.22 msec from Equation (2.5). If a digital transmission system is used then as long as the symbol rate is greater than $\frac{1}{T_C}$ = 454 bps, the channel will not cause distortion due to motion (however distortion could result from multipath time delay spread depending on the channel impulse response). Using the practical formula of equation (2.6), Tc = 6.77 ms and the symbol rate must exceed 150 bits/s in order to avoid distortion due to frequency dispersion.

2.1.6 Types of small scale fading

The previous section demonstrated signal propagating through a mobile radio channel depends on the nature of the transmitted signal with respect to the characteristics of the channel. Depending on the relation between the signal parameters (such as bandwidth, symbol period, etc.) and the channel parameters (such as rms delay spread and Doppler spread), different transmitted signals will undergo different types of fading.

The time dispersion and frequency dispersion mechanisms in a mobile radio channel lead to four possible distinct effects, which are manifested depending on the nature of the transmitted signal, the channel, and the velocity. while multipath delay spread leads to time dispersion sad frequency selective fading. Doppler spread leads to frequency dispersion and time selective fading.

The two propagation mechanisms are independent of one another. Figure 2.4 shows a tree of the four different types of fading.

2.1.6.1 Fading effects due to multipath time delay spread

Time dispersion due to multipath causes the transmitted signal to undergo either flat or frequency selective fading.

2.6.1.1.1 Flat fading

If the mobile radio channel has a constant gain and linear phase response over a bandwidth, which is greater than the bandwidth of the transmitted signal, then the received signal will undergo flat fading. This type of fading is historically the most common type of fading described in the technical literature. In flat fading, the multipath structure of the channel is such that the spectral characteristics of the transmitted signal are preserved at the receiver.

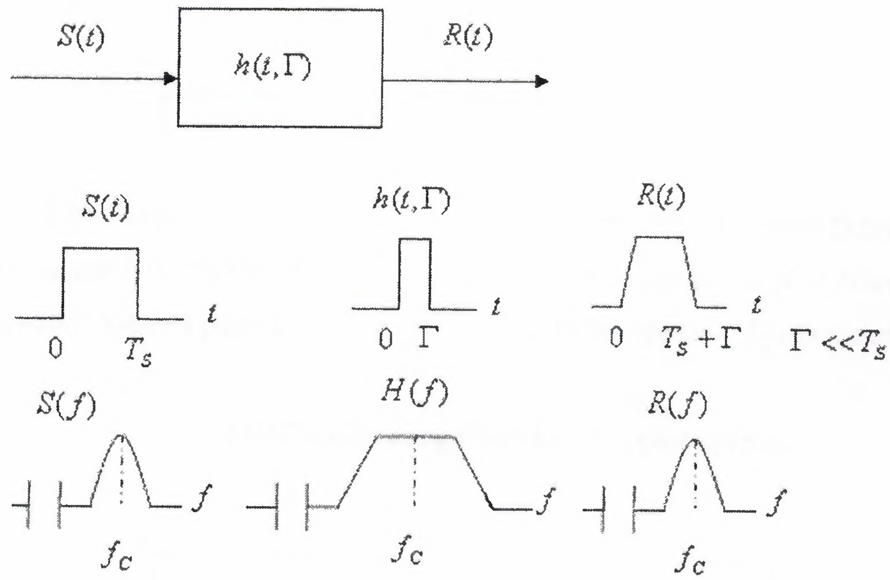


Figure 2.3 Flat fading channel characteristic [6].

However the strength of the received signal changes with time, due to fluctuations in the gain of the channel caused by multipath. The characteristics of a flat fading channel is illustrated in figure 2.3.

It can be seen from figure 2.3 that if the channel gain changes over time, a change of amplitude occurs in the received signal. Over time the received signal R(t) varies in gain, but the spectrum of the transmission is preserved.

Small scale fading (Based on multipath time delay spread)

Flat fading

1.(BW) of signal<BW of channel.

2.Delay spread<Symbol period.

Frequency selective fading

1.BW of signal > BW of channel.

2.Delay spread > Symbol period.

Small scale fading (Based on Doppler spread)

Fast fading

1.High Doppler spread .

2.Coherence time <Symbol period.

3.Channel variation faster than baseband signal variation.

Slow fading

1.BW of signal > BW of channel.

2.Coherence time>Symbol period.

3.Channel variation faster than baseband signal variation .

Figure 2.4 Type of small scale fading [6].

In a flat fading channel, the reciprocal bandwidth of the transmitted signal is much larger than the multipath time delay spread of the channel, and $h_b(t,\tau)$ can be approximated as having no excess delay (a single delta function with $\tau = 0$).

Flat fading channels are also known as amplitude varying channels, and are sometimes referred to as narrowband channels, since the bandwidth of the applied signal is narrow as compared to the channel flat fading bandwidth. Typical fiat fading channels cause deep fades, and thus may require 20 or 30 dB more transmitter power to achieve low bit error rates during times of deep fades as compared to systems operating over non-fading channels.

The distribution of the instantaneous gain of flat fading channels is important for designing radio links, and the most common amplitude distribution is the Rayleigh distribution. The Rayleigh flat fading channel model assumes that the channel induces amplitude, which varies in time according to the Rayleigh distribution

To summarize, a signal undergoes flat fading

$$B_S << B_C \tag{2.7}$$

And

$$T_S << \sigma_\Gamma \tag{2.8}$$

where T_S is the reciprocal bandwidth (symbol period) and B_S is the bandwidth, respectively of the transmitted modulation, and σ_Γ and B_S are the rms delay spread and coherence bandwidth, respectively of the channel.

2.1.6.1.2 Frequency selective fading

If the channel possesses a constant-gain and linear phase response over a bandwidth, that is smaller than the bandwidth of transmitted signal, then the channel creates frequency, selective fading on the received signal. Under such conditions the channel impulse response has a multipath delay spread, which is greater than the reciprocal bandwidth of the transmitted message waveform.

When this occurs the received signal includes multiple versions of the transmitted waveform, which are attenuated (faded) and delayed in time, and hence the received signal is distorted.

Frequency selective fading is due to time dispersion of the transmitted symbols within the channel. Thus the channel induces inter-symbol interference (ISI). Viewed in the frequency domain, certain frequency components in the received signal spectrum have greater gains than others.

Frequency selective fading channels are much more difficult to model than flat fading channels since each multipath signal must be modeled and the channel must be considered to be a linear filter. It is for this reason that wideband multipath measurements are made, and models are developed from these measurements.

When analyzing mobile communication systems, statistical impulse response models such as the two-ray Rayleigh fading model (which considers the impulse response to be made up of two delta functions which independently fade and have sufficient time delay between them to induce frequency selective fading upon the applied signal), or computer generated or measured impulse responses, are generally used for analyzing frequency selective small-scale fading. Figure 2.5 illustrates the characteristics of a frequency selective fading channel.

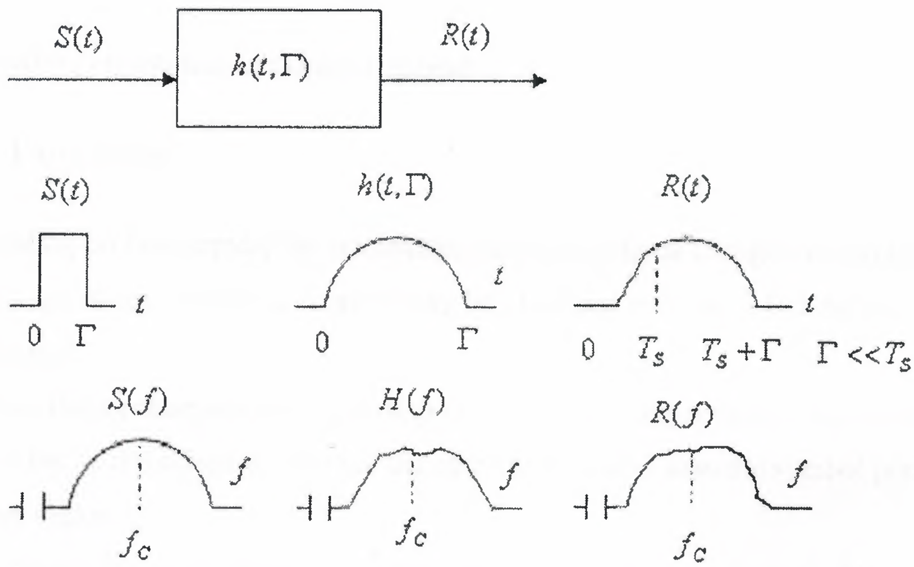


Figure 2.5 Frequency selective fading channel characteristic [6].

For frequency selective fading, the spectrum S(f) of the transmitted signal has a bandwidth which is greater than the coherence bandwidth Br of the channel. Viewed in the frequency domain, the channel becomes frequency selective, where the gain is different for different frequency components. Frequency selective fading is caused by multipath delays, which approach or exceed the symbol period of the transmitted symbol.

Frequency selective fading channels are also known as wideband channels since the bandwidth of the signal s(t) is wider than the bandwidth of the channel impulse response. As time varies, the channel varies in gain and phase across the spectrum of s(t), resulting in time varying distortion in the received signal r(t). To summarize, a signal undergoes frequency selective fading if

$$B_S > B_C \tag{2.9}$$

And
$$T_S < \sigma_\Gamma \tag{2.10}$$

A common rule of thumb is that a channel is flat fading if $T_S \geq 10\sigma_\Gamma$ and a channel is frequency selective if $T_S < 10\sigma_\Gamma$, although this is dependent on the specific type of modulation used.

2.1.6.2 Fading effects due to Doppler spread

2.1.6.2.1 Fast Fading

Depending on how rapidly the transmitted base band signal changes as compared to the rate of change of the channel, a channel may be classified either as a fast fading or as slow fading channel.

In a fast fading channel the channel impulse response changes rapidly within the symbol duration. That is, the coherence time of the channel is smaller than the symbol period of the transmitted signal.

This causes frequency dispersion (also called time selective fading) due to Doppler spreading, which leads to signal distortion. Viewed in the frequency domain, signal distortion due to fast fading increases with increasing doppler spread relative to the bandwidth of the transmitted signal. Therefore a signal undergoes fast fading if

$$T_S << T_C \qquad (2.11)$$

And

$$B_S >> B_D \qquad (2.12)$$

It should be noted that when a channel is specified as a fast or slow fading channel, it does not specify whether the channel is flat fading or frequency selective in nature.

Fast fading only deals with the rate of change of the channel due to motion. In the case of the flat fading channel, we can approximate the impulse response to be simply a delta function (no time delay). Hence, a flat fading, fast fading channel is a channel in which the amplitude of the delta function varies faster than the rate of change of the transmitted base band signal.

In the case of a frequency selective, fast fading channel the amplitudes, phases, and time delays of any one of the multipath components vary faster than the rate of change of the transmitted signal. In practice, fast fading only occurs for very low data rates.

2.1.6.2.2 Slow Fading

In a slow fading channel the channel impulse response changes at a rate much slower than the transmitted base band signal s(t). In this case, the channel may be assumed to be static over one or several reciprocal bandwidth intervals.
In the frequency domain, this implies that the Doppler spread of the channel is much less, than the bandwidths of the baseband signal. Therefore, a signal undergoes slow fading if

$$T_S << T_C \tag{2.13}$$

And

$$B_S >> B_D \tag{2.14}$$

It should be clear that the velocity of the mobile (or velocity of objects in the channel) and the base band signaling determines whether a signal undergoes fast fading or slow fading.

The relation between the various multipath parameters and the type of fading experienced by the signal are summarized in Figure 2.6. It should be emphasized that fast and slow fading deal with the relationship between the time rate of change in the channel and the transmitted signal and not with propagation path loss models.

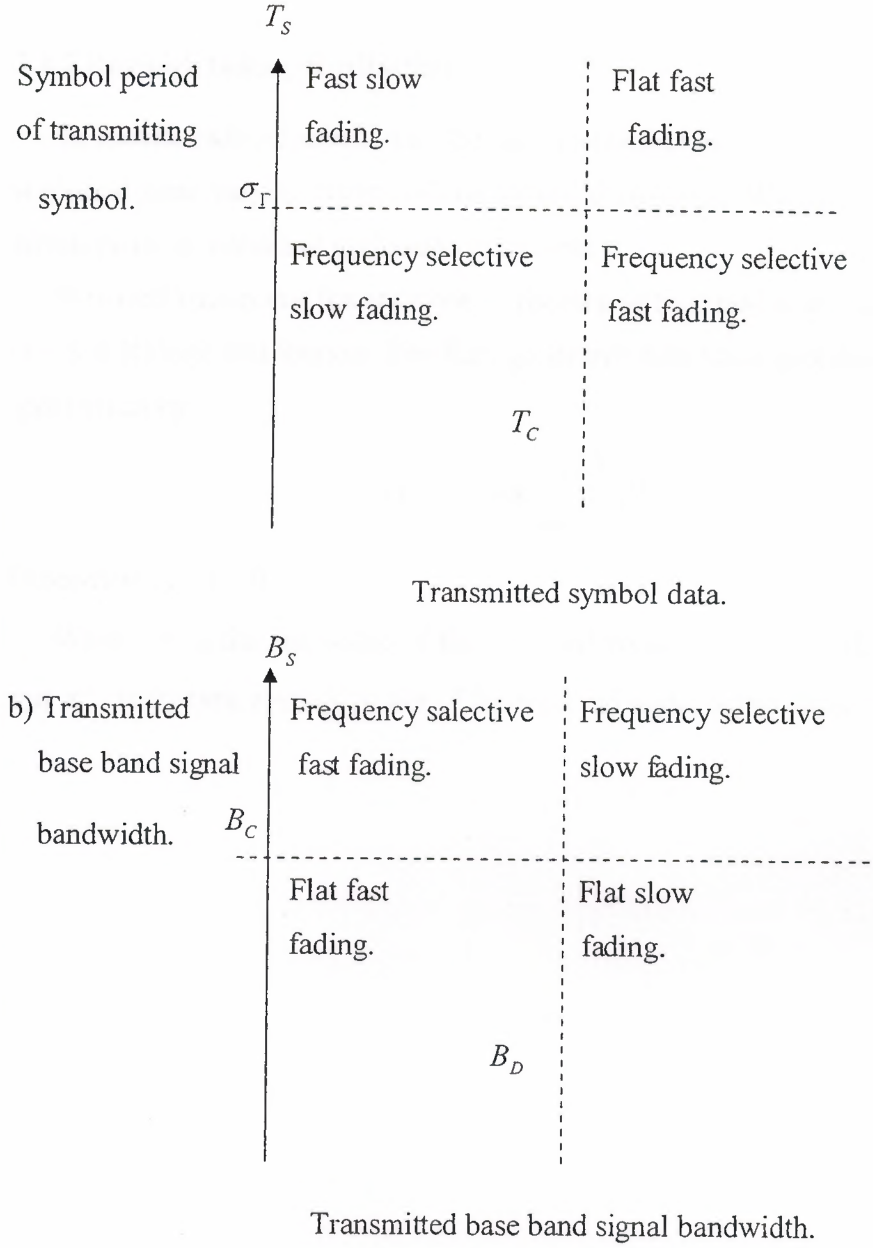


Figure 2.6 Fading as function of a) Symbol period b) Base band signal bandwidth [6].

2.1.7 Rayleigh fading distribution

In mobile radio channels, the Raleigh distribution is commonly used to describe the statistical time varying nature of the received envelope of a flat fading signal, or the envelope of an individual multipath component.

It is well known that the envelope of the sum of two quadrature Gaussian noise signals obeys a Raleigh distribution. The Raleigh distribution has a probability density function (pdf) given by

$$p(r) = \frac{r}{\sigma^2} \exp\left(\frac{-r^2}{2\sigma^2}\right), 0 \le r \le \infty \tag{2.15}$$

Otherwise $p(r) = 0$

Where σ is the rms value of the received voltage signal before envelope detection, and σ^2 is the time average power of the received signal before detection.

2.2 Inter Symbol Interference (ISI)

Reflections cause problems in communication systems since the signals from different echoes arrive at different time instants to the receiver. This means that the received signal is a sum of the signal sent at different time instants, and it is called intersymbol interference (ISI).

When the symbol time is short compared to the excess delay then the ISI spans over several symbols and this effect has to be suppressed by the receiver. Normally this can be accomplished by an equalizer, but when the ISI spans over many symbols the equalization procedure gets quite complicated.

Today there is a growing interest for high speed mobile communication, and when the symbol rate is high, the symbol time is short and there is a risk that the ISI becomes severe. One way to get around this problem is to divide the data stream into several sub-streams and transmit each of them on its own frequency. Then the symbol time on each carrier frequency gets long even though the overall bit rate is high, and the problem with ISI originating from several symbols can be solved.

However new problems arise as the duration of each symbol becomes long compared to the coherence time, The channel variations get fast compared to the symbol duration, which cause problems for the receiver and degrade the performance.

Problems also arise when a transmitter has to transmit several signals at the same time; one of the reasons that single carrier systems are more popular than multi-carrier systems is that the envelope of the sent signal may be constant. This is not the case in multi-carrier modulation and therefore it is hard to design a power amplifier with high efficiency.

In order to get a spectrum efficient multicarrier system the subcarriers have to be placed with minimum frequency distance between them but with enough distance such that they do not interfere with each other. This problem was solved with overlapping spectra between the carriers, but with maintained orthogonality between them and the first "modern" OFDM system was invented.

Figure 2.7 illustrates ISI, and it occurs due to delay spread in the channel caused by multi-path propagation echoes of the same signal arrive with different delays and strength, which results in a spreading of the symbols transmitted. As the symbols spreads, the neighboring symbols will be affected and become correlated with each other.

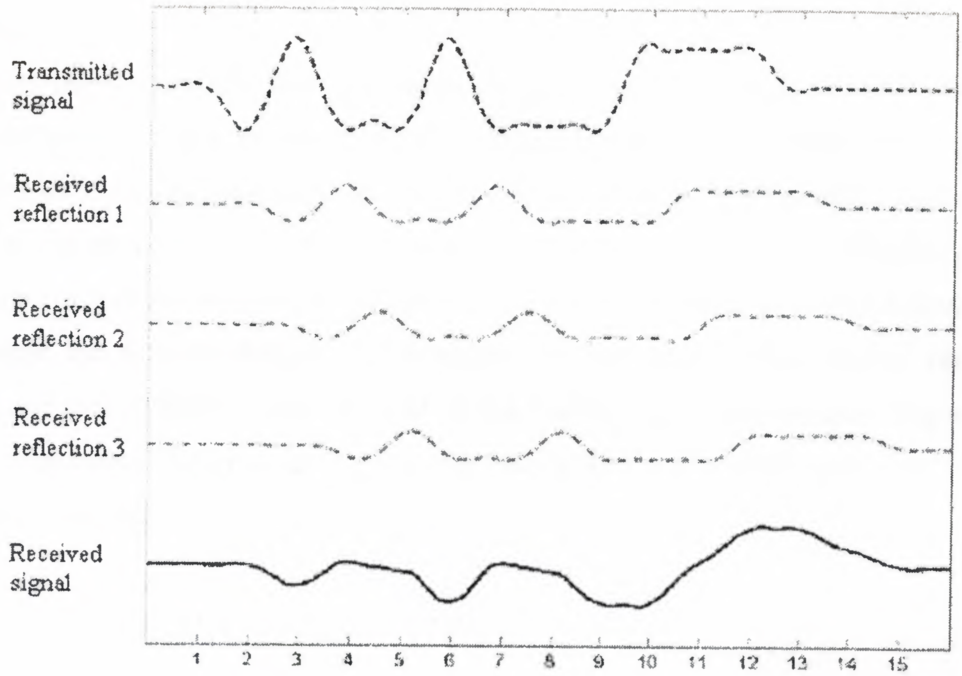


Figure 2.7 An example of the consequences of ISI.

Delay spread is a measurement on how much the signal is spread over time. The delay spread depends on the terrain and the distance between the nodes. Indoor channels the delay spread is typically tens or hundreds of nanoseconds but in outdoor mountainous terrain delay spreads of several microseconds are not uncommon.

The solution to this problem is either to implement an equalizer, a rake receiver or simply to lower the symbol rate.

2.2.1 Equalizer

Knowing the behavior of the channel gives the means of dealing with its problems. If for instance it is known how a symbol is distorted when transmitted over the channel the demodulator could make a new correlation match, using this information, This would completely take care of the ISI.

Highlights the major filtering aspects of a typical baseband digital system, there are circuit reactances throughout the system in the transmitter, in the receiver and in the channel. The pulses at the input might be impulse-like samples, or flat-top samples in either case they are low-pass filtered at the transmitter to confine them to some desired bandwidth.

Channel reactances can cause amplitude and phase variations that distort the pulses. The receiving filter called the equalizing filter should be configured to compensate for the distortion caused by the transmitter and the channel, in a binary system with a commonly used PCM (pulse code modulation) format. Such as non return to zero(NRZ-L). The detector makes symbol decisions by comparing the received bipolar pulses to a threshold. For example the detector decides that a binary one was sent of the received pulse is positive, and that a binary zero was sent if the received pulse is negative. Figure 2.8 illustrates the overall lumping all the filtering effects into one overall equivalent system transfer function, H(f):

$$H_R(f) = H(f)H_T(f)H_C(f)H_R(f) \tag{2.16}$$

$$h_E(t) = h(t) * h_T(t) * h_C(t) * h_R(t) \tag{2.17}$$

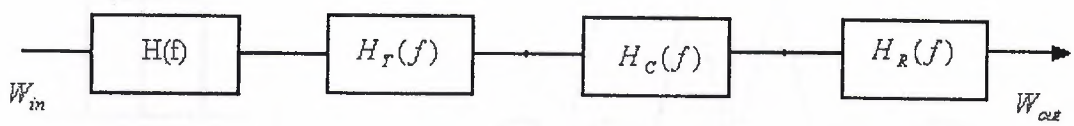


Figure 2.8 Overall equivalent system transfer function.[7]

where $H_T(f)$ characterizes the transmitting filter, $H_C(f)$ the filtering within the channel, and $H_R(f)$ the receiving or equalizing filter, The characteristic H(f) then represents the composite system transfer function due to all of the filtering at various locations throughout the transmitter-channel-receiver chain, due to the effects of system filtering, the received pulses overlap one another the tail of one pulse (smears) into adjacent symbol intervals so as to interfere with the detection process; such interference is termed intersymbol interference (ISI), Even in the absence of noise imperfect filtering and system bandwidth constraints lead to ISI.

In practice, $H_C(f)$ is usually specified, and the problem remains to determine $h_R(f)$ and $H_T(f)$ such that the ISI of the pulses are minimized at the output of $h_R(f)$.

Nyquist investigated the problem of specifying a received pulse shape so that no ISI occurs at the detector. He showed that the theoretical minimum system bandwidth needed to detect Rs symbols/s without ISI, is Rs/2 hertz. This occurs when the system transfer function, H(f) is made rectangular, as shown in figure 2.8.a. When H(f) is such an ideal filter with bandwidth 1/2T, its impulse response, the inverse Fourier transform of H(f) is h(t) = sinc(t/T), shown in Figure 2.8.b Thus h(t) is the received pulse shape resulting from the application of an impulse at the input of such an ideal system.

Nyquist established that if each pulse of a received sequence is of the form h(t), the pulses can be detected without ISI. The bandwidth required to detect 1/T such poises (symbols) per second is equal to 1/2T; in other words, a system with bandwidth

W= 1/2T = Rs hertz can support a maximum transmission rate of 2W=1/T=Rs symbols/s (Nyquist bandwidth contraint) without ISI .

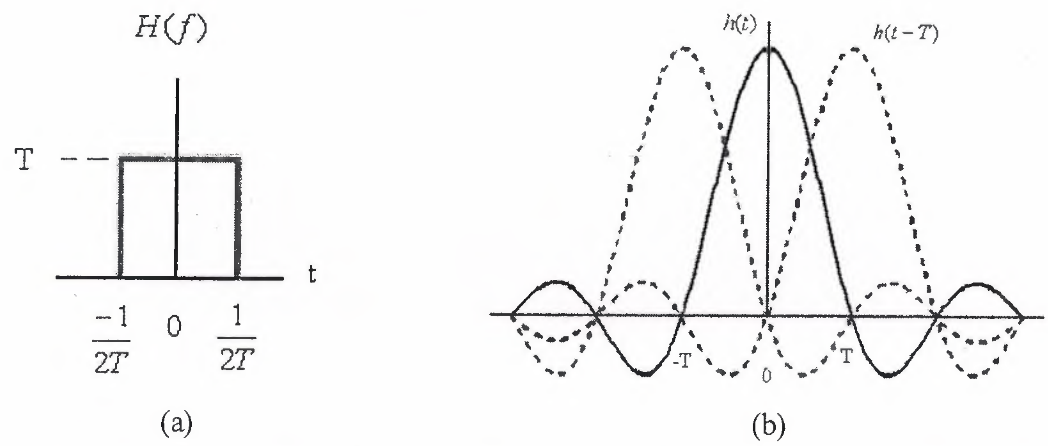


Figure 2.8 Nyquist channels for zero ISI (a) Rectangular system transfer function (b)Received pulse shape h(t)=sinc(t/T) [8]

Figure 2.8.b illustrates how ISI is avoided. Figure 2.8 shows two successive received pulses: $h(t)$ and $h(t-T)$. Even though $h(t)$ has a long tail, it passes through zero at the instant that $h(t-T)$ is sampled (at t = T) and therefore causes no degradation to the detection process. With such an ideal received pulse shape, the maximum possible symbol transmission rate per herts, called the symbol rate packing, is 2 symbol/s/Hz, without ISI.

For most communication systems (with the exception of spread-spectrum systems, our goal is to reduce the required system bandwidth as much as possible; Nyquist has provided us with a basic limitation to such bandwidth reduction.

2.2.2 Symbol rate

In any band limited signal and channel, a certain amount of ISI is inevitable, The correlation between the symbols will however reduce the SNR and make the synchronization for sampling more critical, and when the ISI gets too bad the symbols will finally be completely distorted, At this point the increase of signal power will be futile.

How much ISI the system can handle depends mainly on the symbol rate, A symbol with the duration of several microseconds can easily handle a mean delay spread of a few nano-seconds, when the mean delay spread grows to approximately one tenth of the symbol duration the performance will drop noticeably.

CHAPTER THREE

ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING (OFDM)

3.1 Introduction

Orthogonal Frequency Division Multiplexing (OFDM), is a multicarrier transmission technique used in applications catering to both Wired and Wireless Communications.

However, in the wired case the usage of the term discrete multi-tone is more appropriate. The OFDM technique divides the frequency spectrum available into many closely spaced carriers, which are individually modulated by low-rate data streams. In this sense OFDM is similar to FDMA (The bandwidth is divided into many channels, so that in a multi-user environment each channel is allocated to a user). However, the difference lies in the fact that the carriers chosen in OFDM are much more closely spaced than in FDMA (1kHz in OFDM as opposed to about 30kHz in FDMA), thereby increasing its spectral usage efficiency.

The orthogonality between the carriers is what facilitates the close spacing of carriers. The orthogonality principle essentially implies that each carrier has a null at the center frequency of each of the other carriers in the system while also maintaining an integer number of cycles over a symbol period.

The motivation for using OFDM techniques over TDMA techniques is twofold. First, TDMA limits the total number of users that can be sent efficiently over a channel. In addition, since the symbol rate of each channel is high, problems with multipath delay spread invariably occur.

In stark contrast, each carrier in an OFDM signal has a very narrow bandwidth (i.e. 1 kHz); thus the resulting symbol rate is low. This results in the signal having a high degree of tolerance to multipath delay spread, as the delay spread must be very low to cause significant inter-symbol interference (e.g. > 500 μ sec).

3.2 Multicarrier modulation

The basic idea of multicarrier modulation is to divide the transmitted bitstream into many different substreams and send these over many different subchannels. Typically the subchannels are orthogonal under ideal propagation conditions. The data rate on each of the subchannels is much less than the total data rate, and the corresponding subchannel bandwidth is much less than the total system bandwidth. The number of substreams is chosen to insure that each subchannel has a bandwidth less than the coherence bandwidth of the channel, so the subchannels experience relatively flat fading.

Thus, the ISI on each subchannel is small. The subchannels in multicarrier modulation need not be contiguous, so a large continuous block of spectrum is not needed for high rate multicarrier communications.

Moreover, multicarrier modulation is efficiently implemented digitally. In this discrete implementation, called orthogonal frequency division multiplexing (OFDM), the ISI can be completely eliminated through the use of acyclic prefix.

Multicarrier modulation is currently used in many wireless systems. However, it is not a new technique, it was first used for military HF (high frequency) radios in the late 1950's and early 1960's. Starting around 1990, multicarrier modulation has been used in many diverse wired and wireless applications, including digital audio and video broadcasting in Europe, digital subscriber lines (DSL) using discrete multitone, and the most recent generation of wireless LANs.

There are also a number of newly emerging uses for multicarrier techniques, including fixed wireless broadband services, mobile wireless broadband known as FLASH-OFDM, and even for ultra-wideband standard. Multicarrier modulation is also a candidate for the air interface in next generation cellular systems.

The multicarrier technique can be implemented in multiple ways, including vector coding and OFDM [7], all of which are discussed in this chapter. These techniques have subtle differences, but are all based on the same premise of breaking a wideband channel into multiple parallel narrowband channels by means of an orthogonal channel partition.

There is some debate as to whether multicarrier or single carrier modulation is better for ISI channels with delay spreads on the order of the symbol time. It is claimed in that for some mobile radio applications, single carrier with equalization has roughly the same performance as multicarrier modulation with channel coding, frequency domain interleaving, and weighted maximum-likelihood decoding. Adaptive loading was not taken into account in, which has the potential to significantly improve multicarrier performance [8]. But there are other problems with multicarrier modulation that impair its performance, most significantly frequency offset and timing jitter, which degrade the orthogonality of the subchannels.

In addition, the peak-to-average power ratio of multicarrier is significantly higher than that of single carrier systems, which is a serious problem when nonlinear amplifies are used. Trede offs between multicarrier and single carrier systems, which is a serious problem when nonlinear amplifies are used. Trades offs between multicarrier and single carrier block transmission systems.

Despite these challenges, multicarrier techniques are common in high data rate wireless systems with moderate to large delay spread, as they have significant advantages over time-domain equalization. In particular, the number of taps required for an equalizer with good performance in a high data rate system is typically large. Thus, these equalizers are highly complex. Moreover, it is difficult to maintain accurate weights for a large number of equalizer taps in a rapidly varying channel. For these reasons, most emerging high rate wireless systems use either multicarrier modulation or spread spectrum instead of equalization to compensate for ISI.

3.3 Data Transmission using Multiple Carriers

The simplest form of multicarrier modulation divides the data stream into multiple substreams to be transmitted over different orthogonal subchannels centered at different subcarrier frequencies. The Number of substreams is chosen to make the symbol time on each substream much greater than the delay spread of the channel or, equivalently, to make the substream bandwidth less than the channel coherence bandwidth. This insures that the substreams will not experience significant ISI.

Consider a linearly-modulated system with data rate R and pass band bandwidth B. The coherence bandwidth for the channel is assumed to be $B_c<B$, so the signal experiences frequency selective fading. The basic premise of multicarrier modulation is to break this wideband system into N linearly modulated subsystems in parallel, each with subchannel

bandwidth B_N = B/N and data rate R_N=R/N. For N sufficiently large, the subchannel bandwidth B_N = B/N<<Bc, which insures relatively flat fading on each subchannel. This can also be seen in the time domain, the symbol time T_N of the modulated signal in each subchannel is proportional to the subchannel bandwidth 1/B. So B_N. So B_N<<Bc implies that $T_N \approx B_N$>>1/Bc$\approx$Tm, where Tm denotes the delay spread of the channel. Thus, if N is sufficiently large, the symbol time is much bigger than the delay spread, so each subchannel experiences little ISI degradation.

Figure 3.1 illustrates a multicarrier transmitter. The bit stream is divided into N substreams via a serial to parallel converter. The nth substream is linearly modulated (typically via QAM or PSK) relative to the subcarrier frequency fn and occupies pass band bandwidth Bn. We assume coherent demodulation of the subcarriers so the subcarrier phase is neglected in our analysis. If we assume raised cosine pulses for g(t) we get a symbol time T^N = (1 + β)/B^N for each substream, where β is the roll off factor of the pulse shape. The modulated signals associated with all the subchannels are summed together to form the transmitted signal, given as [10]

$$s(t) = \sum_{i=o}^{N-1} s_i g(t)\cos(2\pi f_i t + \phi_i) \tag{3.1}$$

where s_i is the complex symbol associated with the i-th subcarrier and ϕ_i is the phase offset of the i-th carrier. For non-overlapping subchannels we set f_i= f_o + i(B^N), i=0,...,N-1. The substreams then occupy orthogonal subchannels with pass band bandwidth B^N yielding a total passband bandwidth NB^N = B and data rate NR^N=R. Thus, this form of multicarrier modulation does not change the data rate or signal bandwidth relative to the original system, but it almost completely eliminates ISI for B_N << B_C.

The receiver for this multicarrier modulation is shown in Figure 3.2. Each substream is passed through a narrowband filter to remove the other substreams, demodulated, and combined via a parallel to serial converter to form the original data stream. Note that the ith subchannel will be affected by flat fading corresponding to a channel gain $\alpha_i = |H(f_i)|$.

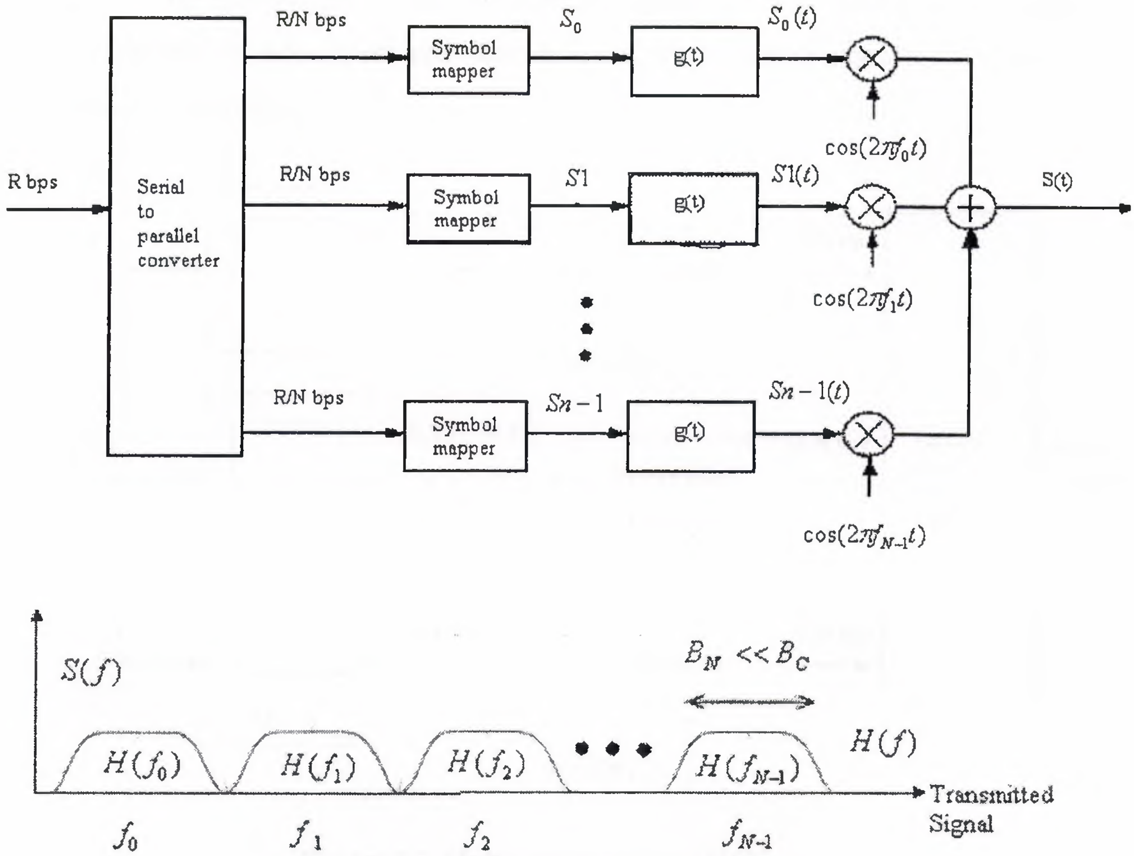


Figure 3.1 Multicarrier transmitter [10].

Although this simple type of multicarrier modulation is easy to understand, it has several significant short comings. First, in a realistic implementation, subchannels will occupy a larger bandwidth than under ideal raised cosine pulse shaping since the pulse shape must be time-limited. Let ε / T_N denote the additional bandwidth required due to time limiting of these pulse shapes. The subchannels must then be separated by $(1+\beta+\varepsilon)/T_N$, and since the multicarrier system has N subchannels, the bandwidth penalty for time limiting is ε / T_N. In particular, the total required bandwidth for non-overlapping subchannels is

$$B = \frac{N(1+\beta+\alpha)}{T_N} \tag{3.2}$$

Thus, this form of multicarrier modulation can be spectrally inefficient Additionally, near-ideal (and hence, expensive) low pass filters will be required to maintain the orthogonality of the subcarriers at the receiver. Perhaps most importantly, this scheme requires N independent modulators and demodulators, which entails significant expense, size, and power consumption.

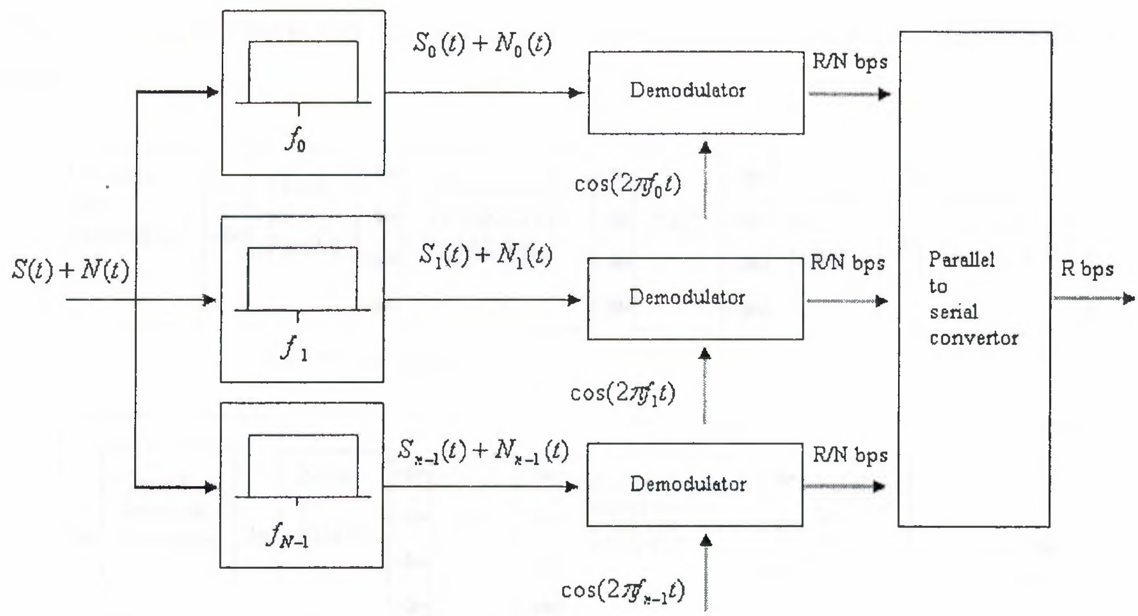


Figure 3.2 Multicarrier receiver [10].

3.4 OFDM in Data Transmission

In high rate single carrier communication systems for, the effects of multipath propagation increases the equalization costs due to short symbol durations and relative long channel delay times. For example, if we want to use a single carrier technique for bit rates of 20 mbps or more, the symbol duration is about 50 ns. Measured indoor channels cause echo delay times of more than 500 ns. In case of QPSK, the viterbi equalizer needs more than 1.000.000 internal states. So a more efficient equalization technique is needed.

Figure 3.3 Block description of multiple carriers we can see in the picture above, Multicarrier technique (MC) is parallel data transmission. On each carrier, we need all elements of a single carrier system. The main advantage of MC technique is a longer symbol duration by factor N (N: number of sub-carrier), OFDM is a special case of the multi-carrier technique, Here rectangular, frequency shifted pulse shaping filters in transmitter and receiver are used, Efficient IFFT/FFT structures are used to approximate the rectangular impulse shaping.

To overcome the effect of multipath propagation, a short guard interval is introduced. With the guard interval, the OFDM system only needs one multiplication on each subcarrier as equalization. On the picture below, we can see a complete discrete OFDM system.

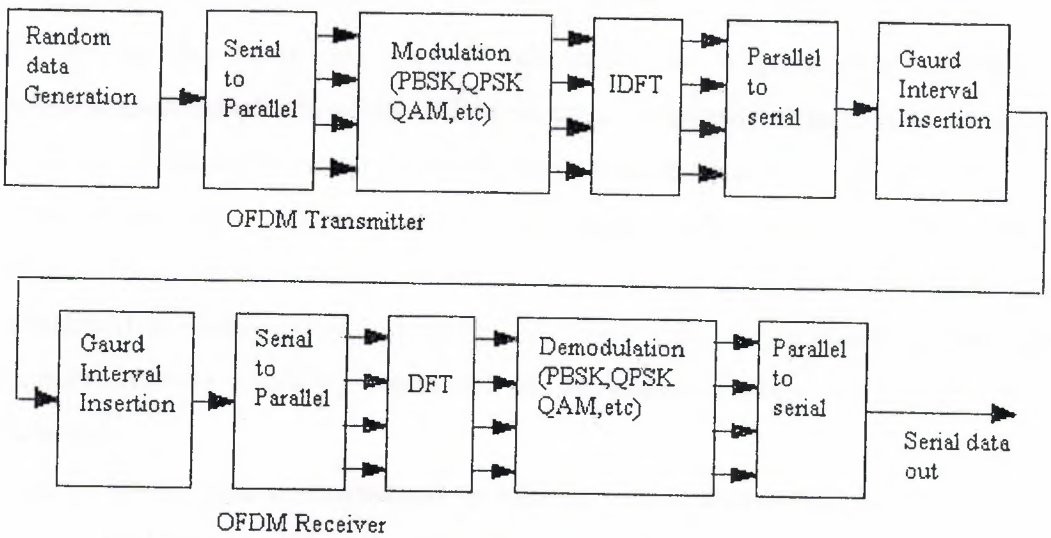


Figure 3.3 Block diagram of a multi-carrier OFDM digital communication system [9].

3.5 Characteristics of 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 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.

In addition, OFDM is a special form of multicarrier transpires where a single carrier high speed data stream is transmitted over a number of parallel data stream is transmitted over a number of lower rate subcarrier. While the concept of parallel data transmission and OFDM can be traced back to the late 1950s [11], its initial use was in several high frequency military systems in the 1960s such as KINEPLEX and KATHRYN.

The discrete fourier transform implementation of pioneers in the early 1970s [17,18,19]. Today, OFDM is a strong candidate for commercial high speed broadband wireless communications, due to recent advances in very large scale integration (VLSI) technology that make high speed, large size fast fourier transform (FFT) chips commercially viable.

In addition, OFDM technology possess a number of unique features that makes it an attractive choice for high speed broadband wireless communication

- OFDM is robust against multipath fading and intersymbol interference because duration increases for the lower rate parallel subcarrier. (for a given delay spread, the implementation complexity of an OFDM receiver complexity of an OFDM receiver is considerably simpler than that of a single carrier with an equalizer).
- OFDM allows for an efficient use of the available radio frequency (RF) spectrum through the use of adaptive modulation and power allocation a cross the subcarriers that are matched to slowly varying channel conditions using programmable digital signal processors, thereby enabiling bandwidth on demand technology and higher spectral efficiency.
- OFDM is robust against narrowband interference since narrowband interference only affects a small fraction of the subcarrier.
- OFDM makes single frequency networks possible, which is particularly attractive for broadcasting applications.

3.5.1 Orthogonality

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

Each carrier in an OFDM signal has a very narrow bandwidth thus the resulting symbol rate is low. This results in the signal having a high tolerance to multipath delay spread, as the delay spread must be very long to cause significant inter-symbol interference.

3.5.2 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 (binary phase shift key), QPSK (quadrature phase shift key), or QAM (quadrature amplitude modulation)).

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 two, 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 bin, then performing the IFFT. Since each bin of an IFFT corresponds to the amplitude and phase of a set of orthogonal sinusoids, the reverse process guarantees that the carriers generated are orthogonal.

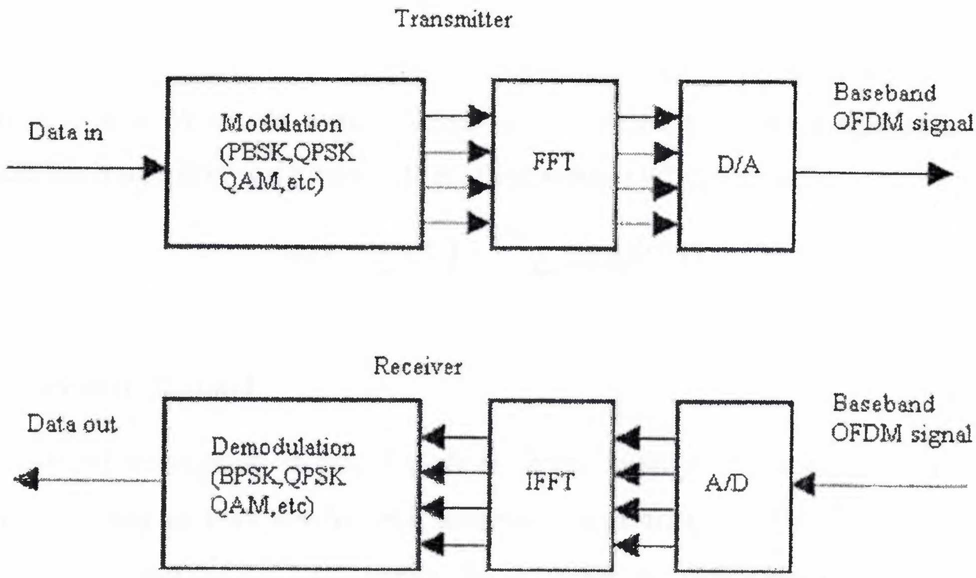


Figure 3.4 Basic FFT, OFDM transmitter and receiver [9].

Figure 3.4 shows the setup for a basic OFDM transmitter and receiver, The signal generated is a baseband, thus the signal is filtered, then stepped up in frequency before transmitting the signal.

3.5.3 Transmitter

We assume an OFDM system with N carriers, a bandwidth of W Hz and a symbol length of T seconds, of which T_{cp} seconds is the length of the cyclic prefix. The transmitter uses the following waveforms:

$$\phi_k(t) = \frac{1}{\sqrt{T - T_{cp}}} e^{j2\pi \frac{W}{N} k(t - T_{cp})} \qquad \forall t \in [0, T] \tag{3.3}$$

= 0 , otherwise where $T = (N/W) + T_{cp}$.

Note that $\phi_k(t) = \phi_k(t + N/W)$ when t is within the cyclic prefix. Since $\phi_k(t)$ is a rectangular pulse modulated on the carrier frequency kW/N, the common interpretation of OFDM is that it uses N carriers, each carrying a low bit-rate. The waveforms $\phi_k(t)$ are used in the modulation and the transmitted base band signal for OFDM symbol number l is

$$s_l(t) = \sum_{k=0}^{k=N-1} x_{k,l}\phi_k(t - lT) \tag{3.4}$$

where $x_{0,l}$, $x_{1,l}$......,$x_{N-1,l}$ are complex numbers obtained from a set of signal constellation points. When an infinite sequence of OFDM symbols is transmitted, the output from the transmitter is a juxtaposition of individual OFDM symbols:

$$s(t) = \sum_{l=-\infty}^{\infty} s_l(t) = \sum_{l=-\infty}^{\infty}\sum_{k=0}^{N-1} x_{k,l}\phi_k(t - lT) \tag{3.5}$$

3.5.4 The Physical Channel

An important assumption is that the effect of the impulse response of the physical channel (which may or may not be time invariant), is restricted to the time period $\tau \in [0,T_{cp}]$, i.e. to the length of the cyclic prefix. The received signal then become

$$r(t) = (g * s)(t) = \int_0^{T_{CP}} g(\tau;t)s(t-\tau)d\tau + n(t) \tag{3.6}$$

where n(t) is additive, white and complex Gaussian noise.

3.5.5 Receiver

A filter bank, matched to the last part $[T_{cp},T]$ of the transmitter waveforms $\Phi_k(t)$, i.e

$$\psi_k(t) = \phi_k^*(T - t)\forall t \in [0, T - T_{cp}] \tag{3.7}$$

$= 0$, otherwise

This operation effectively removes the cyclic prefix in the receiver stage of the system. All the ISI is contained in the cyclic Prefix and does not manifest itself in the sampled output obtained at the receiver filter bank. We can now remove the time index, l, when calculating the sampled output at the kth matched filter

$$y_k = (r * \psi_k)(t)\,|_{t=T} = \int_{-\infty}^{\infty} r(t)\psi_k(T - t)dt \tag{3.8}$$

Considering the channel to be fixed over the OFDM symbol interval and denoting it by g(τ), Eqn.3.8 after simplification gives the following result:

$$y_k = G(\frac{kW}{N})x_k + n'_k \tag{3.9}$$

where G(f) is the Fourier transform of g(τ) and n'_k is additive white Gaussian noise.

3.5.6 Discrete-Time Model

The modulation and demodulation (with $\Phi_k(t)$ and $\psi_k(t)$) in the continuous-time model are replaced by the inverse discrete fourier transform(IDFT) and the discrete fourier (DFT) transform, respectively, while the channel is a Discrete-Time convolution. The cyclic prefix (cp) operates in the same way in this system and calculations are essentially performed in the same fashion. As in all other cases, the integrals are changed to summations when in the Discrete-Time domain.

Using cyclic prefix longer in duration than the channel. Transforms the linear convolution into a cyclic convolution. Denoting the cyclic convolution by *, we can depict the whole OFDM system by the following equation

$$y_l = DFT(IDFT(x_l) * g_l + n_l) = DFT(IDFT(x_l) * g_l) + n'_l \tag{3.10}$$

where y_l contains the N received data points, x_l the N transmitted constellation points, g_l the channel impulse response (padded with zeroes to obtain a length N) and n_l, the channel noise. Since the channel noise is assumed to be white and Gaussian, the term, $n'_l = DFT(n_l)$ represents uncorrelated Gaussian noise. Using the result that the DFT of two cyclically convolved signals is equivalent to the product of their individual DFT's, we obtain

$$y_l = x_l . DFT(g_l) + n'_l = x_l . h_l + n'_l \tag{3.11}$$

Where the symbol "." denotes element-by-element multiplication.

3.6 Cyclic prefix

Even though the symbol time is much longer than the maximum excess delay, there is still some intersymbol interference (ISI).

The receiver can handle echoes within one OFDM symbol since this only result in a phase and amplitude change, but it cannot handle echoes between the symbols. The following describes how the idea of the cyclic prefix can be used to handle multipath delay such that the demodulated signal is free of both intersymbol interference and interchannel interference, ICI. The solution to the problem is to make the echoes affect only one symbol at a time by extending the symbol time.

The last samples from the IFFT are then copied and transmitted before the first IFFT samples. The receiver discards the (just added) first samples of the received symbol, and the original first samples are now affected only by echoes from its own OFDM symbol,. Figure 3.5 illustrates the linear convolution performed by the channel is transferred into a cyclic convolution, which after the FFT acts like a scalar multiplication by the channel transfer function.

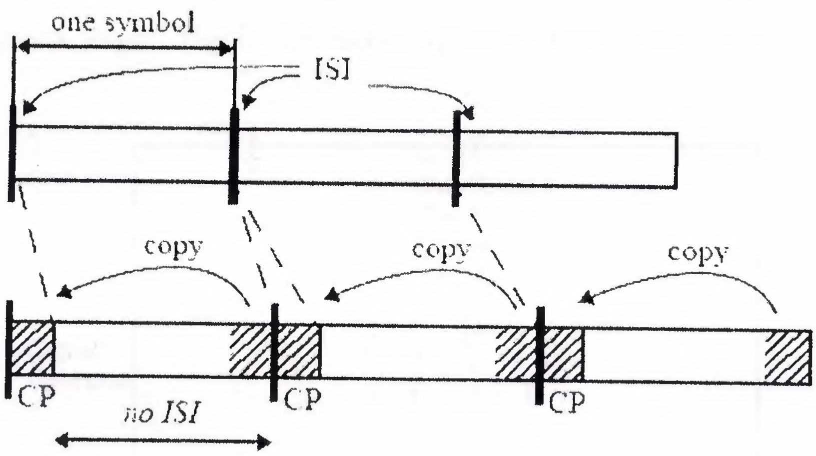


Figure 3.5 Insertion of the cyclic prefix CP, means that the ISI can be removed by the receiver without loss of orthogonality between the symbols [5].

No ISI arises as long as the length of the cyclic prefix exceeds the maximum excess delay, i.e. the difference in delay between the last and first echo. However, the use of the cyclic prefix has some drawbacks. Naturally the total symbol time including the cyclic prefix has to be longer than the maximum excess delay, which in turn limits the signalling rate on each sub-channel. There is also a waste of power since the transmitter discards the energy spent on the cyclic prefix. In the early OFDM systems a guard space was used between the symbols instead of the cyclic prefix. This may eliminate ISI but ICI may still exist due to orthogonal subchannels.

The guard space or transmission of the cyclic prefix takes some extra time and the bit rate is degraded. This can be solved by faster signalling, but then more bandwidth is occupied instead. So one has to choose the "golden mean" between having enough cyclic prefix in order to avoid or suppress ISI, but short enough not to waste too much energy or bandwidth.

3.7 Subchannel bandwidth

In OFDM systems, the total bandwidth can be seen as divided into sub-channels and therefore we can associate a specific sub-channel bandwidth, or sub-channel spacing, with them, In the following we use the term sub-channel bandwidth to denote the distance in frequency between the different sub-channels even though the actual bandwidth of each sub-channel is much larger. Figure 3.6 shows symbolic plot offour sub-carriers in an OFDM system.

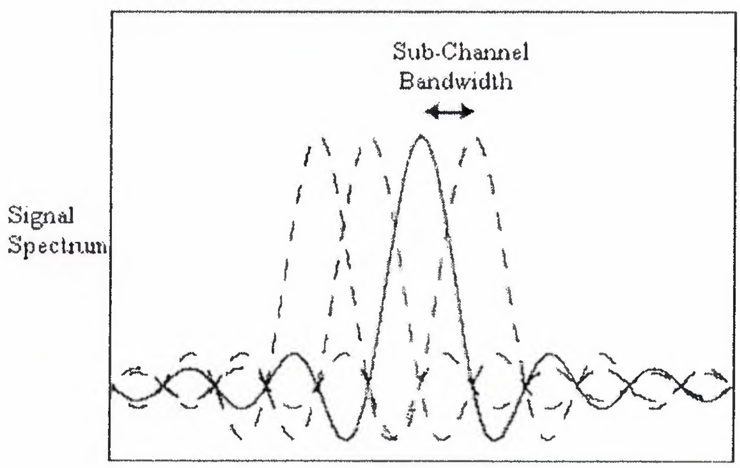


Figure 3.6 Symbolic representation of four sub-channels in an OFDM system [5].

The bandwidth is dependent on the data rate on each of the sub-channels. Many subchannels means that the data rate on them is low and then the corresponding bandwidth becomes low, and vice versa. The advantage of having many sub-channels is that the duration of the cyclic prefix becomes short compared to the symbol time, which reduces the waste of bandwidth and energy. However, the orthogonality between the channels relies on the assumption that the channel characteristic remains constant during a symbol interval.

The sub-channels start to disturb each other when changes of the echo-pattern during a symbol interval becomes perceptible and interchannel interference arises. The symbol time gets long when many sub-carriers are used and the channel changes caused by e.g. movements of the mobile terminal, become evident. Therefore the choice of sub-channel bandwidth is not obvious.

On the one hand, we want to have several sub-channels and long symbols in order to keep the duration of the cyclic prefix small. On the other hand we want to have few sub-channels and short symbols in order to avoid channel changes during a symbol and maintain orthogonality between the sub-channels.

CHAPTER FOUR

RESULTS

4.1 Additive white gaussian noise (AWGN) channel results

4.1.1 Simulation of single carrier communication

Figure 4.1 shows the BER (bit error rate) versus SNR (signal to noise ratio). Performance of communication system using a single carrier on AWGN channel. Binary phase shift key (BPSK) modulation is used.

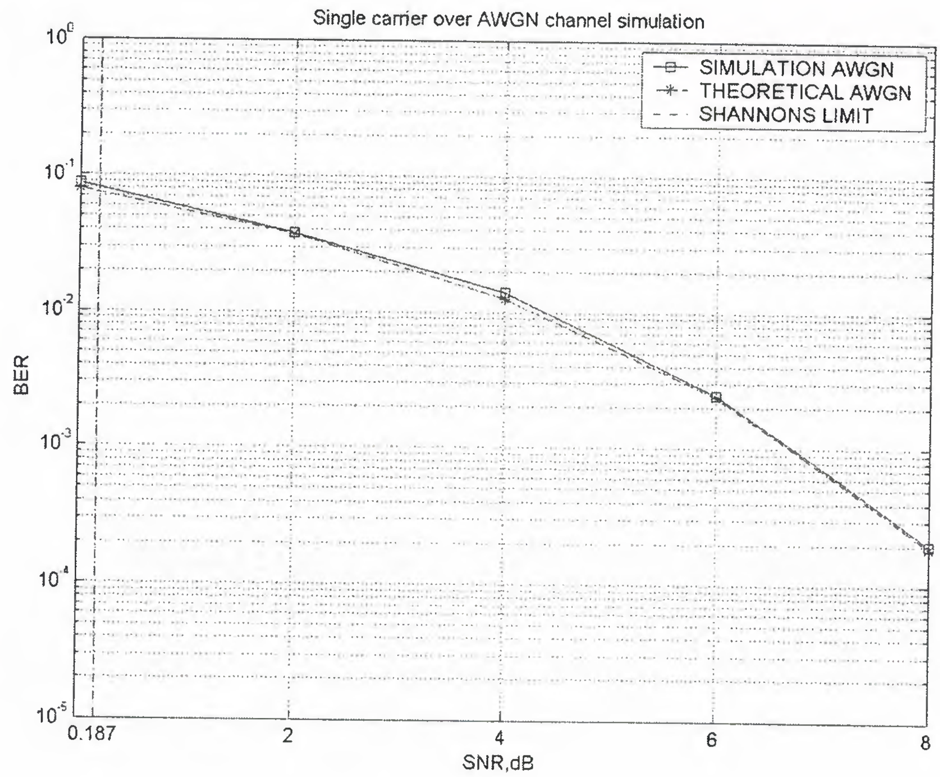


Figure 4.1 Performance of a single carrier communication on AWGN channel.

SNR is inversely proportional to the noise variance, equation (4.1) therefore as SNR increases, the noise corrupts less number of data and BER decreases [1].

$$variance = \frac{1}{2(SNR)} \tag{4.1}$$

4.1.2 Simulation of OFDM

Figure 4.2 shows the performance of OFDM over AWGN channel. The same result as in single carrier communication is obtained. Therefore, it can be argued that OFDM has no effect on the performance over AWGN channel.

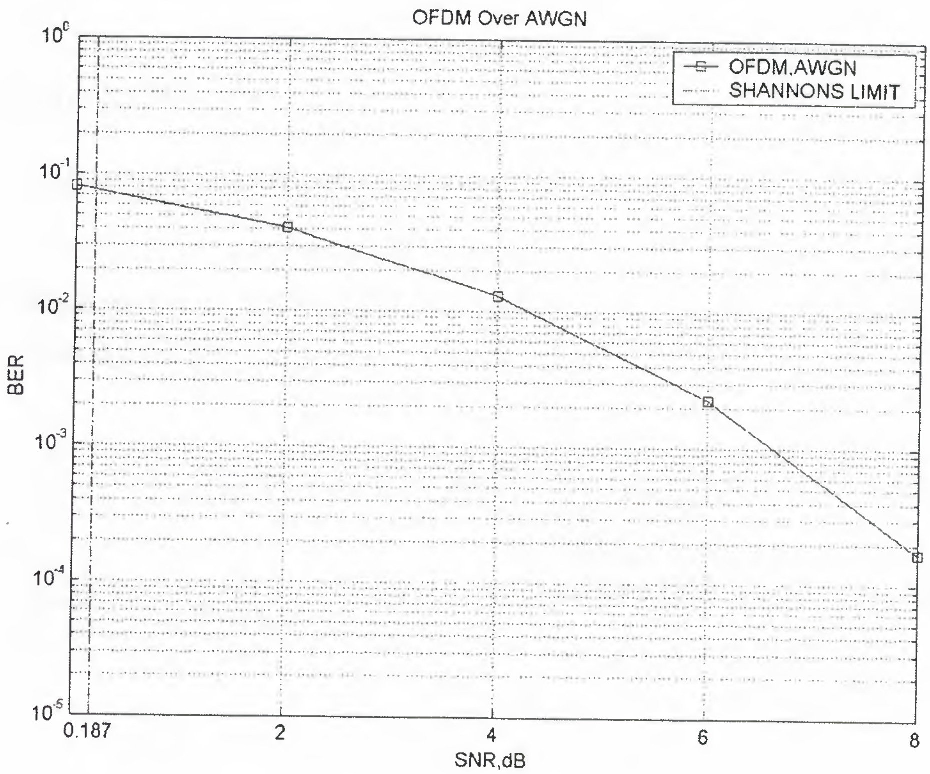


Figure 4.2 Performance of OFDM on AWGN channel.

In OFDM N subcarrier are used. Since fourier transform (FT) is implemented using IFFT/FFT, N needs to be a power of 2 PBSK modulation is used on each carrier.

4.1.3 Theoretical AWGN

Figure 4.1 shows the bit error rate probability for BPSK over AWGN channel. The probability is also given by [6]

$$P_e(AWGN)Q(\sqrt{2*SNR}) \tag{4.2}$$

where the complementary error function Q(x) is defined as [6]

$$Q(x) = \frac{1}{\sqrt{2*\pi}} \int_{x}^{\infty} \exp(\frac{-u^2}{2}) du \qquad (4.3)$$

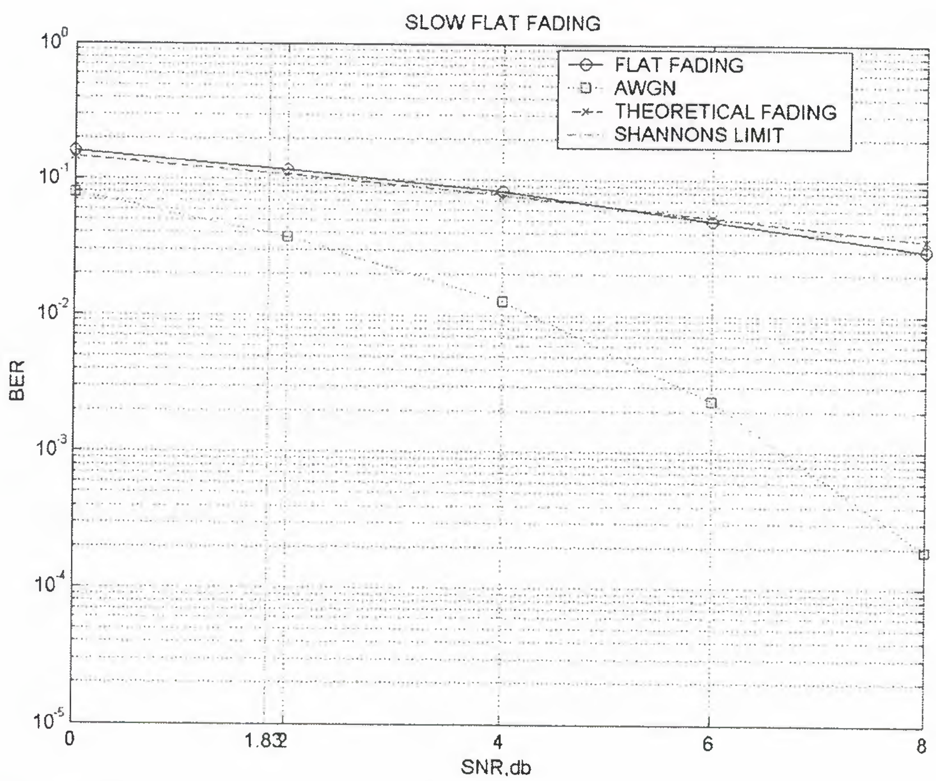
4.1.4 Shannon's limit

Shannon limit (or Shannon capacity) is a signal to noise ratio (SNR) limit required by the ideal system for error free transmission to be possible. For AWGN channel the limit is 0.187 db [1].

4.2 FADING RESULTS

4.2.1 Simulation single carrier communication

Figure 4.3 shows the performance of communication. BPSK modulated single carrier communications over slow, flat fading channel. It is observed that the performance is worse than that obtained over AWGN channel.



s

Figure 4.3 Performance of single carrier communication, over slow flat fading channel.

4.2.2 Simulation of fading with OFDM

Figure 4.4 shows the performance of the flat fading in OFDM over slow, flat fading channel. When compared to the performance of single carrier communication, its observed that the performance has slightly improved. This shows that using multicarrier communication is better to combat against fading than using a single carrier.

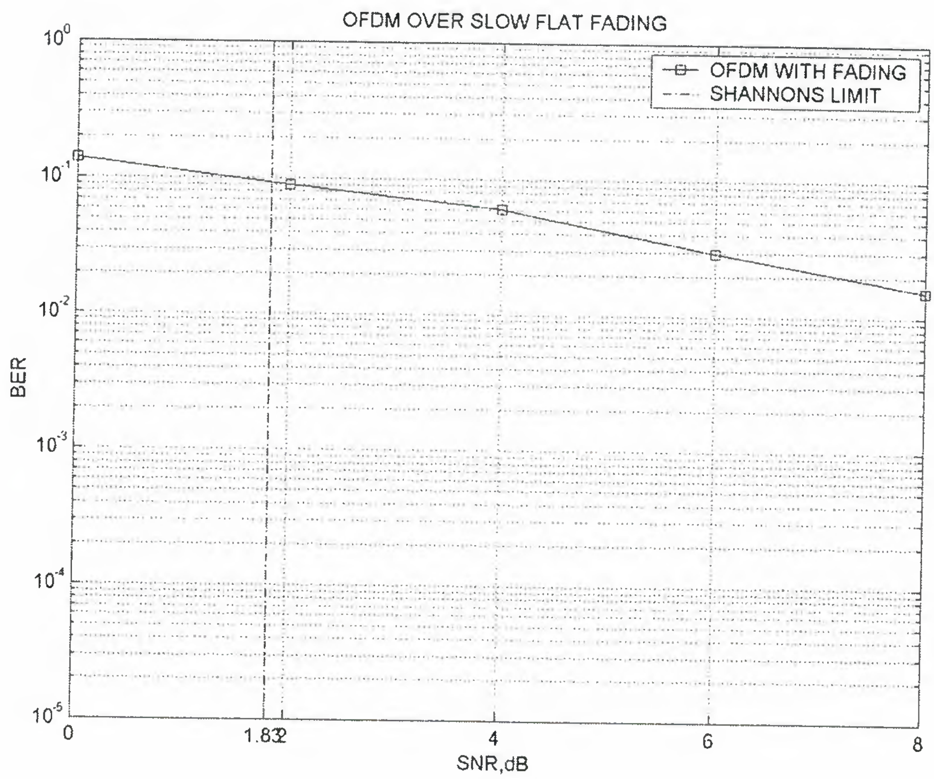


Figure 4.4 Performance of flat fading over OFDM.

4.2.3 Theoretical fading

In figure 4.3 the theoretical fading plot is also given. This performance is also given by

$$P_e(fading) = \frac{1}{2}\left(1 - \sqrt{\frac{SNR}{1+SNR}}\right) \tag{4.4}$$

4.2.4 Shannons limit

The shannons limit for flat fading channel is 1.83 dB [1].

CONCLUSION

Fading and intersymbol interference occurring in wireless and bandlimited channels are major limiting factors in communications systems. Various methods exists for eliminating these limitations, including a multicarrier modulation technique called orthogonal frequency division multiplexing (OFDM).

Performance of single carrier and OFDM modulator communication systems are analyzed in addition white Gaussian noise (AWGN) and flat, slow fading channels. It is shown, through MATLAB simulation, that OFDM should be preferred in flat, slow fading channels over single carrier systems as it helps to increase the performance.

As a future work, guard intervals could be created where cyclic prefix is used to overcome the effect of intersymbol interference.

REFERENCES

[1] Proakis, J.G, Salehi, M., Communication systems Engineering, Upper Saddle River, New Jersy 07458, 2002.

[2] Tranter, W.H., Shanmugan, K., and rappaport, T.S., Simulation of modern communication systems with wireless applications, Prentice Hall, Upper Saddle River, NJ, 2002.

[3] Lee, W. C .Y., Mobile Cellular Telecommunication Systems, McGraw Hill Publications, New York, 1989.

[4] Chuang,J., "The Effects of Time Delay Spread on Portable Communication Channels with Digital Modulation," IEEE Journal on Selected Areas in Communication, Vol.SAC-5,No.5, pp. 879-889, June 1987.

[5] Steele, R. Ed., Mobile Radio Communications, IEEE Press, 1994.

[6] Repapport, T.S., Wireless Communications, Prentice Hall PTR, Upper Saddle River, NJ 07458, 2002.

[7] Proakis, J. G., Digital Communication, McGraw-Hill Book Company, New York, Electr. Eng., vol.47, Apr. 1928, pp. 617-644.

[8] Korn, I., Digital Communication, Van Nostrand Reinhold Company, Inc., New York, 1985.

[9] L.J. cimini, "Anaysis and simulation of a digital mobile channel using orthogonal frequency division multiplexing" IEEE Trans. Infor. Common., Vol. 33, No.7, pp 665-675, July 1985.

[10] P.S. Chow, J.M. Cioffi, and John A.C. Bingham, "A practical discrete multitone transceiver loading algorothim for data transmission over spectrally shaped channels," IEEE Trans. Commun., Vol. 43, No. 2/3/4, Feb.-Apr. 1995.

[11] R. Mosier, R. Clabaugh, “Kineplex: A Bandwidth-efficient Binary Transmission System,” AIEE Trans., Vol. 76, pp 723-728, Jan. 1958.

APPENDIX A

MATLAB CODES DESCRIBTION

Appendix A.1 Simulation of single carrier communication over AWGN channel

snr =0:2:8; % The values of signal to noise ratio(SNR) and it is in decibal values,

for i=1:5 % We make a loop in which each loop has zero error and transmitted data.

err=0;

data=0;

var=1/(2*(10^(snr(i)/10))); % The variance is related with the SNR (nondecibal) values and when its increases the variance decreases which decrease the noise.

q(i)=0.5*erfc(sqrt(10^(snr(i)/10))); % the performance of the theoretical AWGN.

while err<100 %If the error reach this value the loop stops.

data=data+1;

s=rand %We generate a real number from 0 to 1 randomly

if s>.5 %We assume if the random number bigger than 0.5 then this number is equal 1.

s=1

else %Otherwise it equals 0 so we are generating 0's and 1's

s=0

```
end
tx=(2*s)-1                  % We transmit the signal using BPSK(Binary phase
                              shift key) so the values equal 1's and minus 1's
noise=sqrt(var)*randn       %We generate gaussian noise.
rx=tx+noise                 %At the receiver the noise added to the transmitted
                              signal.
if rx>0                     % We assumed if the received message bigger than
                              0 it equals 1 otherwise it equals -1
rx=1
else
rx=-1
end
if rx~=tx                   % We compare the transmitted with the received
                              message if its not equal then an error is detected.
err=err+1                   % We count up the number of error.
end
end                         % End for while loop.
ber(1,i)=err/data           % Bit error rate for individual SNR values.
End                         % End for the FOR loop.
x=0.187*ones(1,6);          % This is the value of Shannon's limit for AWGN.
y=[10^-5 10^-4 10^-3 10^-2 10^-1 10^0];
semilogy(snr,ber,'-s',snr,q,'--*',x,y,'-.')
grid
set(gca,'xtick',[0 .187 2 4 6 8])
xlabel('SNR')
```

```
ylabel('BER')
axis([0 8 10^-5 10^0])
title('AWGN')
legend('PRACTICAL AWGN','THEORETICAL AWGN','SHANNONSLIMIT')
```

Appendix A.2 Simulation of OFDM over AWGN channel

```
snr_db=0:2:8;
N=512;
for i=1:5
err=0;
data=0;
snr=10^(snr_db(i)/10);
variance=1/(2*N*snr);

while err<100
data=data+N;

for k=1:N
s(k)=rand;

if s(k)>0.5
s(k)=1;
```

```
else
s(k)=0;
end
end

tx=2*s-1;
t=ifft(tx,N);
for k=1:N
rand1=sqrt(0.5)*randn;
rand2=sqrt(0.5)*randn;
fade=sqrt(rand1^2+rand2^2);
r(k)=(t(k))+((sqrt(variance)*randn)+(sqrt(variance)*randn*j));
end
rx=fft(r,N);
for k=1:N
if real(rx(k))>0
r(k)=1;
else
r(k)=-1;
end
```

```
if r(k)~=tx(k)

err=err+1;

end

end

end%while

ber(1,i)=err/data;

end%for

x=0.187*ones(1,6);

y=[10^-5 10^-4 10^-3 10^-2 10^-1 10^0];

semilogy(snr_db,ber,'-s',x,y,'-.')

grid

set(gca,'xtick',[0 0.187 2 4 6 8])

xlabel('SNR,dB')

ylabel('BER')

axis([0 8 10^-5 10^0])

title('OFDM Over AWGN')

legend('OFDM,AWGN','SHANNONS LIMIT')
```

Appendix A.3 Simulation of Fading in single carrier

```
snr =0:2:8;

for i=1:5

err=0;

data=0;

snr_db=10^(snr(i)/10)

var=1/(2*snr_db)

q(i)=0.5*erfc(sqrt(snr_db))

thfade(i)=0.5*(1-(sqrt((snr_db)/(1+snr_db))))

while err<100

data=data+1

s=rand

if s>.5

s=1

else

s=0

end

tx=(2*s)-1

noise=sqrt(var)*randn

rand1=sqrt(0.5)*randn
```

```
rand2=sqrt(0.5)*randn
fade=sqrt((rand1^2)+(rand2^2))
rx=(fade*tx)+noise
if rx>0
rx=1
else
rx=-1
end
if rx~=tx
err=err+1;
end
end
ber(1,i)=err/data;
end
x=1.83*ones(1,6);
y=[10^-5 10^-4 10^-3 10^-2 10^-1 10^0];
semilogy(snr,ber,'-o',snr,q,':s',snr,thfade,'--x',x,y,'-.')
grid
set(gca,'xtick',[0 1.83 2 4 6 8])
xlabel('SNR,db')
```

```
ylabel('BER')
axis([0 8 10^-5 10^0])
title('SLOW FLAT FADING')
legend('FLAT FADING','AWGN','THEORETICAL FADING','SHANNONS LIMIT')
```

Appendix A.4 Simulation Fading over OFDM

```
snr_db=0:2:8;
N=512;
for i=1:5
err=0;
data=0;
snr=10^(snr_db(i)/10);
variance=1/(2*N*snr);
while err<100
data=data+N;
for k=1:N
s(k)=rand;
        if s(k)>0.5
            s(k)=1;
        else
s(k)=0;
```

```
End
End
tx=2*s-1;
t=ifft(tx,N);
for k=1:N
rand1=sqrt(0.5)*randn;
rand2=sqrt(0.5)*randn;
fade=sqrt(rand1^2+rand2^2);
r(k)=(fade*t(k))+((sqrt(variance)*randn)+(sqrt(variance)*randn*j));
end
rx=fft(r,N);
for k=1:N
if real(rx(k))>0
r(k)=1;
else
r(k)=-1;
end
if r(k)~=tx(k)
err=err+1;
end
```

```
End

end%while

ber(1,i)=err/data;

end%for

x=1.83*ones(1,6);

y=[10^-5 10^-4 10^-3 10^-2 10^-1 10^0];

semilogy(snr_db,ber,'-s',x,y,'-.')

grid

set(gca,'xtick',[0 1.83 2 4 6 8])

xlabel('SNR,dB')

ylabel('BER')

axis([0 8 10^-5 10^0])

title('OFDM OVER FLAT FADING')

legend('OFDM with fading','SHANNONS LIMIT')
```