NEAR EAST UNIVERSITY

Faculty of Engineering

Department of Electrical and Electronic Engineering

Code Division Multiple Access (CDMA)

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ABSTRACT

Sending and receiving data is very important to the human race nowadays, the main problem that faces this operation is fading and interferences due to the channel or multi user cases, because of that the transmitted data corrupts and the receiver couldn't receive it correctly, so the scientists developed some ways to prevent this errors and to fix it when this errors occurs such as DSSS (Direct Sequence spread spectrum) & CDMA

The main goal of this project is to investigate characteristics of CDMA (Code division Multiple Access), we investigate only one user case DSSS, for CDMA it's the multi user case and it gives every user a special code that only the receiver knows to prevent being jammed, and gives the most important thing that the users looking for more security, and the simulation shown the affect of Additive White Gaussian Noise with different amplitudes in narrow band and channel interferences and pulse jamming in a duty cycle for multiple ρ's,

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INTRODUCTION

Nowadays the development of communication and digital communication is moving rapidly; looking for more accurate ways to send and receive data without being disturbed by the channel or the users; so some development on using DSSS and CDMA allow the senders to send their data preventing interference and fading from happening or being jammed by a jammer who wants to distort the sending data, the receiver will have a code for each sender that gives security to the users, in this project we investigated some of these developments and its characteristics.

Chapter one gives a general overview of communication systems, characteristics of the communication systems, and talks about the transmitter, channel and the receiver, and ways of propagation, then explains the storage channels and shows some mathematical models for the communication channel.

Chapter two describes fading and interference and important variables, and how does it affect the transmitted data

Chapter three overview Spread Spectrum, multiplexing, and multiple access, mainly talks about CDMA, and some major differences between the other multiple access methods and the classification of the different access methods.

Chapter four talks about the results of the simulation and shows some theoretical formulas related to the figures obtained from the simulation, and give the theoretical calculations for the Additive White Gaussian Noise and the pulsed jamming.

Simulation codes can be found in the appendix in the last section of this project

CHAPTER ONE

COMMUNICATION SYSTEMS OVERVIEW

1.1 Communication Systems

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; that is, the output of a source is not deterministic. Otherwise, there would be no need to transmit the message.

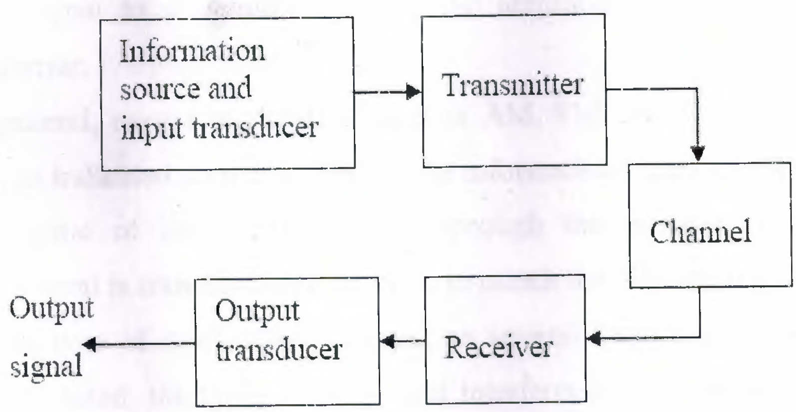

Figure1.1: Functional block diagram of communication system [1].

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; for example, 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 below.

1.1.1 The Transmitter

A transmitter converts the electrical signal into a form that is suitable for transmission through the physical channel or transmission medium. For example, 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 to be transmitted into the appropriate frequency range that 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 the information signal to systematically vary the amplitude, frequency, or phase of a sinusoidal carrier.

In general, carrier modulation such as AM, FM, and PM is performed at the transmitter, as indicated above, to convert the information signal to a form that matches the characteristic 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 in transmission over the channel, and the electronic devices that are available for signal amplification prior to transmission.

In any case, the modulation process makes it possible 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 wire lines, 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.

Signal distortions are usually characterized as random phenomena and described in statistical terms. The effect of these signal distortions must be taken into account in 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 case, 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 to extract the message from the sinusoidal carrier. Since the signal demodulation is performed in the presence of additive noise and possibly other signal distortions, 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 Systems

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

Figure 1.2 illustrates the functional diagram and the basic elements of a digital communication system. The source output may be either an analog signal, such as audio or video signal, or a digital signal, such as the output of a Teletype machine, which is 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.

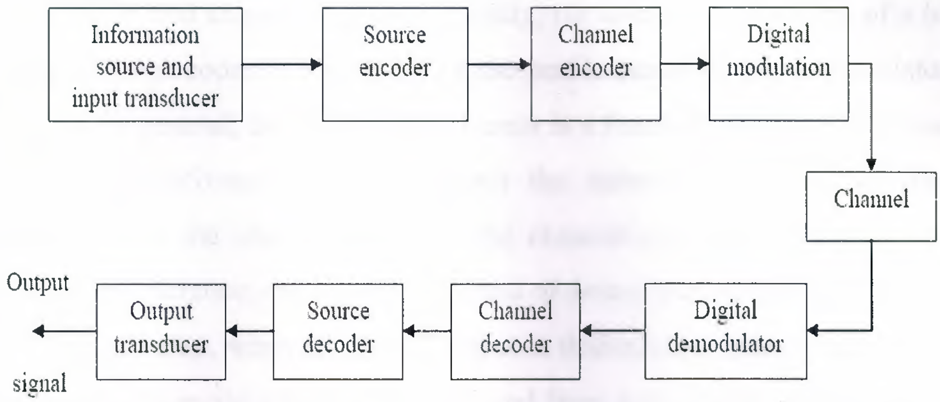

Figure 1.2: Basic elements of digital communication system

Ideally, we would like to represent the source output (message) by as few binary digits as possible. In other words, we seek an efficient representation of the source output that results in little or no redundancy. The process of efficiently converting the output of either an analog or a digital source into a sequence of binary digits is called source encoder 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. For example, 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. 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.

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. 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 demodulator-decoder combination. In general, the probability of error is a function of the code characteristics, the types of waveforms used to transmit the information over the channel, the transmitter power, the characteristics of the channel (i.e., the amount of noise), the nature of the interference, etc., and the method of demodulation and decoding.

As a final step, 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

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 multipath distortion.

Increasing the power in the transmitted signal may minimize the effects of noise. 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 be 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 wirelines for voice signal transmission, as well as data and video transmission. Twisted pair wirelines 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 kilo-hertz (kHz). On the other hand, coaxial cable has a usable bandwidth of several megahertz (MHz). Figure 1.3 illustrates the frequency range of guided electromagnetic channels, which includes waveguides and optical fibers.

Signals transmitted through such channels are distorted in both amplitude and phase and further corrupted by additive noise. Twisted-pair wireline channels are also prone to crosstalk interference from physically adjacent channels. Because wireline 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 communications 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 that have relatively low signal attenuation, and highly 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 trans-pacific communications. With the large bandwidth available on fiber optic channels it is possible for 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 light 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.

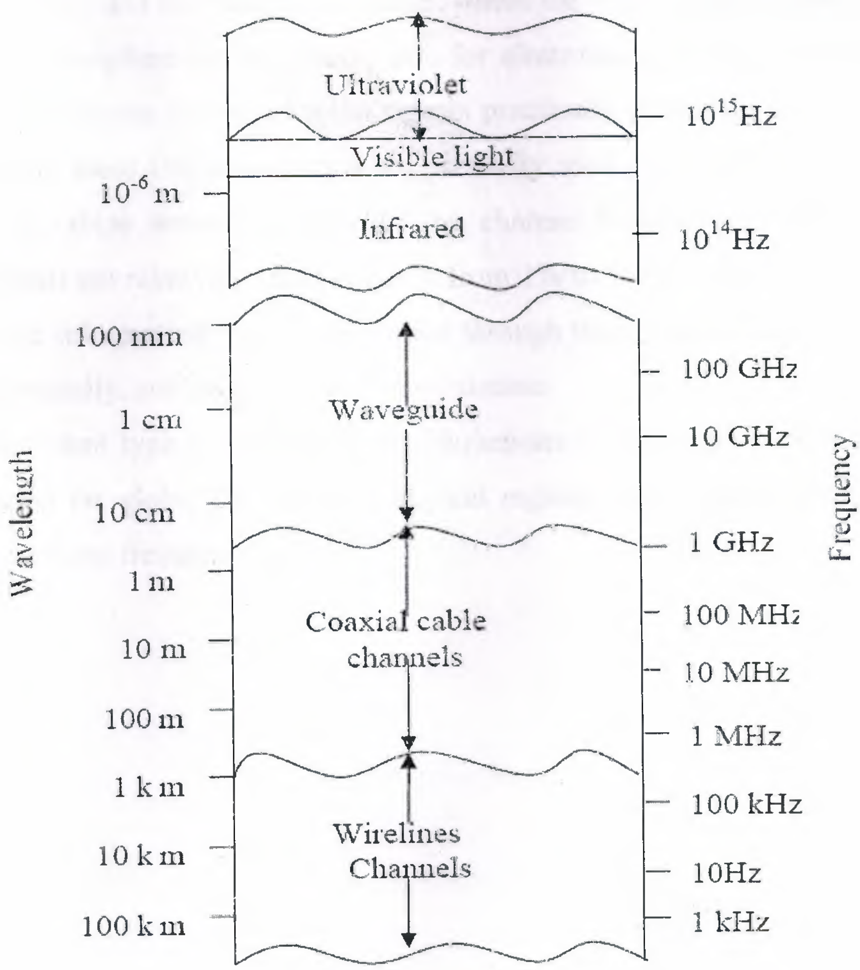

Figure 1.3: Frequency ranges for guided wire channel

1.3.3 Wireless Electromagnetic Channels

In wireless 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 transmitting in the AM frequency band, say at 1 MHz (corresponding to a wavelength of $\lambda = c/f_c = 300\text{m}$), requires an antenna of at least 30 meters.

Figure 1.4 illustrates the various frequency bands of the electromagnetic spectrum. 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 bandwidth available in these frequency bands are relatively small (usually from 1% to 10% of the center frequency), and hence, the information that is transmitted through these channels is relatively slow 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.

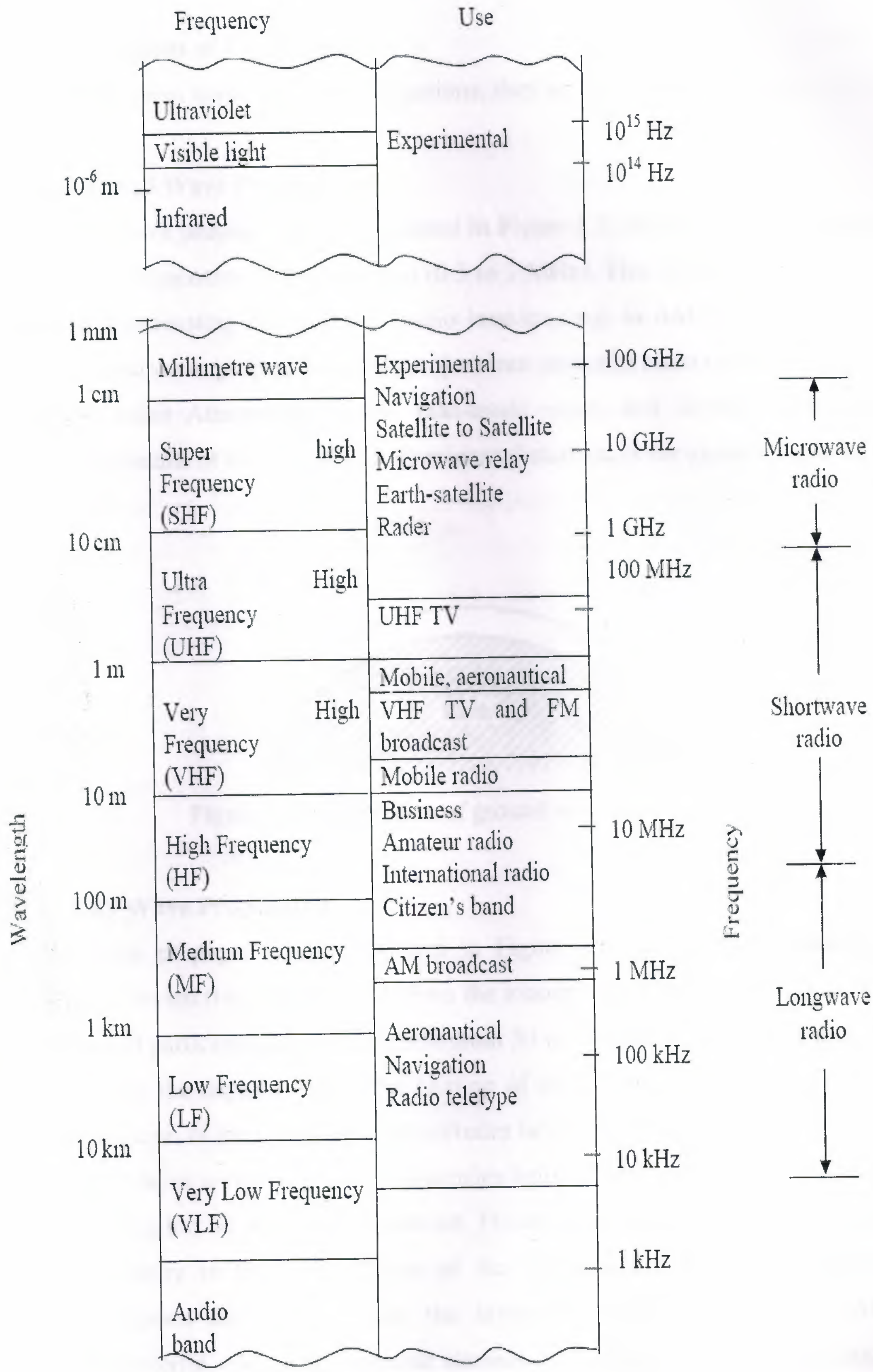

Figure 1.4: Frequency ranges for wireless electromagnetic channel

1.3.3.1 Classifications of Propagations

There are three main ways of propagations, they are explained below.

1.3.3.1.1 Ground Wave Propagation

Ground-wave propagation, as illustrated in Figure 1.5, is the dominant mode of propagation for frequencies in the MF band (0.3 to 3 MHz). This is the frequency band used for AM broadcasting and maritime radio broadcasting. In AM broadcasting, the range with ground wave propagation of even the more powerful radio stations 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.

Figure 1.5: Illustration of ground wave propagation

1.3.3.1.2 Sky Wave Propagation

Sky-wave propagation, as illustrated in Figure 1.6, 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 to 250 miles above the surface of the earth. During the daytime hours, the heating of the lower atmosphere by the sun causes the formation of the lower layers at altitudes below 75 miles. These lower layers, especially the D-layer serves 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 daytime 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 miles to 250 miles above the surface of the earth.

A frequently occurring problem with electromagnetic wave propagation via sky wave in the HF frequency range is signal multipath. Signal multipath occurs when the signal multipath generally results in intersymbol 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 sky wave is the dominant propagation Mode. Additive noise at HF is a combination of atmospheric noise and thermal voice.

Sky-wave ionospheric 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 MHz to 60MHz, resulting from signal scattering from the lower ionosphere. It is also possible to communicate over distances of several hundred miles by use of troposphere scattering at frequencies in the range of 40 MHz to 300MHz. 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.

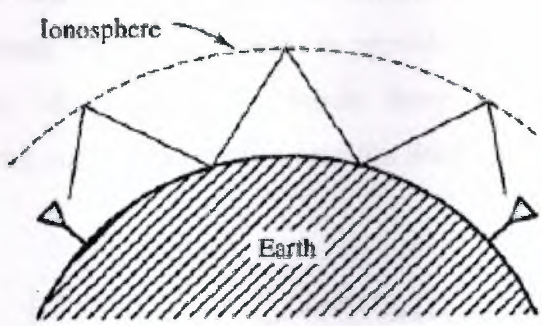

Figure 1.6: Illustration of sky wave propagation

1.3.3.1.3 Line Of Sight (LOS)

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-sight (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 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 feet above the surface of the earth, the distance to the radio horizon, assuming no physical obstructions such as mountains, is approximately $\sqrt{2h}$ miles. For example, a TV antenna mounted on a tower of 1000 ft. in height provides coverage of approximately 50 miles. As another 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.

The dominant noise limiting the performance of communication systems in the VHF and UHF frequency ranges is thermal noise generated in the receiver front end and cosmic noise picked up by the antenna. At frequencies in the SHF band above 10 MHz, atmospheric conditions play a major role in signal propagation. Figure1.7 illustrates the signal attenuation in dB/mile due to precipitation for frequencies in there range of 10 to 100GHz. We observe that heavy rain introduces extremely high propagation losses that can result in service outages (total breakdown in the communication system).At frequencies above the EHF band, we have the infrared and visible light regions of the electromagnetic spectrum, which can be used to provide LOS optical communication in free space. To date, these frequency bands have been used in experimental communication systems, such as satellite-to-satellite links.

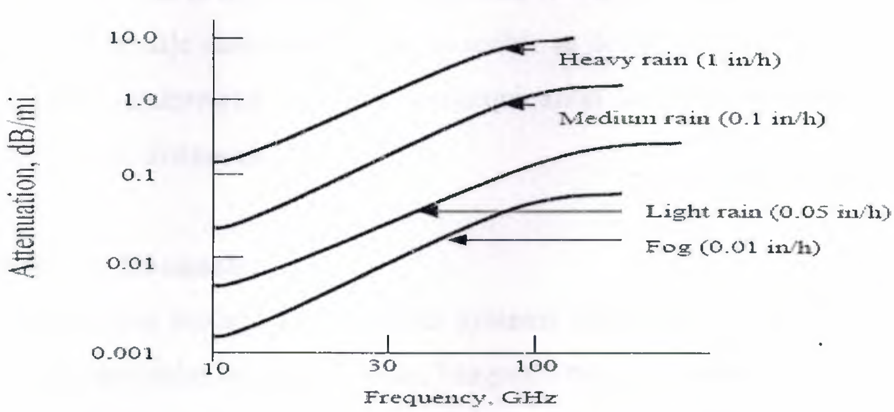

Figure 1.7: Signal attenuation due to precipitation

1.3.4 Underwater Acoustic Channels

Over the past few decades, ocean exploration activity has been steadily increasing. Coupled with this increase is the need to transmit data collected by sensors placed under water 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 under water 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

$$\delta = 250 / \sqrt{f} \tag{1.1}$$

where f is expressed in Hz and δ is in meters. For example, at 10 kHz, the skin depth is 2.5 meters. In contrast, acoustic signals propagate over distances of tens and even hundreds of kilometers.

An underwater acoustic channel is characterized as a multipath channel due to signal reflections from the surface and the bottom of the sea. Because of 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 harbours, there is also man-made acoustic noise in addition to the ambient noise. 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.

1.3.5 Storage Channels

Information storage and retrieval systems constitute a very significant part of 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, optical disks used for computer data storage and compact disks 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 feedback 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 read back 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 per square inch 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 are also limited by the associated mechanical and electrical subsystems that constitute information storage system.

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

1.4.1 The Additive Noise Channel

The simplest mathematical model for a communication channel is the additive noise channel, illustrated in Figure 1.8. In this model, the transmitted signal $s(t)$ is corrupted by an additive random noise process *n(t)*. 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 the 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 a Gaussian noise process. Hence, the resulting mathematical model applies to a broad class of physical communication channels, and because of its mathematical tractability this is the predominant channel model used in the channel is usually called the additive Gaussian noise channel.

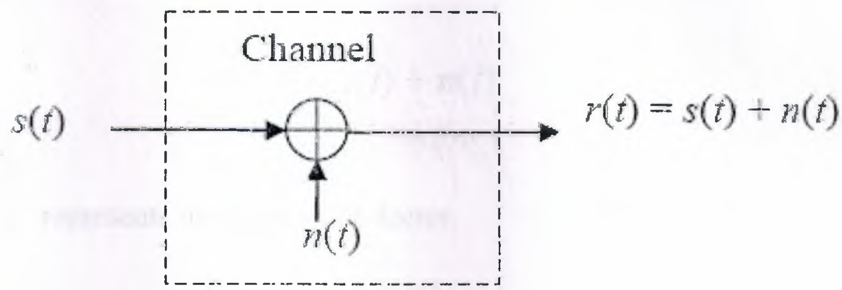

Figure 1.8: The additive noise channel

1.4.2 The Linear Filter Channel

In some physical channels such as wireline 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, see Figure 1.9.Hence, if the channel input is the signal *s*(*t*) the channel output is the signal

$$\begin{aligned} r(t) &= s(t) * h(t) + n(t) \\ &= \int_{-\infty}^{+\infty} h(\tau)s(t-\tau)d\tau + n(t) \end{aligned} \tag{1.2}$$

where $h(\tau)$ is the impulse response of the linear filter and denotes convolution.

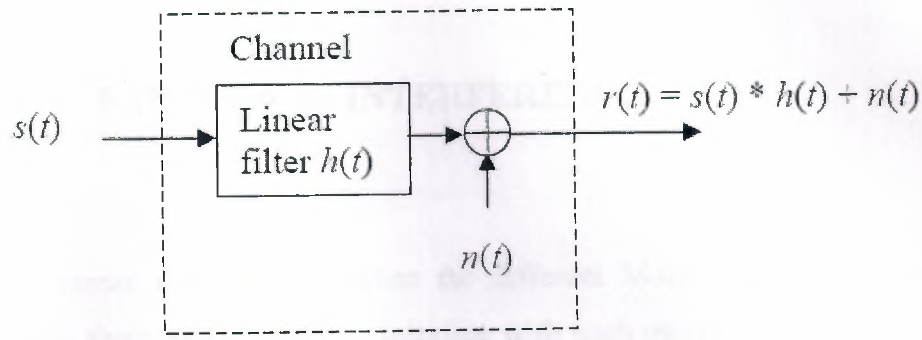

Figure 1.9: The linear filter channel with additive noise

When the signal undergoes attenuation in transmission through the channel, the received signal is

$$r(t) = \alpha\, s(t) + n(t) \tag{1.3}$$

where α represents the attenuation factor.

CHAPTER TWO

FADING AND INTERFERENCE

2.1 Fading

A simple RX cannot distinguish between the different Multi Path Components (MPCs); it just adds them up, so that they interfere with each other. The interference between them can be constructive or destructive, depending on the phases of the MPCs, (Figure 2.1). The phases, in turn, depend mostly on the run length of the MPC, and thus on the position of the mobile station and the IOs. For this reason, the interference, and thus the amplitude of the total signal, changes with time if either TX, RX, or IOs are moving. This effect-- namely, the changing of the total signal amplitude due to interference of the different MPCs – is called small-scale fading. At 2-GHz carrier frequency, a movement by less than 10 cm can already effect a change from constructive to destructive interference and vice versa. In other words, even a small movement can result in a large change in signal amplitude. A similar effect is known to all owners of car radios – moving the car by less than 1 meter (e.g., in stop-and-go traffic) can greatly affect the quality of the received signal. For cell phones, it can often be sufficient to move one step in order to improve signal quality.

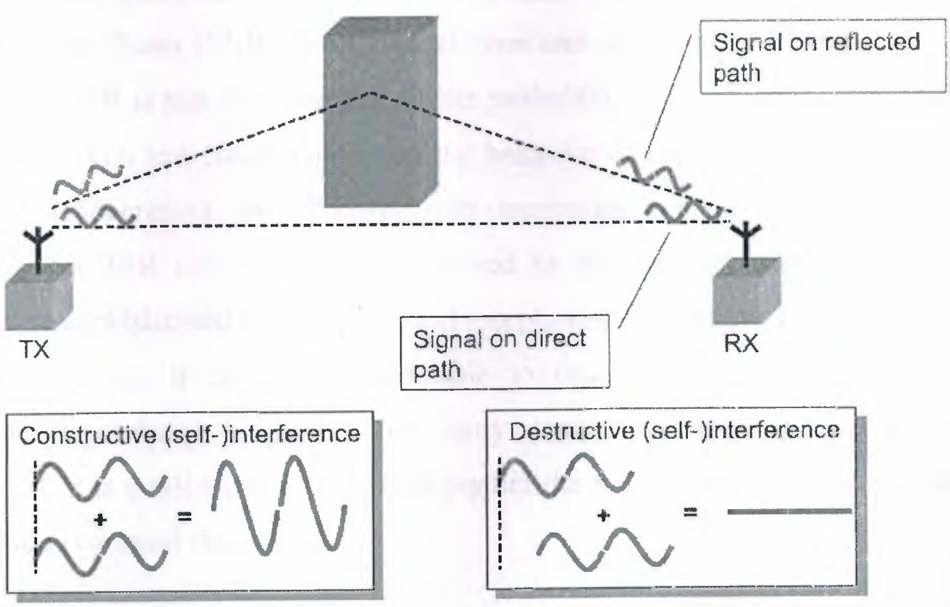

Figure 2.1: Principle of small-scale fading

As an additional effect, the amplitudes of each separate MPC change with time (or with location). Obstacles can lead to a shadowing of one or several MPCs. Imagine, for example, the MS (Mobile Station) in Figure 2.2 that at first (at position A) has LOS to the Base Station (BS). As the MS moves behind the high-rise building (at position B), the amplitude of the component that propagates along the direct connection (LOS) between BS and MS greatly decreases. This is due to the fact that the MS is now in the radio shadow of the high-rise building, and any wave going through or around that building is greatly attenuated – an effect called shadowing. Of course, shadowing can occur not only for a LOS component, but for any MPC. Note also that obstacles do not throw ''sharp'' shadows: the transition from the ''light'' (i.e., LOS) zone to the ''dark'' (shadowed) zone is gradual. The MS has to move over large distances (from a few meters, up to several hundreds of meters) to move from the light to the dark zone. For this reason, shadowing gives rise to large-scale fading.

Large-scale and small-scale fading overlap, so that the received signal amplitude can look like the one depicted in Figure 2.3. Obviously, the transmission quality is low at the times (or places) with low signal amplitude. This can lead to bad speech quality (for voice telephony), high Bit Error Rate (BER) and low data rate (for data transmission), and – if the quality is too low for an extended period of time – to termination of the connection.

It is well known from conventional digital communications that for non-fading communications links, the BER decreases approximately exponentially with increasing Signal-to-Noise Ratio (SNR) if no special measures are taken. However, in a fading channel, the SNR is not constant; rather, the probability that the link is in a fading dip (i.e., location with low SNR) dominates the behavior of the BER. For this reason, the average BER decreases only linearly with increasing average SNR. Consequently, improving the BER often cannot be achieved by simply increasing transmit power. Rather more sophisticated transmission and reception schemes have to be used.

Due to fading, it is almost impossible to exactly predict the received signal amplitude at arbitrary locations. For many aspects of system development and deployment, it is considered sufficient to predict the mean amplitude, and the statistics of fluctuations around that mean.

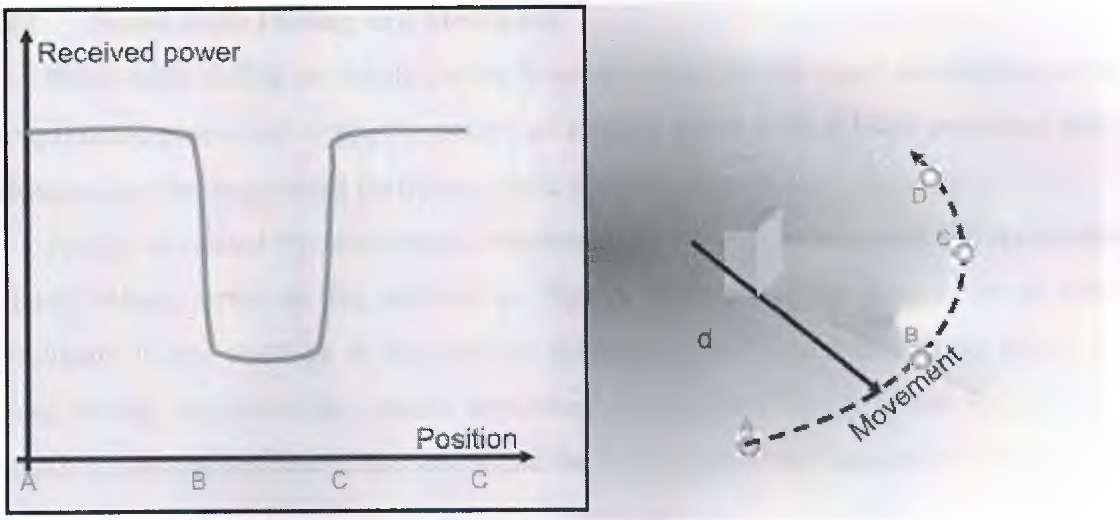

Figure 2.2: Principle of shadowing

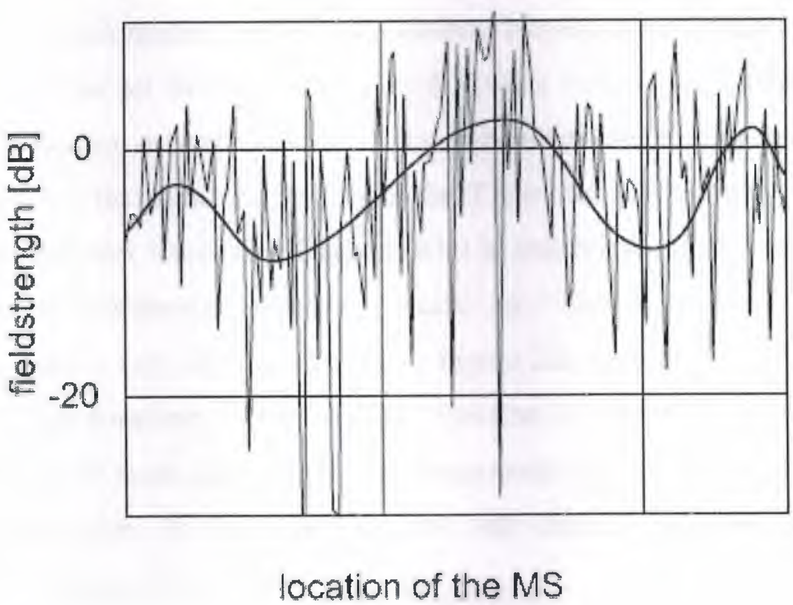

Figure 2.3: Typical example of fading, the thin line is the (normalized) instantaneous field strength; the thick line is the average over a 1-m distance.

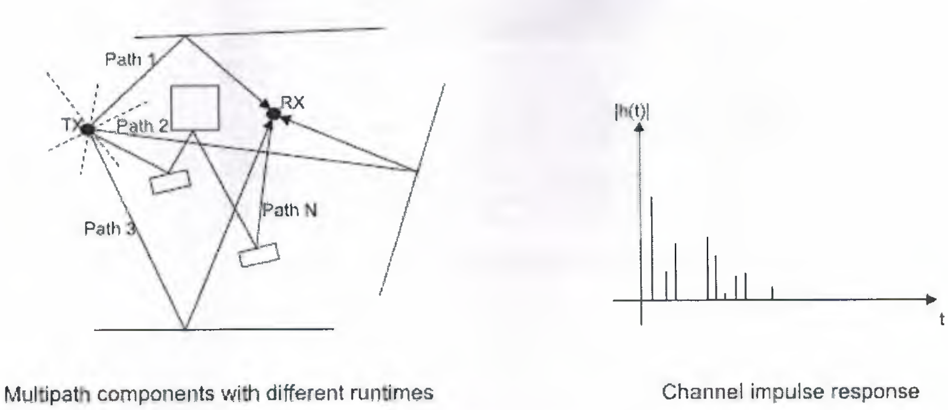

Figure 2.4: Multipath propagation and resulting impulse response

2.2 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 amplitude and phase, depending on the distribution of the intensity and relative propagation time of the waves and the bandwidth of the transmitted signal.

2.2.1 Multipath Propagation

The transmission medium is the radio channel between transmitter TX and receiver RX. The signal can get from the TX to the RX via a number of different propagation paths. In some cases, a Line Of Sight (LOS) connection might exist between TX and RX. Furthermore, the signal can get from the TX to the RX by being reflected at or diffracted by different Interacting Objects (IOs) in the environment: houses, mountains (for outdoor environments), windows, walls, etc. The number of these possible propagation paths is very large. As shown in Figure 2.5, each of the paths has a distinct amplitude, delay (runtime of the signal), direction of departure from the TX, and direction of arrival; most importantly, the components have different phase shifts with respect to each other. In the following, we will discuss some implications of the multipath propagation for system design.

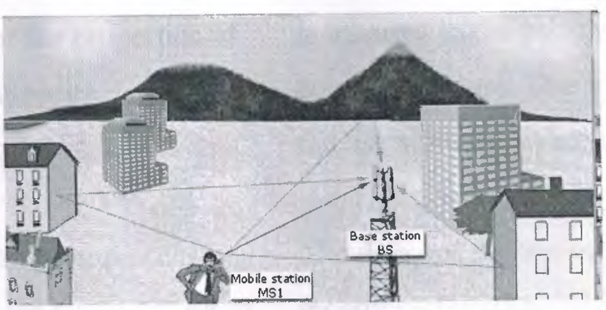

Figure 2.5: Multipath propagation

Wired communications	Wireless communications
The communication takes place over a more or less stable medium like copper wires or optical fibers. The properties of the medium are well-defined, and time-invariant.	Due to user mobility as well as multipath propagation, the transmission medium varies strongly with time.
Increasing the transmission capacity can be achieved by using a different frequency on an existing cable, and/or by stringing new cables.	Increasing the transmit capacity must be achieved by more sophisticated transceiver concepts and smaller cell sizes (in cellular systems), as the amount of available spectrum is limited.
The range over which communications can be performed without repeater stations is mostly limited by attenuation by the medium (and thus noise); for optical fibers, the distortion of transmitted pulses can also limit the speed of data transmission.	The range that can be covered is limited both by the transmission medium (attenuation, fading, and signal distortion) and by the requirements of spectral efficiency (cell size).
Interference and crosstalk from other users either do not happen, or the properties of the interference are stationary.	Interference and crosstalk from other users is inherent in the principle of cellular communications. Due to the mobility of the users, they also are time-variant.
The delay in the transmission process is also constant, determined by the length of the cable, and the group delay of possible repeater amplifiers.	The delay of the transmission depends mostly on the distance between base station and mobile station, and is thus time-variant.

The Bit Error Rate (BER) decreases strongly (approximately exponentially) with increasing Signal-to-Noise Ratio (SNR). This means that a relatively small increase in transmit power can greatly decrease the error rate.	For simple systems, the average BER decreases only slowly (linearly) with increasing average SNR. Increasing the transmit power usually does not lead to a significant reduction in BER. However, more sophisticated signal processing helps.
Due to the well-behaved transmission medium, the quality of wired transmission is generally high.	Due to the difficult medium, transmission quality is generally low unless special measures are used.
Jamming and interception of wired transmission is almost impossible without consent by the network operator.	Jamming a wireless link is straightforward, unless special measures are taken. Interception of the on-air signal is possible. Encryption is therefore necessary to prevent unauthorized use of the information.
Establishing a link is location-based. In other words, a link is established from one outlet to another, independent of which person is connected to the outlet.	Establishing a connection is based on the (mobile) equipment, usually associated with a specific person. The connection is not associated with a fixed location.
Power is either provided through the communications network itself (e.g., for traditional landline telephones), or from traditional power mains (e.g., fax). In neither case is energy consumption a major concern for the designer of the device.	Mobile stations use rechargeable or one-way batteries. Energy efficiency is thus a major concern.

2.2.2 Small-Scale Multipath Propagation

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

- Rapid changes in signal strength of 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 light 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 reflection from the ground and surrounding structures. The incoming radio waves arrive in different directions with different propagations delays. The signal received by the mobile at any point in space may consist of a large number of plane waves having a 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 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 a high speed can pass through several fades in a same period of time.

In a more serious case a receiver may stop at a particular application at which the receiver signal is in a deep fade. Maintaining good communication can then become very difficult, although passing vehicles or people walking in the vicinity of the mobile can often disturb the field pattern, there by diminishing the likelihood of the received signal remaining in a deep null for a long 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 the arrival of the received multipath waves.

2.2.2.1 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 scatters 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 the 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 enter symbol interference.

- Speed of the mobile

  The relative motion between the base station and the mobile result is 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 the 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 maybe ignored, and only the speed of the mobile need to be considered. The coherence time defines the "staticsness" of the channel, and is directly impacted by 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 will be significant).

2.2.2.2 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 decreases, this is called Doppler Effect; this effect becomes important when developing mobile radio systems.

The amount of the frequency changes due the Doppler Effect depends on the relative motion between the source and the receiver, and on the speed of propagation of the wave.

Doppler shift can cause significant problems if the transmission technique is sensitive to carrier frequency of sets, or the relative speed is very high as is the case for low earth orbiting satellites.

2.3 Intersymbol interference

The runtimes for different MPCs are different. We have already mentioned above that this can lead to different phases of MPCs, which leads to interference in narrowband systems. In a system with large bandwidth, and thus good resolution in the time domain, the major consequence is signal dispersion: in other words, the impulse response of the channel is not a single delta pulse, but rather a sequence of pulses (corresponding to different MPCs), each of which has a distinct arrival time in addition to having a different amplitude and phase (see Fig. 2.6). This signal dispersion leads to intersymbol interference at the RX. MPCs with long runtimes, carrying information from bit k, and MPCs with short runtimes, carrying contributions from bit k + 1 arrive at the RX at the same time, and interfere with each other.

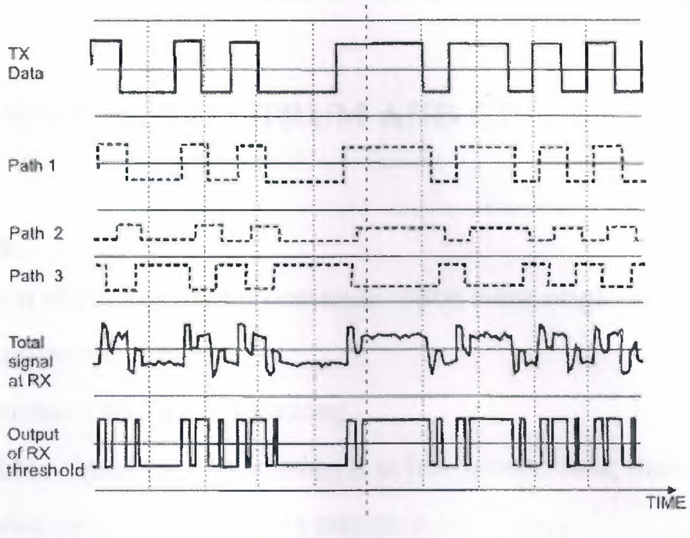

Figure 2.6: Intersymbol interference

Assuming that no special measures are taken, this Inter Symbol Interference (ISI) leads to errors that cannot be eliminated by simply increasing the transmit power, and are therefore often called irreducible errors.

ISI is essentially determined by the ratio between symbol duration and the duration of the impulse response of the channel. This implies that ISI is not only more important for higher data rates, but also for multiple access methods that lead to an increase in transmitted peak data rate.

Finally, it is also noteworthy that ISI can even play a role when the duration of the impulse response is shorter (but not much shorter) than bit duration.

CHAPTER THREE

SPREAD SPECTRUM AND CDMA

3.1 Spread Spectrum

Spread spectrum signal for digital communication were originally developed and used for military communication either

- To provide resistance to hostile jamming.
- To hide signal the signal by transmitting it at low power, thus, making it difficult for an unintended listener to detect its presence in noise or,
- To make it possible for multiple users to communicate through the same channel.

Today, however, spread spectrum signals are being used to provide reliable communication in a variety of commercial applications, including mobile vehicular communication and interoffice wireless communications

The basic element of a spread spectrum digital communication systems are illustrated in Figure 3.1. We observe that the channel encoder and decoder and the modulator and demodulator are the basic elements of conventional digital communication systems. In addition to these elements, a spread spectrum system employs two identical pseudorandom sequence generators, one of which interfaces with the modulator at the transmitting end and the second of which interfaces with the demodulator at the receiving end. These two generators produce a pseudorandom or pseudo noise (PN) binary valued sequence that is used to spread the transmitted signal in frequency at the modulator and to despread the received signal at the demodulator.

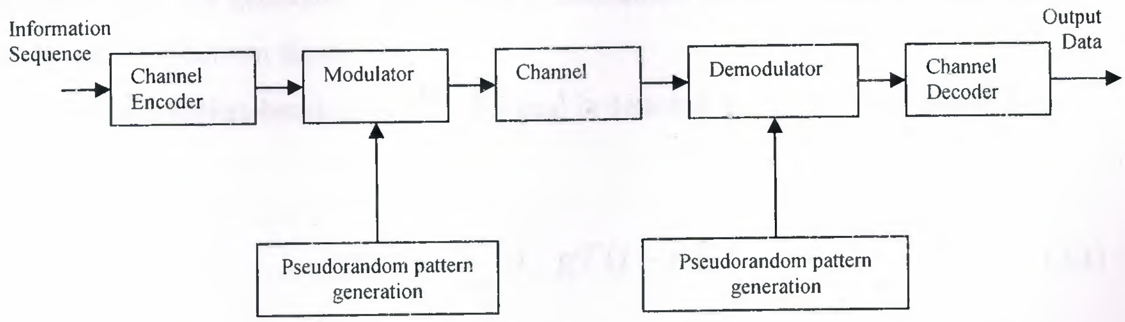

Figure 3.1: Model of spread spectrum digital communication system

NEAR EAST UNIVERSITY

Faculty of Engineering

Department of Electrical and Electronic Engineering

Code Division Multiple Access (CDMA)

Graduation Project
EE-400

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Nicosia-2008

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ABSTRACT

Sending and receiving data is very important to the human race nowadays, the main problem that faces this operation is fading and interferences due to the channel or multi user cases, because of that the transmitted data corrupts and the receiver couldn't receive it correctly, so the scientists developed some ways to prevent this errors and to fix it when this errors occurs such as DSSS (Direct Sequence spread spectrum) & CDMA

The main goal of this project is to investigate characteristics of CDMA (Code division Multiple Access), we investigate only one user case DSSS, for CDMA it's the multi user case and it gives every user a special code that only the receiver knows to prevent being jammed, and gives the most important thing that the users looking for more security, and the simulation shown the affect of Additive White Gaussian Noise with different amplitudes in narrow band and channel interferences and pulse jamming in a duty cycle for multiple ρ's,

TABLE OF CONTENTS

INTRODUCTION

Nowadays the development of communication and digital communication is moving rapidly; looking for more accurate ways to send and receive data without being disturbed by the channel or the users; so some development on using DSSS and CDMA allow the senders to send their data preventing interference and fading from happening or being jammed by a jammer who wants to distort the sending data, the receiver will have a code for each sender that gives security to the users, in this project we investigated some of these developments and its characteristics.

Chapter one gives a general overview of communication systems, characteristics of the communication systems, and talks about the transmitter, channel and the receiver, and ways of propagation, then explains the storage channels and shows some mathematical models for the communication channel.

Chapter two describes fading and interference and important variables, and how does it affect the transmitted data

Chapter three overview Spread Spectrum, multiplexing, and multiple access, mainly talks about CDMA, and some major differences between the other multiple access methods and the classification of the different access methods.

Chapter four talks about the results of the simulation and shows some theoretical formulas related to the figures obtained from the simulation, and give the theoretical calculations for the Additive White Gaussian Noise and the pulsed jamming.

Simulation codes can be found in the appendix in the last section of this project

CHAPTER ONE

COMMUNICATION SYSTEMS OVERVIEW

1.1 Communication Systems

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; that is, the output of a source is not deterministic. Otherwise, there would be no need to transmit the message.

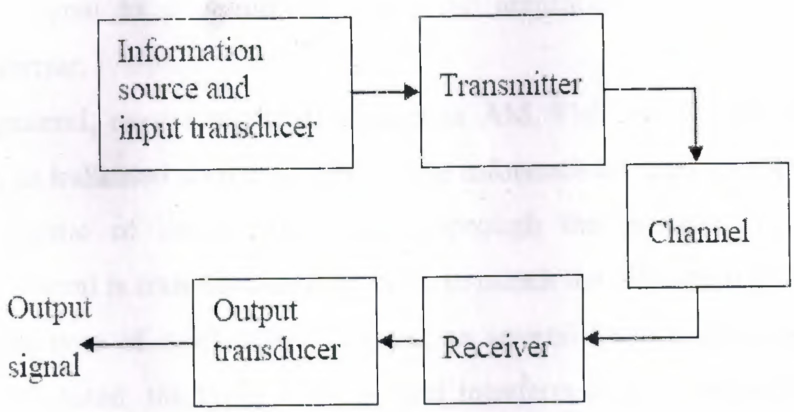

Figure1.1: Functional block diagram of communication system [1].

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; for example, 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 below.

1.1.1 The Transmitter

A transmitter converts the electrical signal into a form that is suitable for transmission through the physical channel or transmission medium. For example, 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 to be transmitted into the appropriate frequency range that 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 the information signal to systematically vary the amplitude, frequency, or phase of a sinusoidal carrier.

In general, carrier modulation such as AM, FM, and PM is performed at the transmitter, as indicated above, to convert the information signal to a form that matches the characteristic 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 in transmission over the channel, and the electronic devices that are available for signal amplification prior to transmission.

In any case, the modulation process makes it possible 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 wire lines, 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.

Signal distortions are usually characterized as random phenomena and described in statistical terms. The effect of these signal distortions must be taken into account in 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 case, 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 to extract the message from the sinusoidal carrier. Since the signal demodulation is performed in the presence of additive noise and possibly other signal distortions, 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 Systems

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

Figure 1.2 illustrates the functional diagram and the basic elements of a digital communication system. The source output may be either an analog signal, such as audio or video signal, or a digital signal, such as the output of a Teletype machine, which is 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.

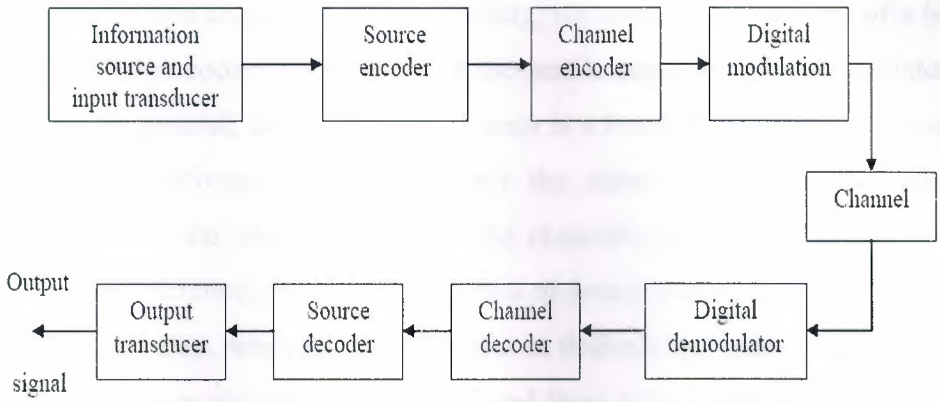

Figure 1.2: Basic elements of digital communication system

Ideally, we would like to represent the source output (message) by as few binary digits as possible. In other words, we seek an efficient representation of the source output that results in little or no redundancy. The process of efficiently converting the output of either an analog or a digital source into a sequence of binary digits is called source encoder 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. For example, 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. 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.

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. 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 demodulator-decoder combination. In general, the probability of error is a function of the code characteristics, the types of waveforms used to transmit the information over the channel, the transmitter power, the characteristics of the channel (i.e., the amount of noise), the nature of the interference, etc., and the method of demodulation and decoding.

As a final step, 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

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 multipath distortion.

Increasing the power in the transmitted signal may minimize the effects of noise. 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 be 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 wirelines for voice signal transmission, as well as data and video transmission. Twisted pair wirelines 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 kilo-hertz (kHz). On the other hand, coaxial cable has a usable bandwidth of several megahertz (MHz). Figure 1.3 illustrates the frequency range of guided electromagnetic channels, which includes waveguides and optical fibers.

Signals transmitted through such channels are distorted in both amplitude and phase and further corrupted by additive noise. Twisted-pair wireline channels are also prone to crosstalk interference from physically adjacent channels. Because wireline 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 communications 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 that have relatively low signal attenuation, and highly 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 trans-pacific communications. With the large bandwidth available on fiber optic channels it is possible for 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 light 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.

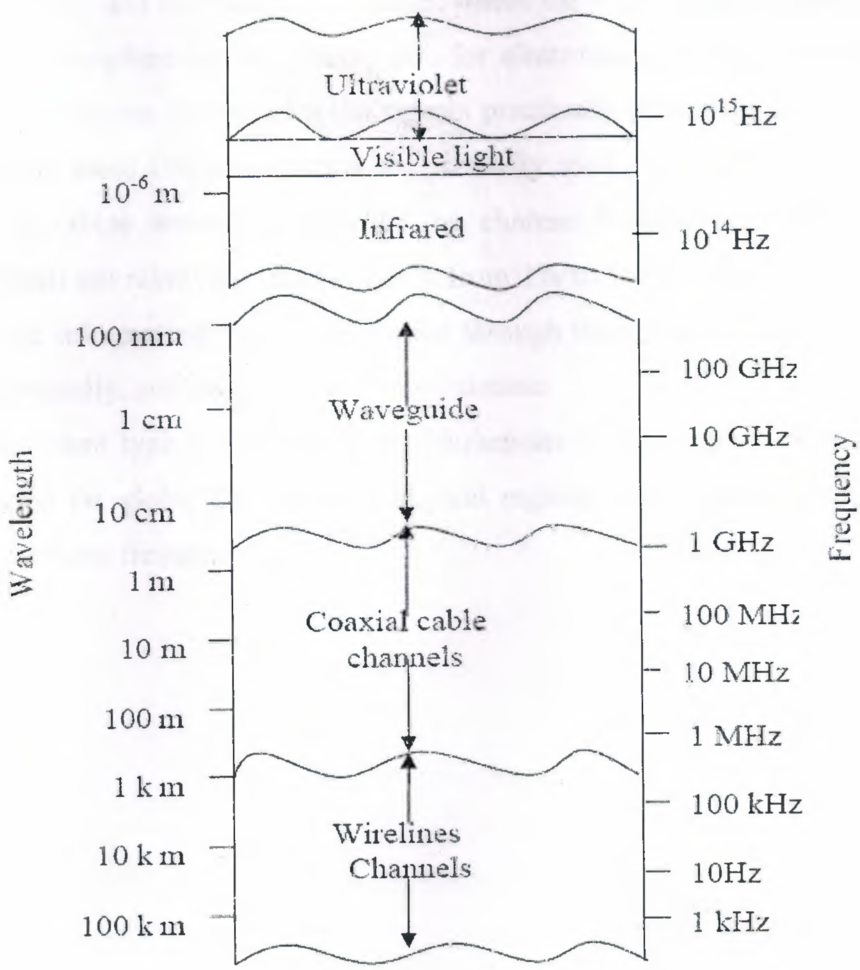

Figure 1.3: Frequency ranges for guided wire channel

1.3.3 Wireless Electromagnetic Channels

In wireless 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 transmitting in the AM frequency band, say at 1 MHz (corresponding to a wavelength of $\lambda = c/f_c = 300$m), requires an antenna of at least 30 meters.

Figure 1.4 illustrates the various frequency bands of the electromagnetic spectrum. 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 bandwidth available in these frequency bands are relatively small (usually from 1% to 10% of the center frequency), and hence, the information that is transmitted through these channels is relatively slow 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.

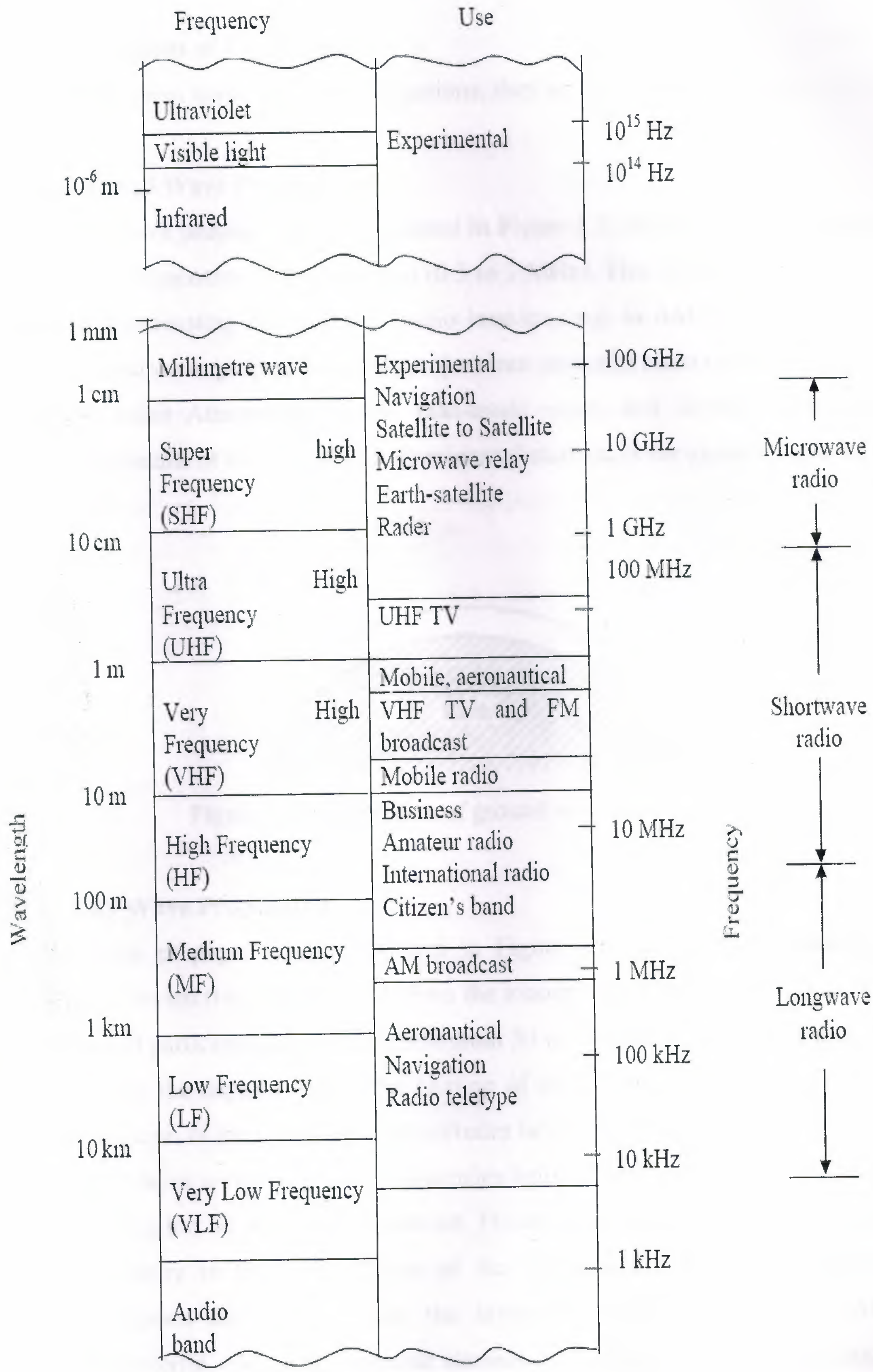

Figure 1.4: Frequency ranges for wireless electromagnetic channel

1.3.3.1 Classifications of Propagations

There are three main ways of propagations, they are explained below.

1.3.3.1.1 Ground Wave Propagation

Ground-wave propagation, as illustrated in Figure 1.5, is the dominant mode of propagation for frequencies in the MF band (0.3 to 3 MHz). This is the frequency band used for AM broadcasting and maritime radio broadcasting. In AM broadcasting, the range with ground wave propagation of even the more powerful radio stations 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.

Figure 1.5: Illustration of ground wave propagation

1.3.3.1.2 Sky Wave Propagation

Sky-wave propagation, as illustrated in Figure 1.6, 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 to 250 miles above the surface of the earth. During the daytime hours, the heating of the lower atmosphere by the sun causes the formation of the lower layers at altitudes below 75 miles. These lower layers, especially the D-layer serves 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 daytime 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 miles to 250 miles above the surface of the earth.

A frequently occurring problem with electromagnetic wave propagation via sky wave in the HF frequency range is signal multipath. Signal multipath occurs when the signal multipath generally results in intersymbol 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 sky wave is the dominant propagation Mode. Additive noise at HF is a combination of atmospheric noise and thermal voice.

Sky-wave ionospheric 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 MHz to 60MHz, resulting from signal scattering from the lower ionosphere. It is also possible to communicate over distances of several hundred miles by use of troposphere scattering at frequencies in the range of 40 MHz to 300MHz. 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.

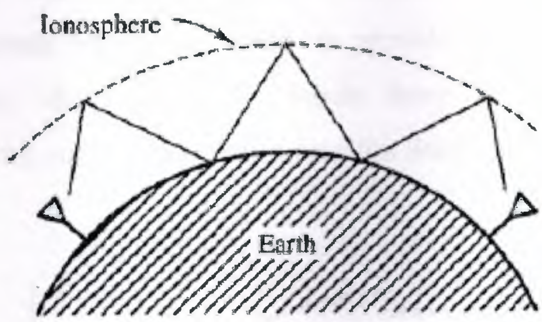

Figure 1.6: Illustration of sky wave propagation

1.3.3.1.3 Line Of Sight (LOS)

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-sight (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 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 feet above the surface of the earth, the distance to the radio horizon, assuming no physical obstructions such as mountains, is approximately $\sqrt{2h}$ miles. For example, a TV antenna mounted on a tower of 1000 ft. in height provides coverage of approximately 50 miles. As another 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.

The dominant noise limiting the performance of communication systems in the VHF and UHF frequency ranges is thermal noise generated in the receiver front end and cosmic noise picked up by the antenna. At frequencies in the SHF band above 10 MHz, atmospheric conditions play a major role in signal propagation. Figure1.7 illustrates the signal attenuation in dB/mile due to precipitation for frequencies in there range of 10 to 100GHz. We observe that heavy rain introduces extremely high propagation losses that can result in service outages (total breakdown in the communication system).At frequencies above the EHF band, we have the infrared and visible light regions of the electromagnetic spectrum, which can be used to provide LOS optical communication in free space. To date, these frequency bands have been used in experimental communication systems, such as satellite-to-satellite links.

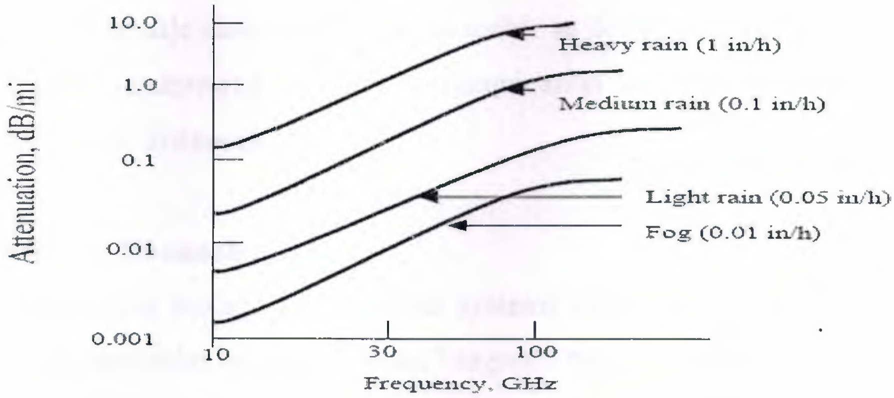

Figure 1.7: Signal attenuation due to precipitation

1.3.4 Underwater Acoustic Channels

Over the past few decades, ocean exploration activity has been steadily increasing. Coupled with this increase is the need to transmit data collected by sensors placed under water 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 under water 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

$$\delta = 250 / \sqrt{f} \tag{1.1}$$

where f is expressed in Hz and δ is in meters. For example, at 10 kHz, the skin depth is 2.5 meters. In contrast, acoustic signals propagate over distances of tens and even hundreds of kilometers.

An underwater acoustic channel is characterized as a multipath channel due to signal reflections from the surface and the bottom of the sea. Because of 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 harbours, there is also man-made acoustic noise in addition to the ambient noise. 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.

1.3.5 Storage Channels

Information storage and retrieval systems constitute a very significant part of 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, optical disks used for computer data storage and compact disks 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 feedback 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 read back 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 per square inch 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 are also limited by the associated mechanical and electrical subsystems that constitute information storage system.

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

1.4.1 The Additive Noise Channel

The simplest mathematical model for a communication channel is the additive noise channel, illustrated in Figure 1.8. In this model, the transmitted signal $s(t)$ is corrupted by an additive random noise process $n(t)$. 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 the 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 a Gaussian noise process. Hence, the resulting mathematical model applies to a broad class of physical communication channels, and because of its mathematical tractability this is the predominant channel model used in the channel is usually called the additive Gaussian noise channel.

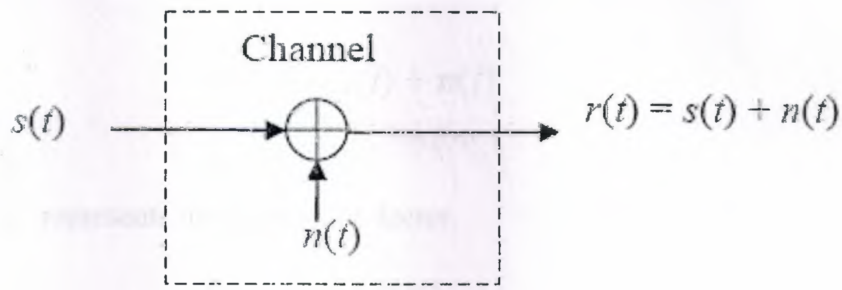

Figure 1.8: The additive noise channel

1.4.2 The Linear Filter Channel

In some physical channels such as wireline 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, see Figure 1.9.Hence, if the channel input is the signal $s(t)$ the channel output is the signal

$$\begin{aligned} r(t) &= s(t) * h(t) + n(t) \\ &= \int_{-\infty}^{+\infty} h(\tau)s(t-\tau)d\tau + n(t) \end{aligned} \tag{1.2}$$

where $h(\tau)$ is the impulse response of the linear filter and denotes convolution.

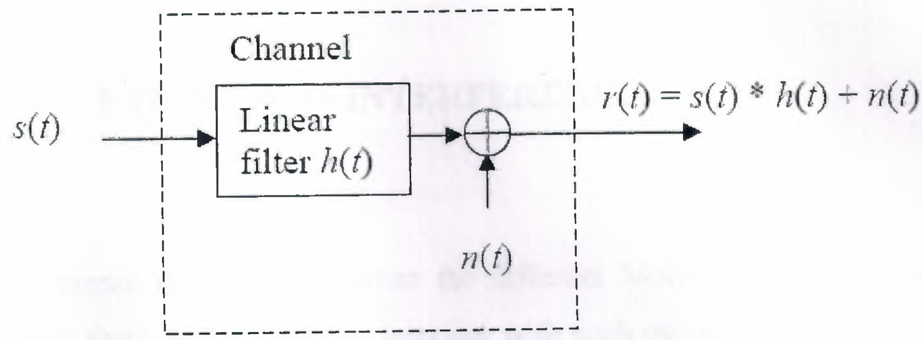

Figure 1.9: The linear filter channel with additive noise

When the signal undergoes attenuation in transmission through the channel, the received signal is

$$r(t) = \alpha\, s(t) + n(t) \tag{1.3}$$

where α represents the attenuation factor.

CHAPTER TWO

FADING AND INTERFERENCE

2.1 Fading

A simple RX cannot distinguish between the different Multi Path Components (MPCs); it just adds them up, so that they interfere with each other. The interference between them can be constructive or destructive, depending on the phases of the MPCs, (Figure 2.1). The phases, in turn, depend mostly on the run length of the MPC, and thus on the position of the mobile station and the IOs. For this reason, the interference, and thus the amplitude of the total signal, changes with time if either TX, RX, or IOs are moving. This effect-- namely, the changing of the total signal amplitude due to interference of the different MPCs – is called small-scale fading. At 2-GHz carrier frequency, a movement by less than 10 cm can already effect a change from constructive to destructive interference and vice versa. In other words, even a small movement can result in a large change in signal amplitude. A similar effect is known to all owners of car radios – moving the car by less than 1 meter (e.g., in stop-and-go traffic) can greatly affect the quality of the received signal. For cell phones, it can often be sufficient to move one step in order to improve signal quality.

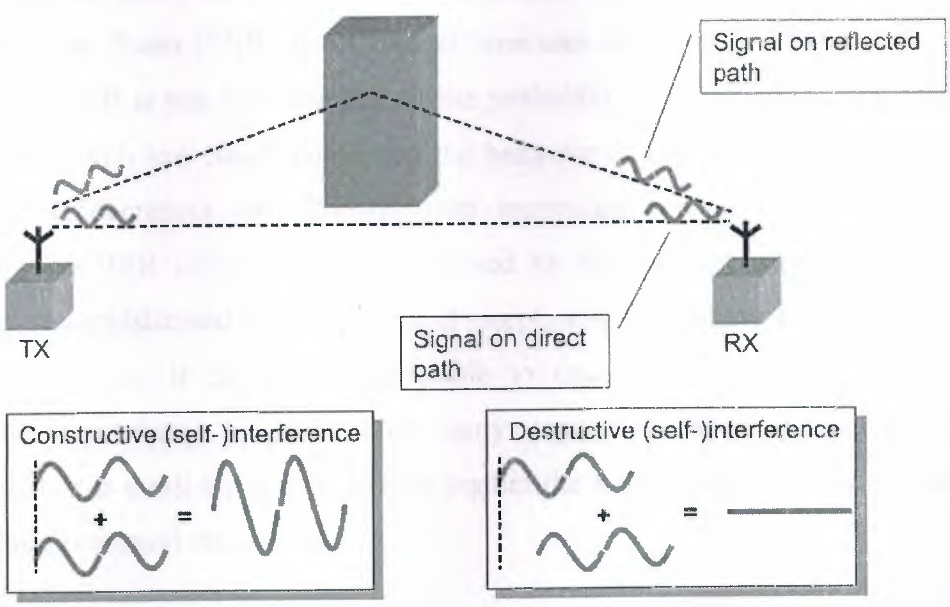

Figure 2.1: Principle of small-scale fading

As an additional effect, the amplitudes of each separate MPC change with time (or with location). Obstacles can lead to a shadowing of one or several MPCs. Imagine, for example, the MS (Mobile Station) in Figure 2.2 that at first (at position A) has LOS to the Base Station (BS). As the MS moves behind the high-rise building (at position B), the amplitude of the component that propagates along the direct connection (LOS) between BS and MS greatly decreases. This is due to the fact that the MS is now in the radio shadow of the high-rise building, and any wave going through or around that building is greatly attenuated – an effect called shadowing. Of course, shadowing can occur not only for a LOS component, but for any MPC. Note also that obstacles do not throw ''sharp'' shadows: the transition from the ''light'' (i.e., LOS) zone to the ''dark'' (shadowed) zone is gradual. The MS has to move over large distances (from a few meters, up to several hundreds of meters) to move from the light to the dark zone. For this reason, shadowing gives rise to large-scale fading.

Large-scale and small-scale fading overlap, so that the received signal amplitude can look like the one depicted in Figure 2.3. Obviously, the transmission quality is low at the times (or places) with low signal amplitude. This can lead to bad speech quality (for voice telephony), high Bit Error Rate (BER) and low data rate (for data transmission), and – if the quality is too low for an extended period of time – to termination of the connection.

It is well known from conventional digital communications that for non-fading communications links, the BER decreases approximately exponentially with increasing Signal-to-Noise Ratio (SNR) if no special measures are taken. However, in a fading channel, the SNR is not constant; rather, the probability that the link is in a fading dip (i.e., location with low SNR) dominates the behavior of the BER. For this reason, the average BER decreases only linearly with increasing average SNR. Consequently, improving the BER often cannot be achieved by simply increasing transmit power. Rather more sophisticated transmission and reception schemes have to be used.

Due to fading, it is almost impossible to exactly predict the received signal amplitude at arbitrary locations. For many aspects of system development and deployment, it is considered sufficient to predict the mean amplitude, and the statistics of fluctuations around that mean.

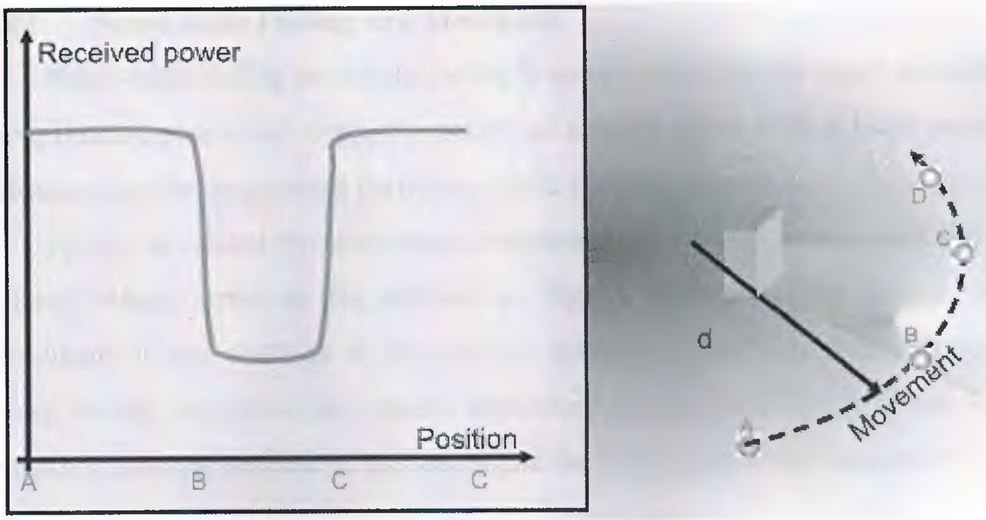

Figure 2.2: Principle of shadowing

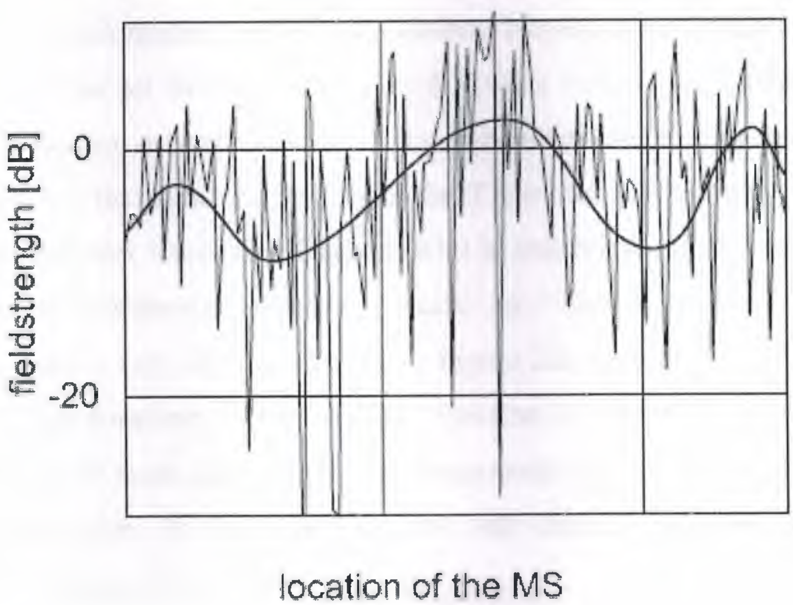

Figure 2.3: Typical example of fading, the thin line is the (normalized) instantaneous field strength; the thick line is the average over a 1-m distance.

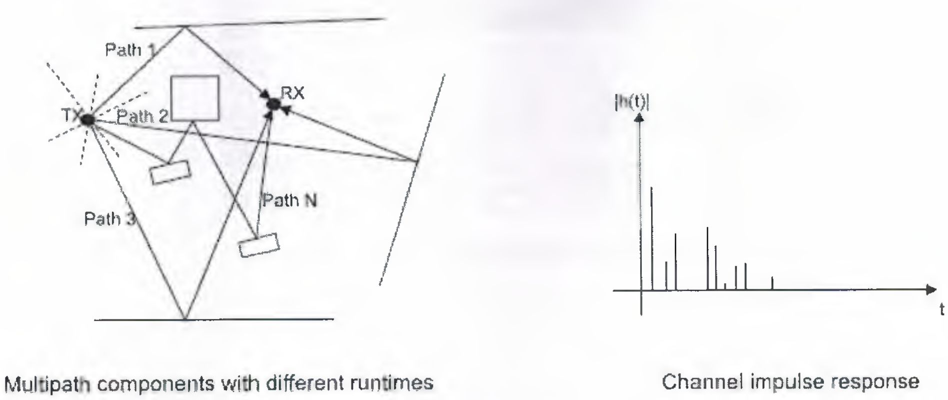

Figure 2.4: Multipath propagation and resulting impulse response

2.2 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 amplitude and phase, depending on the distribution of the intensity and relative propagation time of the waves and the bandwidth of the transmitted signal.

2.2.1 Multipath Propagation

The transmission medium is the radio channel between transmitter TX and receiver RX. The signal can get from the TX to the RX via a number of different propagation paths. In some cases, a Line Of Sight (LOS) connection might exist between TX and RX. Furthermore, the signal can get from the TX to the RX by being reflected at or diffracted by different Interacting Objects (IOs) in the environment: houses, mountains (for outdoor environments), windows, walls, etc. The number of these possible propagation paths is very large. As shown in Figure 2.5, each of the paths has a distinct amplitude, delay (runtime of the signal), direction of departure from the TX, and direction of arrival; most importantly, the components have different phase shifts with respect to each other. In the following, we will discuss some implications of the multipath propagation for system design.

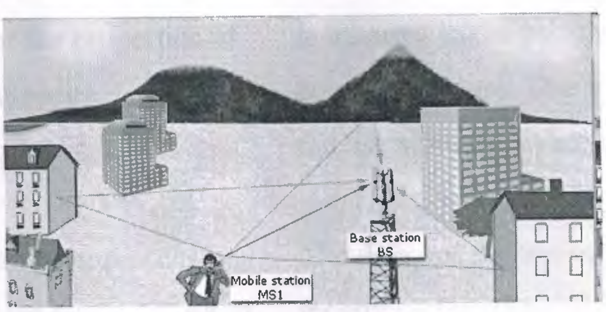

Figure 2.5: Multipath propagation

Wired communications	Wireless communications
The communication takes place over a more or less stable medium like copper wires or optical fibers. The properties of the medium are well-defined, and time-invariant.	Due to user mobility as well as multipath propagation, the transmission medium varies strongly with time.
Increasing the transmission capacity can be achieved by using a different frequency on an existing cable, and/or by stringing new cables.	Increasing the transmit capacity must be achieved by more sophisticated transceiver concepts and smaller cell sizes (in cellular systems), as the amount of available spectrum is limited.
The range over which communications can be performed without repeater stations is mostly limited by attenuation by the medium (and thus noise); for optical fibers, the distortion of transmitted pulses can also limit the speed of data transmission.	The range that can be covered is limited both by the transmission medium (attenuation, fading, and signal distortion) and by the requirements of spectral efficiency (cell size).
Interference and crosstalk from other users either do not happen, or the properties of the interference are stationary.	Interference and crosstalk from other users is inherent in the principle of cellular communications. Due to the mobility of the users, they also are time-variant.
The delay in the transmission process is also constant, determined by the length of the cable, and the group delay of possible repeater amplifiers.	The delay of the transmission depends mostly on the distance between base station and mobile station, and is thus time-variant.

The Bit Error Rate (BER) decreases strongly (approximately exponentially) with increasing Signal-to-Noise Ratio (SNR). This means that a relatively small increase in transmit power can greatly decrease the error rate.	For simple systems, the average BER decreases only slowly (linearly) with increasing average SNR. Increasing the transmit power usually does not lead to a significant reduction in BER. However, more sophisticated signal processing helps.
Due to the well-behaved transmission medium, the quality of wired transmission is generally high.	Due to the difficult medium, transmission quality is generally low unless special measures are used.
Jamming and interception of wired transmission is almost impossible without consent by the network operator.	Jamming a wireless link is straightforward, unless special measures are taken. Interception of the on-air signal is possible. Encryption is therefore necessary to prevent unauthorized use of the information.
Establishing a link is location-based. In other words, a link is established from one outlet to another, independent of which person is connected to the outlet.	Establishing a connection is based on the (mobile) equipment, usually associated with a specific person. The connection is not associated with a fixed location.
Power is either provided through the communications network itself (e.g., for traditional landline telephones), or from traditional power mains (e.g., fax). In neither case is energy consumption a major concern for the designer of the device.	Mobile stations use rechargeable or one-way batteries. Energy efficiency is thus a major concern.

2.2.2 Small-Scale Multipath Propagation

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

- Rapid changes in signal strength of 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 light 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 reflection from the ground and surrounding structures. The incoming radio waves arrive in different directions with different propagations delays. The signal received by the mobile at any point in space may consist of a large number of plane waves having a 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 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 a high speed can pass through several fades in a same period of time.

In a more serious case a receiver may stop at a particular application at which the receiver signal is in a deep fade. Maintaining good communication can then become very difficult, although passing vehicles or people walking in the vicinity of the mobile can often disturb the field pattern, there by diminishing the likelihood of the received signal remaining in a deep null for a long 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 the arrival of the received multipath waves.

2.2.2.1 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 scatters 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 the 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 enter symbol interference.

- Speed of the mobile

The relative motion between the base station and the mobile result is 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 the 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 maybe ignored, and only the speed of the mobile need to be considered. The coherence time defines the "staticsness" of the channel, and is directly impacted by 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 will be significant).

2.2.2.2 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 decreases, this is called Doppler Effect; this effect becomes important when developing mobile radio systems.

The amount of the frequency changes due the Doppler Effect depends on the relative motion between the source and the receiver, and on the speed of propagation of the wave.

Doppler shift can cause significant problems if the transmission technique is sensitive to carrier frequency of sets, or the relative speed is very high as is the case for low earth orbiting satellites.

2.3 Intersymbol interference

The runtimes for different MPCs are different. We have already mentioned above that this can lead to different phases of MPCs, which leads to interference in narrowband systems. In a system with large bandwidth, and thus good resolution in the time domain, the major consequence is signal dispersion: in other words, the impulse response of the channel is not a single delta pulse, but rather a sequence of pulses (corresponding to different MPCs), each of which has a distinct arrival time in addition to having a different amplitude and phase (see Fig. 2.6). This signal dispersion leads to intersymbol interference at the RX. MPCs with long runtimes, carrying information from bit k, and MPCs with short runtimes, carrying contributions from bit k + 1 arrive at the RX at the same time, and interfere with each other.

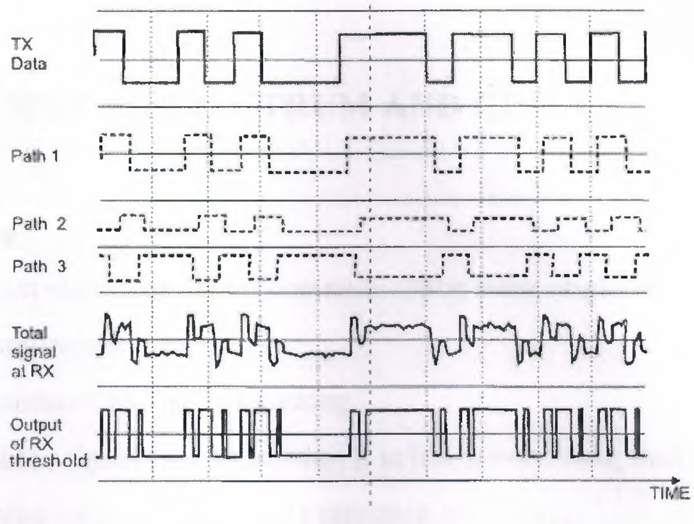

Figure 2.6: Intersymbol interference

Assuming that no special measures are taken, this Inter Symbol Interference (ISI) leads to errors that cannot be eliminated by simply increasing the transmit power, and are therefore often called irreducible errors.

ISI is essentially determined by the ratio between symbol duration and the duration of the impulse response of the channel. This implies that ISI is not only more important for higher data rates, but also for multiple access methods that lead to an increase in transmitted peak data rate.

Finally, it is also noteworthy that ISI can even play a role when the duration of the impulse response is shorter (but not much shorter) than bit duration.

CHAPTER THREE

SPREAD SPECTRUM AND CDMA

3.1 Spread Spectrum

Spread spectrum signal for digital communication were originally developed and used for military communication either

- To provide resistance to hostile jamming.
- To hide signal the signal by transmitting it at low power, thus, making it difficult for an unintended listener to detect its presence in noise or,
- To make it possible for multiple users to communicate through the same channel.

Today, however, spread spectrum signals are being used to provide reliable communication in a variety of commercial applications, including mobile vehicular communication and interoffice wireless communications

The basic element of a spread spectrum digital communication systems are illustrated in Figure 3.1. We observe that the channel encoder and decoder and the modulator and demodulator are the basic elements of conventional digital communication systems. In addition to these elements, a spread spectrum system employs two identical pseudorandom sequence generators, one of which interfaces with the modulator at the transmitting end and the second of which interfaces with the demodulator at the receiving end. These two generators produce a pseudorandom or pseudo noise (PN) binary valued sequence that is used to spread the transmitted signal in frequency at the modulator and to despread the received signal at the demodulator.

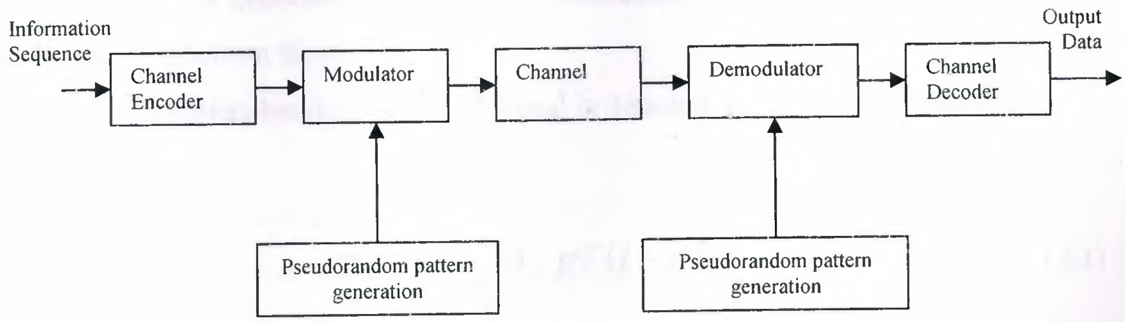

Figure 3.1: Model of spread spectrum digital communication system

Time synchronization of the PN sequence generated at the receiver with the PN sequence contained in the received signal is required to properly despread the received spread spectrum signal. In a practical system, synchronization is established prior to the transmission of information by transmitting a fixed PN bit pattern that is designed so that the receiver will detect it with high probability in the presence of interference. After time synchronization of the PN sequence generators is established, the transmission of information commences. In the data mode, the communication system usually tracks the timing of the incoming received signal and keeps the PN sequence generator in synchronism.

Two types of digital modulation are considered in conjunction with spread spectrum— namely, PSK and FSK. PSK modulation is generally used with DS spread spectrum and is appropriate for applications where phase coherence between the transmitted signal and the received signal can be maintained over a time interval that spans several symbol (or bit) intervals. On the other hand, FSK modulation is commonly used with FH spread spectrum and is appropriate in applications where phase coherence of the carrier cannot be maintained due to time variations in the transmission characteristics of the communications channel.

3.2 Direct-Sequence Spread Spectrum Systems

Let us consider the transmission of a binary information sequence by means of binary PSK. The information rate is R bits per second, and the bit interval is $T_b = l/R$ seconds. The available channel bandwidth is B_c hertz, where $B_c >> R$. At the modulator, the bandwidth of the information signal is expanded to $W = B_c$ hertz by shifting the phase of the carrier pseudo randomly at a rate of W times per second according to the pattern of the PN generator. The resulting modulated signal is called a direct-sequence (DS) spread spectrum signal.

The information-bearing baseband signal is denoted as $v(t)$ and is expressed as

$$v(t) = \sum_{n=-\infty}^{\infty} a_n \; gT(t - nT_b) \qquad (3.1)$$

where $[a_n = \pm 1, -\infty < n < \infty]$ and $gT(t)$ is a rectangular pulse of duration T_B,. This signal is multiplied by the signal from the PN sequence generator, which may be expressed as

$$c(t) = \sum_{n=-\infty}^{\infty} c_n p(t - nT_c) \qquad (3.2)$$

Where $[C_n]$ represents the binary PN code sequence of ±l's and p(t) is a rectangular pulse of duration T_c.

3.3 Two Application of DS Spread Spectrum Signals

In this subsection, we briefly describe the use of DS spread spectrum signals in two applications. First, we consider an application in which the signal is transmitted at very low power, so that a listener would encounter great difficulty in trying to detect the presence of the signal. A second application is multiple-access radio communications.

- **Low-Detect ability Signal Transmission**

In this application, the information-bearing signal is transmitted at a very low power level relative to the background channel noise and thermal noise that are generated in the front end of a receiver. If the DS spread spectrum signal occupies a bandwidth W and the power-spectral density of the additive noise is n_o watts/hertz, the average noise power in the bandwidth W is $p_n = WN_y$. The average received signal power at the intended receiver is P_R. If we wish to hide the presence of the signal from receivers that are in the vicinity of the intended receiver, the signal is transmitted at a power level such that $P_R/P_N << 1$. The intended receiver can recover the weak information-bearing signal from the background noise with the aid of the processing gain and the coding gain. However, any other receiver that has no knowledge of the PN code sequence is unable to take advantage of the processing gain and the coding gain. Consequently, the presence of the information-bearing signal is difficult to detect. We say that the transmitted signal has a low probability of being intercepted (LPI), and it is called an LPI signal.

- **Code Division Multiple Access**

The enhancement in performance obtained from a DS spread spectrum signal through the processing gain and the coding gain can be used to enable many DS spread spectrum signals to occupy the same channel bandwidth, provided that each signal has its own pseudorandom (signature) sequence. Thus, it is possible to have several users transmit messages simultaneously over the same channel bandwidth. This type of digital communication, in which each transmitter/receiver user pair has its own distinct signature code for transmitting over a common channel bandwidth, is called code division multiple access (CDMA).

In digital cellular communications, a base station transmits signals to N_u mobile receivers using N_u orthogonal PN sequences, one for each intended receiver. This Nu signals are perfectly synchronized at transmission so that they arrive at each mobile receiver in synchronism. Consequently, due to the orthogonality of the N_u PN sequences, each intended receiver can demodulate its own signal without interference from the other transmitted signals that share the same bandwidth. However, this type of synchronism cannot be maintained in the signals transmitted from the mobile transmitters to the base station (the uplink, or reverse link). In the demodulation of each DS spread spectrum signal at the base station, the signals from the other simultaneous users of the channel appear as additive interference. Let us determine the number of simultaneous signals that can be accommodated in a CDMA system. We assume that all signals have identical average powers at the base station. In many practical systems, the received signal's power level from each user is monitored at the base station, and power control is exercised over all simultaneous users by use of a control channel that instructs the users on whether to increase or decrease their power levels. With such power control, if there are N_u simultaneous users, the desired signal-to-noise interference power ratio at a given receiver is

$$\frac{P_s}{P_n} = \frac{P_s}{(N_u - 1)P_s} = \frac{1}{N_u - 1} \tag{3.3}$$

From this relation, we can determine the number of users that can be accommodated simultaneously. In determining the maximum number of simultaneous users of the channel, we implicitly assumed that the pseudorandom code sequences used by the various users are orthogonal and that the interference from other users adds on a power basis only. However, orthogonality of the pseudorandom sequences among the N_u users generally is difficult to achieve, especially if N_u is large.

3.4 Multiplexing

Multiplexing is not only a fundamental mechanism in communication systems but also in everyday life. Multiplexing describes how several users can share a medium with minimum or no interference. One example is highways with several lanes. Many users (car drivers) use the same medium (the highways) with hopefully no interference (i.e., accidents). This is possible due to the provision of several lanes (space division multiplexing) separating the traffic. In addition, different cars use the same medium (i.e., the same lane) at different points in time (time division multiplexing).

While this simple example illustrates our everyday use of multiplexing, the following examples will deal with the use of multiplexing in wireless communications.

3.4.1 Characteristics of Multiplexing

3.4.1.1 Space Division Multiplexing

For wireless communication, multiplexing can be carried out in four dimensions: space, time, frequency, and code. In this field, the task of multiplexing is to assign space, time, frequency, and code to each communication channel with a minimum of interference and a maximum of medium utilization. The term communication channel here only refers to an association of sender(s) and receiver(s) who want to exchange data.

Figure 3.2 shows six channels k_i, and introduces a three dimensional coordinate system. This system shows the dimensions of code c, time t and frequency f. For this first type of multiplexing, space division multiplexing (SDM), the (three dimensional) space s_i is also shown. Here space is represented via circles indicating the interference range. The channels k_1 to k_2 can be mapped onto the three 'spaces' s_1 to s_3 which clearly separate the channels and prevent the interference ranges from overlapping. The space between the interference ranges is sometimes called guard space. Such a guard space is needed in all four multiplexing schemes presented.

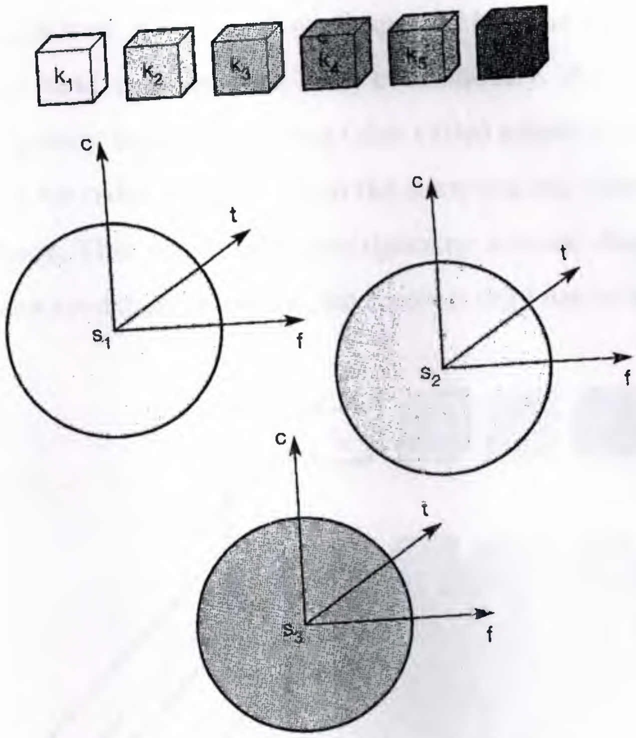

Figure 3.2: Space division multiplexing (SDM)

For the remaining channels (k_4 to k_6) three additional spaces would be needed. In our highway example this would imply that each driver had his or her own lane. Although this procedure clearly represents a waste of space, this is exactly the principle used by the old analog telephone system: each subscriber is given a separate pair of copper wires to the local exchange. In wireless transmission, SDM implies a separate sender for each communication channel with a wide enough distance between senders. This multiplexing scheme is used, for example, at FM radio stations where the transmission range is limited to a certain region many radio stations around the world can use the same frequency without interference. Using SDM, obvious problems arise if two or more channels were established within the same space, for example, if several radio stations want to broadcast in the same city. Then, one of the following multiplexing schemes must be used (frequency, time, or code division multiplexing).

3.4.1.2 Frequency Division multiplexing

Frequency division multiplexing (FDM) describes schemes to subdivide the frequency dimension into several non-overlapping frequency bands as shown in Figure 3.3. Each channel k, is now allotted its own frequency band as indicated. Senders using a certain frequency band can use this band continuously. Again, guard spaces are needed to avoid frequency band overlapping (also called adjacent channel interference). This scheme is used for radio stations within the same region, where each radio station has its own frequency. This very simple multiplexing scheme does not need complex coordination between sender and receiver: the receiver only has to tune in to the specific sender.

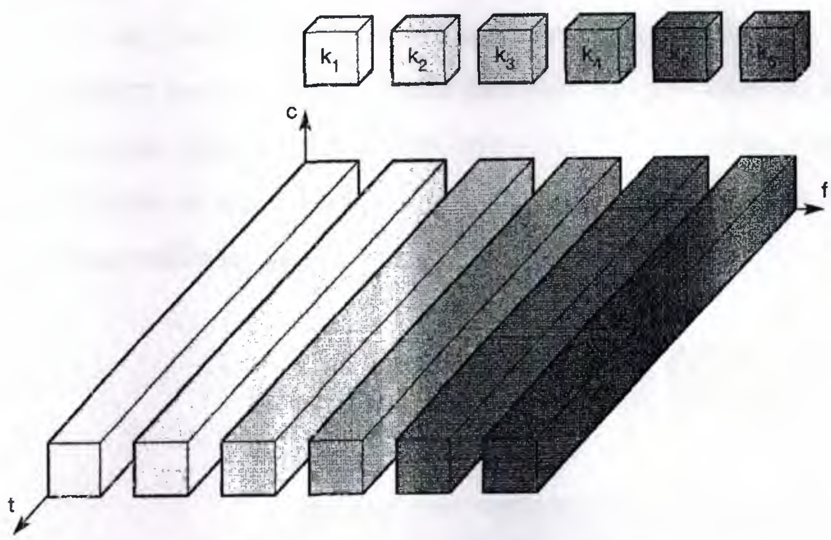

Figure 3.3: Frequency Division multiplexing (FDM)

However, this scheme also has disadvantages. While radio stations broadcast 24 hours a day, mobile communication typically takes place for only a few minutes at a time. Assigning a separate frequency for each possible communication scenario would be a tremendous waste of (scarce) frequency resources. Additionally, the fixed assignment of a frequency to a sender makes the scheme very inflexible and limits the number of senders.

3.4.1.3 Time Division Multiplexing

A more flexible multiplexing scheme for typical mobile communications is time division multiplexing (TDM). Here a channel k, is given the whole bandwidth for a certain amount of time, i.e., all senders use the same frequency but at different points in time. Again, guard spaces, which now represent time gaps, have to separate the different periods when the senders use the medium. In our highway example, this would refer to the gap between two cars. If two transmissions overlap in time, this is called co-channel interference. (In the highway example, interference between two cars results in an accident.) To avoid this type of interference, precise synchronization between different senders is necessary. This is clearly a disadvantage, as all senders need precise clocks or, alternatively, a way has to be found to distribute a synchronization signal to all senders. For a receiver tuning in to a sender this does not just involve adjusting the frequency, but involves listening at exactly the right point in time. However, this scheme is quite flexible as one can assign more sending time to senders with a heavy load and less to those with a light load.

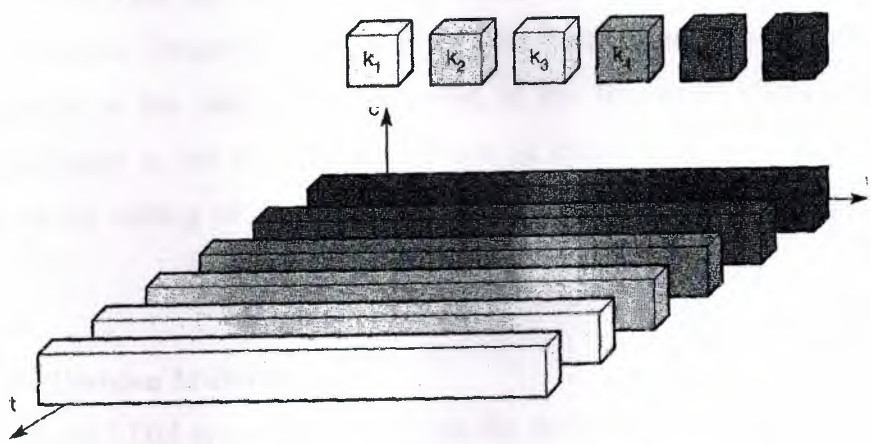

Figure 3.4: Time division multiplexing (TDM)

Frequency and time division multiplexing can be combined, i.e., a channel k_i can use a certain frequency band for a certain amount of time as shown in Figure 3.5. Now guard spaces are needed both in the time and in the frequency dimension. This scheme is more robust against frequency selective interference, i.e., interference in a certain small frequency band. A channel may use this band only for a short period of time. Additionally, this scheme provides some (weak) protection against tapping, as in this case the sequence of frequencies a sender uses has to be known to listen in to a channel.

The mobile phone standard GSM uses this combination of frequency and time division multiplexing for transmission between a mobile phone and a so-called base station.

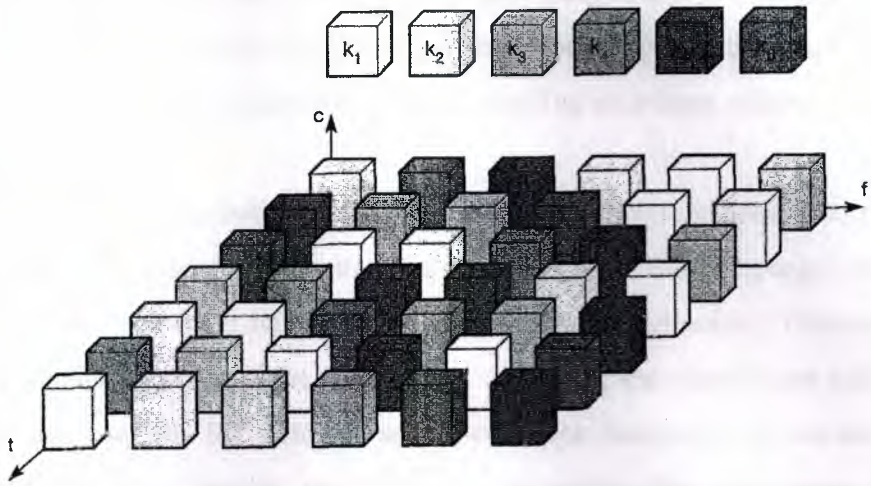

Figure 3.5: Frequency and time division multiplexing

A disadvantage of this scheme is again the necessary coordination between different senders. One has to control the sequence of frequencies and the time of changing to another frequency. Two senders will interfere as soon as they select the same frequency at the same time. However, if the frequency change (also called frequency hopping) is fast enough, the periods of interference may be so small that, depending on the coding of data into signals, a receiver can still recover the original data.

3.4.1.4 Code Division Multiplexing

SDM and FDM are well known from the early days of radio transmission and TDM is used in connection with many applications, code division multiplexing (CDM) is a relatively new scheme in commercial communication systems. First used in military applications due to its inherent security features, it now features in many civil wireless transmission scenarios thanks to the availability of cheap processing power. Figure 3.6 shows how all channels k, use the same frequency at the same time for transmission. Separation is now achieved by assigning each channel its own 'code', guard spaces are realized by using codes with the necessary 'distance' in code space, e.g., orthogonal codes.

The typical everyday example of CDM is a party with many participants from different countries around the world who establish communication channels, i.e., they talk to each other, using the same frequency range (approx. 300-6000 Hz depending on a person's voice) at the same time. If everybody speaks the same language, SDM is needed to be able to communicate (i.e., standing in groups, talking with limited transmit power).

But as soon as another code, i.e., another language, is used, one can tune in to this language and clearly separate communication in this language from all the other languages. (The other languages appear as background noise.) This explains why CDM has built-in security: if the language is unknown, the signals can still be received, but they are useless. By using a secret code (or language), a secure channel can be established in a 'hostile' environment. (At parties this may cause some confusion.) Guard spaces are also of importance in this illustrative example. Using, e.g., Swedish and Norwegian does not really work; the languages are too close. But Swedish and Finnish are 'orthogonal' enough to separate the communication channels.

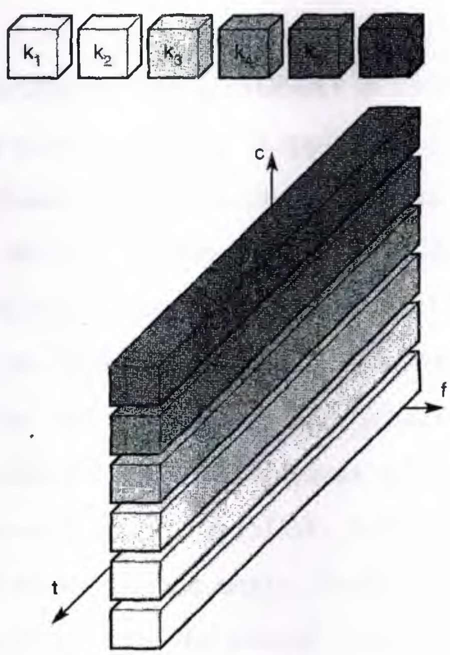

Figure 3.6: Code division multiplexing

The main advantage of CDM for wireless transmission is that it gives good protection against interference and tapping. Different codes have to be assigned, but code space is huge compared to the frequency space. Assigning individual codes to each sender does not usually cause problems. The main disadvantage of this scheme is the relatively high complexity of the receiver; a receiver has to know the code and must separate the channel with user data from the background noise composed of other signals and environmental noise. Additionally, a receiver must be precisely synchronized with the transmitter to apply the decoding correctly. The voice example also gives a hint to another problem of CDM receivers. All signals should reach a receiver with almost equal strength; otherwise some signals could drain others. If some people close to a receiver talk very loudly the language does not matter. The receiver cannot listen to any other person. To apply CDM, precise power control is required.

3.5 Multiple Access

The idea of multiple access is to permit earth station to transmit to the same satellite without interfering with other.

3.5.1 Classifications of Multiple Access

3.5.1.1 SDMA

Space Division Multiple Access (SDMA) is used for allocating a separated space to users in wireless networks. A typical application involves assigning an optimal base station to a mobile phone user. The mobile phone may receive several base stations with different quality. A MAC algorithm could now decide which base station is best, taking into account which frequencies (FDM), time slots (TDM) or code (CDM) are still available (depending on the technology). Typically, SDMA is never used in isolation but always in combination with one or more other schemes. The basis for the SDMA algorithm is formed by cells and sectorized antennas which constitute the infrastructure implementing space division multiplexing (SDM). A new application of SDMA comes up together with beam-forming antenna arrays. Single users are separated in space by individual beams. This can improve the overall capacity of a cell (e.g., measured in bit/s/m^2 or voice calls/m^2) tremendously.

3.5.1.2 FDMA

Frequency division multiple access (FDMA) comprises all algorithms allocating frequencies to transmission channels according to the frequency division multiplexing (FDM) scheme. Allocation can either be fixed (as for radio stations or the general planning and regulation of frequencies) or dynamic (i.e., demand driven).

Channels can be assigned to the same frequency at all times, i.e., pure FDMA, or change frequencies according to a certain pattern, i.e., FDMA combined with TDMA. The latter example is the common practice for many wireless systems to circumvent narrowband interference at certain frequencies, known as frequency hopping. Sender and receiver have to agree on a hopping pattern, otherwise the receiver could not tune to the right frequency. Hopping patterns are typically fixed, at least for a longer period. The fact that it is not possible to arbitrarily jump in the frequency space (i.e., the receiver must be able to tune to the right frequency) is one of the main differences between FDM schemes and TDM schemes.

Furthermore, FDM is often used for simultaneous access to the medium by base station and mobile station in cellular networks. Here the two partners typically establish a duplex channel, i.e., a channel that allows for simultaneous transmission in both directions. The two directions, mobile station to base station and vice versa are now separated using different frequencies. This scheme is then called frequency division duplex (FDD). Again, both partners have to know the frequencies in advance; they cannot just listen into the medium. The two frequencies are also known as uplink, i.e., from mobile station to base station or from ground control to satellite, and as downlink, i.e., from base station to mobile station or from satellite to ground control.

3.5.1.3 TDMA

Compared to FDMA, time division multiple access (TDMA) offers a much more flexible scheme, which comprises all technologies that allocate certain time slots for communication, i.e., controlling TDM. Now tuning in to a certain frequency is not necessary, i.e., the receiver can stay at the same frequency the whole time. Using only one frequency, and thus very simple receivers and transmitters, many different algorithms exist to control medium access. As already mentioned, listening to different frequencies at the same time is quite difficult, but listening to many channels separated in time at the same frequency is simple. Almost all MAC schemes for wired networks according to this principle, e.g., Ethernet, Token ring ATM etc. Now synchronization between sender and receiver has to be achieved in the time domain. Again this can be done by using a fixed pattern similar to FDMA techniques, i.e, allocating a certain time slot for a channel, or by using a dynamic allocation scheme. Dynamic allocation scheme require an identification for each transmission as this is the case for typical for wired MAC schemes, or the transmission has to be announced beforehand. MAC addresses are quite often used as identification. This enables a receiver in a broadcast medium to recognize if it really is the intend receiver message. Fixed schemes do not need identification, but are not as flexible considering varying bandwidth requirements.

3.5.1.4 CDMA

Finally, codes with certain characteristics can be applied to the transmission to enable the use of code division multiplexing (CDM). Code division multiple access (CDMA) systems use exactly these codes to separate different users in code space and to enable access to a shared medium without interference. The main problem is how to find "good" codes and how to separate the signal from noise generated by other signals and the environment.

A code for a certain user should have a good autocorrelation and should be orthogonal to other codes. Orthogonal in code space has the same meaning as in standard space (i.e., the three dimensional space). Think of a system of coordinates and vectors starting at the origin, i.e., in (0, 0, O).[3] Two vectors are called orthogonal if their inner product is 0, as is the case for the two vectors (2, 5, 0) and (0, 0, 17): (2, 5, 0)*(0, 0, 17) = 0 + 0 + 0 = 0. But also vectors like (3, -2, 4) and (-2, 3, 3) are orthogonal: (3, -2, 4)*(-2, 3, 3) = -6 - 6 + 12 = 0. By contrast, the vectors (1,2,3) and (4,2, -6) are not orthogonal (the inner product is -10), and (1, 2, 3) and (4, 2, -3) are "almost" orthogonal, with their inner product being -1 (which is "close" to zero). This description is not precise in a mathematical sense. However, it is useful to remember these simplified definitions when looking at the following examples where the original code sequences may be distorted due to noise. Orthogonality cannot be guaranteed for initially orthogonal codes.

Now let us translate this into code space and explain what we mean by a good autocorrelation. The Barker code (+1, -1, +1, +1, -1, +1, +1, +1, -1, -1, -1), for example, has a good autocorrelation, i.e., the inner product with itself is large, the result is 11.

This code is used for ISDN and IEEE 802.11. But as soon as this Barker code is shifted 1 chip further (think of shifting the 11 chip Barker code over itself concatenated several times), the correlation drops to an absolute value of 1. It stays at this low value until the code matches itself again perfectly. This helps, for example, to synchronize a receiver with the incoming data stream. The peak in the matching process helps the receiver to reconstruct the original data precisely, even if noise distorts the original signal up to a certain level.

After this quick introduction to orthogonality and autocorrelation, the following (theoretical) example explains the basic function of CDMA before it is applied to signals

- Two senders, A and B, want to send data. CDMA assigns the following unique and orthogonal key sequences: key A_k = 010011 for sender A, key B_k = 110101 for sender B. Sender A wants to send the bit $A_d = 1$, sender B sends $B_d = 0$. To illustrate this example, let us assume that we code a binary 0 as -1, a binary 1 as +1. We can then apply the standard addition and multiplication rules.
- Both senders spread their signal using their key as chipping sequence (the term 'spreading' here refers to the simple multiplication of the data bit with the whole chipping sequence). In reality, parts of a much longer chipping sequence are applied to single bits for spreading. Sender A then sends the signal $A_s = A_d*A_k = +1*(-1, +1, -1, -1, +1, +1) = (-1, +1, -1, -1, +1, +1)$. Sender B does the same with its data to spread the signal with the code:

  $B_s = B_d*B_k = -1*(+1, +1, -1, +1, -1, +1) = (-1, -1, +1, -1, +1, -1)$.
- Both signals are then transmitted at the same time using the same frequency, so, the signals superimpose in space (analog modulation is neglected in this example). Discounting interference from other senders and environmental noise from this simple example, and assuming that the signals have the same strength at the receiver, the following signal C is received at a receiver: $C = A_s + B_s = (-2, 0, 0, -2, +2, 0)$.
- The receiver now wants to receive data from sender A and, therefore, tunes in to the code of A, i.e., apply A's code for dispreading:

  $C*A_k = (-2, 0, 0, -2, +2, 0)*(-1, +1, -1, -1, +1, +1) = 2 + 0 + 0 + 2 + 2 + 0 = 6$.

  As the result is much larger than 0, the receiver detects a binary 1. Tuning in to sender B, i.e., applying B's code gives $C*B_k$ = (-2, 0, 0, -2, +2, 0)* (+1, +1, -1, +1, -1, +1) = -2 + 0 + 0 - 2 - 2 + 0 = -6. The result is negative, so a 0 has been detected.

This example involved several simplifications. The codes were extremely simple, but at least orthogonal. More importantly, noise was neglected. Noise would add to the transmitted signal C, the results would not be as even with -6 and +6, but would maybe be close to 0, making it harder to decide if this is still a valid 0 or 1. Additionally, both spread bits were precisely superimposed and both signals are equally strong when they reach the receiver. What would happen if, for example, B was much stronger? Assume that B's strength is five times A's strength. Then, $C' = A_s + 5*B_s$ = (-1, +1, -1, -1, +1, +1) + (-5, -5, +5, -5, +5, -5) = (-6, -4, +4, -6, +6, -4). Again, a receiver wants to receive B: $C'*B_k$ = -6-4-4-6-6-4= -30. It is easy to detect the binary 0 sent by B. Now the receiver wants to receive A: $C'*A_k$=6-4-4+6+6-4=6. Clearly, the (absolute) value for the much stronger signal is higher (30 compared to 6). While -30 might still be detected as 0, this is not so easy for the 6 because compared to 30, 6 is quite close to zero and could be interpreted as noise. Remember the party example. If one person speaks in one language very loudly, it is of no more use to have another language as orthogonal code - no one can understand you, your voice will only add to the noise. Although simplified, this example shows that power control is essential for CDMA systems. This is one of the biggest problems CDMA systems face as the power has to be adjusted over one thousand times per second in some systems - this consumes a lot of energy.

3.6 Comparison between SDMA, TDMA, FDMA CDMA

To conclude the chapter, a comparison of the four basic multiple access versions is given in Table 3.1. The table shows the MAC schemes without combination with other schemes. However, in real systems, the MAC schemes always occur in combinations. A very typical combination is constituted by SDMA/TDMA/FDMA as used in IS-54, GSM, DECT, PHS, and PACS phone systems, or the Iridium and ICO satellite systems. CDMA together with SDMA is used in the IS-95 mobile phone system and the Global star satellite system

Although many network providers and manufacturers have lowered their expectations regarding the performance of CDMA compared to the early 1980s (due to experiences with the IS-95 mobile phone system) CDMA is integrated into almost all third generation mobile phone systems either as W-CDMA (FOMA, UMTS) or cdma2000. CDMA can be used in combination with FDMA/TDMA access schemes to increase the capacity of a cell. In contrast to other schemes, CDMA has the advantage of a soft handover and soft capacity. Handover, describes the switching from one cell to another, i.e., changing the base station that a mobile station is connected to. Soft handover means that a mobile station can smoothly switch cells. This is achieved by communicating with two base stations at the same time. CDMA does this using the same code and the receiver even benefits from both signals. TDMA/FDMA systems perform a hard handover, i.e., they switch base station and hopping sequences (time/frequency) precisely at the moment of handover. Handover decision is based on the signal strength, and oscillations between base stations are possible.

Soft capacity in CDMA systems describes the fact that CDMA systems can add more and more users to a cell, i.e., there is no hard limit. For TDMA/FDMA systems, a hard upper limit exists - if no more free time/frequency slots are available, the system rejects new users. If a new user is added to a CDMA cell, the noise level rises and the cell shrinks, but the user can still communicate. However, the shrinking of a cell can cause problems, as other users could now drop out of it. Cell planning is more difficult in CDMA systems compared to the more fixed TDMA/FDMA schemes.

Approach	SDMA	TDMA	FDMA	CDMA
Idea	Segment space Into cells/sectors	Segment sending time into disjoint time-slots, demand driven or fixed patterns	Segment the frequency band into disjoint sub-bands	Spread the spectrum using orthogonal codes
Terminals	Only one terminal can be active in one cell/one sector	All terminals are active for short periods of time on the same frequency	Every terminal has its own frequency, uninterrupted	AH terminals can be active at the same place at the same moment, uninterrupted

Signal separation	Cell structure directed antennas	Synchronization in the time domain	Filtering in the frequency domain	Code plus special receivers
Advantages	Very simple, increases capacity per km^2	Established, fully digital, very flexible	Simple, established, robust	Flexible, less planning needed, soft handover
Disadvantages	Inflexible, antennas typically fixed	Guard space needed (multi-path propagation), synchronization difficult	Inflexible, frequencies are a scarce resource	Complex receivers, needs more complicated power control for senders
Comment	Only in combination with TDMA, FDMA or GDMA useful	Standard in fixed networks, together with FDMA/SDMA used in many mobile networks	Typically combined with TDMA (frequency hopping patterns) and SDMA (frequency reuse)	Used in many 3G systems, higher complexity, lowered expectations; integrated with TDMA/FDMA

CHAPTER FOUR

RESULTS

4.1 Additive White Gaussian Noise (AWGN) and Channel Interference

4.1.1 Simulation of Transmitted Data and Varying Amplitudes With AWGN Noise and Channel Interference

In Figure 4.1 the performance of the transmitted data with the error probability versus the SNR (Signal to noise ratio) shows that the perfect state is when the amplitude is equal to zero.

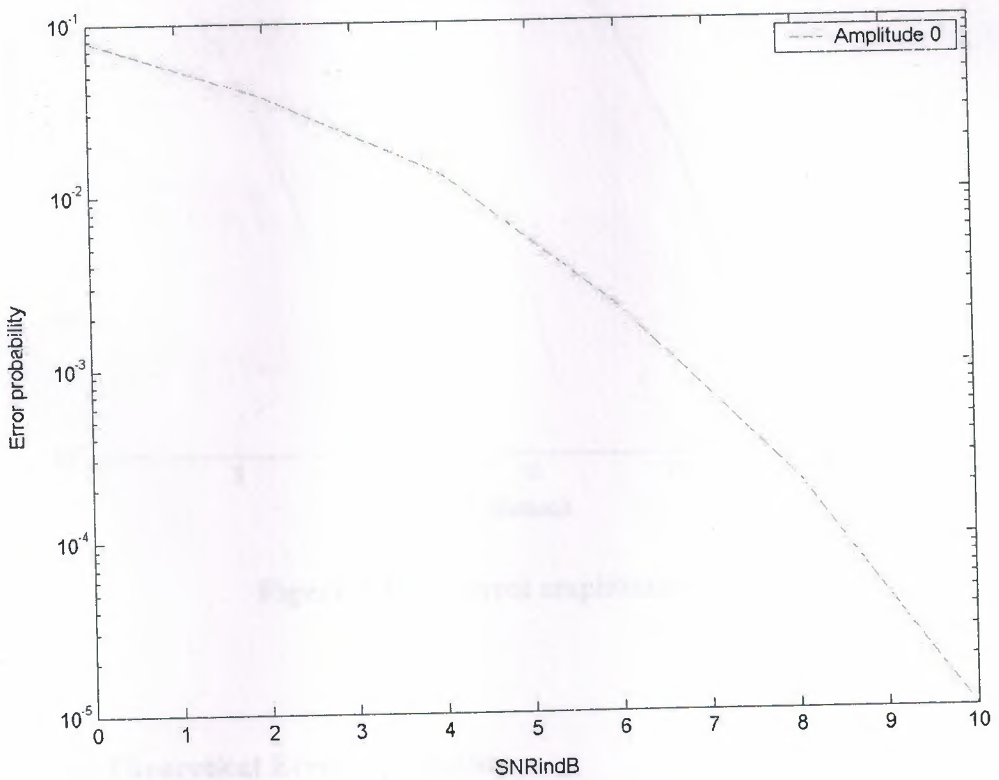

Figure 4.1: The perfect performance of the transmitted data

In Figure 4.2 we can see the multi amplitudes and the differences between the variations that means whenever the SNR (Signal to noise ratio) increases the probability of error increases slightly.

Here we have three different amplitudes (0, 3, 7 and 12), after investigating these multiple amplitudes and running the simulation in Appendix (step 1 and 2) we obtain these four slopes.

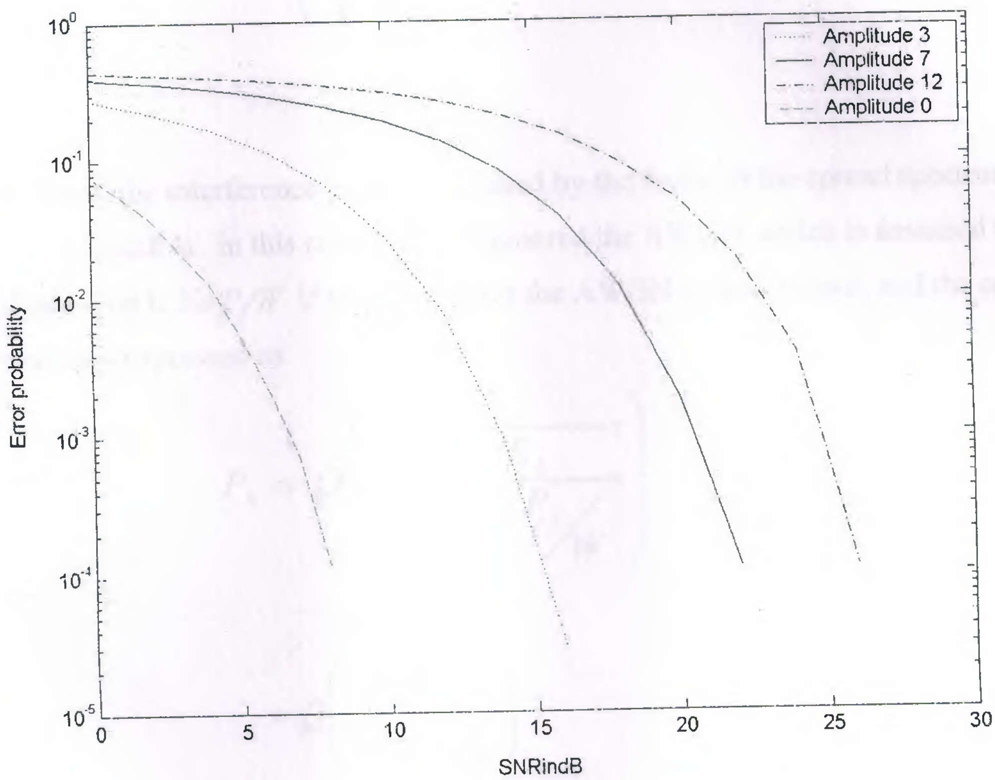

Figure 4.2: Different amplitudes slopes

4.1.2 The Theoretical Error Probability

In an AWGN channel, the probability of error for a DS spread spectrum system employing binary PSK (Phase shift keying) is identical to the probability of error conventional binary PSK

$$P_b = Q\left(\sqrt{\frac{2E_b}{N_0}}\right) \tag{4.1}$$

where P_b = Probability of error, E_b = Energy per bit

On the other hand, if the interference is sinusoidal signal with power P_j, the probability of error is (approximately)

$$P_b = Q\left(\sqrt{\frac{2E_b}{P_j/W}}\right) = Q\left(\sqrt{\frac{2E_b}{J_0}}\right) \tag{4.2}$$

Thus, the interference power is reduced by the factor of the spread spectrum signal bandwidth W. in this case, we have ignored the AWGN, which is assumed to be negligible- that is *No P_j/W.* if we account for the AWGN in the channel, and the error probability is expressed as

$$P_b = Q\left(\sqrt{\frac{2E_b}{N_0 + P_j/W}}\right)$$

(4.3)

$$= Q\left(\sqrt{\frac{2E_b}{N_0 + J_0}}\right)$$

4.2 Additive Noise and Interference and Pulse Jamming

4.2.1 Simulation of Transmitted Data With Respect to Different Rho's With Interference and Noise and Pulse Jamming

In this simulation we investigate about changing different ρ's (duty cycle) so we can obtain the differences between the pulse jammer and the main signal which has been send.

In Figure 4.3 we took $\rho = 0.3$ to see what will happen to the error of probability versus the SIR (Signal to interference ratio) between the theoretical and the simulation value, and as we can see they are approximately the same.

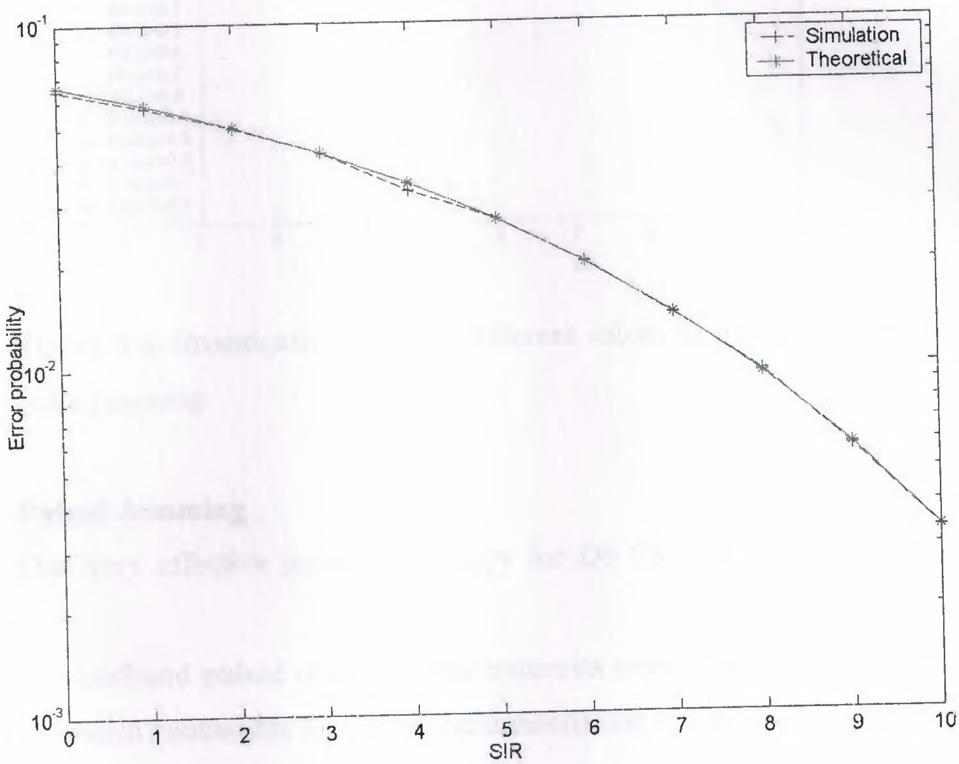

Figure 4.3: The variance between the theoretical and simulation ρ slopes

In figure 4.4 we compared multiple ρ's together between the simulation slopes values and the theoretical slopes value and we can see the effect of the duty cycle on it.

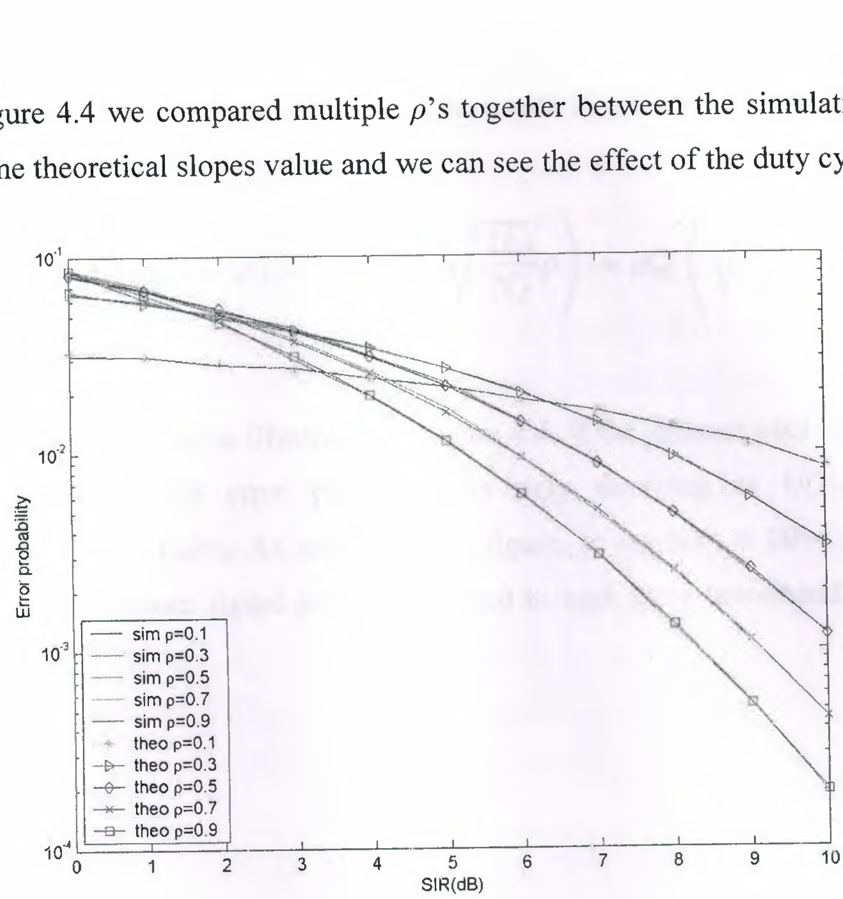

Figure 4.4: Investigation between different values of ρ due to the duty cycle for pulse jamming

4.2.2 Pulsed Jamming

One very effective jamming strategy for DS-SS is a broadband pulsed noise jammer.

A broadband pulsed noise jammer transmits noise whose power is spread over the entire system bandwidth. However, the transmission is only on for a fraction ρ of the time (i.e., ρ is the duty cycle of the jammer transmission and $0 < \rho \leq 1$). This allows the jammer to transmit with a power of J/ρ when it is transmitting (remember that J is the average received jammer power), and the equivalent spectral height of the noise is $N_J/2\rho$.

To make a simple analysis of the impact of a pulsed jammer we start by assuming that the jammer affects an integer number of information bits. That is, during the transmission of a certain information bit, the jammer is either on (with probability ρ) or off (with probability $1 - \rho$). Furthermore, if we assume that the jammer waveform is Gaussian noise and ignores all other noise and interference.

The bit error probability for a DS-SS system with BPSK modulation is

$$P_b = (1-\rho)\times 0 + \rho Q\left(\sqrt{\frac{2E_b}{N_J}\rho}\right) = \rho Q\left(\sqrt{\frac{2E_b}{N_J}\rho}\right) \tag{4.4}$$

This situation is illustrated in Figure 4.4. If the jammer uses the worst-case duty cycle, then the bit error probability is only decaying as $1/(E_b/N_J)$ rather than exponentially in E_b/N_J. As seen from the figure, to reach $P_b = 10{-}4$ we have to spend almost 21 dB more signal power compared to case for a broadband continuous noise jammer.

CONCLUSION

In this project we saw how fading and interference effects the transmitted data which can make the receiver has errors in it, and how the signal can be jammed.

After that we saw the CDMA and how it can make users use the same channel easily with less interference. And make them capable to have more security via the channel that made from the code that only the receiver have and only known by the receiver.

The simulation shows how we can avoid the channel interference and the additive noise by varying the amplitudes and using different ρ's due to the duty cycle.

In the first and the second result we saw if we increases the amplitude the performance of the sending data will have more errors, in the third and the forth we saw that the theoretical and simulation result are approximately the same.

REFERENCES

[1] Proakis, J.G, Salehi, M, Communication systems engineering, Prentice hall, Upper saddle river, New Jersey 07458, 2002.

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

[3] Rappaport, T.S., Wireless Communication, Prentice Hall PTR, Upper saddle river, NJ 07458, 2002 .

[4] Proakis, J.G, Digital Communication, McGraw-Hill, New York, 2000.

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

APPENDIX A

MATLAB CODES DESCRIPTION

APPENDIX A.1

Simulation of transmitted data and varying amplitudes with AWGN noise and channel interference

Step 1

```
function [p]=ss_Pe94(snr_in_dB, Lc, A, w0)
     %[p]=ss_Pe94(snr_in_dB, Lc, A, w0)
     %SS_PE94 finds the measured error rate. The function
     %that returns the measured probability of error for
     %the given value of
     %the snr_in_dB, Lc, A and w0.
snr=10^(snr_in_dB/10);
sgma=1;
     %Noise standard deviation is fixed.
Eb=2*sgma^2*snr;
     %signal level required to achieve the given

     %signal-to-noise ratio
E_chip=Eb/Lc;
     %energy per chip
N=10000;
     %number of bits transmitted
     %The generation of the data, noise, interference,
     %decoding process and error
     %counting is performed all together in order to
     %decrease the run time of the
     %program. This is accomplished by avoiding very large
     %sized vectors.
num_of_err=0;
for i=1:N,
     %Generate the next data bit.
```

```
temp=rand;
if (temp<0.5),
  data=-1;
else
  data=1;
end;
    %Repeat it Lc times, i.e. divide it into chips.
for j=1:Lc,
  repeated_data(j)=data;
end;
    %pn sequence for the duration of the bit is generated
    %next
for j=1:Lc,
  temp=rand;
  if (temp<0.5),
    pn_seq(j)=-1;
  else
    pn_seq(j)=1;
  end;
end;
    %the transmitted signal is
trans_sig=sqrt(E_chip)*repeated_data.*pn_seq;
    %AWGN with variance sgma^2
noise=sgma*randn(1,Lc);
    %interference
n=(i-1)*Lc+1:i*Lc;
interference=A*sin(w0*n);
    %received signal
rec_sig=trans_sig+noise+interference;
    %Determine the decision variable from the received
    %signal.
temp=rec_sig.*pn_seq;
decision_variable=sum(temp);
```

```
    %making decision
 if (decision_variable<0),
   decision=-1;
 else
   decision=1;
 end;
    %If it is an error, increment the error counter.
 if (decision~=data),
   num_of_err=num_of_err+1;
 end;
end;
    %then the measured error probability is
p=num_of_err/N;
```

```
Step 2
echo on
Lc=20;
    %number of chips per bit
A1=3;
    %amplitude of the first sinusoidal interference
A2=7;
    %amplitude of the second sinusoidal interference
A3=12;
    %amplitude of the third sinusoidal interference
A4=0;
    %fourth case: no interference
w0=1;
    %frequency of the sinusoidal interference in radians
SNRindB=0:2:30;
for i=1:length(SNRindB),
     %measured error rates
 smld_err_prb1(i)=ss_Pe94(SNRindB(i),Lc,A1,w0);
```

```
 smld_err_prb2(i)=ss_Pe94(SNRindB(i),Lc,A2,w0);
 smld_err_prb3(i)=ss_Pe94(SNRindB(i),Lc,A3,w0);
 echo off ;
end;
echo on ;
SNRindB4=0:1:8;
for i=1:length(SNRindB4),
     %measured error rate when there is no interference
 smld_err_prb4(i)=ss_Pe94(SNRindB4(i),Lc,A4,w0);
 echo off ;
end;
echo on ;
semilogy(SNRindB,smld_err_prb1,'b:',SNRindB,smld_err_prb2,'
y-',...
    SNRindB,smld_err_prb3,'r-.',SNRindB4,smld_err_prb4,'g--
');
legend
('Amplitude3','Amplitude7','Amplitude12','Amplitude0')
xlabel('SNRindB');
ylabel('Error probability');
```

APPENDIX A.2

Simulation of transmitting data for different rho's with interference and noise and pulse jamming

Step 1

```
function
[p,q]=pulsed_interference(sir_in_dB,snr_in_dB,Lc,rho)

sir=10^(sir_in_dB/10);
snr=10^(snr_in_dB/10);

sgma_int=1;
     %Noise standard deviation is fixed.
Eb=2*sgma_int^2*sir*rho;
     %signal level required to achieve the given
     %signal-to-noise ratio

sgma_noise=sqrt(Eb/(2*snr));

E_chip=Eb/Lc;
     %energy per chip

     %The generation of the data, noise, interference,
     %decoding process and error
     %counting is performed all together in order to
     %decrease the run time of the
     %program. This is accomplished by avoiding very large
     %sized vectors.
num_of_err=0;
num_of_trans=0;

while (num_of_err<2000)
     %Generate the next data bit.
  temp=rand;
```

```
  if (temp<0.5),
    data=-1;
  else
    data=1;
  end;
    %Repeat it Lc times, i.e. divide it into chips.
  for j=1:Lc,
    repeated_data(j)=data;
  end;
    %pn sequence for the duration of the bit is generated
next
  for j=1:Lc,
    temp=rand;
    if (temp<0.5),
      pn_seq(j)=-1;
    else
      pn_seq(j)=1;
    end;
  end;

    %the transmitted signal is
  trans_sig=sqrt(E_chip)*repeated_data.*pn_seq;

  num_of_trans=num_of_trans+1;

    %AWGN with variance sgma^2
  noise=sgma_noise*randn(1,Lc);

    %pulsed interference
  interference=(rand<rho)*sgma_int*randn(1,Lc);
```

```
    %received signal
  rec_sig=trans_sig+noise+interference;

    %Determine the decision variable from the received
    %signal.
  temp=rec_sig.*pn_seq;
  decision_variable=sum(temp);

    %making decision
  if (decision_variable<0),
    decision=-1;
  else
    decision=1;
  end;

    %If it is an error, increment the error counter.
  if (decision~=data),
    num_of_err=num_of_err+1;
  end;

end;

     %then the measured error probability is

sir_in_dB
num_of_err
num_of_trans

     %simulation
p=num_of_err/num_of_trans

     %theoretical
%q=rho*0.5*erfc(sqrt(sir*rho)) %no noise, only interference
```

```
q=rho*0.5*erfc(sqrt(rho*sir*snr/((rho*sir)+snr)))
%noise+interference



Step 2
clear
clc

Lc=20;
     %number of chips per bit
SIRindB=0:10;
SNRindB=13;

for i=1:length(SIRindB),
    j=1;
    for rho=0.1:0.2:1,


[err_prb_sim(i,j),err_prb_theo(i,j)]=pulsed_interference(SI
RindB(i),SNRindB,Lc,rho);
    j=j+1;
end;
end;
     %Plotting commands follow.
semilogy(SIRindB,err_prb_sim, SIRindB,err_prb_theo)
     %semilogy(SIRindB,err_prb_sim)
xlabel('SIR(dB)')
ylabel('Error probability')
legend('sim\rho=0.1','sim\rho=0.3','sim\rho=0.5','sim\rho=0
.7','sim\rho=0.9','theo\rho=0.1','theo\rho=0.3','theo\rho=0
.5','theo\rho=0.7','theo\rho=0.9',3)
```