



Budapest University of Technology and Economics
Faculty of Electrical Engineering and Informatics
Department of Measurement and Information Systems

İlker DİNDORUK

**PERFORMANCE EVALUATION AND
SIMULATION OF THE PHYSICAL LAYER OF
AN EMBEDDED COMMUNICATIONS SYSTEM
IN MATLAB**

SUPERVISOR

Asst. Lecturer Tamás Krébesz

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Summary

Nowadays, the wireless communication systems have become an important part of our everyday life. In wireless communications analogue signals used to carry information suffer from distortion as they propagate due to interference, noise, security breach, etc. One way to transmit digital data is to map the information into the properties of the built-in radio frequency waveforms used as carrier signals. Binary phase shift keying (BPSK) is one of those methods in digital communication. This modulation is accomplished by changing the sign of sinusoidal signals according to the information. It is widely used for wireless LANs, RFID and Bluetooth. The thesis focuses on the effect of noise on BPSK modulated signal and simulation of BPSK-based wireless digital communication in embedded systems.

The aim of this work is the simulation of a BPSK-based transmission through noisy channel including the generation and reception of signals, i.e., modulation and demodulation.

To summarize, I generated BPSK modulated signal and transmitted it into a channel characterized by Additive Gaussian White Noise (AWGN). Then, I applied different filtering methods to minimize AWGN and fed the signal into the receiver to detect and recover the binary information. Finally, based on results, I compared theoretical and simulated bit-error-rates of the simulated BPSK wireless digital communication system.

Összefoglalás

Manapság a vezeték nélküli kommunikációs rendszerek életünk minden területén fellelhetőek. A vezeték nélküli adatátviteli rendszerekben használt analóg jelek az adatátviteli csatornában az interferencia, zaj vagy biztonsági támadások miatt torzulnak. A vezeték nélküli adatok továbbításának egyik módja, hogy hasznos információt a nagyfrekvenciás vivő valamely jellemzőjébe képezik le. A bináris fázisbillentyűzés (BPSK) a digitális kommunikáció egyik ilyen módszere. Ez a moduláció a szinuszos jelek előjelébe képezi le az információt. Széles körben használják vezeték nélküli LAN-hoz, RFID-hez és Bluetooth-hoz. A dolgozat a zajnak a BPSK modulált jelre gyakorolt hatását vizsgálja egy Matlabban megépített szimulátor alkalmazásával.

A Diplomamunka célja a BPSK modulált jelek szimulátoron alapuló vizsgálata zajos csatornában, amely szimulátor implementálja az adót és a vevőt, azaz a modulációs és demodulációs algoritmust valamint az AWGN csatornát.

Összefoglalva, előállítottam a BPSK jelet és az additív Gauss-i fehér zajjal jellemezhető csatornába küldtem ki. Ezután különféle szűrési módszereket alkalmaztam az AWGN minimalizálására, és a vett jelből a vevő által detektáltam és visszaállítottam a bináris információt. Végül az eredmények alapján összehasonlítottam a BPSK vezeték nélküli adatátviteli rendszer elméleti teljesítőképességét a szimulációs eredményekkel.

1 INTRODUCTION

The main purpose of communication is to exchange information. The main components of a communication system seen in Figure 1 (whether analogue or digital) are the source of information; a transmitter that converts information into a suitable form for transmission that occurs in a channel, a receiver that performs the reverse function of the transmitter, and finally the recovery of the information. The source of information is usually people (in audio or picture format) or computers. Basic features that differentiate analogue and digital communication are that the information is continuous in amplitude as well as in time or it is binary and discrete in the latter.

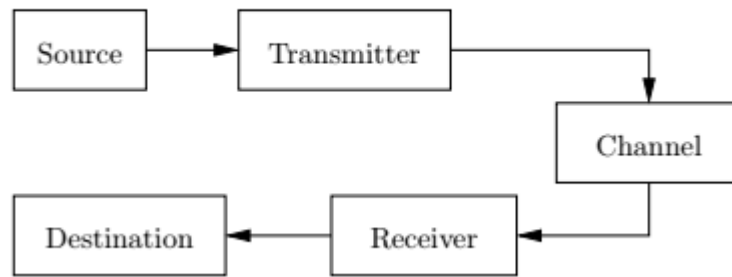


Figure 1: Basic block diagram of a communication system [1].

The main advantages of digital communication over analogue communication are listed below:

- Errors can be detected and corrected which makes this system more reliable.
- Error detection and correction are a unique feature for digital communication.

The benefit of a software-based application does not require emphasis. In addition to providing flexibility, it is much more reliable than hardware application. Complex signal processing techniques can be used to achieve optimum performance.

- Spread spectrum technique is used to prevent signal jamming.
- Other important features of digital communication include data compression and the ability to encrypt data. The compression process involves removing the excess in the data. This makes the transmission of data more efficient. Encrypting the data ensures that the intruder acquires an immune response against attack. This is useful in military applications.

Figure 2 shows the extended block diagram of a simple communication system used in our approach. Transmitter sends generated signal to communication channel where Additive

Gaussian Noise is added to pure form of the signal. Communication channel is the natural environment in which the information propagates. The signal arrives at the receiver to be demodulated.

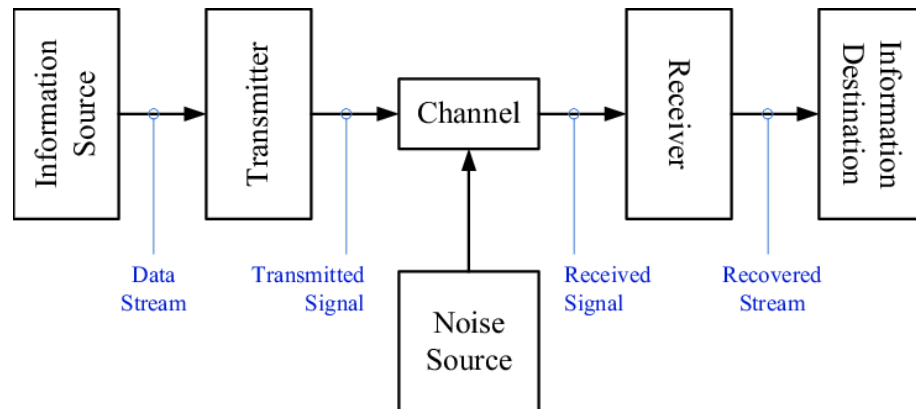


Figure 2: Block diagram of a simple digital communication system [2].

In a digital communication system, the logic information is mapped as the functional blocks of the transmitter and receiver starting from the far end of the channel with followings:

- source encoder-decoder
- channel encoder-decoder
- modulator – demodulator.

The sequence of analogue symbols generated by the modulator is called a waveform suitable for transmission over the channel. At the receiver, the channel output is processed by the transmitter, thereby creating a recoverable version of the original message signal. The recovered message signal is finally sent to user at the destination.

1.1 Advantages of Digital Communication

Design Efficiency: Digital is naturally more efficient than analogue when it comes to power transfer for bandwidth. For any performance required, there is a three-way balance between power, bandwidth and complexity [3].

Error Handling: Due to the channel coding technique in digital communication, it is possible to detect and correct the errors occurring during data transmission. The presence of noise as well as the different propagation characteristics of some communication channels make the correct handling of errors essential. Some of the ways in which errors are handled include error detection, error correction, data confirmation, and data repetition.

Typically, a system based on the combination of error detection and acknowledgment can perform well in a stable environment where data delays are not a problem. The main advantages of this type of system are the high data obtained in terms of data rate for simplicity and good SNR conditions. However, such a system does not work well in poor SNR conditions due to multiple retransmissions. Error detection and correction are often used to improve the typical BER of a mobile communication system. This includes the transmission of some additional data to ensure that errors are detected, identified and corrected.

New and improved services: In today's widespread lifestyle, internet services such as surfing, e-mailing, video games, texting, e-commerce and interactive multimedia services have become possible and even indispensable. It is also easier to integrate different services with different modalities into the same transmission scheme or to improve the services by transmitting some additional information. Researches in 5g help a lot to transfer data in fields that huge capacity of data transfer is needed.

Variety of Signal Processing Types: Computing data through Digital Signal Processing (DSP) provides variety of functionality. Digital Signal Processing deals with the representation of signals by a sequence of numbers or symbols and the processing of these signals. Digital signal processing is a branch of signal processing. Digital Signal Processing includes subfields such as audio and speech signal processing, sonar and radar signal processing, system control, image and video coding, sensor array processing, spectral estimation, statistical signal processing, digital image processing, signal processing for communication, biomedical signal processing, seismic data processing, etc.

DSP is also much more precise than the analogue ones. Settings are not changed with time and temperature changes, since they are only controlled by the DSP clock and the software, not the values of the caps or resistors, which change over time and temperature [4].

Quality Control: A desired level of distortion can be initially set and then kept virtually constant at every stage of the digital communication path. This reconstruction of the digital signal is performed by sufficiently spaced regenerative repeaters that do not allow the interference and accumulation of noise. On the other hand, if the analogue signal is distorted, the distortion cannot be eliminated and a repeater in the analogue system regenerates the distortion with the signal [3].

1.2 Why Digital Communications?

The mid-19th century had lots of indentations like telegraph which was the precursor of digital communication. Now, however, we can assert emphatically that digital is the ubiquitous technology of the twenty-first century and beyond, since the first generation of mobile phones was the last major analogue communications invention in the late 1970s. Over the last three decades, communications networks, systems and devices have all moved in the digital direction. It will continue to expose smart infrastructures and sophisticated end-user devices with many applications in the fields of digital communication technology such as entertainment, education, information, and business. Thus, the field of digital communication will continue to affect almost all aspects of our contemporary life.

1.3 Layered Structure of Digital Communications Systems

There are three layers which can influence the design of digital communication systems.

Physical layer: Physical mechanism is involved in the transmission of bits between any node pair in the network. The communication between the two nodes is achieved by modulation in the transmitter, transmission over the channel and demodulation in the receiver. This layer is in our focus. We worked in physical layer.

Data link layer: Communication channels are broken due to inevitable noise and interference. Therefore, one purpose of the data-link layer is to perform error correction or detection, even if this function is shared with the physical layer. Typically, the data link layer incorrectly resends received packets, but discards them for some applications. This layer is also responsible for the way different users share the transmission medium. A portion of the data link layer, called the middle access control (MAC) sublayer, is responsible for allowing frames to be sent over the shared transmission medium without unnecessary interference with other nodes. This feature is called multi-network communication.

Network layer: This layer has various functions to determine the orientation of information, to remove it from the source to its final destination. A second function is to determine the quality of service. A third function is flow control to ensure that the network is not blocked [5].

1.4 Radio Frequency

Radio spectrum refers to a portion of the electromagnetic spectrum corresponding to radio frequencies which are around 20 kHz to around 300 GHz as shown in Figure 3. This is approximately between the upper limit of the audio frequencies and the lower limit of the infrared frequencies. These are the frequencies at which energy can be emitted from an oscillating current as radio waves from a conductor into space. Different sources define different upper and lower limits established for the frequency range.

The energy of a radio frequency current can be radiated as electromagnetic radio waves from a conductor into space. This is the basis of radio technology. In radio communication, the signal is transmitted using an antenna that emits energy at a carrier frequency and is received by another antenna. The propagation characteristics of the electromagnetic waves used in radio channels strongly depend on the operating frequency. Under similar propagation conditions, the higher frequency signal is attenuated faster than the low frequency signal and becomes too weak to be detected at the receiver. An RF power amplifier is used to boost the power level of such a transmitted signal so that it can travel longer distances with less attenuation. The higher the operating frequency, the more directional the antenna beam becomes.

Radio frequency power transmission is the transmission of the output power of a transmitter to an antenna. If the antenna is not near the transmitter, special transmission lines are required.

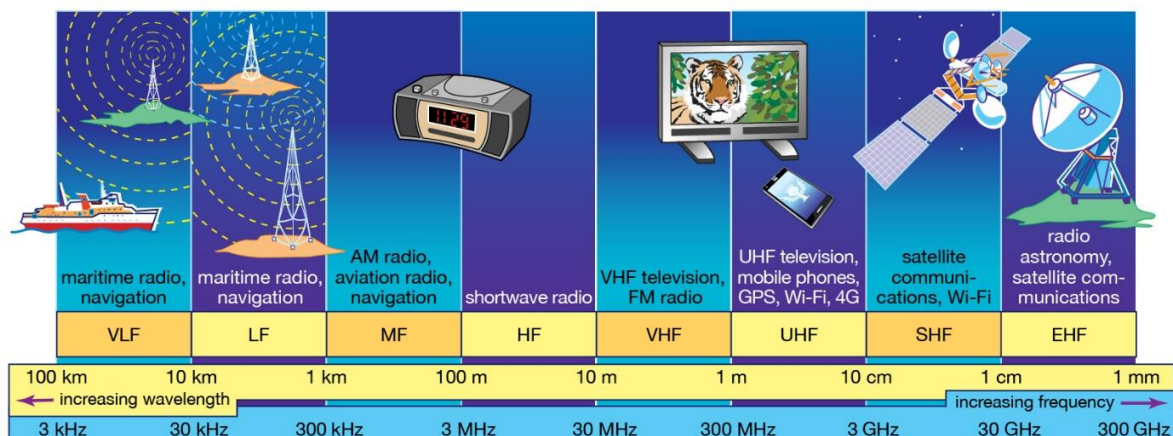


Figure 3: Commercially exploited bands of the radio frequency spectrum [6].

2 WIRELESS DIGITAL COMMUNICATION

A basic definition of digital is the transmission of a message using binary numbers (bits) or symbols from a finite alphabet over a finite time interval (bit or symbol duration). A bit or symbol that occurs in each interval is mapped to a continuous waveform that is transmitted along the channel. In each finite domain, the continuous time waveform at the channel output belongs to a finite set of possible waveforms. This contrasts with analogue communications where the output can take any possible waveform.

Wireless communication is sometimes referred to as mobile communication because most of new technical problems are caused by the motion of the transmitter or receiver. There are two major new wireless communication issues that do not occur with cables. The first is that the communication channel often varies with time. The second is that there are often interferences between multiple users. Modulation and coding techniques are ways to combat noise in communication channels. In wireless systems, these techniques must also deal with time-dependent variability and interference. This will result in major changes in both the channel modelling, the type of modulation and coding [7].

Over the past century, many types of wireless systems have evolved and often disappeared. For example, television broadcasts were broadcast in the early days by wireless radio transmitters, which were increasingly replaced by cable or satellite broadcasts. Similarly, point-to-point microwave circuits that previously formed the backbone of the telephone network are replaced by optical fiber. In the first example, wireless technology became obsolete when a wired distribution network was established; In the latter, a new wired technology replaced the old wireless technology. The purpose of these examples is that there are many situations in which to choose between wireless and wire technologies, and the choice changes by launching new technologies.

2.1 Distortion

Distortion in electronics is any change in a signal that alters the basic waveform or the relationship between the various frequency components; this is usually a degradation of the signal. In many practical cases, there are restrictions that force coding to be imperfect, thereby

causing inevitable distortion. For example, restrictions imposed by a communication channel may impose an upper limit on the allowed code rate, and therefore on the average code word length assigned to the information source. As another example, the information source may have a continuous amplitude, such as in the case of speech and the requirement is to measure the amplitude of each sample generated by the source to allow it to be represented by a finite length code word, such as in pulse code modulation. Distortion types on transmitted signal are shown in Figure 4.

In such cases, the problem is called source coding, which is the criterion of fidelity, and the branch of information theory dealing with it is called the rate-distortion theory. Rate distortion theory finds application in two different cases:

- Source coding in which the allowed coding alphabet does not fully represent the source of information; in this case, we must perform lossy data compression.
- Information is transmitted at a rate greater than the channel capacity.

Channels can be categorized as

- Distortionless: The impulse response can be modelled as a Dirac-delta function.
- Distorting: The impulse response cannot be modelled as a Dirac-delta function.

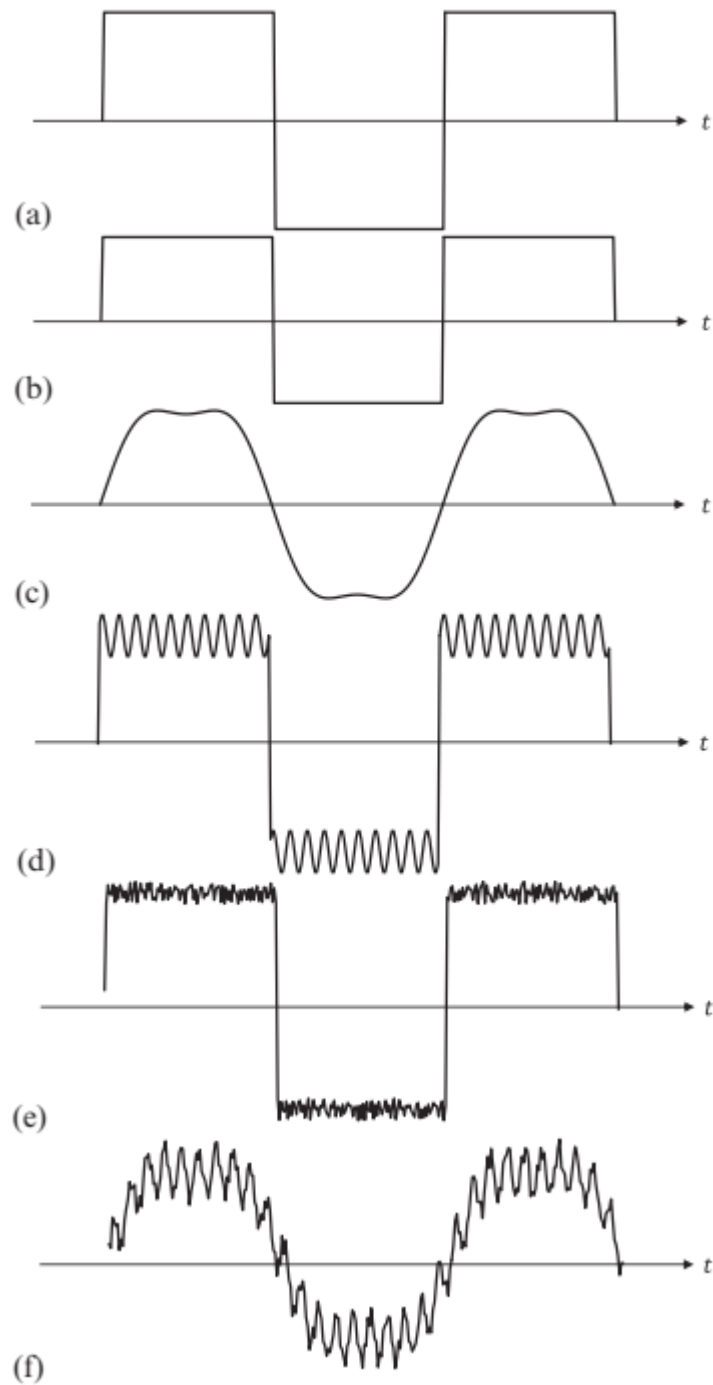


Figure 4: Transmission impairments: (a) transmitted signal; (b) effects of attenuation; (c) effects of distortion; (d) effects of interference; (e) effects of noise; and (f) aggregate effects of impairments [3].

2.1.1 Interference

Interference refers to the energy that appears in the receiver from sources outside its transmitter. The interference may be generated by other users of the same frequency or by devices that inadvertently transmit power outside of their band and into bands of adjacent channels or systems.

In most cases, there are various ways to virtually eliminate or significantly minimize radio interference. These include adequate filtering with stringent requirements, appropriately placed physical barriers, high directivity transmit and receive antennas, spatial separation between desired and interfering sources.

2.1.1.1 Intersymbol Interference

Intersymbol Interference (ISI) is a form of distorted signal in which one or more symbols interfere with subsequent signals, generating noise or transmitting a weak signal to output. The main reasons of ISI mostly are multi-path propagation and non-linear frequency in communication channels. Pulse shaping should include the transmit signal within the specified band while minimizing the possibility of receiver errors. On the other hand, pulse shaping causes Intersymbol Interference, which reduces the detection process and thus the overall system's error performance. The ISI is caused by the spread of the received pulses over adjacent time intervals. As a manifestation of channel distortion and defective filtering, the ISI can cause a significant number of errors if large enough. In digital transmission, designing pulses to eliminate ISI is thus the primary objective.

2.1.2 Channel Noise

Noise refers to unwanted, random waves that tend to interfere with the transmission and processing of signals in a communication system, thereby causing a corrupted version of the transmitted signal. It is generally assumed that noise is added to the signal. The filtering can be used to maximize the SNR at the receiver, but there is inevitably some noise that cannot be eliminated. Moreover, due to the central limit theorem, it can be assumed that there are several different noise sources end up in a Gaussian distribution. It is also assumed that the noise is white and therefore emits an equal amount of noise power per unit band at all frequencies. For this purpose, Additive White Gaussian Noise (AWGN) is the most common type of noise considered in digital communication systems and constitutes one of the most

basic system limitations. The effects of noise cannot be eliminated, but proper filtering can maximize the SNR value.

Communication noise influences the effectiveness of communication. It can have a profound impact on both our perception of our interactions with others and our ability to analyse our communication competence. Physical noise is any external stimulus that slightly or fully blocks us from receiving the intended message sent by a transmitter.

AWGN is generally used as a channel model where the only distortion in communication is a linear addition of a broadband or white noise with constant band spectrum density and a Gaussian amplitude distribution. The model is not responsible for fading, frequency selectivity, interference, non-linearity or distribution. However, it produces simple and traceable mathematical models that are useful to learn about the underlying behaviours of a system without considering other phenomena.

In modern communication systems, band limited AWGN cannot be ignored. While the band limited AWGN is being modelled in the phasor field, statistical analysis shows that the amplitude of the real and imaginary contributions are independent variables that follow the Gaussian distribution model. When combined, the magnitude of the resulting phase is a Rayleigh distributed random variable, evenly distributed between phases 0 to 2π .

In our approach, the noise added to signal is AWGN with zero mean and power spectral density $\frac{N_o}{2}$.

There is a fundamental difference among interference, ISI and channel noise, which can be summarized as follows:

- Channel noise is independent of the transmitted signal. The effect on data transmission over the band-limited channel is displayed at the receiver input when the data transmission system is switched on.
- ISI disappears when the transmitted signal is switched off.

2.2 Signal to Noise Ratio (SNR)

There are many different measures of signal power, noise power, energy per symbol, energy per bit, and so on. A criterion is needed to analyse the effects of noise on receiver performance and to perform a comparative analysis of the noise performance of different

modulation schemes. An intuitive measurement may be the signal-to-noise ratio, which is defined as the ratio of the mean power of the signal to the mean power of the noise.

Quadrature Amplitude Modulation (QAM) digital modulation method commonly used in modern telecommunications to transmit information, and a related family of analogue modulation techniques. Pulse Amplitude Modulation (PAM) is a form of signal modulation in which message information is encoded within the amplitude of a series of signal pulses.

Each QAM symbol U_k can be used to be with signal energy $E_s = [|E_k|^2]$. Signal power is given by $P = E_s W$. In the case of baseband PAM, there is a $2W$ degree of freedom per second, so the signal power is $P = 2E_s W$. This way the SNR will be;

$$\text{SNR} = \frac{P}{N_o W} \text{ for QAM and PAM} \quad (2.3.1)$$

PAM and QAM each use the same signal energy for each degree of freedom (or at least for each complex degree of freedom), while other systems may use the available degrees of freedom differently. For example, a PAM with bandwidth W occupies $2W$ bandwidth when modulated with bandwidth and uses only half of the available degrees of freedom. In these situations, SNR can be defined in different ways depending on the context. As another example, frequency hopping is a method used in both wireless and secure communications. This is the same as QAM, except that the carrier frequency f_c changes at random intervals over a period of time relative to the symbol interval. In this case, the bandwidth W can be considered as the bandwidth of the underlying QAM system, or as the general bandwidth within which f_c jumps [7].

The appearance of W in the denominator of the SNR expression (2.3.1) is at first surprising and disturbing. It says that if more bandwidth is allocated to a communication system with the same available power, then the SNR will decrease. This is best interpreted by observing the SNR to the extent that the signal-to-noise energy is considered by the degree of freedom. As the number of degrees of freedom increases per second, the SNR decreases, but the number of degrees of freedom available increases.

Another important parameter is the speed R ; this is the transmitted bit number per second, the bit number per symbol, $\log_2|A|$, times the number of symbols per second. Therefore;

$$R = W \log_2|A|, \text{ for QAM} \quad R = 2W \log_2|A|, \text{ for PAM} \quad (2.3.2)$$

An important parameter is the spectral efficiency of the system defined by $\rho = R / W$. This is the number of bits / secs transmitted in each frequency unit. Therefore;

$$\rho = W \log_2 |A|, \text{ for QAM} \quad \rho = 2W \log_2 |A|, \text{ for PAM} \quad (2.3.3)$$

One of the most basic quantities of communication is the $\frac{E_b}{N_o}$ ratio. E_b over N_o ratio is measured in the same way, so the ratio is dimensionless. Finding ways to reduce $\frac{E_b}{N_o}$ is especially important where transmitter uses batteries. Therefore, we can formulate that;

$$\frac{E_b}{N_o} = \frac{SNR}{\rho} \quad (2.3.3)$$

The denominator in (2.3.1) can be interpreted as the total noise power over the full bandwidth W , so SNR is also considered as the signal power divided by the noise power in the nominal band. For those who want to reduce the number of formulas in their memory, each of these SNR equations follow from the basic definition as signal energy per degree divided by noise energy per degree of freedom.

A key performance criterion for the accuracy of the received signal is the output SNR defined as the ratio of the mean power of the demodulated signal to the mean power of the noise measured at both the receiver output. The SNR depends on the modulation type used in the system and the type of demodulation technique used on the receiver. To make a fair comparison of all modulation schemes, they all need to work in the same environment. More specifically, in order to have a meaningful comparison, the transmitted signal strength and the noise power in the bandwidth of the signal must be the same for all modulation types.

The signal-to-noise ratio is calculated by taking the decibel level of the signal and subtracting the decibel level of the noise signal can be seen in Figure 5. The higher value of the signal-to-noise ratio, the better the transmission of information, because the larger decibel value of desired signal that it records or the less noise it receives.

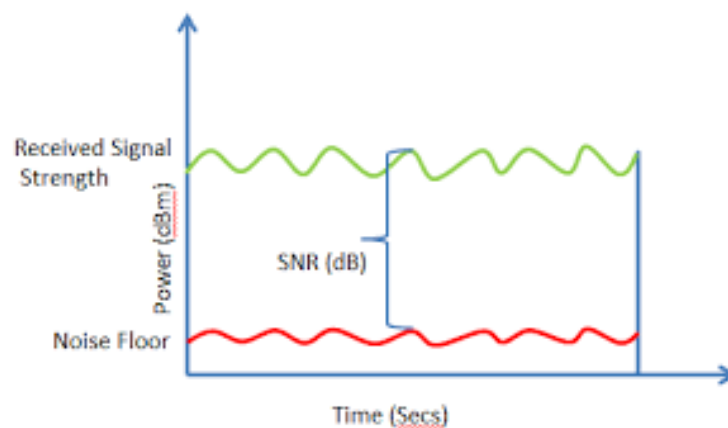


Figure 5: SNR value detection between signal and noise [8].

2.3 Digital Communication Design

In a communication system, there are two main sources: transmitted power and channel bandwidth. The transmitted power which consumes the battery is the average power of the transmitted signal. The channel bandwidth reflects the band of important frequency components allocated for the transmission of the input signal. One of these two sources may be more significant than the other. Therefore, communication channels are generally classified as band-limited or power-limited like in satellites. Each of these two main sources can have a significant impact on many system aspects.

Communication design is an interdisciplinary field between design and knowledge development. A communication design approach does not only refer to the development of the transmission, regardless of the aesthetics in the media, but also to the creation of new media channels to ensure that the message reaches the target clearly. Some designers use graphic design and communication design interchangeably because of overlapping capabilities. The goals of digital communication design are numerous, but not all are of equal importance since some of them are necessary, some are desirable.

3 BAND-PASS TRANSMISSION

In transmitting the baseband pulse, a data stream represented as a discrete pulse amplitude modulated signal is transmitted over a low pass channel.

In the case of digital transmission band transmission, the incoming data stream is modulated to a fixed frequency carrier and then transmitted over a band-pass channel.

In Figure 6, there is first a message source to generate symbols periodically with the symbol of m . Then sequence of symbols generated in message source are presented to a signal transmission encoder where the symbols are converted to corresponding set of N elements. Modulator by using set of N elements constructs a signal with time domain $s_i(t)$ which is transmitted every T second. Communication channel is where the original signal is distorted and becomes a noisy signal. We used AWGN to simulate this process. Receiver part has a detector to receive a signal and convert it to symbol [9].

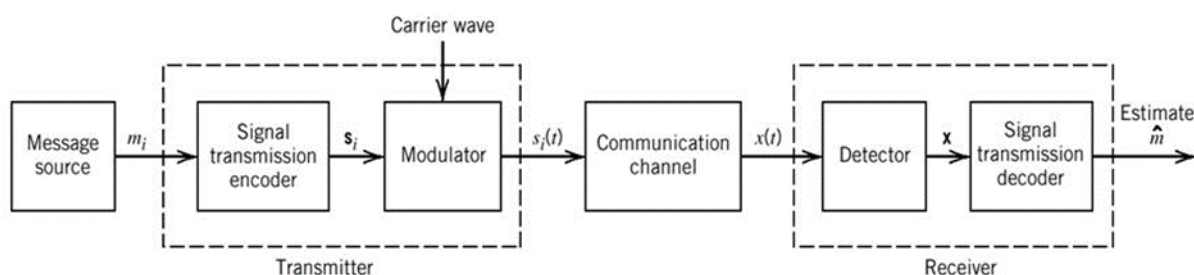


Figure 6: Block diagram of band-pass transmission system [5].

There are three basic signalling schemes:

Amplitude-shift keying (ASK)

Frequency-shift keying (FSK)

Phase-shift keying (PSK)

PSK and FSK are preferred over ASK signals for band-pass data transmission over non-linear channels such as microwave and satellite channels.

3.1 M-ary transmission

M-ary data transmission is a type of digital modulation in which two or more bits are transmitted simultaneously rather than one bit at a particular time. With binary signalling, each symbol represents one bit that is transmitted in every T_b seconds. For binary transmission, the symbol rate is equal to $\frac{1}{T_b}$ and according to Nyquist frequency, the minimum bandwidth requirement is $\frac{1}{2T_b}$.

Some outstanding features of the M-ary PSK:

- The envelope is set with more phase probabilities.
- More efficient than ASK and FSK.
- Minimum phase prediction error at the receiver.
- By increasing M, the bandwidth efficiency of the M-ary PSK decreases and system performance increases. M represents number of digits corresponding to number of conditions.

3.2 Binary Digital Modulation

In a binary digital modulation technique, a sinusoidal signal is transmitted during each individual bit interval. The amplitude, frequency or phase of the sine carrier is modified in accordance with the input bits to produce the modulated signal. There are, therefore, three types of binary digital modulation techniques: binary amplitude shift keying (BASK), binary frequency shift keying (BFSK), and binary phase shift keying (BPSK). Assuming that the average energy per bit is the same for these modulation schemes, the bit error rates are determined. Their waveforms are shown in Figure 7.

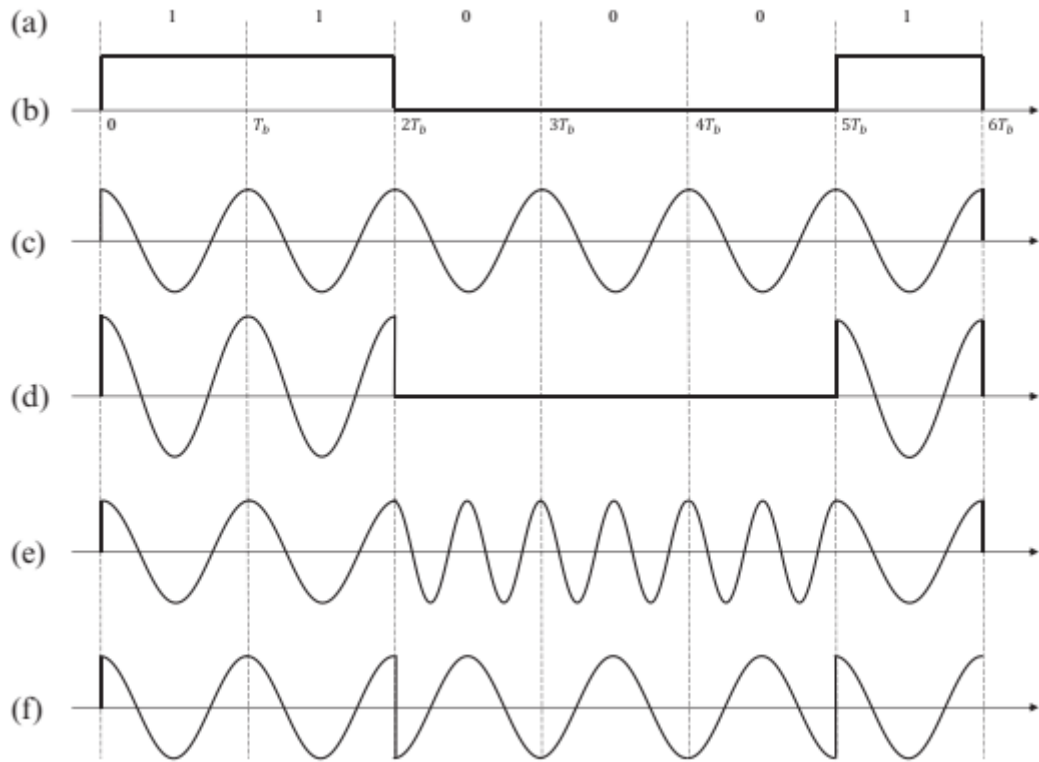


Figure 7: Binary digital modulation schemes: (a) binary data, (b) modulating signal, (c) carrier wave, (d) BASK signal, (e) BFSK signal, and (f) BPSK signal [3].

3.3 Amplitude-Shift Keying

Amplitude-shift keying (ASK) is a form of amplitude modulated wave representing digital data as variations in the amplitude of a carrier wave. In an ASK system, the binary symbol 1 is represented by a constant amplitude carrier wave and the constant frequency being transmitted for a period of T seconds. If the signal value is 1, the carrier signal will be transmitted. Otherwise, the signal value 0 will be transmitted.

Any digital modulation scheme uses a limited number of different signals to represent digital data. The ASK uses a limited number of amplitudes, each assigned a unique binary digit pattern. Each bit pattern creates the symbol represented by a given amplitude.

3.4 Frequency-Shift Keying

Frequency-shift keying (FSK) is a form of frequency modulated wave representing digital data as variations in the frequency of a carrier wave. FSK is the frequency modulation scheme.

In principle, the FSK can be implemented using independent oscillators and switching between them at the beginning of each symbol period. Generally, the independent oscillators will not be at the same phase and therefore will not have the same amplitude at the time of switching and will cause sudden discontinuities in the transmitted signal.

In digital communication, minimum shift switching (MSK) is a form of frequency modulation based on a system called continuous phase frequency shift switching. Desirably, the pulse shaping filter must meet the following three conditions:

- Frequency response with narrow bandwidth and sharp cut off;
- Pulse response with relatively low overshoot
- The evolution of a phase lattice, with the carrier phase of the modulated signal, assumes that the $\pm\pi/2$ at odd multiples of the bit duration T_b , and that the two values 0 and π are even at the multiples of T_b , as in the MSK.

The frequency-response condition is necessary to suppress the high frequency components of the modified frequency modulated signal. The impulse response condition prevents excessive deviations in the instantaneous frequency of the changed frequency modulated signal. Finally, the condition applied on phase-lattice evolution allows the modified frequency modulated signal to be detected in the same manner as the MSK signal, or if desired, can be consistently detected as a simple binary FSK signal.

3.5 Phase-Shift Keying (PSK)

PSK is the digital modulation technique in which the phase of the carrier signal is altered. The PSK technique is widely used for wireless LANs, biometrics, contactless processes, RFID and Bluetooth communication. The phase of the output signal varies depending on the input.

Phase-shift keying is widely used in many wireless communication systems. It is particularly suitable for the growing data communication area. PSK allows data to be transported more efficiently over a radio communication signal than frequency-shift keying, FSK and some other modulation formats.

There are various types for phase-shift keying. Each form has its own advantages and disadvantages, and the optimum format should be selected according to designed radio communication system. To make the right choice, one should have knowledge and

understanding of how the PSK works. These are mainly two types, namely Binary Phase Shift Keying (BPSK) and Quadrature Phase Shift Keying (QPSK), according to the number of phase shifts.

It is generally appropriate to show a phase shift keyed signal and sometimes other signal types using a phasor diagram. Using this scheme, the phase of the signal is represented by amplitude with the angle around the circle and the distance from the centre. With this way, the signal may be subdivided into quadratic components represent the cosine for the quadrature component.

Each phase shift switching form has its own advantages and disadvantages. In general, high modulation forms allow for higher data rates to be carried within a given bandwidth. Many radio communication systems can dynamically select the modulation pattern based on the current conditions and requirements.

The following list provides some of the more commonly used phase-shift keying formats, PSK, and the corresponding modulation formats used:

- PSK - Phase Shift Keying
- BPSK - Binary Phase Shift Keying
- QPSK - Quadrature Phase Shift Keying
- O-QPSK - Offset Quadrature Phase Shift Keying
- 8 PSK - 8 Point Phase Shift Keying
- 16 PSK - 16 Point Phase Shift Keying [7].

3.5.1 Binary Phase-Shift Keying

BPSK is called a 2-phase PSK or Phase Reversal switching. In this technique, the sine wave carrier receives two phase inverses, in the range of 0° and 180° .

The block diagram of the Binary phase shift keying shown in Figure 8 consists of a balance modulator including the carrier sine wave as an input and the binary array as the other input.

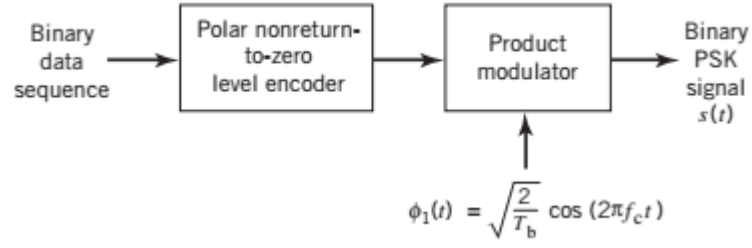


Figure 8: Block diagram of binary PSK transmitter [5].

To generate a binary PSK signal, we have to represent the input binary sequence in a polar form with symbols 1 and 0 represented by constant amplitude levels of $+\sqrt{E_b}$ and $-\sqrt{E_b}$, respectively. Where carrier frequency is $f_c = \frac{n}{T_b}$, E_b is transmitted energy per bit, n is number of integers, and T_b is bit duration.

$$s_1(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t) \quad \text{if } s_i = +\sqrt{E_b}$$

$$s_2(t) = -\sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t) \quad \text{if } s_i = -\sqrt{E_b}$$

3.6 Sampling Theorem

Sampling is usually defined in time domain. Therefore, it is a process that is essential for digital signal processing and digital communication. During the sampling process, an analogue signal is converted to a corresponding sample sequence, which is usually distributed over time as shown in Figure 9. For such a procedure, the sampling rate must be selected in proportion to the bandwidth of the message signal, so that the sequence of samples uniquely identifies the original analogue signal and Nyquist rule must be always met.

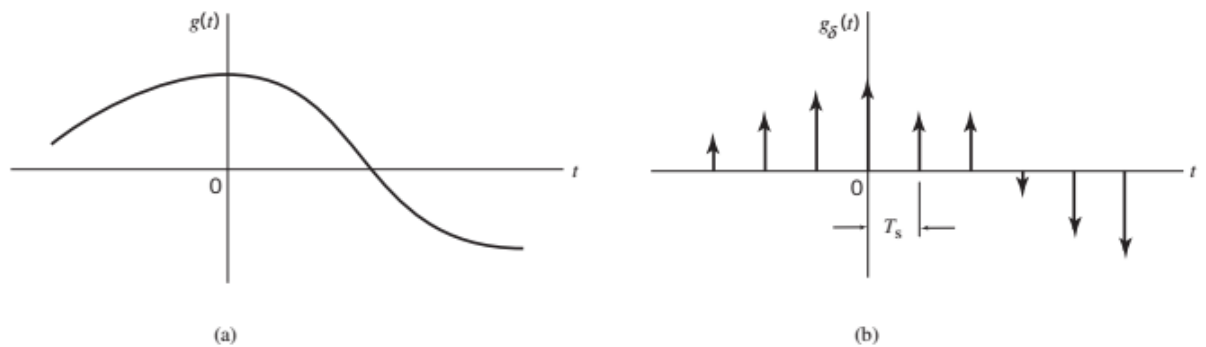


Figure 9: Sampling process. (a) Analogue signal. (b) The instantaneous sampling of the analogue signal [5].

Sampling is the process of converting a signal (a function of continuous time and/or space) into a numerical sequence (a function of a separate time and /or gap). Threshold $2B$ is called the Nyquist rate and the continuous time input to be sampled is a property of $x(t)$. B is the largest frequency of the signal to be sampled. The sample rate must be greater than the Nyquist rate. Threshold $f_s / 2$ is called Nyquist frequency and is an attribute of sampling equipment. All the meaningful frequency components of appropriately sampled $x(t)$ are found under the Nyquist frequency.

3.7 Pulse Shaping

Pulse shaping is the process of adjusting the waveform of transmitted pulses to the channel properties. Its purpose is to make the transmitted signal more appropriate the communication channel, basically by limiting the effective bandwidth of the transmission. By filtering the impulses that transmitted, the intersymbol interference caused by the channel can be controlled. In RF communication, pulse formation is essential for the signal to fit into the frequency band.

In communication systems, two important requirements of a wireless communication channel require the use of a pulse shaping filter. These conditions are limiting the signal bandwidth to be radiated and reducing interference (ISI) from multi-path signal reflections. Both requirements can be achieved with a pulse shaping filter applied to each symbol. ^[8]

Not every filter is suitable to be used as a pulse shaping filter. The filter itself must not cause intersymbol interference. The Nyquist ISI criterion is a commonly used criterion for evaluation because it correlates the frequency spectrum of the transmitter signal to intersymbol interference. The Nyquist channel cannot be physically performed because it has a rectangular bandwidth property and an infinite time delay. The detection process will be very sensitive to the small timing errors. Solution is Raised-Cosine Filter due to its ability to minimize intersymbol interference (ISI).

Examples of pulse shaping filters commonly used in communication systems:

- Raised-cosine filter
- Truncated Sinc shaped filter
- Butterworth filter

3.7.1 Sinc Shaped Filter

It is also called the Boxcar filter, because the frequency domain equivalent is a rectangular shape. Theoretically the best pulse shaping filter in the frequency domain is the sinc filter, but it cannot be implemented in real systems. It is also problematic in terms of synchronization because any phase error causes a sharp increase in intersymbol interference. Waveform of the normalized sinc function can be seen in Figure 10.

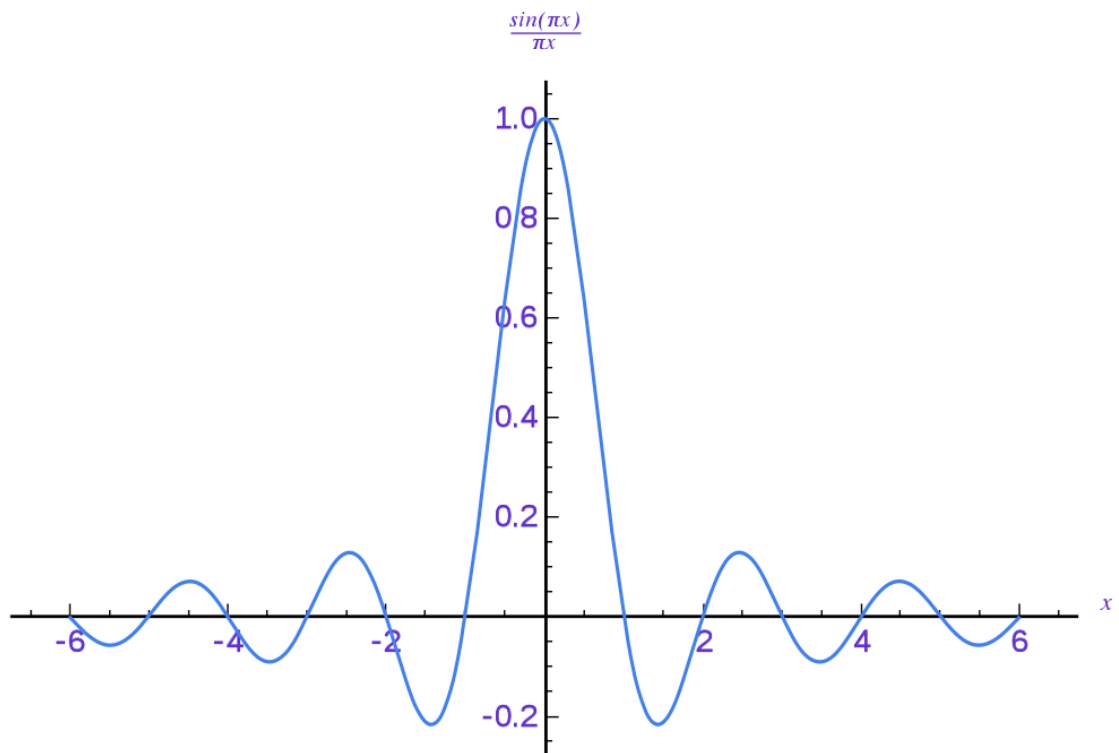


Figure 10: The normalized sinc function [10].

3.7.2 Raised-Cosine Filter

The raised-cosine filter is a filter commonly used for pulse shaping in digital modulation because of its ability to minimize intersymbol interference as shown in Figure 11. The equations that define the filter contain an α parameter known as a roll-off factor or excess bandwidth. α changes between 0 and 1.

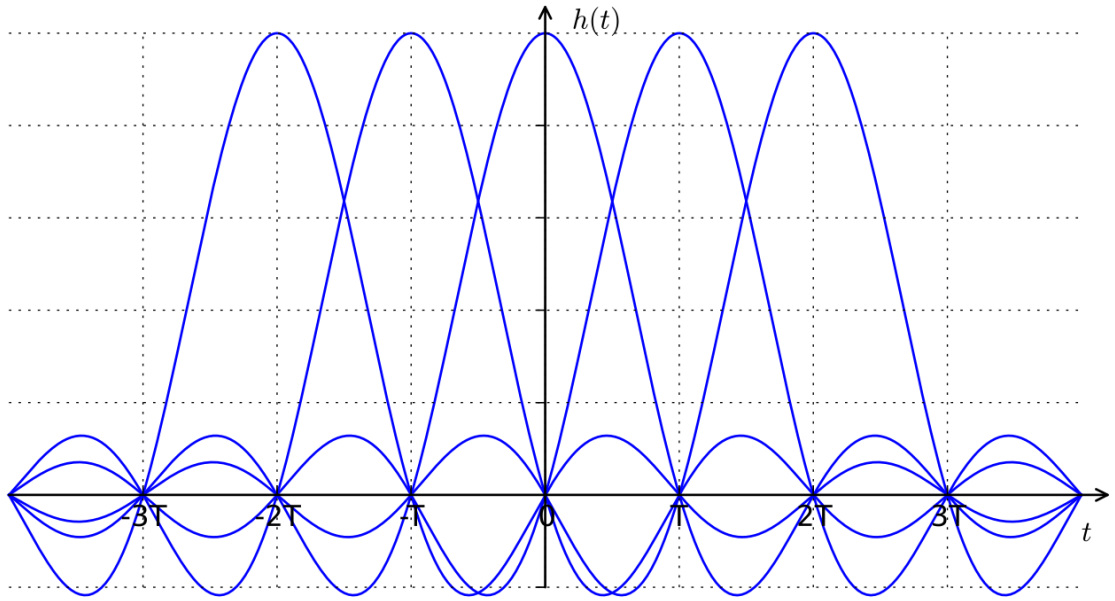


Figure 11: Consecutive raised-cosine impulses, demonstrating zero-ISI property [11].

3.7.3 Zero Phase Filter

A signal $x(t)$ is said to be,

Even

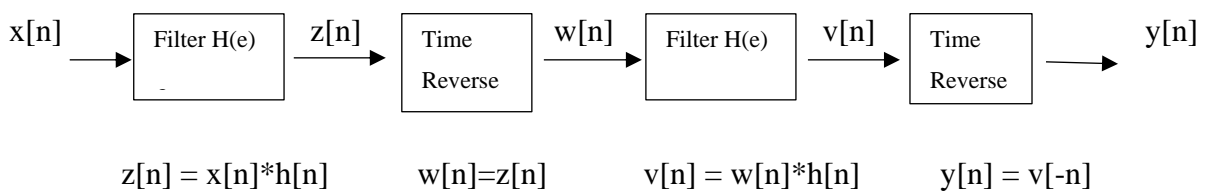
if $x(t) = x(-t)$

Odd

if $x(t) = -x(-t)$

The real impulse response $x(t)$ of a zero-phase filter is even [12]. Zero phase filtering helps to maintain filtered time wave characteristics at the point where they occur in the unfiltered signal. In our approach zero phase filter is used to optimize BPSK signal and noise. Both of BPSK signal and noise are filtered, reversed, filtered and reversed, respectively so that end to end optimization is carried out by implemented zero phase filter. That results with no phase distortion, nor time delay of linear phase.

Zero phase filter is used to limit the noise in our approach:



3.7.4 Butterworth Filter

A Butterworth filter is a type of signal processing filter designed to provide the most direct frequency response in the transition band. An ideal signal processing filter should not only totally reject unwanted frequencies, but also should have the same sensitivity for the desired frequencies. Such an ideal filter cannot be achieved, but Butterworth has shown that successive approximations have been achieved with the increase in the number of filter elements with the correct values. At that time, the filters created a significant fluctuation in the passband and the choice of component values was very interactive. The frequency response of the Butterworth filter approximation is also called the “maximally flat” response. The effect of the Butterworth filter in the time domain on a noisy signal can be seen in Figure 12.

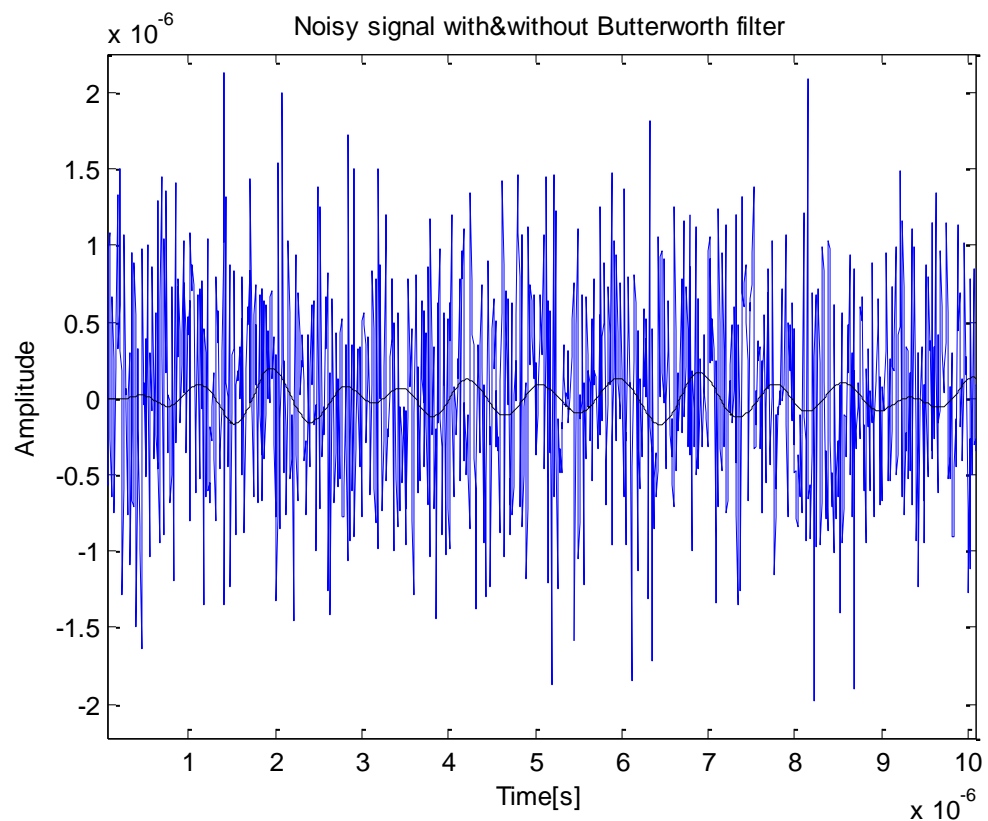


Figure 12: Noisy signal with and without Butterworth filtering: blue curve shows the noisy signal without an, black curve shows it with Butterworth applied.

3.8 Digital to analogue conversion (DAC) and modulation

DAC means that digital information is represented as analogue waveforms that can be transmitted over physical channels. In our approach, modulation maps the bit stream coming

from message source into the sign of carrier. A simple example is shown below in Figure 13. The input is a set of bits that take the values 1 and 0. Then these input values are transformed as the bit 1 to the symbol -1 and bit 0 to the symbol +1. In the end these values are drawn as analogue waveform.

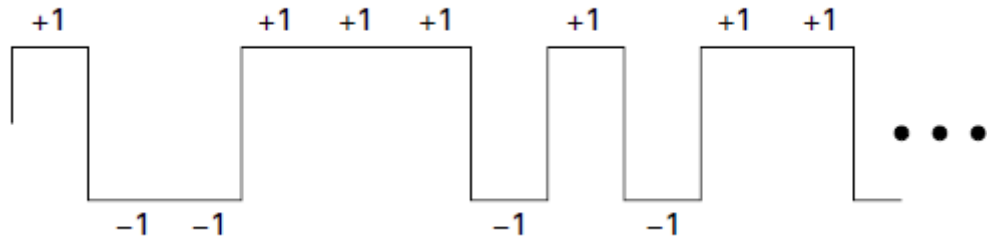


Figure 13: A simple example of binary modulation [13].

The purpose of DAC is to transmit a digital bit stream over an analogue communication channel.

Basically, we can divide modulation into steps. Each input bits are transmitted to corresponding symbols and corresponding parameters such as carrier frequency, frequency sampling, constants values and etc. are applied to the sequence of symbols. After adding a noise we generated, filtering methods are applied to limit the bandwidth and optimize the noise in order to surpass bit error rate (BER).

3.9 Demodulation

Demodulation is the corresponding process at the receiver to convert the received waveform to a duplicate of the input bit sequence. The demodulator specially designed for the symbol set used by the modulator determines the phase of the received signal and matches it to the symbol it represents, thus recovering the original data. Figure 14 shows the block diagram of demodulator for BPSK.

A suitable method for representing PSK schemes is on the constellation diagram. This shows the points in the complex plane. In this context, which are called in-phase and quadrature axes, respectively, due to the 90° separation of the real and imaginary axes. Such a representation in the vertical axes gives itself to a straightforward application. The common example is BPSK which used 2 phases. Because the data to be transferred is usually binary,

the PSK scheme is usually designed with the number of constellation points being a power of two.

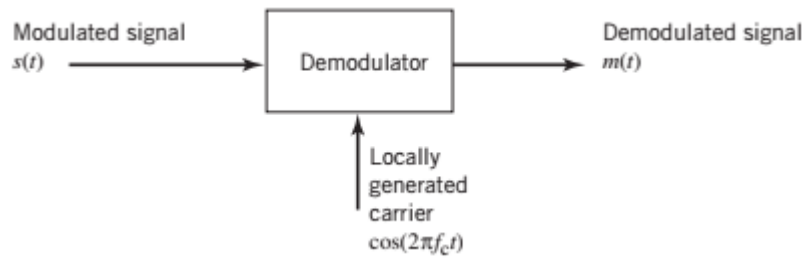


Figure 14: Block diagram of demodulator [5].

3.10 Error Correction

Error probability often associated with simple modulation and demodulation techniques is too high. One possible solution is to divide the channel coder into two layers, first an error-correcting code and then a simple modulator.

As a very simple example, the bit rate to the channel encoder can be reduced 3 times, and then each binary input can be repeated 3 times before being input to the modulator. If at most one of the 3 binary digits of the demodulator are incorrect, it can be corrected by the majority rule in the decoder, thereby reducing the error probability of a system error at a significant data rate cost.

In digital communication, synchronization problem and error correction are one of the most important technical issues. The probability of an error that occurs when using simple modulation and demodulation methods is often too high. One possible solution is to divide the channel encoder into two layers, first the error correction code and then a simple modulator. As a very simple example, the bit rate of the channel encoder can be reduced by 3 times, and then each binary input can be repeated 3 times before entering the modulator. If no more than one of the 3 binary digits of the demodulator is malfunctioning, this can be corrected using the majority rule in the decoder, which reduces the likelihood of a system error at significant costs for the data transfer speed.

The early use of error correction codes was part of a two-layer system where the digital error correction encoder is followed by a modulator. At the receiver, the waveform is first demodulated to a noisy version of the encoded sequence, and then this noisy version is decoded by the error correction decoder. The present application often provides better performance by combining error correction coding and modulation into coded modulation schemes. Whether error correction and conventional modulation are separate layers or

combined, the combination is generally referred to as a modulator and performs this modulation on data in one direction and demodulation in the other direction is called a modem.

Wireless channels are not only subject to additional interference, but also random power fluctuations on several paths between transmitter and receiver. The interaction of these paths leads to fading and we investigate how this affects coding, signal selection, modulation and detection. Wireless communication also discusses channel measurement and how these measurements can be used for input and output.

In our approach, we applied sinc shaped filter and raised-cosine filter, demonstrated truncated sinc shaped filter in order to minimize number of errors so that bit error rate.

3.11 BPSK Receiver

The BPSK uses coherent or synchronized demodulation, which requires that the information about the phase and frequency of the carrier be found in the receiver. If a low-level pilot carrier signal is transmitted with the BPSK signal, the carrier phase and frequency can be recovered at the receiver using a phase locked loop (PLL). If the pilot carrier is not transmitted, a Costas loop or square loop can be used to synthesize the received phase and frequency from BPSK signal.

A communication channel is the physical environment between the transmitter and receiver where they are physically separated. No communication channel is ideal and therefore a message signal is subjected to various forms of distortion. Sources of degradation may include attenuation, noise, distortion, and interference. Since some or all of these disturbances are present in a physical channel, the most important purpose in the design of a communication system is to overcome the effects of such disturbances.

The function of the receiver is to recover the message signal from the received signal. The main function is to perform the demodulation process with several functions such as amplification and filtering. The complexity of a receiver is usually more important than the transmitter because a receiver must also minimize the effects of channel distortion. Figure 15 shows the basic functional elements of a receiver for BPSK.

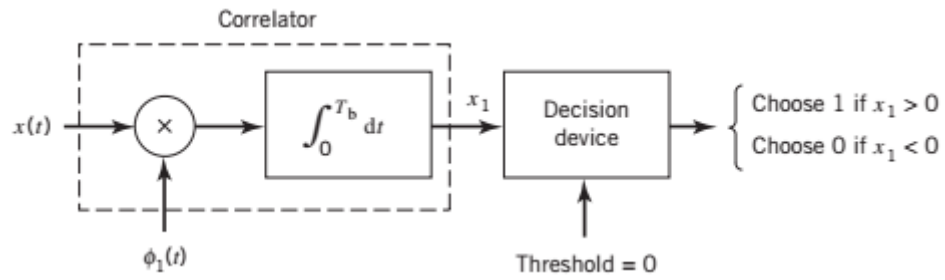


Figure 15: Block diagram of coherent binary PSK receiver [5].

In order to make an optimal decision on the $x(t)$ signal in favour of symbol 1 or symbol 0 (i.e., estimation of the original binary sequence at the transmitter input), it is assumed that the receiver has access to the locally produced replica of the base function. In other words, the receiver is synchronized with the transmitter. Two basic components can be identified in the BPSK receiver:

1. The correlator that associates the received signal $x(t)$ with the base function based on the bit by bit basis.
2. Decision maker comparing the correlator output to the zero threshold assuming that the binary symbols 1 and 0 are equiprobable. If the threshold is exceeded, a decision is made in favour of symbol 1; if not, the decision is made in favour of symbol 0. Correlator equality with the zero threshold is determined by throwing a fair coin (i.e., randomly).

3.12 Bit Error Rate (BER)

When data is transferred over a data connection, there is a chance that errors will appear in the system. If the data fails, the integrity of the system may be compromised. As a result, it is necessary to evaluate the performance of the system, and the bit error rate BER provides an ideal way to accomplish this.

Bit error rate (BER) is the number of bit errors per unit time. It is a measurement scale of the number of bit errors that occur in a given number of bit transmissions. When transmitting data from one point to another over a radio/wireless communication or a wired connection, the key parameter is how many errors will appear in the data that appears at the end.

Unlike many other types of evaluation, the bit error rate BER evaluates the overall end-to-end performance of a system, including the transmitter, receiver, and the environment between them. In this way, the bit error rate BER makes it possible to test the actual performance of a system in operation, rather than testing the components.

Simple formula of BER;

$$\text{BER} = \text{Total Number of Errors} / \text{Total Number of Bits}$$

When the medium between the transmitter and receiver is good and the signal to noise ratio is high, then the bit error rate will be very small. The main reasons for the distortion of a data channel and the associated bit error rate are noise and changes in the propagation path. Both effects have a random element, the propagation model follows a Rayleigh model, while the noise following a Gaussian probability function. This means that the analysis of channel properties is normally performed using statistical analysis techniques.

Digital communication systems lead to performance degradation when the SNR falls below a certain threshold. A slight reduction of SNR may result in a significant increase in the bit error rate (BER), leading to a significant distortion of system performance [3].

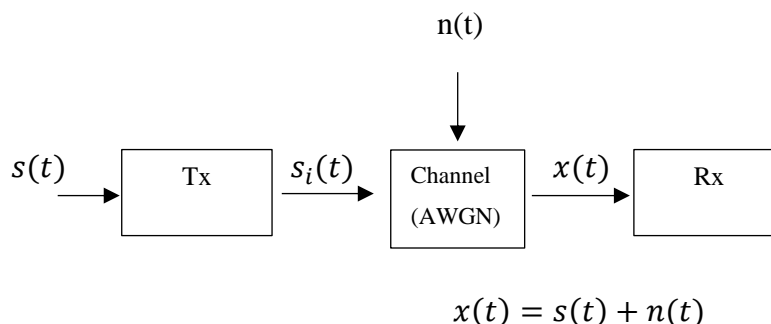
4 MATLAB SIMULATION

MATLAB is a programming language developed by MathWorks. It started as a matrix programming language in which linear algebraic programming was simple. It can be run in interactive sessions and in batch form.

Modelling is a way to create a virtual representation of an actual system that contains software and hardware. If the software components of the model are controlled by mathematical relationships, in many cases you can simulate the virtual representation that should behave.

Modelling and simulation are valuable in the early stages of the design process, especially when hardware cannot be used, under test conditions that may be difficult to reproduce only with hardware prototypes. Choosing between modelling and simulation can improve the quality of the system design, reducing the number of errors detected during the subsequent design process. MATLAB is widely used as a computing tool in science and engineering, encompassing physics, mathematics and all engineering processes. We work on Signal Processing and Communications applications in this work. In transmit filter zero-order hold (ZoH) is used. Additive White Gaussian Noise is added to simulate transmission channel. AWGN is a basic noise model used in communication to mimic the effect of many random processes that occur in nature. The bit energy over the noise is calculated in channel part. Receiver detects information which is intended to transfer and recovers binary information. The performance of digital receiver is measured by a parameter called the Bit-Error Ratio (BER).

The basis of AWGN simulation is following;



From now on, I'm moving step by step according to block diagram of band-pass transmission system shown in Figure 6 in order to explain simulation of the thesis work. As a first step, we generate random binary bits in sinusoidal form (Figure 16):

Code:

```
s_temp= cos(2*pi*fc*t);  
  
for i=0:Nb-1  
    s(i*Ns+1:(i+1)*Ns)= (-1)^b(i+1)*s_temp;  
end
```

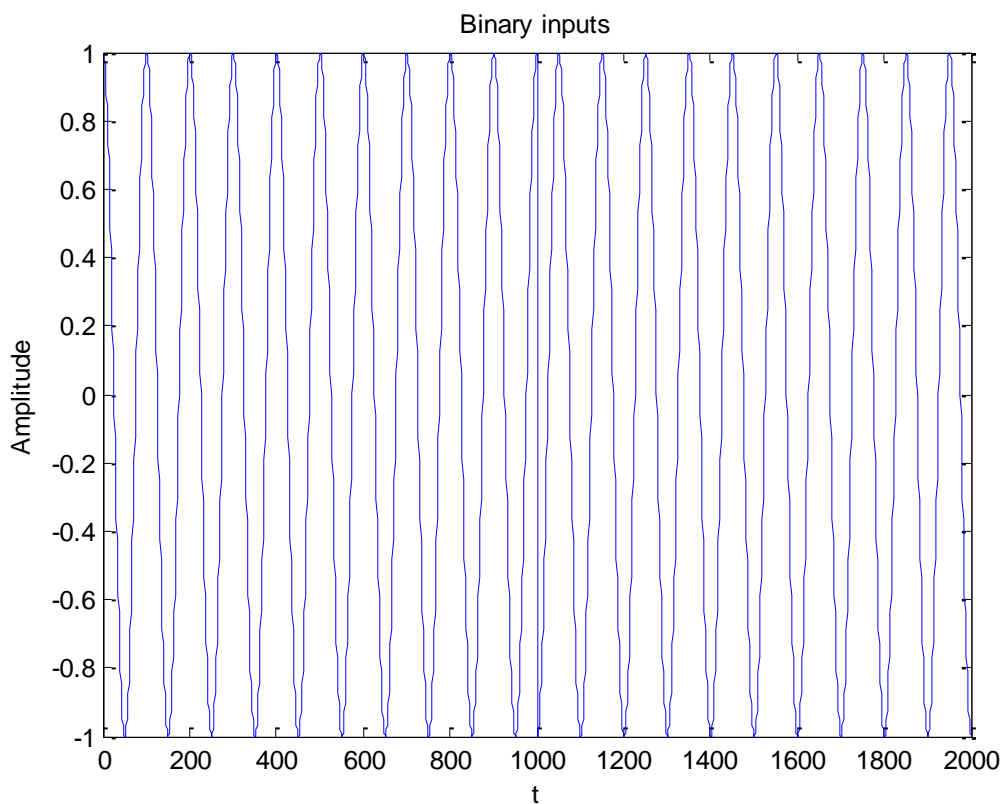


Figure 16: BPSK waveform.

Then generated signal goes through transmitter unit where the modulation of carrier wave is applied and sent to communication channel. Power Spectral Density is the measure of the power of a signal relative to frequency. The PSD amplitude is normalized to the spectral resolution used to digitize the signal. Transmitter encodes generated bits.

We can apply Power Spectral Density to BPSK waveform (Figure 17).

Code:

```
Ns= round(Tb*fs);  
t = [0:1/fs:Tb-1/fs]; %phase  
s_temp= cos(2*pi*fc*t);  
  
for i=0:Nb-1 % system generates the symbol 1 or -1 according to each bit, Nb times  
    s(i*Ns+1:(i+1)*Ns)= (-1)^b(i+1)*s_temp;  
end
```

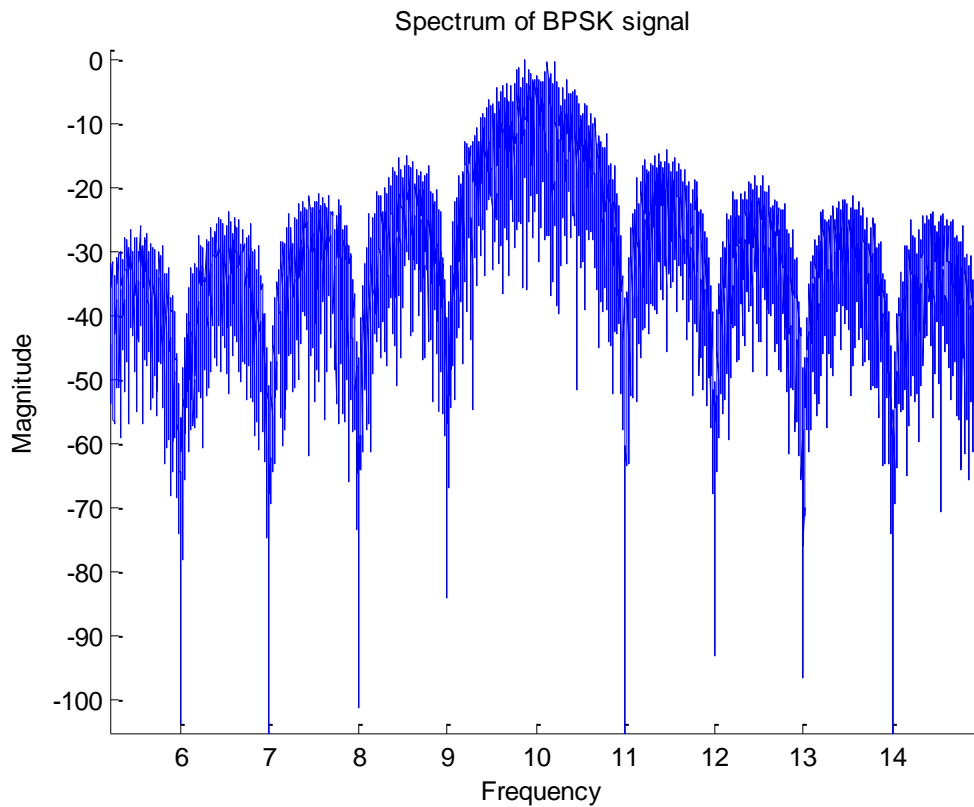


Figure 17: PSD waveform for BPSK.

The sinc filter modulated set of bits are shown in Figure 18.

Code:

```
for i=0:Nb-1
```

```

h=sinc(2*W*(t-T_sinc/2))*(cos(2*pi*alfa*W*(t-T_sinc/2))/(1-16*(alfa*W*(t-
T_sinc/2)).^2));%shifted
end

```

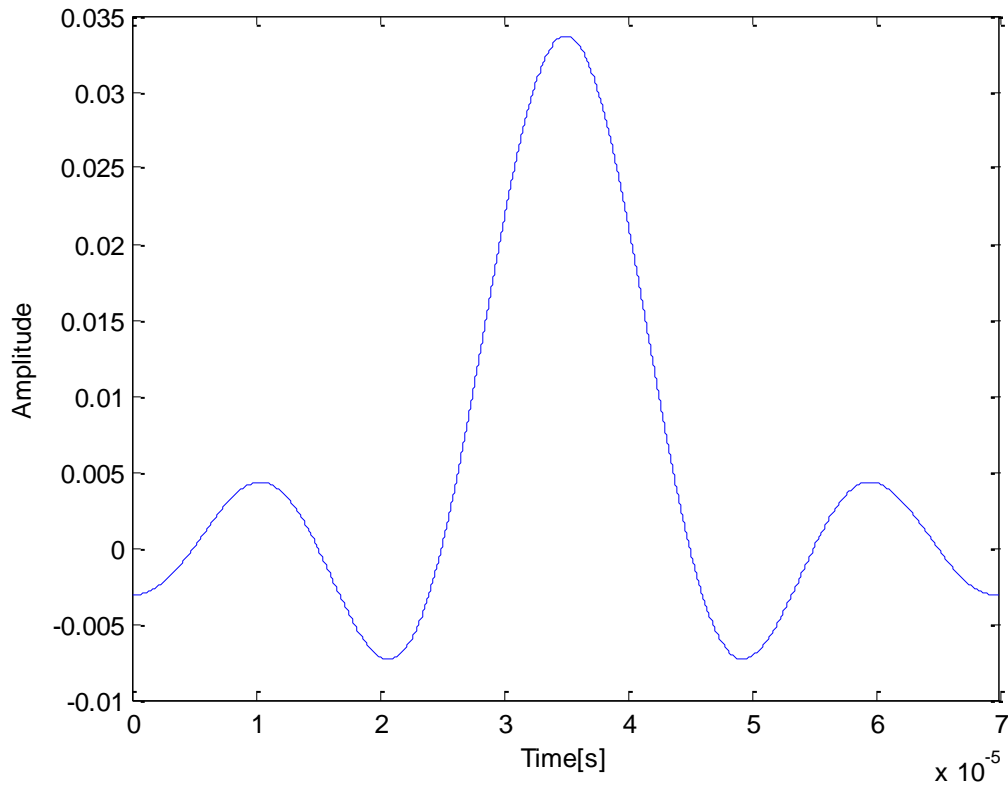


Figure 18: Waveform for Sinc filter.

The raised-cosine filter is commonly used for pulse shaping. The roll-off factor, a measure of the excess factor of bandwidth of the filter is important element of the raised-cosine filter. The raised-cosine filter modulated set of bits with different magnitude of roll-off factor is shown in Figure 20. Waveform is stabilized by roll-off factor.

Code:

```

alfa=0.5;
if abs(t-T_sinc/2)==Tb/(2*alfa)
    h=pi.*sinc(1/(2*alfa))./(4*Tb);
else
    h=sinc((t-T_sinc/2)/Tb).*cos(pi*alfa*(t-T_sinc/2)/Tb)./(Tb*(1-(2*alfa*(t-
T_sinc/2)/Tb).^2));

```

```

end

for i=0:Nb-1
    s_temp(round(i*Tb*fs+1):round((i+1)*Tb*fs+Tb*(L-1)*fs))=(-
1)^(i+1)*h+randn(size(t));
    s=s+s_temp;
    s_temp=zeros;
end

```

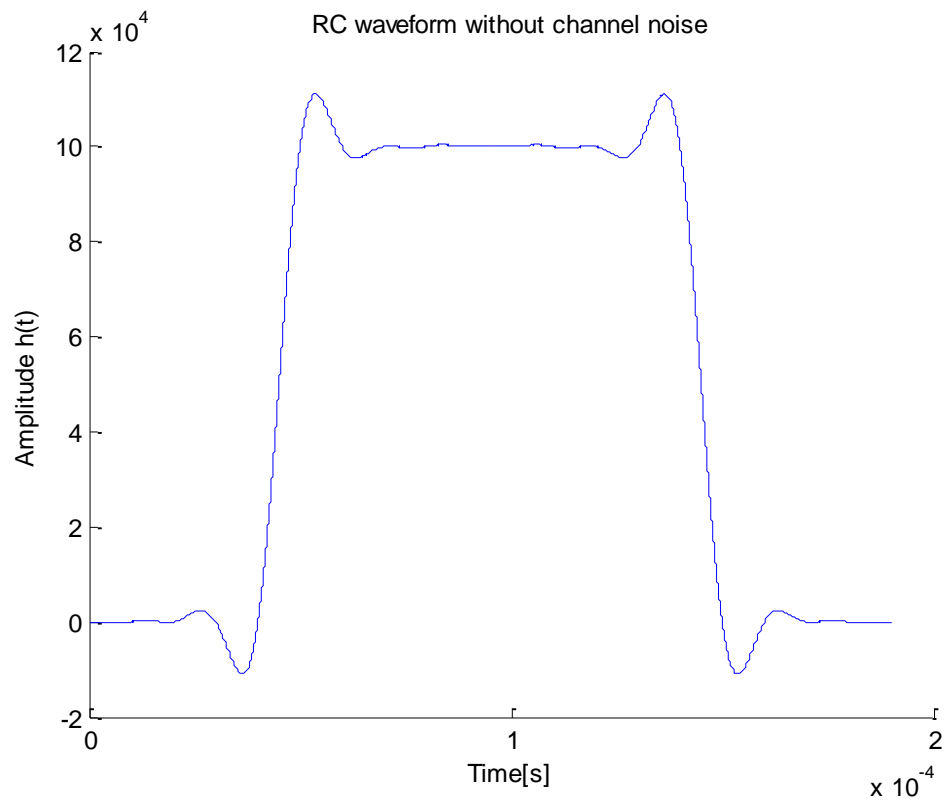


Figure 19: Bit sequence by RC waveform.

Difference between roll-off factor 0 and 1:

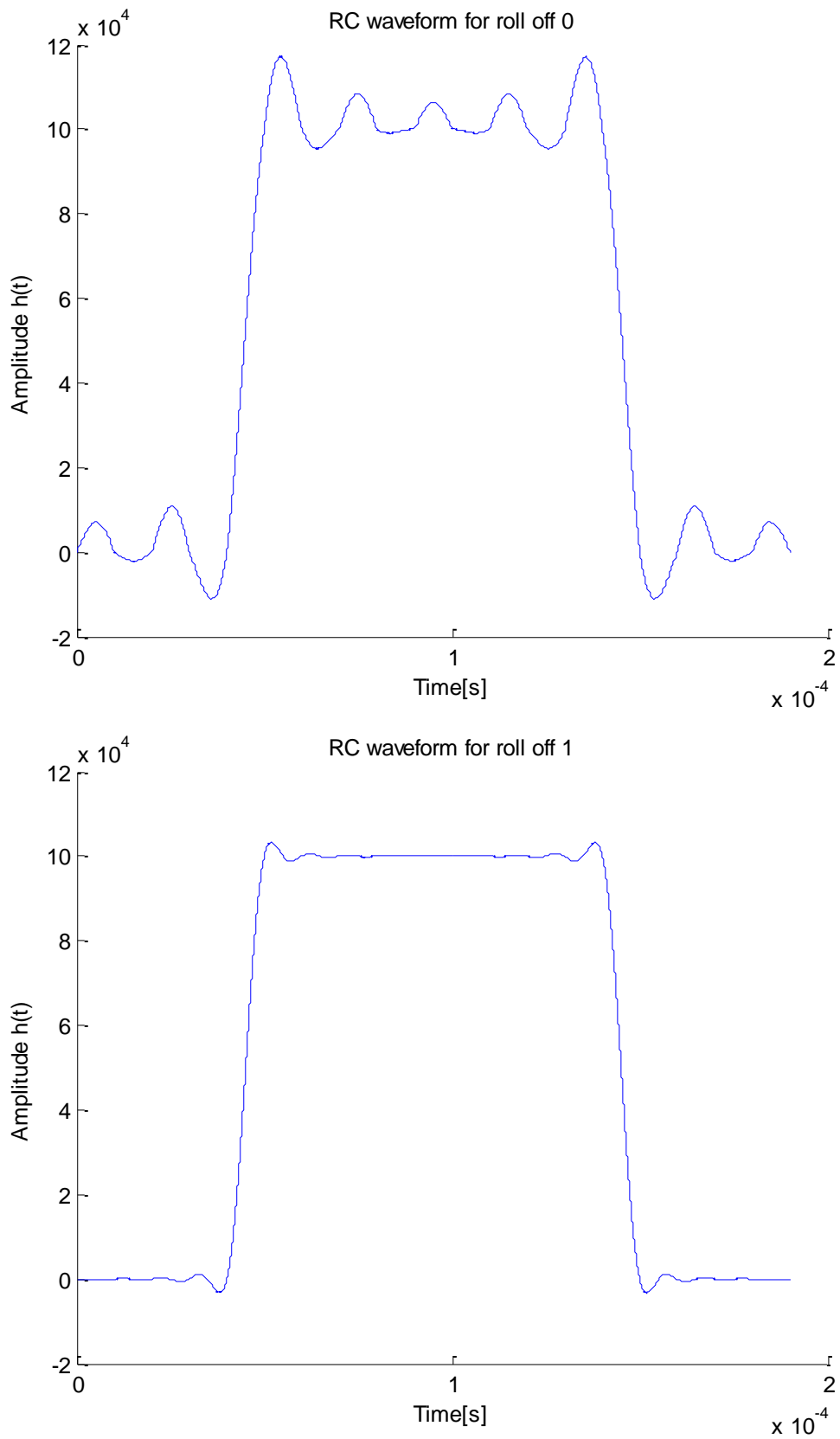


Figure 20: Effect of roll-off factor.

Simulation of the communication channel is necessary part of our design since wireless digital signal is distorted with external noise sources in its environment. To simulate the noise, I have used Additive White Gaussian Noise model which can provide us uniformity to power across the frequency band with normal distribution in time domain.

Additive Gaussian White Noise is model of artificial noise to mimic external processes which occur in nature. Signal to Noise Ratio is set to 10 for simulation of AWGN as a result output waveform of noisy communication channel is distinguishable showed in Figure 21.

Code:

```
n=awgn(s,10,'measured'); % s with noise. 10 represents SNR
```

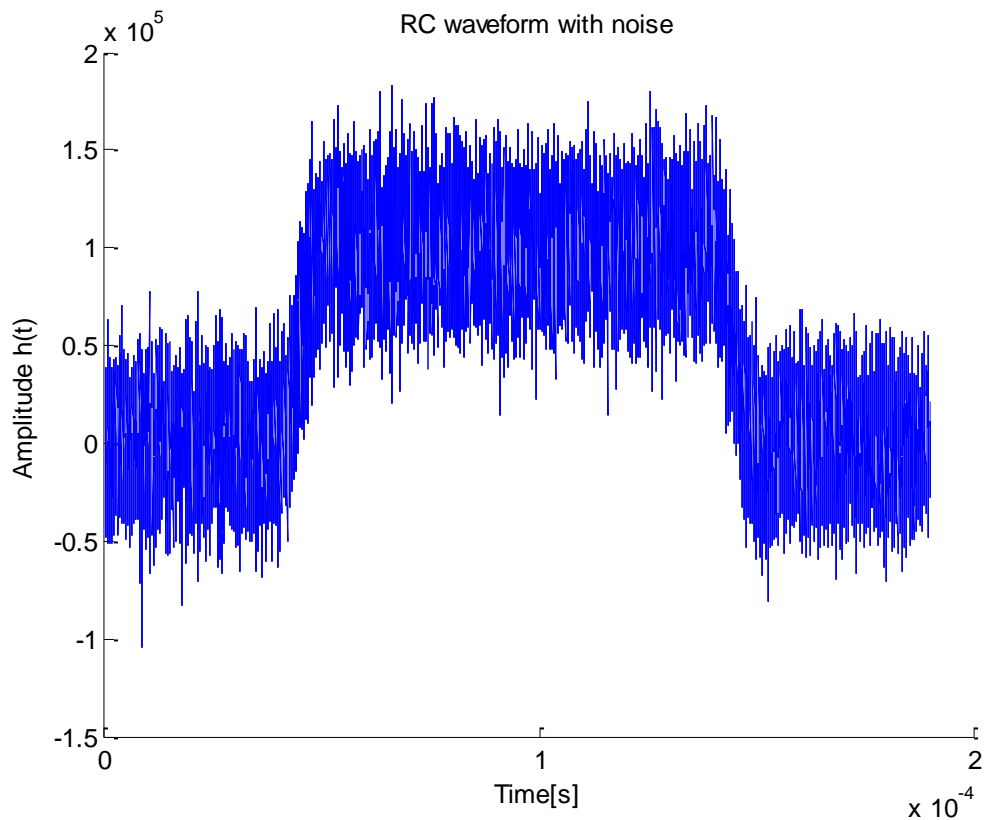


Figure 21: AWGN Noise.

Welch's Power Spectral Density (PSD);

`pxx = pwelch (x)` returns the PSD estimate (`pxx`) of the input signal `x` which is found using Welch's overlapping segment average estimator. Its power spectral density is shown in Figure 22 and Figure 23 for the signal without noise and with noise, respectively.

Welch power signal spectrum without noise:



Figure 22: Welch's power signal spectrum without noise.

Welch power signal spectrum with noise:

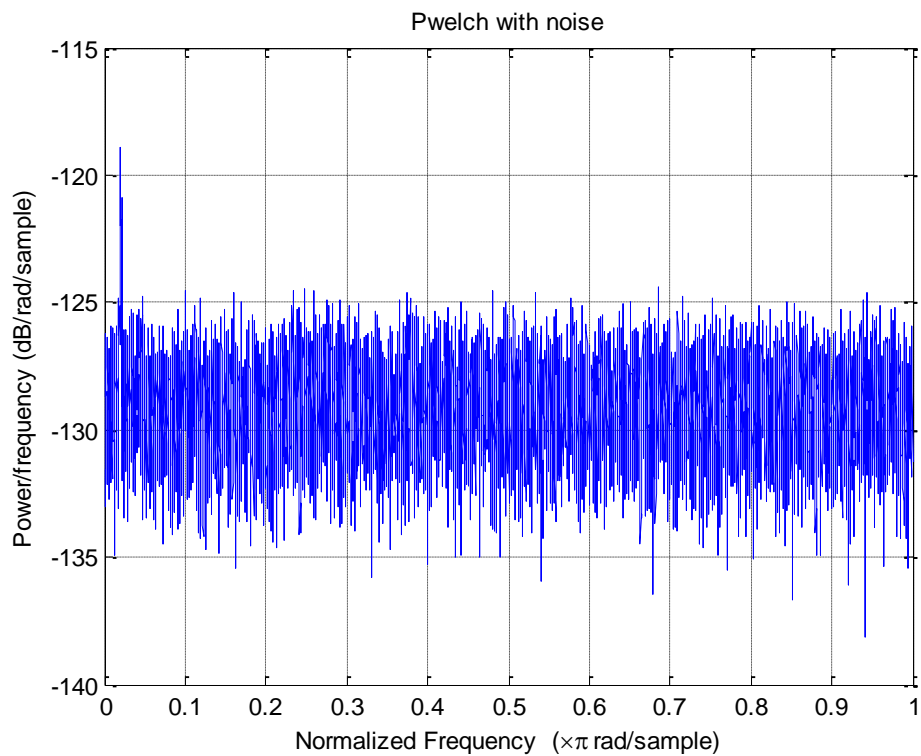


Figure 23: Welch's power signal spectrum with noise.

Power of Radio Frequency for noisy signal is a term that directly related to SNR. Signal level of radio frequency signal is shown in Figure 24 and Figure 25 without noise and with noise, respectively.

Code:

```
t_carrier = [0:1/fs:length(s)/fs-1/fs];  
P_RF = mean(s.^2) %power without noise  
Pn_RF = mean(Noise.^2) %power only noise
```

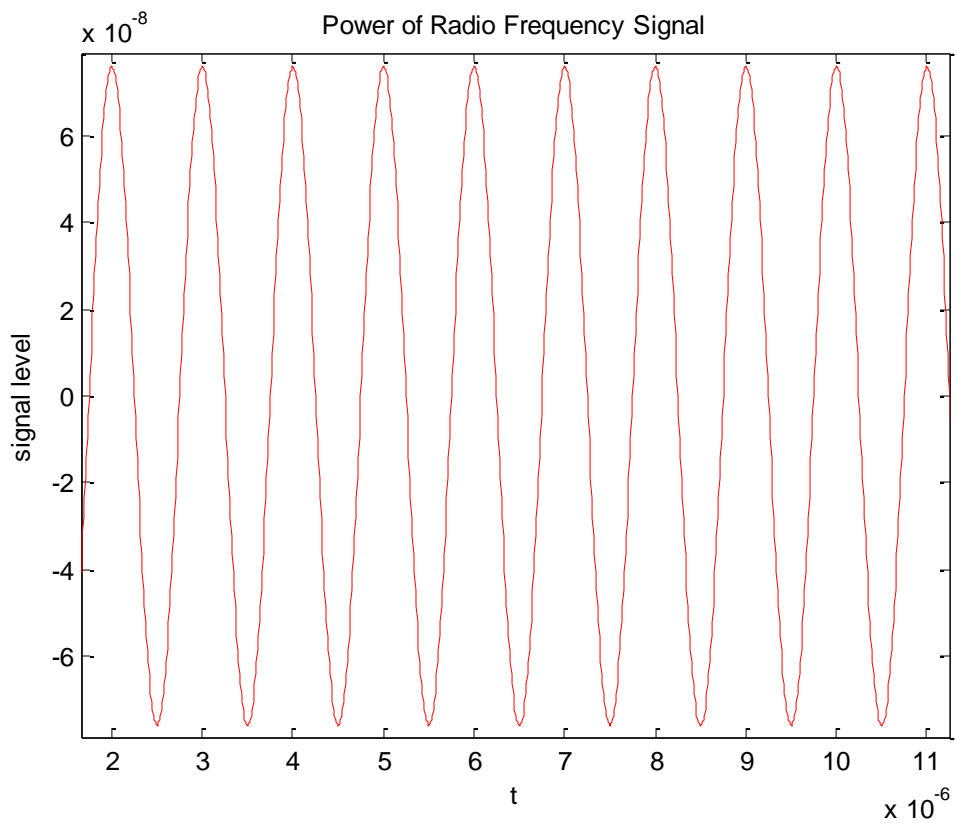


Figure 24: Signal level of radio frequency signal without noise.

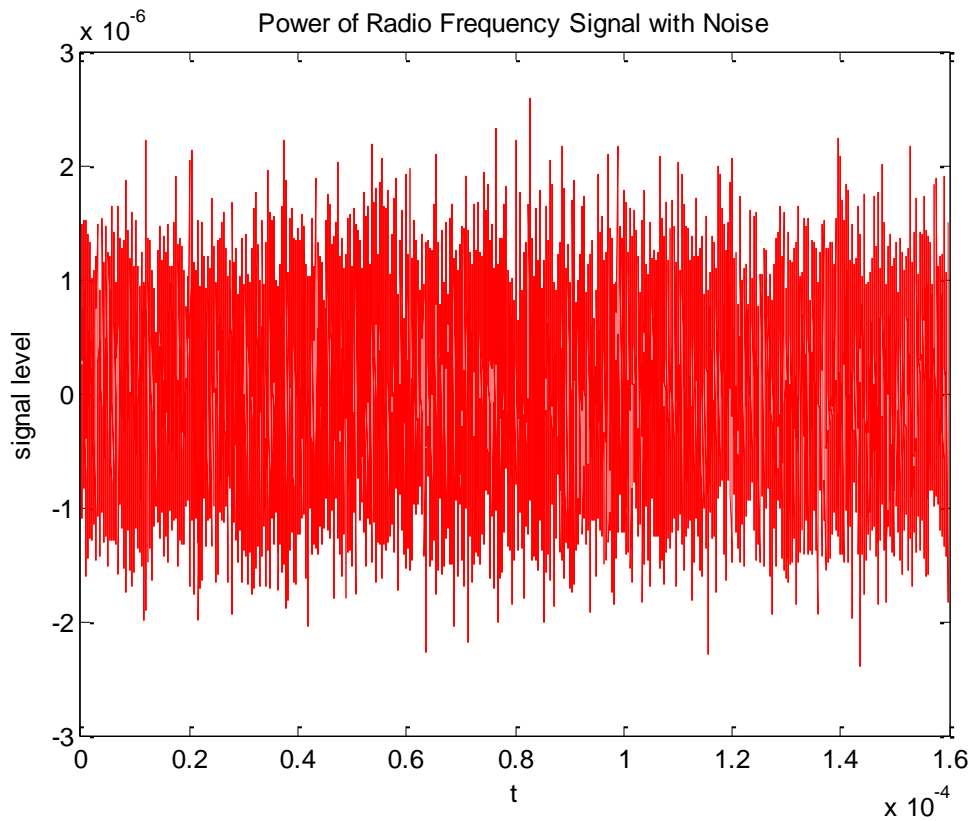
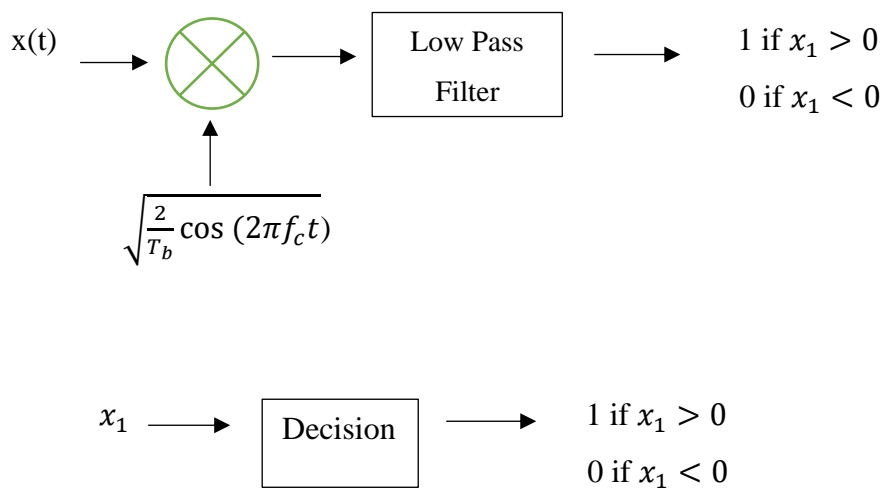


Figure 25: Signal level of radio frequency signal with noise.

When we compare both graphics of the signal without noise and noisy signal, amplitude of signal is increasing with noise. To detect the original binary sequences 1 and 0, the noisy PSK signal is applied to a correlator:



Zero phase filter;

%implementation of zero phase filter:

```
Noise_filtered=filter(c,a,Noise);
```

```
Noise_filtered=Noise_filtered(length(Noise_filtered):-1:1);
```

```
Noise_filtered=filter(c,a,Noise_filtered);
```

```
Noise_filtered=Noise_filtered(length(Noise_filtered):-1:1);
```

%

```
s_filtered=filter(c,a,s);
```

```
s_filtered=s_filtered(length(s_filtered):-1:1);
```

```
s_filtered=filter(c,a,s_filtered);
```

```
s_filtered=s_filtered(length(s_filtered):-1:1);
```

```
r=s_filtered+Noise_filtered;
```

```
P_RF_filtered=mean(s_filtered.^2) %power without noise
```

```
Pn_RF_filtered=mean(Noise_filtered.^2) %power only noise
```

Bit Error Rate simulation plays a significant role of communication. It is basically resulting for net efficiency of components and design methods in system. One of the main goals of passband bandwidth transmission systems is to optimize the receiver to minimize the Additive White Gaussian Noise.

Bit Error Rate;

$$N_b > \frac{10}{BER} \text{ should be valid for each dB value.}$$

During simulation when $N_b < \frac{10}{BER}$, N_b should be increased in order to see reliable results

N_b number of input and output values are compared to detect errors. Number of errors over number of inputs (or outputs) gives BER.

Simulation results of BER and Bit Error Rate Comparison of theory and simulation are shown below (Blue line is theoretical BER, markers are estimated BER).

N_b (Total number of bits)	Errors	$E_b N_o$ (dB)	Bit Error Rate (BER)
10^4	399	2	0.0399
10^4	245	3	0.0245
10^4	116	4	0.0116
10^4	80	5	0.008
10^4	30	6	0.003
10^4	10	7	0.001
10^5	30	8	0.0003
10^6	70	9	0.00007

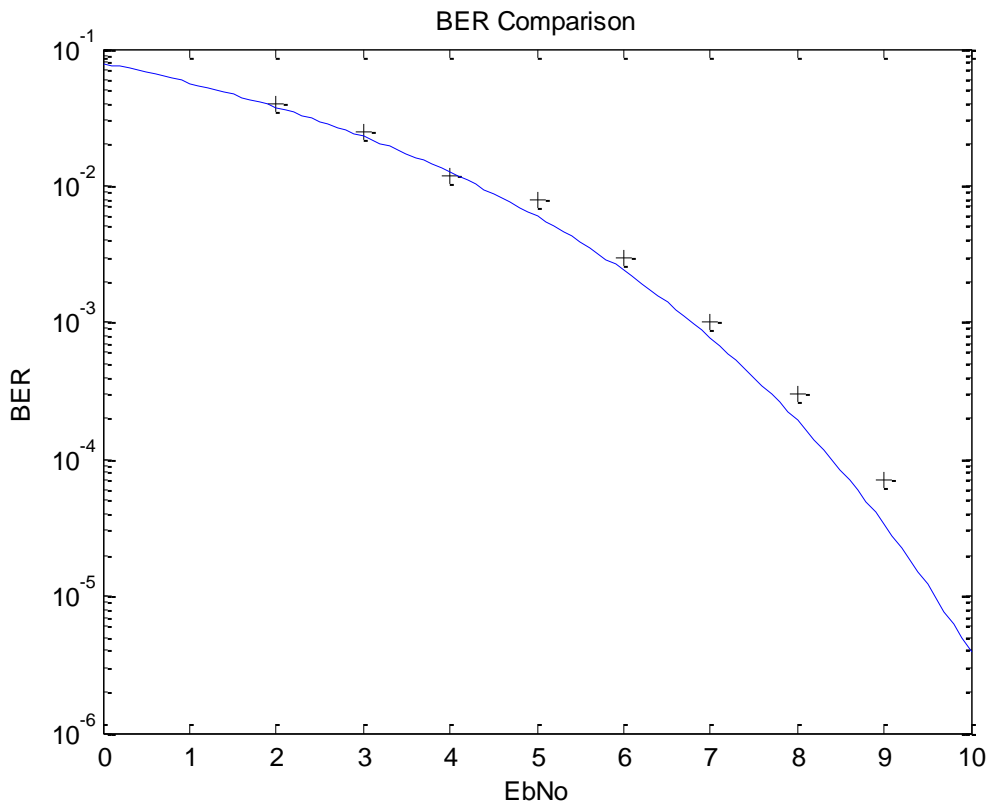


Figure 26: Bit Error Rate Comparison of theory and simulation

5 CONCLUSION AND FUTURE WORK

In this project, different analysis and filtering methods are applied to BPSK signal with AWGN in MATLAB. Performance of the system are tested and compared with different analysis techniques. Even though different techniques are applied, main properties of BPSK stayed same or with negligible changes. In general, raised-cosine filter has better performances since roll-off factor can be applied in this type of filter. This simulation result confirms the result obtained in past work by other authors using similar analyses. The evaluation of different methods to analyse BPSK systems based on our precise analysis provides reliable information for choosing suitable modulation ways in design of BPSK systems.

The design of digital communication systems and networks present many challenges as it is a multidimensional, nonlinear, constrained optimization problem. It is important to emphasize that there are many system variables and parameters that represent a large number of constraints, some of them cannot be easily classified and directly put into a design. Moreover, I introduced output graphics of signal with and without noise. Finally, I represented Bit Error Rate results and comparison of theoretical and practical results. In the future, different kind of signal types beside BPSK can be applied to same communication channel and the system can be implemented accordingly.

6 REFERENCES

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Annex

Figure 1: Basic block diagram of a communication system.

Figure 2: Block diagram of a simple digital communication system.

Figure 3: Commercially exploited bands of the radio frequency spectrum.

Figure 4: Transmission impairments: (a) transmitted signal; (b) effects of attenuation; (c) effects of distortion; (d) effects of interference; (e) effects of noise; and (f) aggregate effects of impairments.

Figure 5: SNR value detection between signal and noise.

Figure 6: Block diagram of band-pass transmission system.

Figure 7: Binary digital modulation schemes: (a) binary data, (b) modulating signal, (c) carrier wave, (d) BASK signal, (e) BFSK signal, and (f) BPSK signal

Figure 8: Block diagram of binary PSK transmitter.

Figure 9: Sampling process. (a) Analogue signal. (b) The instantaneous sampling of the analogue signal.

Figure 10: The normalized sinc function.

Figure 11: Consecutive raised-cosine impulses, demonstrating zero-ISI property.

Figure 12: Noisy signal with and without Butterworth filtering: blue curve shows the noisy signal without an, black curve shows it with Butterworth applied.

Figure 13: A simple example of binary modulation.

Figure 14: Block diagram of demodulator.

Figure 15: Block diagram of coherent binary PSK receiver.

Figure 16: BPSK waveform

Figure 17: PSD waveform for BPSK

Figure 18: Waveform for Sinc filter

Figure 19: Bit sequence by RC waveform

Figure 20: Effect of roll-off factor

Figure 21: AWGN noise

Figure 22: Pwelch without noise

Figure 23: Pwelch with noise

Figure 24: Signal level of radio frequency signal without noise

Figure 25: Signal level of radio frequency signal with noise

Figure 26: Bit Error Rate Comparison of theory and simulation (Blue line is theoretical BER, markers are estimated BER)