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BACHELOR THESIS

Modern Modulation Methods for Underwater Communication

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Thesis Declaration

I certify that this work which is being submitted as a Bachelor Thesis to the Faculty of Electrical Engineering at West Bohemia University in Pilsen is my own work, and the sources of information have been acknowledged. The work was done under the guidance of Ing. Richard Linhart. It is understood that West Bohemia University in Pilsen can use my work under the conditions of § 60 Law č. 121/2000.

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Abstract

Orthogonal Frequency Division Multiplexing (OFDM) is being considered for communication over wireless (acoustic) underwater channels.

This paper is a study about OFDM modulation for underwater communication system. Study of the theory of OFDM modulation is explained, and a basic simulation was made for the modulator and demodulator structure in MATLAB environment as a study experiment for modern modulation method for underwater communication systems. The symbol-error-ratio (SER) is considered as the main parameter. The simulation is performed as an SNR (signal-to-noise-ratio) versus SER comparison for transmission and reception of modulated signals. OFDM block size is changed in the simulation later to realize the behavior for transmission and reception of modulated signals and to find out how error rate changes based on the size of OFDM block size.

Acknowledgments

I am very grateful to all my teachers at University of West Bohemia in Pilsen, who have taught me the skills for this work. I am very thankful to my guide, Ing. Richart Linhart, who has been very cooperative and understanding. And, I would like to thank the authors who write nice scientific books which are a source of knowledge for many people.

Contents

ABSTRACT 4

ACKNOWLEDGEMENT 5

CONTENTS 6

INTRODUCTION 8

1 UNDERWATER CHANNEL 9

- 1.1 ACOUSTICS FOR UNDERWATER COMMUNICATION 9
- 1.2 ACOUSTIC PROPAGATION 9
  - 1.2.1 Attenuation 9
  - 1.2.2 Noise 11
  - 1.2.3 Propagation Delay 12
  - 1.2.4 Multipath 13
  - 1.2.5 Doppler Effect 14
- 1.3 RESOURCE ALLOCATION 15
  - 1.3.1 The AN Product and the SNR 15
  - 1.3.2 Optimal Frequency 16
  - 1.3.3 Transmission Power 17

2 SYSTEM DESIGN 18

- 2.1 BASIC OFDM PRINCIPLE 18
  - 2.1.1 Orthogonality 19
  - 2.1.2 Modulation Using Fast Fourier Transform (FFT) 20
  - 2.1.3 Guard Time 21
  - 2.1.4 Equalization 22
  - 2.1.5 Advantages 22
  - 2.1.6 Disadvantages 23

3 THE OFDM IMPLEMENTATION 24

3.1 OFDM 24

3.1.1 Signal Representation in OFDMl 24

3.2 OFDM TRANSMITTER 25

3.2.1 Random Data Generator 25

3.2.2 Data to Symbol Mapper 25

3.2.3 Channel Model 26

3.2.4 Transfer Implementation 26

3.3 OFDM RECEIVER 28

3.3.1 FFT Demodulation 28

4 SIMULATION RESULTS AND DISCUSSION 31

4.1 CODING 31

4.2 SIMULATION RESULTS AND DISCUSSION 34

CONCLUSION 36

ABBREVIATIONS 37

REFERENCES 38

Introduction

Underwater acoustic communication is a technique of sending and receiving message below water. There are several ways of employing such communication. In the 4th Generation System, the transmission schemes are based on Orthogonal Frequency Division Multiplexing (OFDM) [1]. The idea behind OFDM modulation is to send the data over different sub-channels which are orthogonal to each other.

In this thesis the description is divided into three parts. First part is the background and information about an OFDM modulation. In the second part, the system model is expressed. Last part presents the results and discussion regarding the simulation in MATLAB.

1

Underwater Channel

1.1 Acoustics for Underwater Communication

There are a few means for wireless communication underwater. Radio waves of extra low frequency (30 Hz–300 Hz) are the only waves that can propagate any distance in sea water. But such low frequencies require large antennas and high transmission power. The other common means is optical waves. Even though optical waves do not suffer much attenuation, they are considerably affected by scattering and hence only lasers of extreme intensity can propagate in water. Acoustics remains the best known solution for wireless underwater communication.

The underwater channel is one of the most challenging channels for reasons such as frequency dependent attenuation, distance dependent bandwidth, and accentuated Doppler effect which is non uniform in the signal bandwidth. This is in addition to the common background noise which is predominant and non-negligible, is frequency dependent and site dependent.

1.2 Acoustic Propagation

1.2.1 Attenuation

For a distance *l* and a frequency *f*, the attenuation in an underwater acoustic channel is given as [2]

(1.1)

where *k* is the spreading factor, which describes the geometry of propagation, and *a(f)* is the absorption coefficient. Equation 1.1 is expressed in dB:

$$10\log A(l, f) = k \cdot 10\log l + l \cdot 10\log a(f) \qquad (1.2)$$

The absorption coefficient for frequencies above a few hundred Hz can be expressed empirically, using the Thorp's formula [1] which gives a(*f*) in dB/km for *f* in kHz as:

—— ——— (1.3)

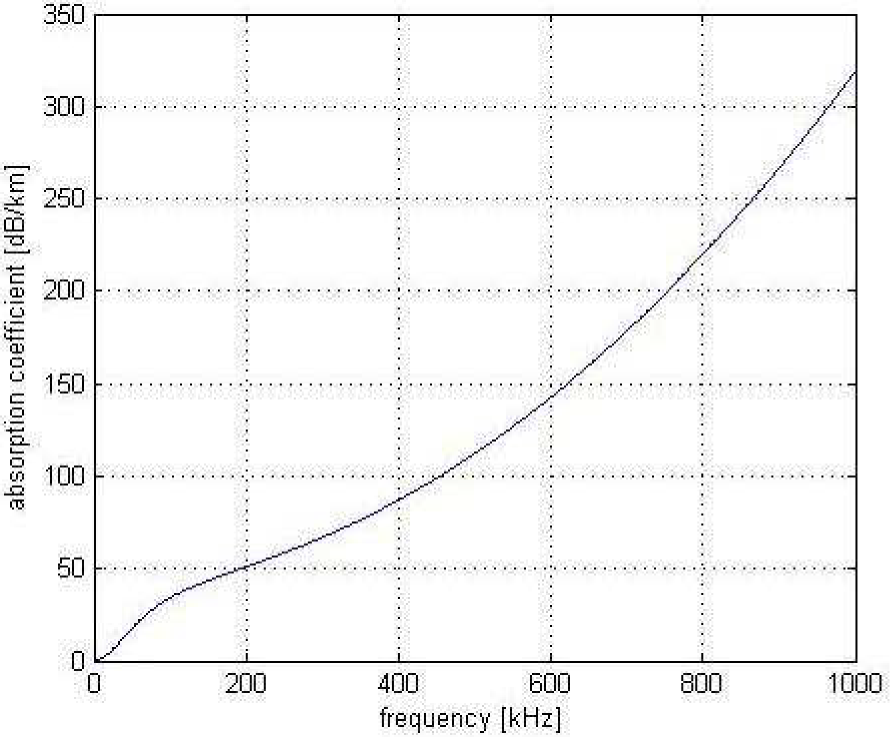

Figure 1.1: Absorption coefficient as a function of frequency.

For lower frequencies, the following formula may be used:

—— (1.4)

The absorption coefficient is the major factor that limits the maximal usable frequency of an underwater system. As it rapidly increases with frequency, the path loss will also increase (see

Figure 1.1), and therefore only the frequencies below a threshold may be used when deploying an underwater communication link.

1.2.2 Noise

The ambient noise in the ocean can be modeled using four sources: turbulence, shipping, waves, and thermal noise. Gaussian statistics and a continuous power spectral density describe the major sources or ambient noise. The following empirical formula gives the p.s.d. of the four noise components in dB re μ Pa per Hz as a function of frequency in kHz [3]:

$$\begin{aligned} 10 log N_t(f) &= 17 - 30 log f \\ 10 log N_s(f) &= 40 + 20(s - 0.5) + 26 log f - 60 log(f + 0.03) \\ 10 log N_w(f) &= 50 + 7.5 w^{1/2} + 20 log f - 40 log(f + 0.4) \\ 10 log N_{th}(f) &= -15 + 20 log f \end{aligned} \tag{1.5}$$

Turbulence noise influences only the very low frequency region, $f < 10$ Hz. Noise caused by distant shipping is dominant in the frequency region 10 Hz – 100 kHz, and it is modeled through the shipping activity factor s, whose value ranges between 0 and 1 for low and high activity, respectively. Surface motion, caused by wind-driven waves (w is the wind speed in m/s) is the major factor contributing to the noise in the frequency region 100 Hz – 100 kHz (which is the operating region used by the majority of acoustic systems). Thermal noise becomes dominant for $f > 100$ kHz.

The overall power spectral density of the noise is given by

$$N(f) = N_t(f) + N_s(f) + N_w(f) + N_{th}(f) \tag{1.6}$$

Figure 1.2 illustrates the power spectral density $N(f)$ of the ambient noise for the case of no wind (solid line) and wind at 10m/s with different shipping activity factors. The noise increases at the low frequency range, thus limiting the useful acoustic bandwidth from below. Due to the linear decayment of the noise p.s.d. on the logarithmic scale (in a certain frequency range), the following approximation may then be useful:

$$10logN(f) \approx N_1 - \eta logf \qquad (1.7)$$

The approximation is shown in Figure 1.2 with $N1 = 50dBre\mu Pa$ and $\eta = 18dB/decade$.

1.2.3 Propagation Delay

The experienced delays in an underwater acoustic communication link are much higher than in an open-air link. The nominal speed of sound in water is 1,500 m/s, which is much lower than the speed of electromagnetic waves in air (3×108 m/s). This fact causes long propagation delays, which becomes a major complication for the application of feedback to correct for the channel distortions. As an example, typical propagation delays in acoustic underwater links can be on the order of several seconds, while the measured coherence time in an underwater channel can be on the order of milliseconds. In contrast with the propagation delays in underwater channels, the open-air radio propagation delay is typically of the order of microseconds.

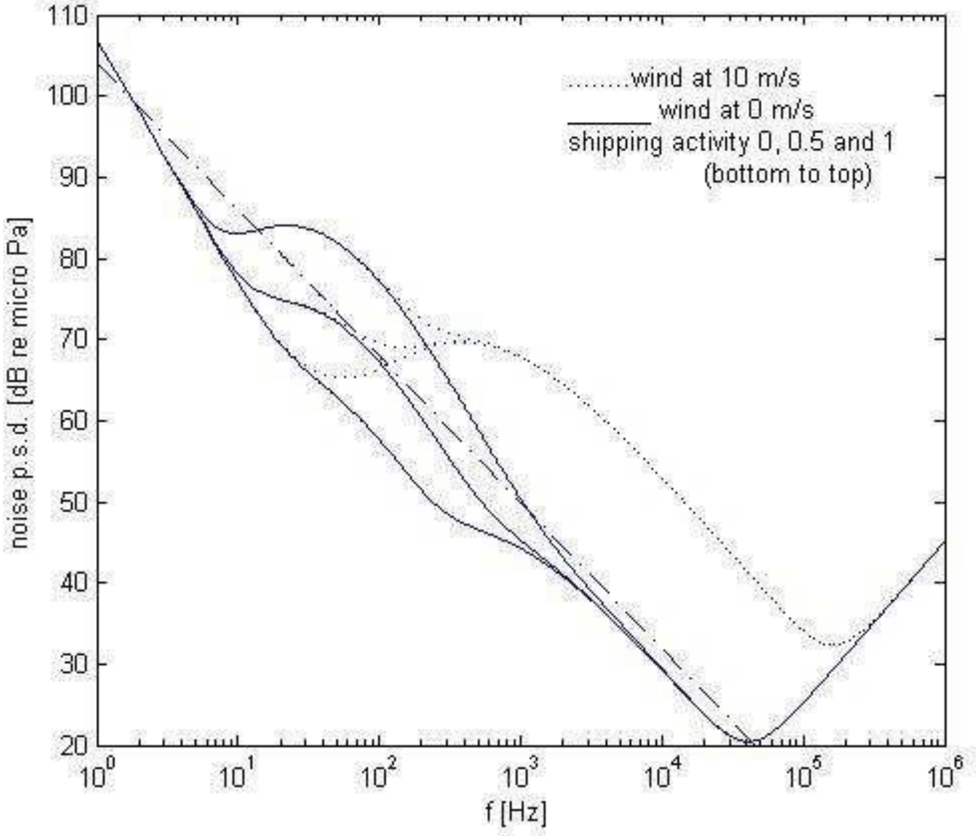

Figure 1.2: Power spectral density of ambient noise $N(f)$[$dBre\mu Pa^2$]

1.2.4 Multipath

Multipath propagation is one of the common problems in communications through underwater acoustic links. This propagation phenomenon results in communication signals reaching the receiving hydrophones by two or more paths. At the receiver, due to the presence of multiple paths, more than one pulse will be received, and each one will arrive at different times. The channel impulse response will be expressed by:

(1.8)

where the channel taps, hp, arriving at τ_p , can be described by an amplitude component ρ_p and a phase shift φ_p.

Underwater multipath can be caused either by reflection or refraction of the acoustic waves. Reflection of the acoustic waves occurs when the wave bounces either at the surface or the bottom and reaches the receiver. Refraction of the communication waves is a typical phenomenon in deep water links, where the speed of sound changes at different depths.

The distortions caused by multipath must be equalized in the receiver. In addition, ways to avoid inter-symbol interference (ISI) must be designed in order to correctly demodulate and detect the transmitted data.

1.2.5 Doppler Effect

The Doppler effect is caused by the relative motion of the transmitter-receiver pair, and it causes a shift in the frequency components of the transmitted signal. The frequency shift is mainly described by the factor v_r/c, where v_r is the relative velocity between transmitter and receiver, and c is the signal propagation speed (the speed of sound underwater in this case). In underwater environments c is approximately 1500m/s which is much lower in comparison with radio communication where c is 3×108m/s. This gives a Doppler factor 5 times order of magnitude greater than radio communication, and so the Doppler effect cannot be ignored. In addition, the fact that underwater systems are wideband causes different Doppler shifts for different frequency components of the transmitted signal. This is typically known as frequency spreading.

It is of key importance that underwater acoustic systems deal with non-uniform Doppler effect. As an example, a very high Doppler effect correction in a multicarrier system could cause inter-carrier interference (ICI), which happens when some distortion due to other subcarriers' information is present in a selected channel [4].

1.3 Resource Allocation

Taking into account the physical models of acoustic propagation loss and ambient noise, the optimal frequency allocation for communication signals can be calculated. Considering optimal signal energy allocation, such frequency band is defined so that the channel capacity is maximized [5].

The results that are assessed suggest that, despite the fact that frequency spectrum is not yet been regulated by the Federal Commission of Communications (FCC) for underwater acoustic communications, the possibilities in terms of usable frequency bands are not numerous, due to acoustic path propagation and noise characteristics.

1.3.1 The AN Product and the SNR

The narrow-band signal to noise ratio (SNR) observed at a receiver over a distance *l* when
the transmitted signal is a tone of frequency f and power P is given by

$$SNR(l, f) = \frac{P(f)/A(l,f)}{N(f)\,\Delta f} = \frac{S(f)}{N(f)/A(l,f)} \qquad (2.9)$$

where Δf is a narrow band around the frequency f, and S(f) is the power spectral density of the transmitted communication signal. Directivity indices and losses other than the path loss are not counted. The AN product, A(l, f)N(f), determines the frequency-dependent part of the SNR. The inverse of the AN product is illustrated in Figure 1.3.

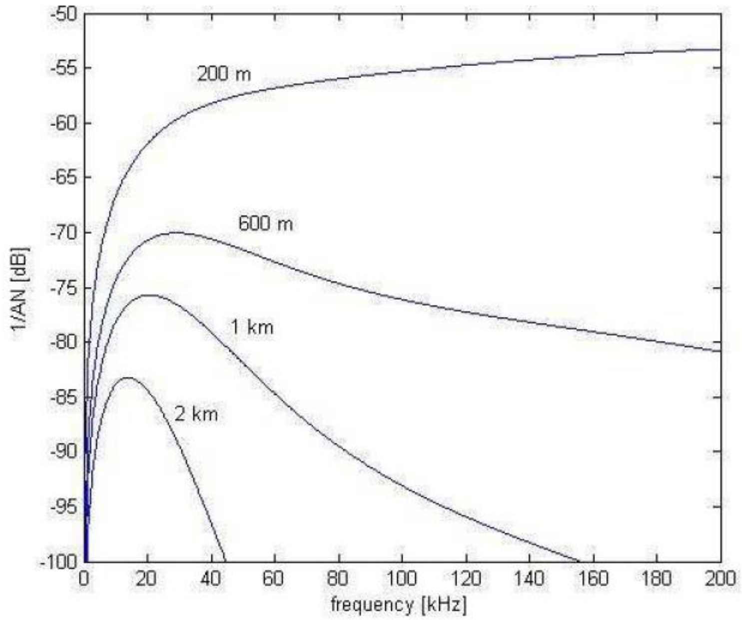

Figure 1.3: Frequency dependent part of SNR ~ 1/A(l, f)N(f), k = 1.5 is used for A(l, f).

1.3.2 Optimal Frequency

Observing the inverse of the AN product, 1/A(l, f)N(f) in Figure 1.3 , it can be concluded that for each distance *l* there exists an optimal frequency $f_0(l)$ for which the maximal narrowband SNR is obtained at the receiver. The optimal frequency is plotted in Figure 1.4 as a function of transmitter receiver distance.

When implementing a communication system, some transmission bandwidth around $f_0(l)$ is chosen. The transmission power is adjusted so as to achieve the desired SNR level throughout the selected frequency band. Practically, the response of the transducers and hydrophones must be taken into account and the optimal transmission frequency may vary.

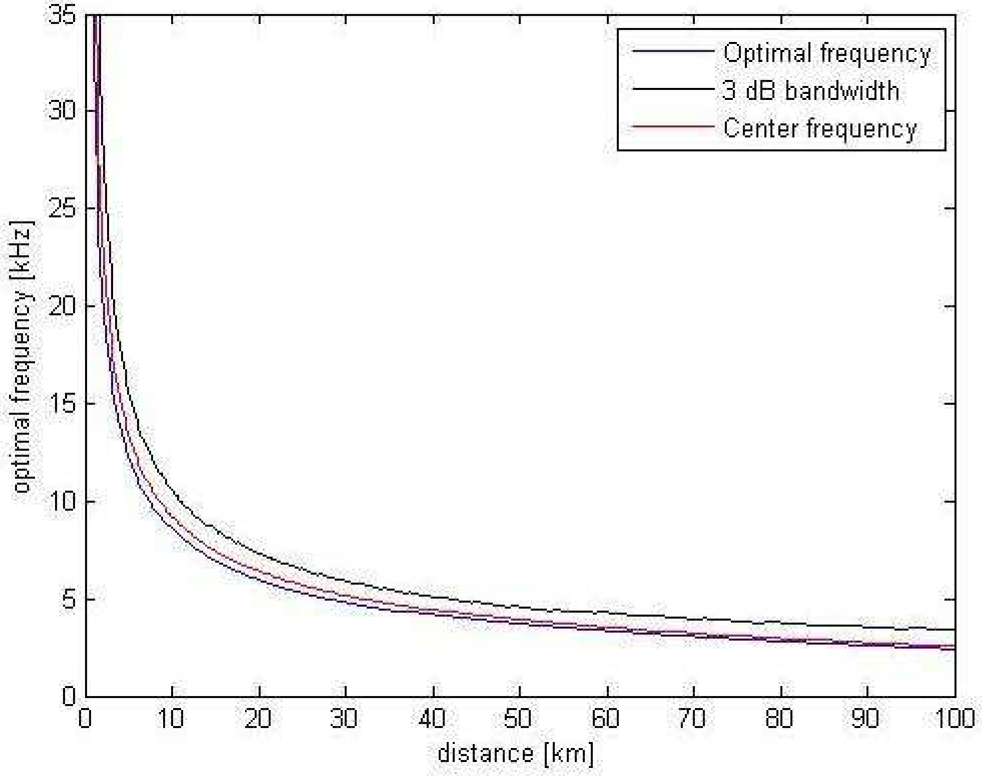

Figure 1.4: Optimal frequency plotted as a function of distance.

1.3.3 Transmission Power

Once the transmission bandwidth is set, the transmission power *P(l)* can be adjusted to achieve a desired narrow-band SNR level corresponding to the bandwidth B(l). If we denote by S_l(f) the p.s.d. of the transmitted signal chosen for the distance *l*, then the total transmitted power is

(2.10)

where the transmitted signal p.s.d. is considered constant in the signal bandwidth.

2

System Design

Multi-carrier modulation is an attractive alternative to single-carrier broadband modulation on channels with frequency-selective distortion. Research in the area of underwater acoustic communications over the past several years has resulted in demonstrating a different type of bandwidth-efficient modulation and detection method, which uses multiple carriers instead of a single carrier. In its basic form, this method is known as Orthogonal Frequency Division Multiplexing (OFDM)[6].

Rectangular pulse shaping combined with multi-carrier modulation and detection can be easily implemented using the Fast Fourier transform, which enables easy channel equalization in the frequency domain, thereby eliminating the need for potentially complex time-domain equalization of a single-carrier system. For this reason OFDM has found application in a number of systems, including the wire-line digital subscriber loops (DSL), wireless digital audio and video broadcast (DAB, DVB) systems, and wireless LAN. It is also considered for the fourth generation cellular systems, and ultra-wideband (UWB) wireless communications in general.

2.1 Basic OFDM Principle

The primary motive of transmitting the data on multiple carriers is to reduce intersymbol interference and, thus, eliminate the performance degradation that occurs in single carrier modulation. Multicarrier modulation is an approach to design a bandwidth efficient digital communication system in the presence of channel distortion, by sub-dividing the available channel bandwidth into a number of sub-channels, such that each channel is nearly ideal. Dividing the available channel bandwidth into sub-bands of relatively narrow width would result in the channel transfer function being constant inside each sub-band, eliminating the need for complex time-domain channel equalization.

OFDM is a frequency-division multiplexing scheme utilized as a digital multi-carrier modulation method. A large number of closely-spaced orthogonal subcarriers are used to carry data. The data is divided into several parallel data streams or channels, one for each subcarrier. Each subcarrier is modulated with a conventional modulation scheme (such as Quadrature Amplitude Modulation -QAM- or Phase Shift Keying -PSK-) at a low symbol rate, maintaining total data rates similar to conventional single-carrier modulation schemes in the same bandwidth. Figure 2.1 shows the utilization of the available bandwidth for a 7 sub-carrier OFDM signal.

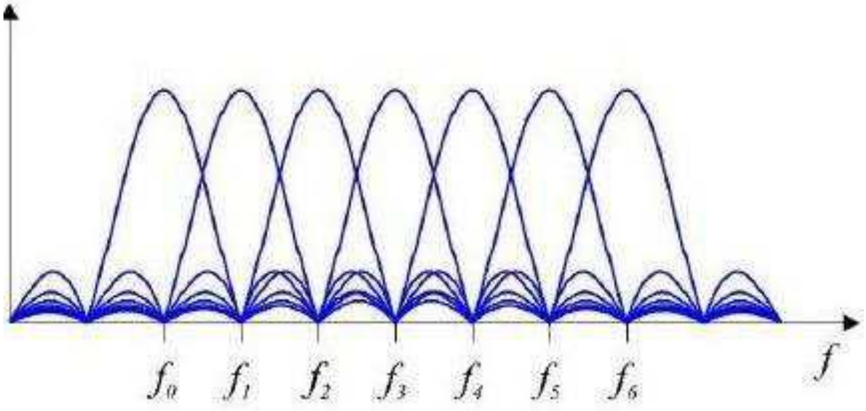

Figure 2.1: Bandwidth utilization of OFDM.

2.1.1 Orthogonality

OFDM is a special type of multicarrier modulation in which the subcarriers of the corresponding subchannels are mutually orthogonal. For an OFDM system with K subcarriers and a total available bandwidth B, the subcarrier separation $\Delta f = B/K$. If this subcarrier separation Δf is small enough, the channel frequency response $C(f)$ essentially constant across each subband.

With each subband a sinusoidal carrier signal is associated which is of the form $s_k(t) = \cos 2\prod f_k t$, where f_k is the center frequency of the k^{th} subchannel. By selecting the symbol rate $1/T$ in each of the subchannels to be equal to the frequency separation Δ*f*, the subcarriers are orthogonal over the symbol interval, independent of the relative phase between the subcarriers.

That is;

(2.1)

One of the most important advantages of orthogonality between subcarriers is that it allows high spectral efficiency, as almost the full available frequency band can be utilized. A disadvantage that results from the use of orthogonality is the need for highly accurate frequency synchronization between the transmitter and the receiver. The frequency deviation that OFDM systems can tolerate is very small, as the subcarriers will no longer be orthogonal, causing intercarrier interference, or cross-talk between subcarriers.

Frequency offsets are typically caused by Doppler shifts due to motion, or mismatched transmitter and receiver oscillators. While Doppler shift alone may be compensated for by the receiver, multipath arrivals with independent Doppler distortions makes the correction more difficult.

2.1.2 Modulation Using Fast Fourier Transform (FFT)

Due to the orthogonality of OFDM subcarriers, the modulator and demodulator can be efficiently implemented using the FFT algorithm on the receiver side, and the inverse FFT, or IFFT, on the transmitter side. On the transmitter side, the IFFT of a signal *U(k)*, where *k* denotes the frequency component index is

—

(2.2)

where K designates the number of frequency components, and u(l) is the resulting sampled signal, which is formed by the sum of the modulated frequency components U(k) (at their corresponding digital frequency k/K) (see Figure 2.2). To retrieve again the digital frequency components, the inverse equation must be used:

$$-$$

(2.3)

which corresponds to the K-point FFT of U(k).

2.1.3 Guard Time

One key principle of OFDM is that since low symbol rate modulation schemes (i.e., where the symbols are relatively long compared to the channel time characteristics) suffer less from intersymbol interference caused by multipath propagation, it is advantageous to transmit a number of low-rate streams in parallel instead of a single high-rate stream. Since the duration of each symbol is long, it is feasible to insert a guard interval between the OFDM symbols, thus eliminating the intersymbol interference. The guard interval also eliminates the need for a pulse-shaping filter, and it reduces the sensitivity to time synchronization problems.

In a cyclic prefix OFDM, the guard interval consists of the end of the OFDM symbol copied into the guard interval. The reason that the guard interval consists of a copy of the end of the OFDM symbol is because it allows the linear convolution of a frequency selective multipath channel to be modeled as a circular convolution which in turn may be transferred to the frequency domain using a discrete Fourier transform.

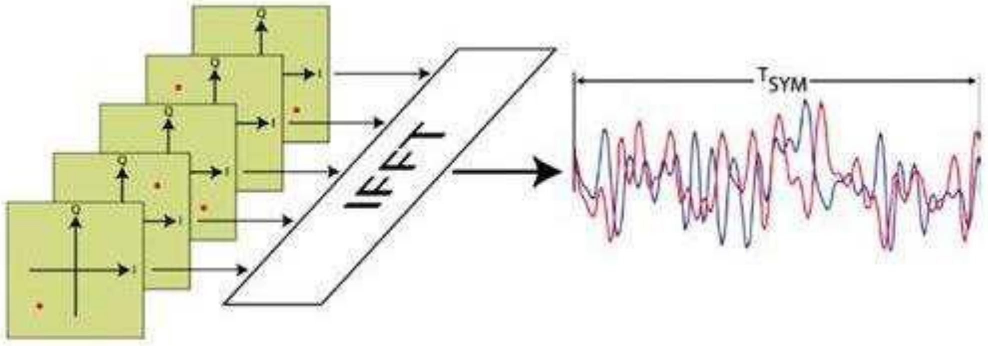

Figure 2.2: Transmitter implementation using IFFT.

2.1.4 Equalization

The effects of the channel conditions, for example fading caused by multipath propagation, can be considered as constant (flat) over an OFDM sub-channel if the sub-channel is sufficiently narrow-band. This makes equalization far simpler at the receiver in OFDM in comparison to conventional single-carrier modulation. The equalizer only has to multiply each detected sub-carrier (each Fourier coefficient) by a constant complex number [7].

If a differential detection and differential modulation (such as DPSK or DQPSK) is applied to the subcarriers, equalization can be completely eliminated, since these non-coherent schemes are insensitive to slowly changing amplitude and phase distortion.

2.1.5 Advantages

The main advantage of the OFDM modulation scheme in terms of practical implementation is that it enables channel equalization in the frequency domain, thus eliminating the need for potentially complex time-domain equalizers.

OFDM modulation techniques have been used both in wired and wireless systems due to its advantages. Among them, the following must be mentioned:

- Simple and effective channel equalization in the frequency domain.
- High spectral efficiency.
- Robustness against inter-symbol interference and fading caused by the multipath channel.
- Efficient implementation using the FFT, avoiding the need for complex subchannel filters.
- Easy equalization in the frequency domain.

2.1.6 Disadvantages

The major disadvantages of OFDM are:

- Sensitivity to frequency offsets.
- High Peak to Average Ratio (PAR), with a subsequent difficulty to optimize the transmission power.

The major difficulty in applying OFDM to an underwater acoustic channel is the signal's sensitivity to frequency offsets, which imposes strict synchronization requirements. Motion-induced Doppler effect in an acoustic channel creates a frequency offset that is not uniform across the signal bandwidth. This fact is in stark contrast to the frequency distortion in radio systems, and, hence, many of the existing synchronization methods cannot be used directly.

Instead, dedicated methods have to be designed. Such methods have been proposed over the past several years, and demonstrated good performance in initial trials with real data transmitted over a few kilometers at bit rates on the order of 10 kbps within comparable acoustic bandwidths.

3

The OFDM Implementation

The system implementation has been discussed in this section. The transmitter set-up is explained in detail followed by the receiver algorithms.

3.1 OFDM

OFDM is a modulation technique that splits a wide-band into several orthogonal narrow sub-bands. OFDM modulation sends the data over different sub-channels which are orthogonal to each other. Each sub-channel has a bandwidth less than the coherence bandwidth of the channel which is called flat-fading channel.

3.1.1 Signal Representation in OFDM

In an OFDM modulation, data is carried on narrow-band sub-carriers in frequency domain. As shown in the figure 3, the input data stream is modulated by QAM-Modulator. Assume that, the modulated complex data stream is :

$$U(0), U(1), ...U(N-1)$$

where N is the number of sub-carriers. Each sub-stream passed through S/P converter (i.e. serial-to-parallel) and was transformed into time-domain using IFFT at the transmitter. The signal after IFFT can be expressed as:

3.2 OFDM Transmitter

The most important parts of the transmitter include the symbol mapping and IFFT.

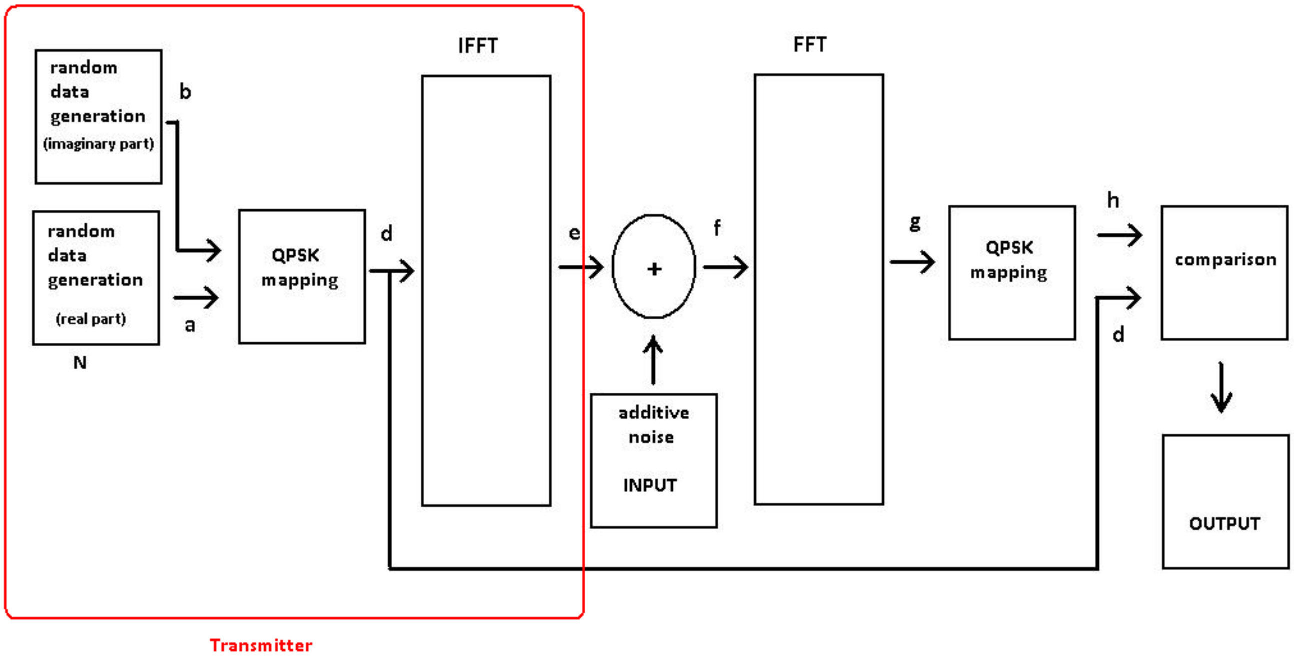

Figure 3: OFDM Transmitter Block Diagram

3.2.1 Random Data Generator:

Random data generator is used to generate a serial random data .This data stream models the raw information that going to be transmitted. The serial data is then fed into OFDM transmitter

3.2.2 Data to symbol Mapper:

This block does modulation of QPSK. The data on each symbol is mapped based on the modulation method used.

3.2.3 Channel model:

Additive white Gaussian Noise (AWGN) is a channel model in which the only impairment to communication is a linear addition of wideband or white noise with a constant spectral density (expressed as watts par hertz of bandwidth) and a Gaussian distribution of amplitude. The model does not account for fading, frequency, selectivity, interference, nonlinearity or dispersion. However, it produces simple and tractable mathematical models which are useful for gaining insight into the underlying behavior of a system before these other phenomena are considered. Wideband Gaussian noise comes from many natural sources, such as the thermal vibrations of atoms in conductors, shot noise, black body radiation from the earth and other warm objects and from celestial sources such as the sun.

3.2.4 Transmitter Implementation

The transmitter implementation is discussed in this section.

Symbol Mapping

Phase shift keying has been found to be a suitable modulation method for this project. Phase-shift keying (PSK) is a digital modulation scheme that conveys data by changing, or modulating, the phase of a reference signal (the carrier wave). In particular Quadrature Phase Shift Keying (QPSK) has been used. QPSK uses four points on the constellation diagram, equispaced around a circle. With four phases, QPSK can encode two bits per symbol, shown in the diagram with gray coding to minimize the bit error rate (BER). Figure shows the constellation of a QPSK signal.

The scaling factor of is for normalizing the average energy of the transmitted symbols to 1, assuming that all the constellation points are equally likely.

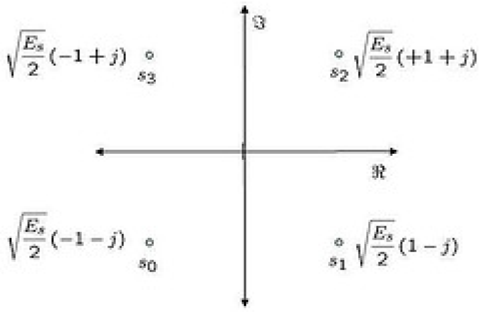

Figure 3.1: Constellation plot for QPSK (4-QAM) constellation

Noise model

Assuming that the additive noise n follows the Gaussian probability distribution function,

$$p(x) = \frac{1}{\sqrt{2\pi\sigma^2}} e^{\frac{-(x-\mu)^2}{2\sigma^2}} \text{ with } \mu = 0 \text{ and } \sigma 2 = \frac{N_0}{2}.$$

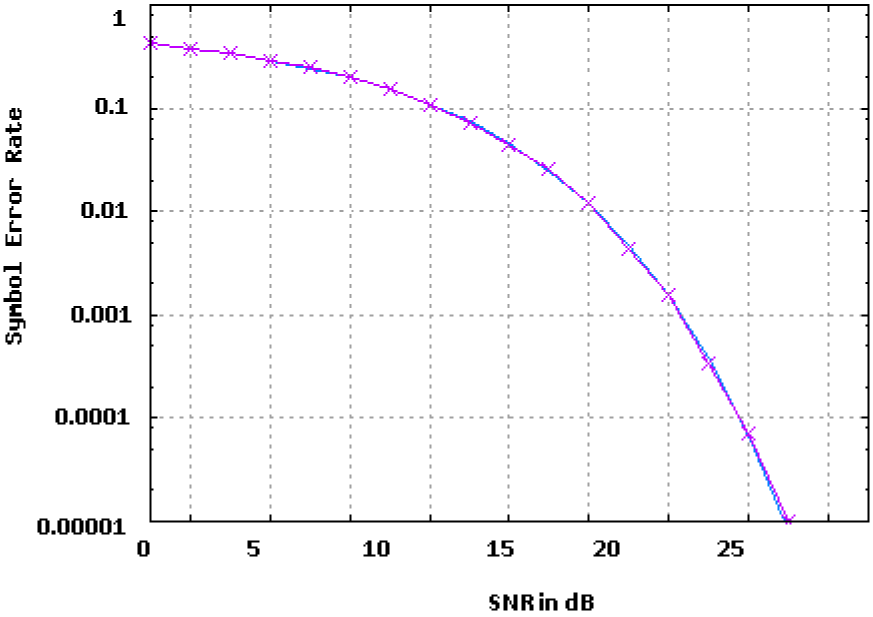

Figure 3.2: Symbol Error Rate for QPSK modulation

3.3 OFDM Receiver

The most important parts of the transmitter include the FFT and symbol de-mapping.

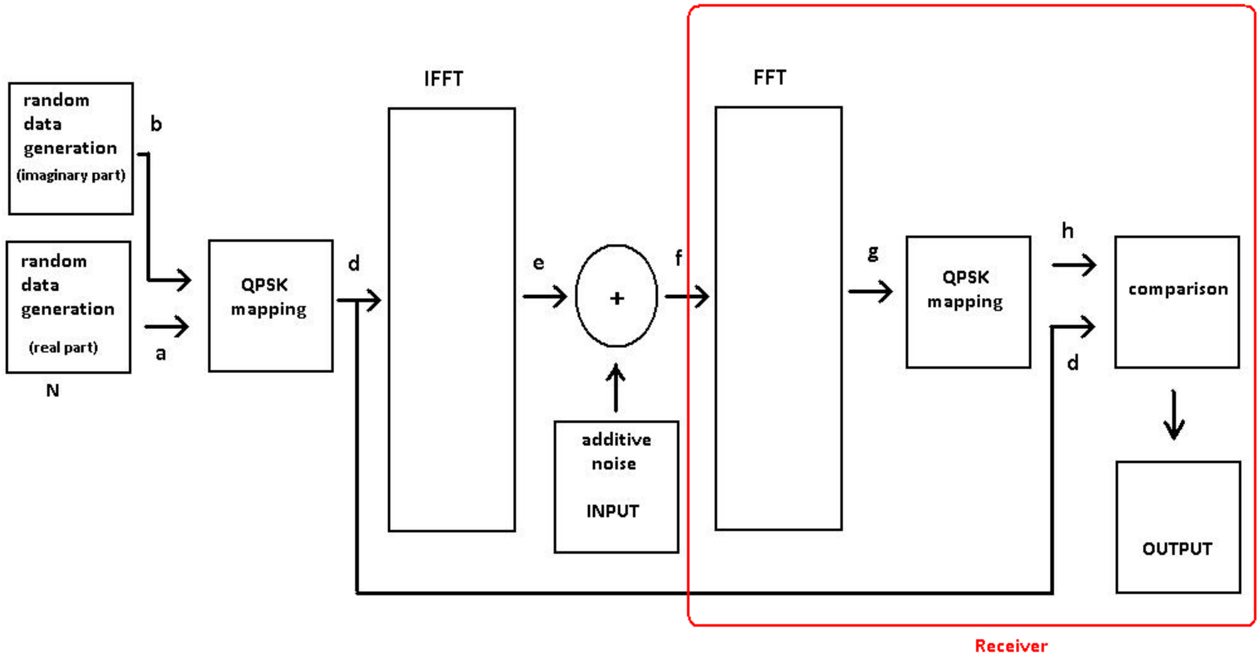

Figure 3.3: The OFDM receiver block diagram.

There is no synchronization in receiver, because of the simplicity of channel model - no delay of signal and the transmited block is directly processed in receiver. Also no channel parameters estimation and equalization is there, because the frequency characteristic of channel is flat.

3.3.1 FFT Demodulation

At this stage, the OFDM blocks have been obtained. These OFDM blocks are then passed through a FFT to get the raw constellation points.

The FFT algorithm is used to retrieve the subcarriers' received symbols before channel treatment, by using the complementary method to the IFFT modulator. The applied equation is

—

(4.3)

Signal-to-noise ratio (SNR)

Signal-to-noise ratio (SNR) is broadly defined as the ratio of the desired signal power to the noise power. SNR estimation indicates the reliability of the link between the transmitter and receiver. In adaptive system design, SNR estimation is commonly used for measuring the quality of the channel. Then, the system parameters are changed adaptively based on this measurement. For example, if the measured channel quality is low, the transmitter adds some redundancy or complexity to the information bits (more powerful coding), or reduces the modulation level, or increases the spreading rate for lower data rate transmission. Therefore, instead of fixed information rate for all levels of channel quality, variable rates of information transfer can be used to maximize system resource utilization with high quality of user experience.

CNR (carrier-to-noise ratio) and E_b/N_{0}: their relation to SNR

E_b/N_0 (the energy per bit to noise power spectral density ratio) is an important parameter in digital communication or data transmission. It is a normalized signal-to-noise ratio (SNR) measure, also known as the "SNR per bit". It is especially useful when comparing the bit error rate (BER) performance of different digital modulation schemes without taking bandwidth into account.

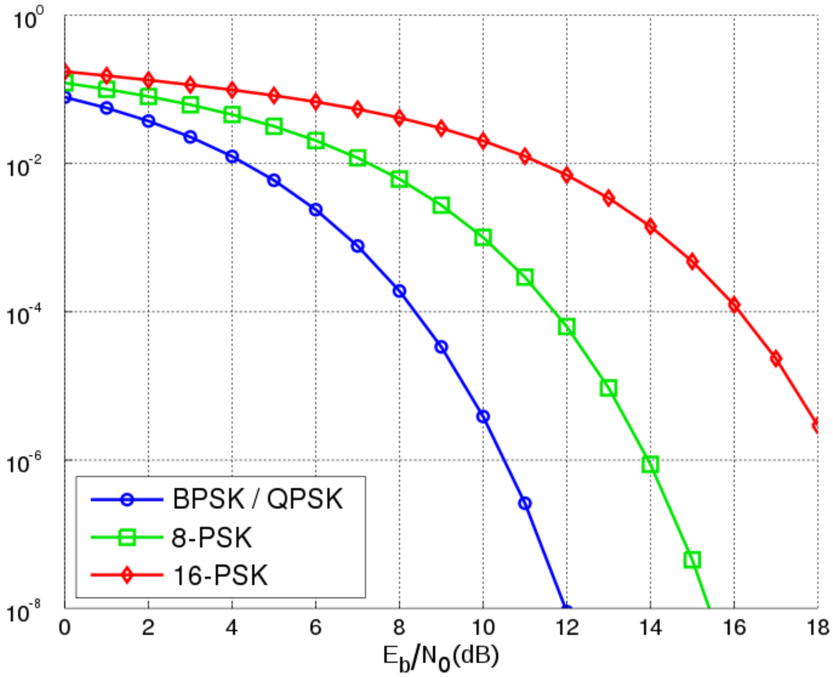

Figure 3.4: Bit-error rate (BER) vs E_b/N_0 curves for different digital modulation methods is a common application example of E_b/N_0. Here an AWGN channel is assumed

E_b/N_0 is closely related to the carrier-to-noise ratio (CNR or C/N), i.e. the signal-to-noise ratio (SNR) of the received signal, after the receiver filter but before detection:

where

f_b is the channel data rate (net bit rate), and

B is the channel bandwidth

The equivalent expression in logarithmic form (dB):

4

Simulation Results and Discussion

4.1 Coding

Based on the block diagram shown below, this section presents the coding part that is simulated in MATLAB:

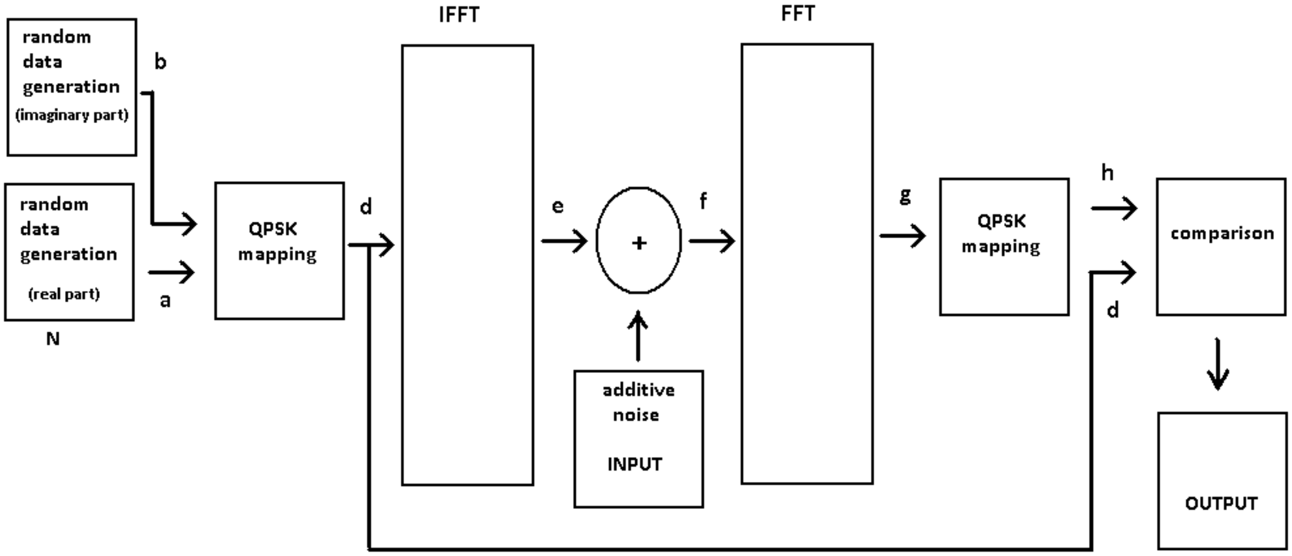

Figure 4.1: Block Diagram

```
function y=mapping (x)     % for QPSK mapping

%x = input signal of data (size 1xN)
%y = output signal of data (size 1xN)

%splitting data into real and imaginary
rex = real (x); % real part of the signal
imx = imag (x); % imaginary part of the signal
```

```
rey = quantiz (rex, [0]); % for dividing axis into two parts due to 4 QPSK


imy = quantiz (imx, [0]);  % for dividing axis into two parts due to 4 QPSK

rey = ((rey .* 2) -1) .* sqrt (2);
imy = ((imy .* 2) -1) .* sqrt (2);
y = rey + j.*imy;

%%%

% comparing correct signal with received signal
function error = compare (correct, received)

e =(correct ~= received) ; % when error is not equal to received signal

error = sum (e) ; % sum of errors

%%%

function ser = structure (snr) %main function
n = 128 ; % block size of OFDM
%snr = 10 ;
m = 10; % averaging
ser = 0 ;

for     ii=1:m

a = rand (1,n); % generating random data into real part of the signal
b = rand (1,n); % generating random data into imaginary part of the signal
a = ((a .* 2) - 1); %making bipolar signal to unipolar signal
b = ((b .* 2) - 1); %making bipolar signal to unipolar signal
c = a + j .* b ; %making complex signal

d = mapping (c) ;  %Transmitted signal before IFFT (QPSK mapping)

e = ifft (d,n) ; % OFDM Modulation
f = awgn (e, snr) ;  %Additive noise is added
```

```
g = fft( f,n); % OFDM Demodulation
h = mapping (g); % Signal after QPSK de-mapping

ser = ser + compare (h,d); %comparing signal h (Signal after QPSK de-mapping)
to signal d (Transmitted signal before IFFT)
end
ser = ser ./ m; %division in averaging
ser = ser ./ n; % division in by number of subcarriers

%%%

function loop %looping function
ii=1;
for     snr = 0:5:50 %sweeping over snr values


    SER (ii) = structure (snr);
    SNR (ii) = snr ;
    ii = ii + 1;
end
semilogy (SNR, SER) %y axis values from 0 to 1
xlabel('SNR in dB'); %labeling x axis
ylabel('SER'); %labeling y axis
```

4.2 Simulation Results and Discussion

Simulation result of the above MATLAB codes is implemented in the figure below. The x-axis represents SNR in dB and the y-axis represents SER [-].

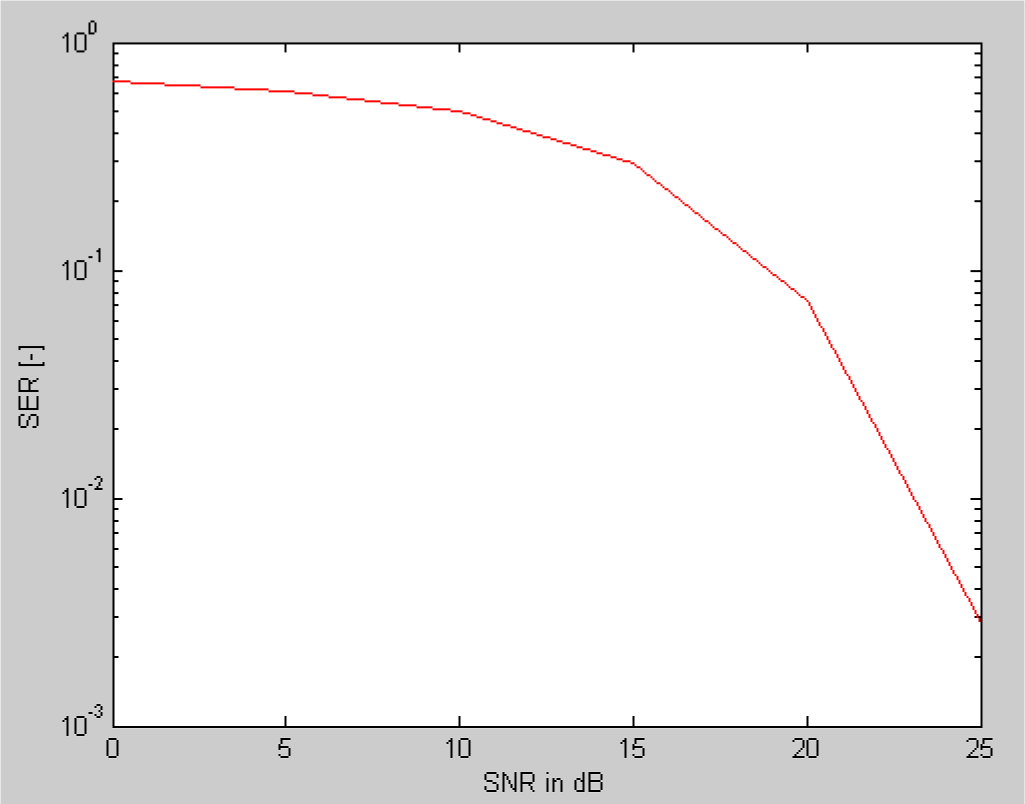

Figure 4.2: SER vs SNR (when n=128)

As seen in the figure above, the demonstrated simulation result of SER versus SNR is very close in terms of comparison to the theoretical result (see Figure 3.2: Symbol Error Rate for QPSK modulation).

In regards to OFDM block size *n*, if errors are zero, then *SER* is zero [——]. When all data is wrong, then *SER* will be equal to 1, where 1 represents 100% error.

When *SNR* is zero, it is shown that *SER* is very close to 1. Once SNR is increased from 5 to 10, 10 to 15, 15 to 20 and so on (until after SNR > 25 db where SER = 0), SER values decreases. In order words, when *SNR* increases in dB, error and *SER* is decreasing.

As shown in the Figure 4.3 below, OFDM block size *n* is 128 - illustrated with red. When OFDM block size *n* is increased to 256 (illustrated with blue) and then changed to 512 (illustrated with green), *SER* increases as well as error does increase.

For the given values;

In case of OFDM block size n=128, SNR in dB is 25 when SER is zero;

In case of OFDM block size n=256, SNR in dB is 29 when SER is zero;

In case of OFDM block size n=512, SNR in dB is 31 when SER is zero.

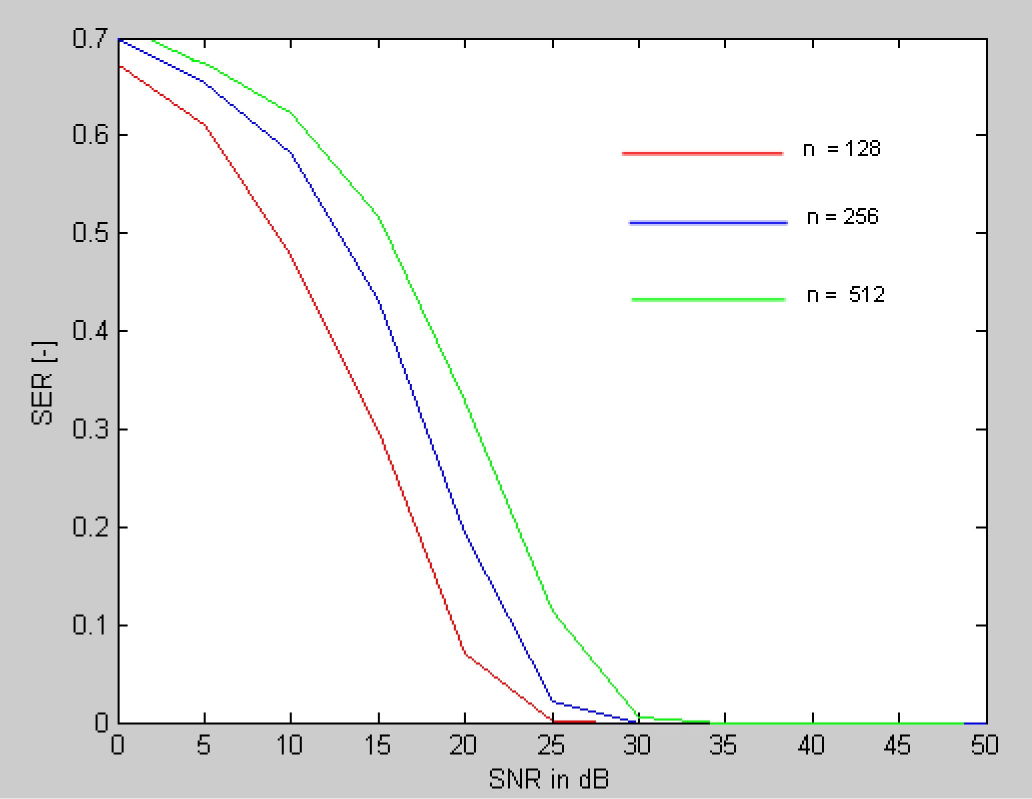

Figure 4.3: SER vs SNR (when n= 128, n=256, n=512)

As seen above, when subcarriers *n* increases, error increases – and in relation to that *SER* increases.

Conclusion

A study experiment, as well as a study about OFDM modulation for underwater communication system, is made in this paper in regards to modern modulation method for underwater communication systems. The modulator and demodulator structure is shown and a study experiment made for modern modulation method for underwater communication systems. The symbol-error-ratio (SER) is considered as the main parameter. The simulation is performed as an SNR vs SER comparison for transmission and reception of modulated signals.

As a result of the first simulation part, it was shown that the demonstrated simulation result of SER versus SNR is very close in terms of comparison to the theoretical result (shown in Figure 3.2: Symbol Error Rate for QPSK modulation).

As a result of the second simulation when OFDM block size subcarriers n increases, error increases and SER increases. In order to have (SER=0), SNR should be increased. In other words, the more sub-carriers, the more power is to spend. Therefore, there is a higher SNR values needed in order to keep the same quality. We can also say that the smaller the subcarriers, the less the error is. it is found out that data increases, length of symbol gets bigger in time domain (more data, longer) and also velocity of wave increase.

In addition, some parts of this paper may not be full-filled according to the task that was given for this project due to the lack of the time. However, hopefully this may be completed in some future work.

Abbreviations

AWGN : Additive white Gaussian Noise

CNR: Carrier-to-Noise Ratio

FFT : Fast Fourier Transform

IFFT : Inverse Fast Fourier Transform

OFDM: Orthogonal Frequency Division Multiplexing

SER: Symbol Error Rate

SNR: Signal-to-Noise Ratio

PSK: Phase Shift Keying

QAM : Quadrature Amplitude Modulation

QPSK: Quadrature Phase Shift Keying

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