Design and Analysis of Low Frequency Transmitter and Receiver for an Insecure Shallow Water Acoustic Channel of the Persian Gulf

A. Doosti Aref, M. J. Jannati, V. Tabataba Vakili

***Abstract*— underwater acoustic communications presents unique challenges that are being overcome with advances in signal processing algorithms and related hardware technologies. The accurate simulation and performance comparison of various algorithms is essential for building an optimized and robust communications system. This paper describes part of a project designed , based on the results obtained from practical measurements in The Persian Gulf, to examine an underwater acoustic communication system in The Persian Gulf. Transmitted data are acoustic signals to which for more safety in transmission and low frequency bandwidth, a cryptography algorithm and speech coding are applied. In transmitter, quadrature phase shift keying (QPSK) signaling is employed to make efficient use of the available channel bandwidth. In this case, data are transmitted at a rate of 2.4kbps with a carrier frequency of 27 kHz for a maximum range of 2km. In the channel modeling, a comprehensive model for short-range shallow water acoustic channel has been used whose the mathematical modeling of the multipath effects is based on the ray tracing method. In the receiver, decision feedback equalizer (DFE) is applied.**

***Index Terms*— Underwater Acoustic Communication, Persian Gulf, Linear Prediction Coder-10, Decision Feedback Equalizer, Rivest Cipher, QPSK Modulation.**

I. INTRODUCTION

The need for underwater wireless communications exists in applications such as collection of scientific data recorded at ocean-bottom stations and by unmanned underwater vehicles, speech transmission between divers, etc. UWC can be established by transmission of acoustic waves. Radio waves are of little use because they are severely attenuated, while optical waves suffer from scattering and need high precision in pointing the laser beams. Seawater acts as an acoustic waveguide in which sound waves travel. The sound channel, as a sound waveguide, is a channel with random parameters; however, this does not mean that its behaviour is unpredictable. The most important characteristic of the seawater is its inhomogeneous nature, which on the whole, can be classified into regular and random varieties. Regular variations of sound speed in different layers of water lead to the formation of sound channels and this phenomenon facilitates long distance sound propagation. Random inhomogeneities cause the scattering of sound waves and result in sound field fluctuations. This paper is organized as follows. In the second section, transmitter is designed. Because of the strong frequency attenuation, channel bandwidth is limited, therefore in transmitter, we have used LPC-10 (Linear Prediction Coder-10) algorithm to compress speech signal. After that, for more safety in transmission, RC5 (Rivest Cipher) cryptography algorithm has been used to encrypt data. Then, for reduction of bit error rate (BER), channel coding is used. In the last section of transmission, data is modulated. In the third section, a new model for the Persian Gulf channel has been used that we have deeply described it in [1]. In the fourth section, the block diagram of the receiver and its performance are discussed. Finally, the simulation results related to the transmitter, the devised channel and the receiver in the Persian Gulf are presented.

II. TRANSMITTER

On the basis of extensive laboratory and field experiments and the results obtained from deferent simulations, to improve the bandwidth efficiency, using the coherent modulation methods such as Quadrature Amplitude Modulation (QAM) and phase shift keying (PSK) is the best approach in underwater operations [2]. Depending on the method for carrier synchronization, phase-coherent systems fall into two categories; differentially coherent and purely phase coherent. The advantage of using differentially coherent detection is the simple carrier recovery which it allows. Its disadvantage is performance loss as compared to coherent detection.

While bandwidth-efficient methods have successfully been tested on a variety of channels, the real-time systems have mainly been implemented for applications in vertical and very short range channels, where little multipath is observed and the

Manuscript received April 4, 2011.

A. Doosti Aref was with Malek-Ashtar University of Technology, Tehran, Iran. E-mail: doosti.aref@ ieee.org.

M. J. Jannati was with the University of Science and Technology, Tehran, Iran. E-mail: mjannati@ee.iust.ac.ir.

V. Tabataba Vakili is with the Electrical Engineering Department, University of Science and Technology, Tehran, Iran. E-mail: vakily@iust.ac.ir.

phase stability is good [2]. In this paper, for the purpose of compensating for the multipath effects and inter-symbol interference (ISI), since the considered channel is shallow and horizontal and the QPSK modulation method in comparison with other methods has proper bit error rate, despite low bandwidth, we have used the QPSK modulation method, which is purely coherent.

The block diagram of the transmitter is shown in Fig.1. This transmitter includes blocks for producing the QPSK symbols. The resulting QPSK symbols are then passed through a pulse shaping filter. The rectangular pulses are not practical to send and require a large bandwidth. Hence, we replace them with shaped pulses that convey the same information but use smaller bandwidth and have other good properties such as inter symbol interference rejection. In continuation, we completely explain each of the blocks.

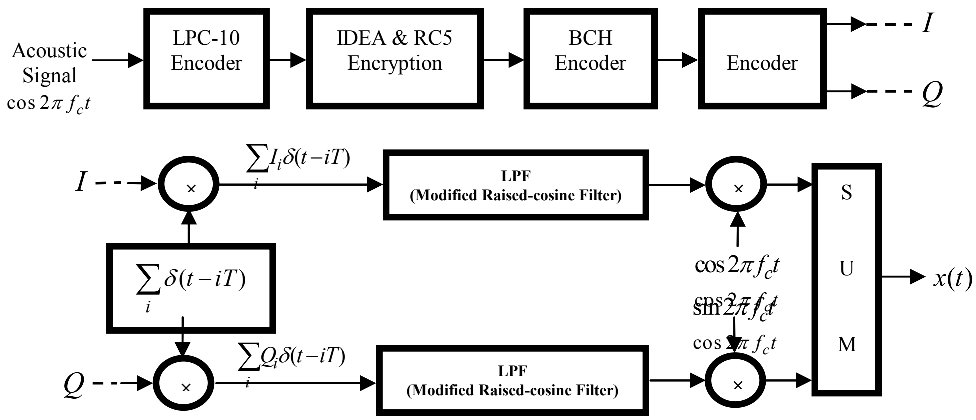


Fig. 1. Block diagram and structure of the transmitter.

phase stability is good [2]. In this paper, for the purpose of compensating for the multipath effects and inter-symbol interference (ISI), since the considered channel is shallow and horizontal and the QPSK modulation method in comparison with other methods has proper bit error rate, despite low bandwidth, we have used the QPSK modulation method, which is purely coherent.

The block diagram of the transmitter is shown in Fig.1. This transmitter includes blocks for producing the QPSK symbols. The resulting QPSK symbols are then passed through a pulse shaping filter. The rectangular pulses are not practical to send and require a large bandwidth. Hence, we replace them with shaped pulses that convey the same information but use smaller bandwidth and have other good properties such as inter symbol interference rejection. In continuation, we completely explain each of the blocks.

A. Acoustic Signal Processing

In the first block of transmitter, acoustic signal is compressed with LPC-10 algorithm. Sampling frequency is 8 KHz and for any sample, 8 bit is appropriated. Therefore sampling bit rate is 64 kbps. LPC-10 algorithm reduce bit rate to 2.4 kbps, consequently small bandwidth is needed and frequency attenuation is reduced [3],[4].

B. Cryptography

In the second block, data is encrypted with RC5 algorithm. RC5 is a symmetric block cipher designed to be suitable for both software and hardware implementation.

It is a parameterized algorithm, with a variable block size, a variable number of rounds and a variable-length key. This provides the opportunity for great flexibility in both performance characteristics and the level of security [5]. In this paper, block size has been selected 32, number of rounds are 16 and key length is 10.

C. Channel Coding

This block is needed for error correction in channel. BCH (Bose, ray-Chaudhuri, Hocquenghem) coding has been selected for channel coding. The inputs of applied BCH code are 12 bits that convert to 32 bits and are transmitted in channel. This code can correct 5 or fewer random errors in receiver [6]. In each data transmission, a training sequence to which allocates 10% of the first transmitted sequences itself is multiplexed with the data sequences before the QPSK modulation. The main purpose of the training sequence is to provide the receiver with a known sequence which can be used for phase estimation and synchronization in decision feedback equalizer.

D. Modulation

The encoder accepts the sequence of the input binary data. It has two outputs; in-phase (I), and quadrature (Q). For each distinct pair of input binary data a unique combination of I $=\{\pm 1\}$ and Q $=\{\pm 1\}$ is assigned. we consider each QPSK symbol as a complex number (I+jQ), whose real and imaginary components are the outputs of the in-phase and quadrature channels, respectively, to describe those four points separated in the complex plane by $\frac{\pi}{4}$, $\frac{3\pi}{4}$, $\frac{5\pi}{4}$, and $\frac{7\pi}{4}$. The encoded data stream of I and Q is then used to modulate a sequence of impulses in which are Transmitted every signaling period; T. To limit their bandwidth such modulated sequences are then filtered by LPFs. The same low pass filters are applied at the receiver. The in-phase and quadrature signals at the output of low pass filter are:

$$S_I(t) = \sum_i I_i h_H(t-iT) \tag{1}$$

$$S_Q(t) = \sum_i Q_i h_H(t - iT) \qquad (2)$$

That $h_H(t)$ is the impulse response of the low pass filter. The filtered signals are then multiplied by a carrier frequency, added and transmitted through the Persian Gulf underwater acoustic channel.

To avoid the disadvantages of the side lobes and to reduce the ISI we transmit data with shaped pulses instead of rectangular pulses. Therefore, the obtained QPSK symbols are passed through a modified raised-cosine filters (LPFs) with a roll-off factor $\beta = 1$ and impulse response; $h_H(t)$. The transfer function of the baseband channel has the form:

$$H(f) = \sqrt{T}\cos(\frac{\pi f T}{2}); \quad |f| < \frac{1}{T} \qquad (3)$$

That T=1/1200s. The transfer function of both the transmitter filter and receiver filter is then $\sqrt{X(f)}$, and the impulse response of the assumed non-fading part of the baseband channel is:

$$x(t) = \int_{-\infty}^{+\infty} X(f) e^{j2\pi f t} df \qquad (4)$$

Thus the impulse response of each filter is obtained through the inverse Fourier transform of (2), which is:

$$h_T(t) = \frac{1}{\sqrt{T}}\left(\frac{\sin\phi}{\phi} + \frac{\sin\varphi}{\varphi}\right) \qquad (5)$$

Where

$$\phi = \pi(\frac{4t+T}{2T}) \qquad \varphi = \pi(\frac{4t-T}{2T})$$

Fig. 2 shows the impulse response of the raised cosine filter. To reduce side lobe levels, the impulse response of the filter is modified by multiplying it with a Hamming window given as:

$$w_H(t) = 0.76 + 0.39\cos(\frac{\pi t}{T}) \qquad (6)$$

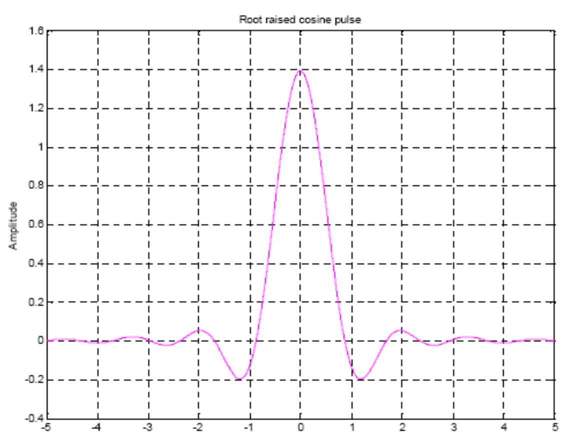


Fig. 2. Impulse response of the pulse shaping filter

The period of Hamming window is $2T$ that T is the period of each symbol. This window has 99.96% of its energy in the main lobe, with side lobes of over 20 dB down from the peak [7],[8]. Thus Impulse response of filters is corrected:

$$h_H(t) = h_T(t).w_H(t) \qquad (7)$$

We use this modified raised cosine filter in the transmitter and receiver. In Fig. 3, for the assumed sequence of "00110001101111011", the output of the I, Q channels, and the QPSK signals are shown.

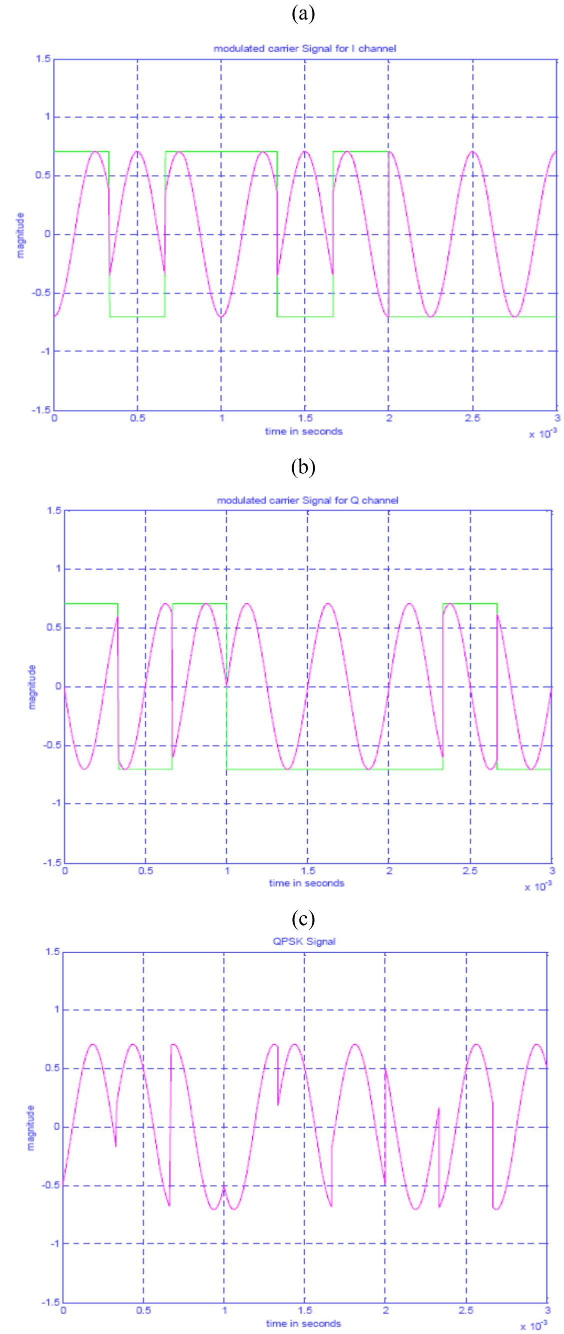


Fig. 3. (a). I channel - (b). Q channel (c). - QPSK signal

III. Channel

The model considered for channel is based on the performed laboratory measurements and results which we have deeply described in [1]. Hence, in loss modelling, the attenuation due to the absorption effects of boric acid ($B(OH)_3$), magnesium sulphate ($MgSO_4$), and pure water (H_2o) is considered and the total loss is the sum of individual losses due to each material. The mathematical modelling of the multipath effects is on the basis of ray tracing method and image theory [9]-[14]. In the modelling of surface scattering and bottom reflections, to calculate the loss due to wave scattering in the surface, we use the probability density function of the Gaussian Normal distribution for the surface displacement variable, and likewise, for the calculation of bottom reflection coefficient, we use the Jackson pattern to select the bottom water type which was simulated based on the Strait of Hormuz conditions and the Hamilton-Bachman model [15]. The model considered for noise is the combination of ambient noises such as, thermal noise, shipping noise, sea state noise, turbulences and the non-Gaussian impulsive noise, due napping shrimps, which is very dominant [16]-[22].

For the Persian Gulf channel, according to Fig. 4, we concluded that from the fifth path on, due to strong attenuation of the transmitted wave, there was no signal reception. Hence, a four-path channel pattern is suitable.

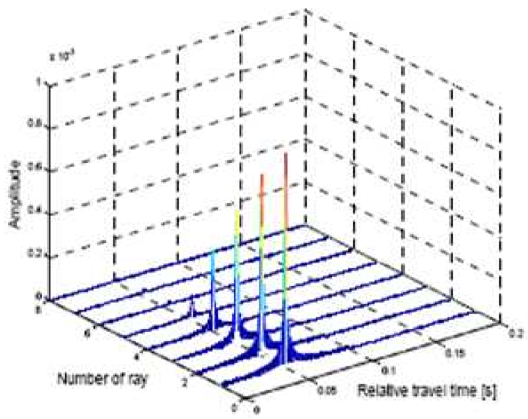


Fig. 4. Received signals from different paths of channel. In this model, channel has 8 paths. The first signal is for the direct path and the delays of other paths are calculated based on the travelling time of the first path.

IV. RECEIVER

Digital acoustic communication is limited by severe ISI associated with shallow underwater channels. The ISI is caused by multipath propagation resulting from surface and bottom reflections. To overcome the effects of the ISI; an adaptive equalization method employing a Mean Square Error (MSE) criterion is introduced. Computer simulation is carried out to verify the effectiveness of the equalization scheme for high data rate communication. In the selection of an equalizer we consider convergence rate, processing complexity and the ability to track changes in the channel characteristics. There are a variety of algorithms which have been investigated to be used in underwater acoustic communication. In this paper we have applied DFE equalizer for acoustic signal transmitting in the Persian Gulf channel. A decision feedback equalizer is a nonlinear equalizer that contains a forward filter and a feedback filter. The forward filter is fractionally spaced with spacing *T/2*; while the feedback filter contains a tapped delay line whose inputs are the decisions made on the equalized signal. The purpose of a DFE is to cancel inter symbol interference while minimizing noise enhancement. Fig. 5.a. shows structure of decision feedback equalizer.

The criterion used in the optimization of the equalizer coefficients is the minimization of the mean square error (MSE) by use of the least mean square (LMS) algorithm. The output of DFE at $t=kT$ is given as [23]-[25]:

$$z(kT) = \sum_{n=0}^{N-1} c_n(kT) y(kT - n\frac{T}{2}) - \sum_{m=1}^{M} b_m(kT) d(KT - m) \quad (8)$$

and error is:

$$e(kT) = d(kT) - z(kT) \quad (9)$$

New set of coefficients is obtained iteratively as:

$$c_n\left[(k+1)T\right] = c_n(kT) + \Delta e(kT) y^*(kT - \frac{nT}{2}) \quad (10)$$

$$b_m\left[(k+1)T\right] = b_m(kT) + \Delta e(kT) d^*(kT - mT) \quad (11)$$

Structure of the receiver has been shown in Fig. 5.b. We can replace adaptive equalizer block with nonlinear equalizer (DFE). DFE has 84 weights in forward filter and 42 weights in backward filter.

V. SIMULATION RESULTS

In this section, the simulation results are presented. For the channel whose speed profile is shown in Fig. 6, the simulated channel characteristics are given in Table 1.In this case, the transmitter and receiver use the QPSK modulation with a bandwidth of 5 KHz and a carrier frequency of 27 KHz. Also, the transmitter and receiver are placed at a depth of 5m and 70m from the surface, respectively. Fig. 7 illustrates the constellation diagram of QPSK symbols and the power spectrum density of the band pass signal with a 27 KHz carrier frequency before the transmission through the channel.

TABLE1

Channel characteristics based on the data measured at 56.7°E and 25.4°N by the ADCP measurement tool aboard the NOAA submarine, in the August of 2009.

Characteristic	Value
Depth	85 m
Range	2000 m
Temperature	33.5609 °C
Salinity	38.3773 ppt
PH	7.2

The constellation diagram of the received QPSK symbols after entering the fading channel is obtained according to Fig. 8. As shown, the amount of Inter Symbol Interference is high in this figure.

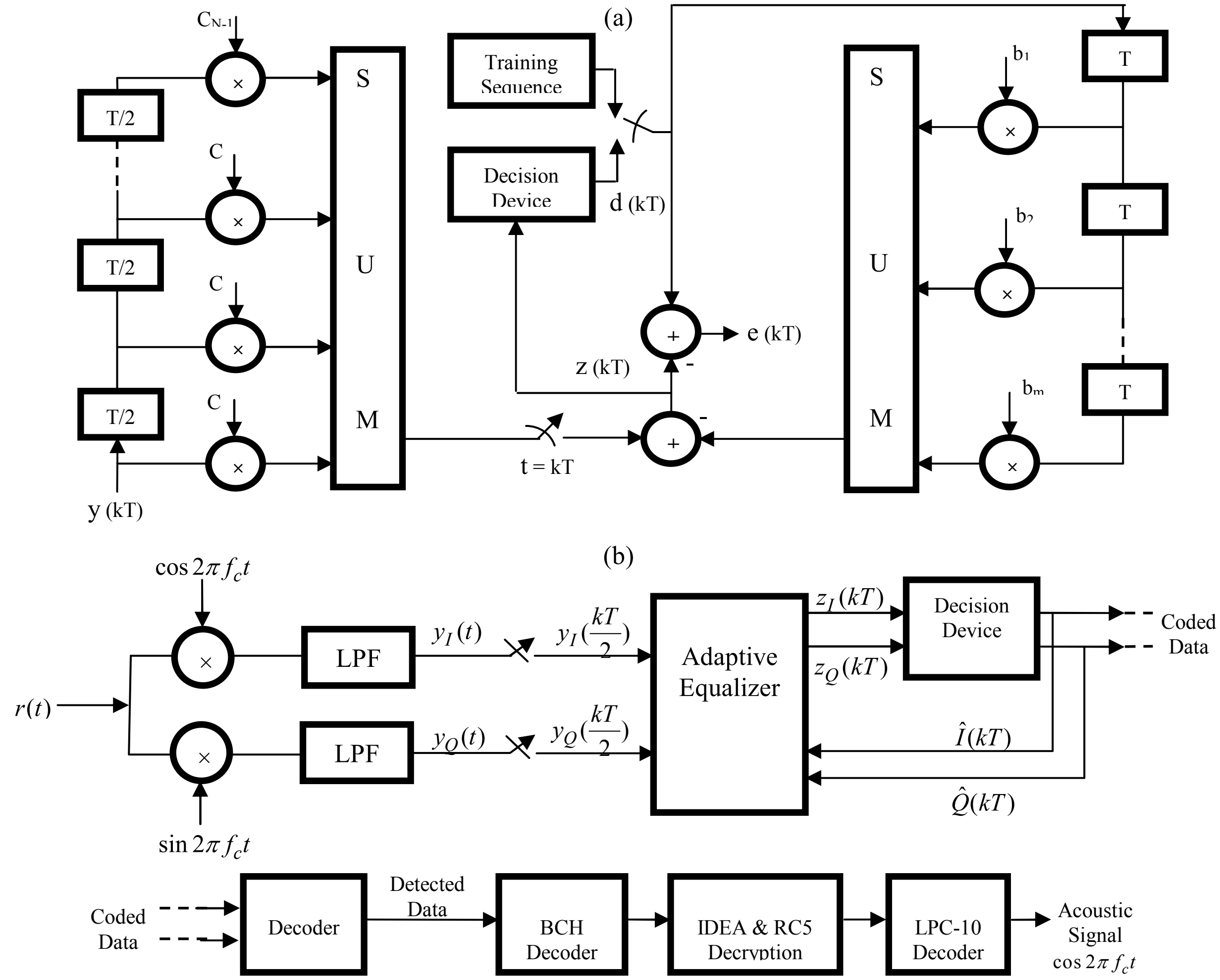


Fig. 5. (a): Decision Feedback Equalizer. (b): Block diagram and structure of the receiver

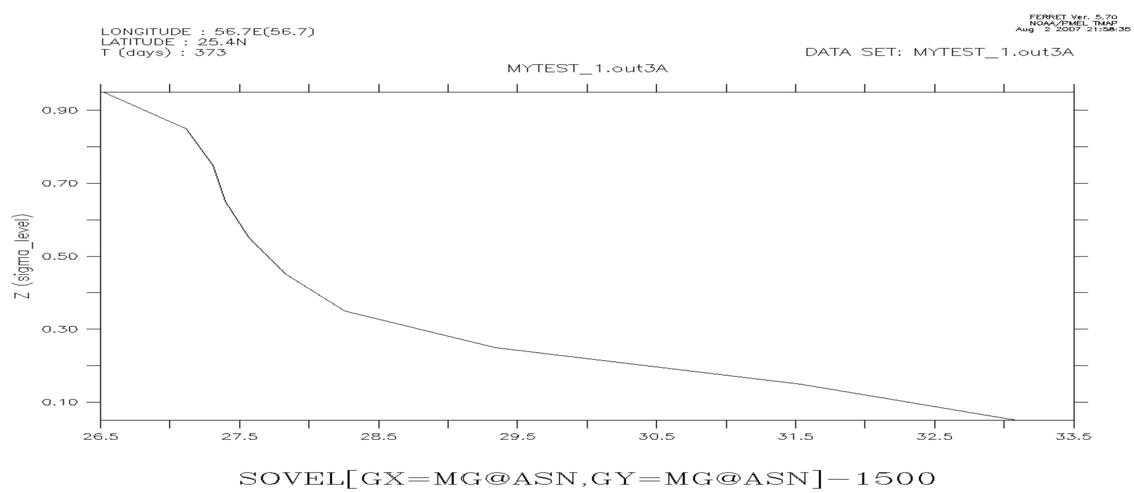
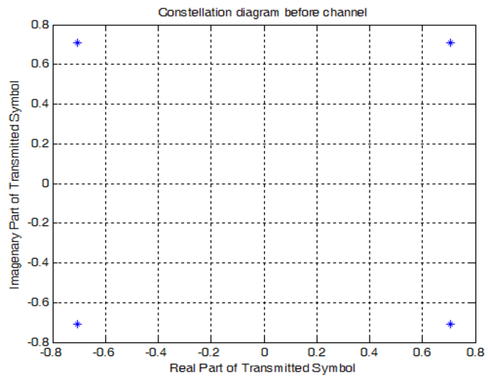


Fig. 6. a profile of sound speed variation with depth for the Strait of Hormuz, in 56.7°E (Lon) and 25.4°N (Lat). The water depth was 85m, and the ADCP measurement tool, belonging to the NOAA submarine, was placed at a depth of 10 m.

Also, Fig. 9 illustrates the power spectrum density of the transmitted signal in the channel for each of the special paths. As expected, in the RSRBR path (Fig.9.d), the largest attenuation takes place, and the power level in this path experiences a 23dB loss, in comparison with the direct path. In addition, the signals received at the channel exit from each of the 8 paths are shown in Fig. 4. It can be seen that for the fifth path and the remaining paths after it, the signal is strongly attenuated.

Fig. 10 shows the bit error rates of the received signals at different signal to noise ratio (SNR). In the simulation, the number of transmitted bits in each repeat is 5000, and the

(a)



(b)

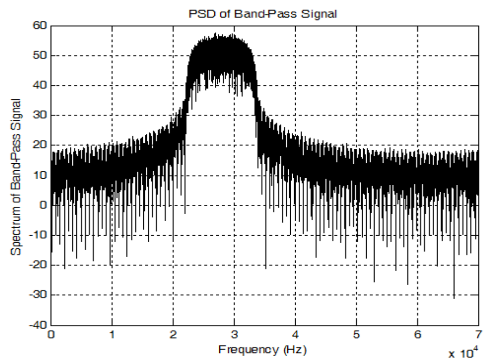


Fig. 7. (a): Constellation diagram of QPSK symbols. (b): power spectrum density of band pass signal before entering the channel

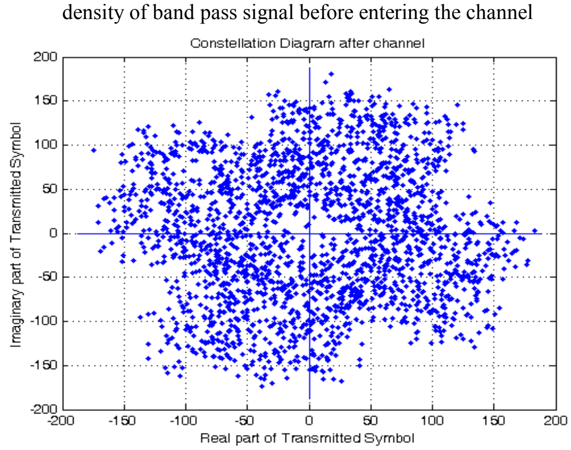
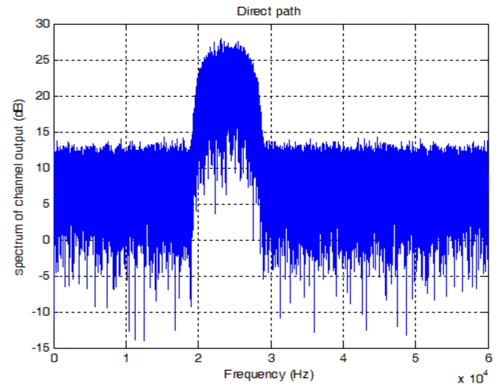


Fig. 8. Impulse response of the pulse shaping filter

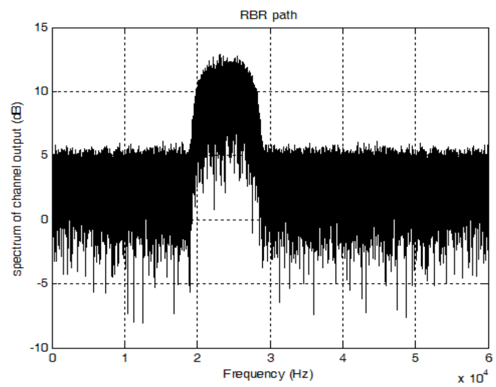
repeat number of training sequence for obtaining the BER in a specified SNR is 100. Also 2500 symbol is applied for equalizer training. For convergence, the equalizer requires over 120 iterations.

Fig. 11 shows The constellation diagrams of symbols detected by the simple receiver which had been used in [1], in this case we concluded that in applications in which the SNR was more than 6dB, (Fig. 11.a), the implementation of a transmitter and receiver using the QPSK modulation for the mentioned channel was appropriate and for lower SNR, e.g. 3dB or less, (Fig. 11.b), the use of an equalizer to compensate the ISI is necessary. On the other hand, Fig. 12 shows the constellation diagram of symbols detected by the receiver that has been described in this paper which has a DFE in its structure. As a result, the SNR is improved, as well as the constellation diagram is clearer than Fig. 11.b.

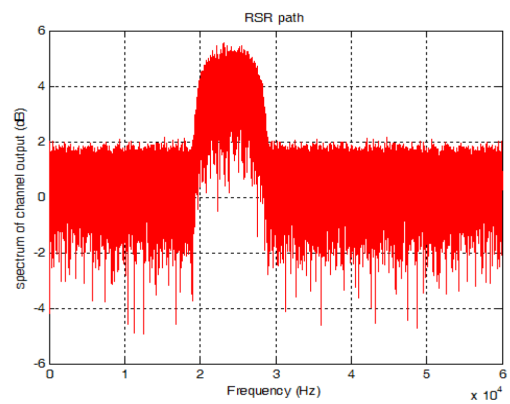
(a)



(b)



(c)



(d)

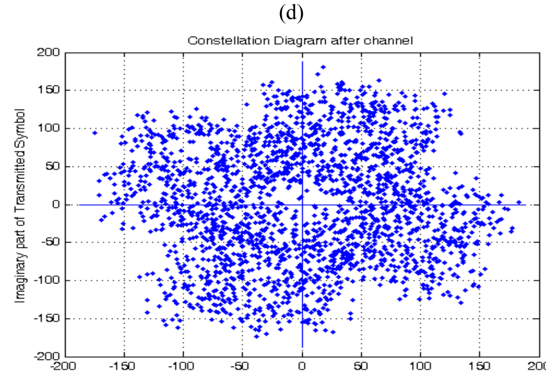


Fig. 9. Power spectrum density of the four special paths used in the simulation. (a): Direct Path (DP). (b): Refracted Bottom Reflected (RBR). (c): refracted Surface Reflected (RSR). (d): Refracted Surface Reflected Bottom Reflected (RSRBR).

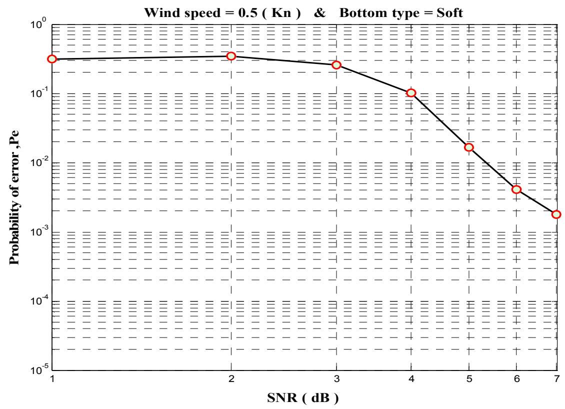
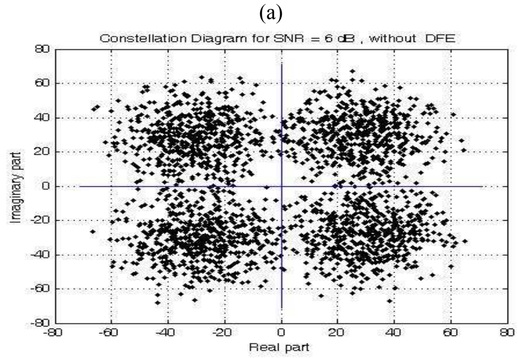


Fig. 10. Probability of error of detected symbols versus SNR under the condition of wind speed=0.5Kn, and soft bottom

(a)



(b)

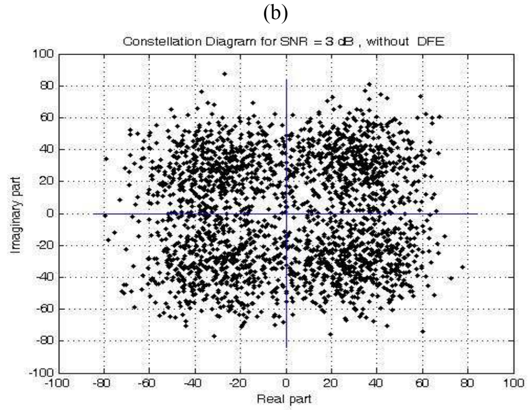


Fig. 11. The constellation diagram of symbols detected by the simple receiver that has no DFE in its structure

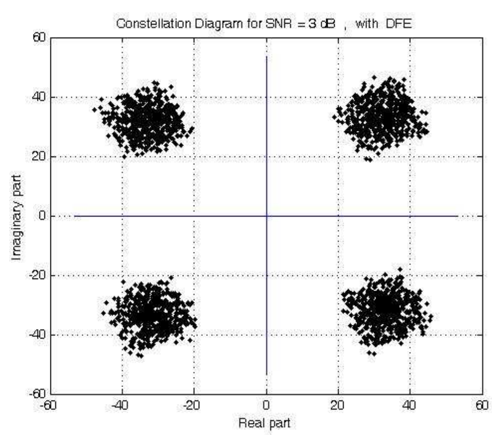


Fig. 12. The constellation diagram of symbols detected by the receiver that has a DFE in its structure for SNR=3 dB

I. CONCLUSION

Based on the results obtained from extensive field experiments in the Persian Gulf and the simulations carried out in this paper, using LPC-10 algorithm to compress acoustic signals and RC5 cryptography algorithm, because of the opportunity for great flexibility in both performance characteristics and the level of security, is suitable in transmitter. In addition, employing a DFE with mentioned structure in receiver has a good performance in the Persian Gulf.

ACKNOWLEDGMENT

We really appreciate the efforts by the personnel of the Hydro-physics research institute of Shiraz for preparing and providing us with the experimental measurements.

REFERENCES

[1] **A**. Doosti Aref and B. Abbasi Arand, "Design and simulation of a new model for shallow water multipath acoustic channel in the Persian Gulf," presented at the 5th International Symposium on Telecommunications, Iran, December 2010.

[2] M. Stojanovic, "Recent Advances in High-Speed Underwater Acoustic Communications," IEEE Journal of Oceanic Engineering, Vol.121, No.2, pp.125-136, April 1996.

[3] A. S. Spanias, " Speech Coding," Proc. IEEE, Vol. 82, No. 10, pp. 1541-1582 Oct. 2001.

[4] T. Painter and A. S. Spanias, "Perceptual Coding of Digital Audio," Proc. IEEE, Vol. 88, No. 4. pp. 451-515, Apr. 2002.

[5] M. Y. Rhee, *Internet Security*, Vol. 2, New York: John Wiley & Sons Ltd, 2003, 1950, p. 123.

[6] B. Stephen, *Error Control Systems for Digital Communication and Storage*, Upper Saddle River, N.J., Prentice Hall, 1995, p. 110.

[7] M. C. Jericho, P. Balaban and K. S. Shanmugan, *Simulation of Communication Systems*, Plenum Press, New York, 1992, p. 38.

[8] A. Falahati, B. Woodward and S. C. Bateman, "Underwater Acoustic Channel Models For 4800 b/s QPSK Signals," IEEE Journal of Oceanic Engineering, Vol. 16, No. 1, pp. 12-20, Jan.

[9] L. M. Brekhovskikh, Yu. P. Lysanov, "Fundamentals of Ocean Acoustics'', second edition. Springer- Verlag. pp. 130-155, 1999.

[10] K. C. Hegewisch, N. R. Cerruti and S. Tomsovic, "Ocean acoustic wave propagation and ray method correspondence: internal wave fine structure," The *Journal of the Acoustical Society of America*, Vol. 117, No.3, pp. 1582-1594, Oct. 2009.

[11] R. J. Urick, ''Principles of Underwater Sound''. McGraw-Hill Book Company, third edition, 1983, p. 130.

[12] W. E. Johns, F.Yao, D. B. Olson, S. A. Josey, J. P Grist, and D. A. Smeed, "Observations of seasonal exchange through the Straits of Hormuz and the inferred freshwater budgets of the Persian Gulf," J. Geophys. Res.108 (C12), 2007.

[13] M. Sadrinasab, K. Kenarkoohi, " Numerical Modelling of sound Velocity profile in different layers in the Persian Gulf ", Asian journal of applied sciences, 232- 239, Knowledgia Review, Malaysia 2009.

[14] E.Seibold, J. Ulrich, " Zur Bodengestalt des nordwestlichen Gulfs von Oman," .Meteor. Forsch. Ergebnisses, Reihe C, 3, 1–14, 1970.

[15] R. M. Reynolds, " Physical oceanography of the Gulf, Strait of Hormuz, and the Gulf of Oman" Results from the Mt Mitchell expedition, Mar. Pollution Bull., 27, 35–59, 2006.

[16] W. E. Johns, F.Yao, D. B. Olson, S. A. Josey, J. P Grist, and D. A. Smeed, " Observations of seasonal exchange through the Straits of Hormuz and the inferred freshwater budgets of the Persian Gulf," J. Geophys. Res.,108(C12), 3391, doi:10.1029/2003JC001881, 2007.

[17] S. A. Swift, A. S. Bower, “ Formation and circulation of dense water in the Persian/Arabian Gulf ” J. Geophys. 2003.
[18] M. Sadrinasab, K. ampf, “ Three-dimensional flushing times in the Persian Gulf,” Letters, 31, L201, doi:10.19/204GL025, 2004.
[19] A. Doosti. Aref, “ Survey underwater acoustic communication with design and simulation of an Underwater acoustic communication system in the Persian Gulf,” MSc thesis, Malek-e Ashtar Industrial University, Tehran, Iran, 2010.
[20] H. Medwin, C.S. Clay, “ Fundamentals of Acoustical Oceanography’’, Academic Press, San Diego, pp. 220-228, 1998.
[21] R.J. Urick, “Ambient Noise in the sea’’, eninsula publishing, P.O.Box 867, California. 94023, third edition, pp. 155-205, 1984.
[22] K. C. Hegewisch, N. R. Cerruti and S. Tomsovic, “Ocean acoustic wave propagation and ray method correspondence: internal wave fine structure,” The Journal of the Acoustical Society of America, Vol. 114, Issue 4, p.2428, Oct.2003.
[23] J. P. Gomes and V. Barroso, “SDR Underwater Acoustic Modem,” Ph.D Thesis, Instituto de Sistemas e Robotica, Lisboa, Portugal, Apr. 2005.
[24] J. G. Proakis, M. Stojanovic and J. Catipovic, “Phase coherent digital communications for underwater acoustic communications,” IEEE Journal of Oceanic Engineering. Vol. 19, pp. 100-111, Jan. 1994.
[25] A. Masoomzadeh-Fard and S. Pasupathy, “Nonlinear equalization of multipath fading channels with noncoherent demodulation,” *IEEE J. Sel. Areas Commun.*, vol. 14, no. 3, pp. 512–520, Apr. 1996.

Abdollah Doosti Aref, was born in Tehran, Iran on 7 Mar, 1983. He received B.Sc. degree in Electrical and Electronic Engineering from the Shahed University, Tehran, Iran in 2007 and M.Sc. degree in Communication Systems from the Malek-Ashtar University of Technology, Tehran, Iran in 2009. His research interests include digital communications theory and its applications to mobile radio, satellite, radar and especially underwater acoustic communication.

Mohammad Javad Jannati, was born in Tehran, Iran on 21 May, 1983. He received B.Sc. degree in Electrical and Electronic Engineering from the Shahed University, Tehran, Iran in 2007 and M.Sc. degree in Communication Systems from the University of Science and Technology, Tehran, Iran in 2010. His research interests include Speech processing, underwater acoustic communication, cooperative communication and MIMO-OFDM systems.

Professor Vahid Tabataba Vakili, was born in Tehran, Iran. He received B.Sc. degree in Electrical and Electronic Engineering from the sharif university of Tehran , Tehran, Iran . He received M.Sc. and PHD degrees in Communication Systems from the university Bradford , Bradford, England .

Prof. Vakili is a faculty member of Iran university of science and technology. His research interests include digital communications theory, satellite communication, Signal processing, mobile and cooperative communication and MIMO-OFDM systems.