Modulation and Error Correction in the Underwater Acoustic Communication Channel

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Summary
We analyze the use of 8PSK and 16QAM Modulation Techniques along with Turbo Codes in the underwater acoustic channel. Turbo Codes have been used successfully to correct errors especially with higher order digital modulations. The channel model we will use is a Rician Multipath Fading.

Key words:
Acoustic Channel, 8PSK, 16QAM, Turbo Codes, Rician Model

1. Introduction

The design and implementation of means of communications over long distances has been one of the greatest achievements of modern science. The majorities of these systems uses electromagnetic waves to transmit information and are used through out the planet (Microwaves, Radio, Television, Satellite, etc), these waves have certain properties that make them ideal for such applications, and also for other types of applications such as the exploration and monitoring of our environment.

Even though the electromagnetic waves are so widely used there are some portions of our planet that are not suitable for these types of transmissions. The underwater world is an important part of our planet, which covers 70 % of it. In this realm the use of electromagnetic waves is very difficult, if not impossible, due to large attenuations of the electric and magnetic filed, this limits it range and effectiveness to transmit data. The other types of wave known are mechanical waves, for the underwater channel acoustics waves are the only solution for wireless communication. Acoustics waves can travel so much easily especially in sea water, where salinity shows strong conductivity.

The principal difference between the use of electromagnetic waves in open air and acoustics waves under the sea is observed in the constrains showed by the propagation channel. The underwater channel is favorable to the range an acoustic wave can cover but it still has many problems that have to be carefully examined when designing an acoustic based transmission system. Some of these limitations or problems are listed below:

- Attenuation because of the absorption of the acoustic waves in water, this limit the distance the sound can cover.
- Small propagation speed of the sound, approximately 1,500 m/s
- Multipaths due to the reflection on the bottom of the sea and sea surface, multipaths cause delayed echoes and interference.
- The transmitted signal suffers transformations related to the heterogeneous characteristics of the underwater channel as well as Doppler Effect caused by the movement of transmitter and receiver.
- Noise in the ocean. The noise level can mask the portion of the signal that can be use to receive the data.

The properties of the underwater medium are also extremely varied, and change both in space and time. Fluctuations due to environmental characteristics include, seasonal changes, geographical variations both in temperature and salinity, seabed relief, currents, tides, internal waves, movement of the acoustic systems and their targets. All this make the underwater acoustic signal to be randomly fluctuating.

Because all of the above mentioned limitations of the underwater channel the selection of the type of modulation and error correction techniques has to be carefully analyzed. We use 8PSK and 16QAM modulation techniques for our paper as well as Turbo Codes for the correction of errors during transmission. Our channel model is a Rician Multipath Fading [3].

The paper is divided in six sections, and is presented as follows: Section 2 covers the fundamental characteristics of the underwater channel, as well as the model we selected. Section 3 deals with the properties and differences between the two modulations techniques we will later use in our simulation; in section 4 we intended to cover the most important part of Turbo Codes and its difference with other method of error correction. Section 5 shows the results of our simulation and analyzes its

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performance. Finally, in Section 6 we cover what can be improved and future work related to this paper.

2. The Underwater Communication Channel

The underwater channel is limited by two well define interfaces, the bottom of the sea and the sea surface. The transmission of an acoustic signal is always accompanied by multiple paths due to reflection in both of these interfaces. These multiple paths appear as bursts of replicas of the main signal (in high frequencies) or as spatial field of stable interferences (low frequencies). They both are sources of problems when the receiver is trying to get the data from the transmitted signal.

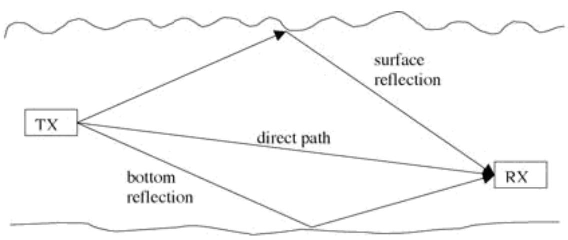

Fig. 1 Multipath in the underwater communication channel.

Acoustic signals are not instantaneous perturbations; instead they can be described as maintained vibrations. The range of frequencies used for communications in the underwater medium is from 10 Hz to 1 MHz depending of the type of application. The period varies from 0.1 second to 1 microsecond.

The most visible problem during the propagation of an acoustic wave is its loss of intensity due to geometric spreading and absorption of acoustic energy by the medium itself. This loss is a determinant element in the acoustic channel, it will attenuate the signal amplitude at the receiver site and will affect its signal to noise ratio.

The spreading transmission loss is given by [1]:

$$TL = 20\log\left(\frac{R}{R1m}\right)$$

Where, R is the radial distance from the source, and R1m is the distance at 1 meter from the source.

The effects of attenuation also occur. Sea water absorbs part of the energy of the transmitted wave, which is dissipated. The acoustic pressure decreases exponentially with distance [1]. This will add to the spreading losses given:

$$TL = 20\log R + \alpha R$$

where α is the absorption coefficient.

For frequencies of 1 kHz and less acoustic attenuation is of a few hundredths of dB per km and not a limiting factor. At 10 kHz, a coefficient of 1 dB/km limits the range to less than tens of kilometers. At 100 kHz, the attenuation coefficient reaches several tens of dB/km and the reach cannot be more than 1 km. If the system uses frequencies in the MHz range the range will be limited to less than 100 m, with attenuation of several hundreds of dB/km.

Multiple paths occur when a given signal propagates from a source to a destination along multiple distinct paths, corresponding to different durations and directions. The direct path arrives at the receiver along with a series of alternate paths due to reflections. The number of alternate paths is variable depending on the medium and topographic factors.

In radio transmissions multiple paths interfere and cause fading, but the delayed time echo is not always problematic when the receiver demodulates and decodes the signal, this is because the speed of light (300,000 m/s) induces very short delays. However, in underwater acoustic signals the propagation velocity is considerable slow compared to the speed of light, so the delays are much more important, creating considerable echoes and reverberation effects.

Another considerable problem in the acoustic wave is the Doppler Shift. The Doppler Effect is presented as a shift of the apparent frequency after propagation due to the change in the duration of the transmitter-receiver paths during the transmission, it is caused bye the relative movement of the transmitter and receiver.

The Doppler Effect [2] makes the signal processing harder, especially in data transmission. But it can be used effectively in some applications; its measurement can be used to find the speed of a vessel relative to the bottom of the water column.

The channel model we will use in our simulations is the Rician Multipath Fading. A channel is said to be Multipath Rician is there exist a direct line of sight (LOS) between transmitter and receiver. A parameter to describe a Rician Channel is the ratio K, of the power of the direct unfaded path to the overall power of the fading paths.

Other parameters of the Rician channel are the delay spread Tm and Doppler fading spectrum bandwidth Bd.

Typical values are, K=2, Tm=10 ms and Bd=10 Hz. These values will be used in the Simulation Section.

3. Modulation Techniques

Underwater Communication Systems use a signal that have to be chosen for how well they can carry data. These signals are not much different from the electromagnetic waves use in radio transmissions for atmosphere and space links. The different appear from the differences in the physics of the surrounding underwater environment, such as propagation, noise, transducer types.

There exists two fundamental aspects to the good performance of an underwater communication link, they are:

- use and definition of signals that perform well, and the environmental conditions
- the use of good processing techniques in the receiving site, taking into account the characteristics of the signal and perturbations in the medium, and very important is to consider the level of complexity and cost.

The signal to noise ratio (SNR) expresses the relative importance of the power contribution of the signal expected and the perturbing noise. The SNR is the principal parameter that affects the performance of a receiver in almost all applications being acoustic or electromagnetic transmissions. We have to make sure that the SNR and the Bit Error Rate (BER) allows us to maintain a robust communication link in which the data transmitted is protected from ambient noise as well as other impairments. BER is a parameter that describes the way a digital communication link behaves and how we can rely on it. Another fundamental parameter in a digital communication link is the Eb/No which will be described in detail in the simulation section of this paper.

In our simulations we will use two modulation techniques and compare the results of each one of them. We have selected 8PSK and 16QAM modulations as they can achieve high data rate with smaller bandwidth. In the acoustic communication channel it is very important to have a robust transmission and also an efficient modulation technique that allows us to improve the amount of data transmitted over the limited bandwidth offered by the medium.

The demodulation of a signal around its carrier frequency aims at keeping only the meaningful frequency content of the signal, and therefore limiting the amount of digital data to process at larger states.

If the signal has a certain bandwidth BW centered around fc, the Shannon criterion means the time sampling must have a frequency at least twice the higher frequency of the spectrum (2fc+BW) [1]. If the signal is demodulated around fc, it is possible to sample it at frequency BW.

The effects and impairments mentioned earlier can be simplified in the sonar equation. The sonar equation can be expressed very simply, to give the probability of detection by an underwater communication system:

$$Signal - Noise + Gain > Threshold$$

This is obviously an equation on energy conservation used to evaluate the system performance. The parameters of the equation are described below:

- Signal. Is the signal level at the receiver site
- Noise. Is the sum of all noise contribution, ambient noise and system noise
- Gain. Is the array directivity index and the receiver processing gain
- Threshold. Considered at the receiver output, depends on performance determined by the ultimate detection or measurement operations.

We now describe the two modulation techniques used.

8PSK is a higher order MPSK modulation which uses 8 phases to modulate the incoming data stream, usually is the higher order PSK modulation used; with more than 8 phases the error rate becomes too high and there are better and also more complex modulation techniques to handle higher data requirements. Although in MPSK any number of phases is allowed normally we used a power of two because the incoming data is binary, this allows a equal number of bits per symbol.

The symbol error probability for a MPSK modulation can be found by [7]:

$$P_s = 1 - \int_{-\frac{\pi}{M}}^{\frac{\pi}{M}} p_{\Theta_r}\left(\Theta_r\right) d\Theta_r$$

where:

$$p_{\Theta_r}\left(\Theta_r\right) = \frac{1}{2\pi} e^{-2\gamma_s \sin^2 \Theta_r} \int_0^{\infty} V e^{-\left(V - \sqrt{4\gamma_s} \cos \Theta r\right)^2/2}$$

$$V = \sqrt{r_1^2 + r_2^2}$$
$$\Theta_r = \tan^{-1}(r_2/r_1)$$
$$r_1 \sim N\left(\sqrt{E_s}, N_0/2\right)$$
$$\gamma_s = \frac{E_s}{N_0}$$
$$r_2 \sim N(0, N_0/2)$$

This for high M and high Eb/No can be reduced to:

$$P_s \approx 2Q\left(\sqrt{2\gamma_s}\sin\frac{\pi}{M}\right)$$

A typical constellation for 8PSK is shown below:

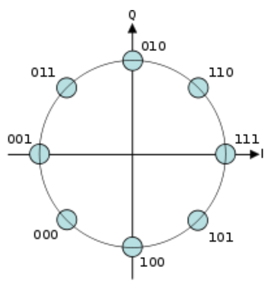

Fig.2 Constellation of 8PSK Modulation [7]

16QAM is higher order QAM Modulation that uses rectangular constellation which are in general suboptimal in the sense they don't maximally space the constellation for a given energy. However they have the advantage that can be transmitted as two PAM signals and can be demodulated without further complications.

Particularly 16QAM is the first order QAM modulation; 2QAM and 4QAM are just BPSK and QPSK Modulations not exactly square constellations. The error rate of 8QAM is almost as the 16QAM but its data rate is much less, ¾ of 16QAM data rate; that is the reason we choose 16QAM in our simulations.

Symbol error rate [7] can be found using the following expression:

$$P_{sc} = 2\left(1 - \frac{1}{\sqrt{M}}\right) Q\left(\sqrt{\frac{3}{M-1}\frac{E_s}{N_0}}\right)$$

$$P_s = 1 - (1 - P_{sc})^2$$

BER will be affected by the assignments of the bits to symbols. For a Gray Coded assignment with equal bit per carrier we have:

$$P_{bc} = \frac{2}{k}\left(1 - \frac{1}{\sqrt{M}}\right) Q\left(\sqrt{\frac{3k}{M-1}\frac{E_b}{N_0}}\right)$$
$$P_b = 1 - (1 - P_{bc})^2$$

A typical constellation for 16 QAM is shown below:

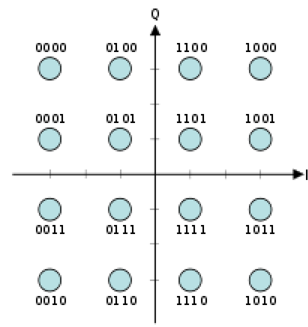

Fig. 3 Constellation of 16QAM Modulation [7]

4. Forward Error Correction and Turbo Codes

There are many FEC algorithms that can be used for wireless communications, each one of them performs different and have different characteristics. One can be strong reconstructing bit loss and other can be very god if packets losses have occurred.

There are also two main types of coding:

-**Block Codes**: this algorithm applies to transmission of constant data.

-**Convolution Codes**: applies to streams of variable data.

The choice of which algorithm has to be made looking at the number and types of errors and also the delay due to the encoding and decoding process in transmitter and receiver respectively.

Now let's take a look at both Block codes and Convolution codes.

Block codes are FEC codes that enable a limited number of errors to be detected and corrected without retransmission. Block codes can be used to improve the performance of a communications system when other means of improvement (such increasing transmitter power or using a stronger modulator) are impractical.

In block codes, parity bits are added to blocks of messages bits to construct codewords or code blocks. In a block encoder, k information bits are encoded into n code bits. A total of n-k redundant bits are added to the k information bits for the purpose of detecting and correcting errors. The block code is referred to as an (n, k) code, and the rate of the code is defined as **RC = k/n** and is equal to the rate of information divided by the raw channel rate.

Examples of Block Codes that are currently used are the following:

1-Hamming Codes
2-Hadamard Codes
3-Golay Codes
4-Cyclic Codes
5-BCH Codes (Reed Solomon Code)

As mentioned above block code requires that the data stream be portioned into blocks of bits. If each block contains k bits, then theses k bits define a data word. The number of data words is 2(exp) k. There is no redundancy in the system, meaning that even a single error in transmission would convert one data word into another, which of course would constitute an error.

The dataword can be encoded into codewords, which consist of n bits, where the additional n-k bits are derived from the message bits but are not part of the message. The number of possible codeword is 2(exp)n but only 2(exp)k of these will contain dataword, and these are the ones that are transmitted. So the rest of the codewords are redundant, but only in the sense that they do not contribute to the message.

The n-k bits are referred as the parity check bits. If errors occur in the transmission, there is high probability that they will convert the permissible codewords into one or another of the redundant words that the decoder at the receiver is design to recognize as an error. It will be noted that the term high probability is used. There is always the possibility that enough errors occur to transform a transmitted codeword into another legitimate codeword in error.

Convolutional Codes are fundamentally different from block codes in that information sequences are not grouped into distinct blocks and encoded. Instead a continuous sequence of information bits is mapped into a continuous sequence of encoder output bits. This mapping is highly structured, enabling a decoding method considerably different from that of block codes to be employed. T can be argued that convolutional coding can achieve a larger coding gain than can be achieve using a block coding with the same complexity.

A convolutional code is generated by passing the information sequence through a finite state shift register. In general, the shift register contains N k-bits stages and m linear algebraic function generators based on the generator polynomials. The input data is shifted into and along the shift register, k bits at a time. The number of output bits for each k bit user input data sequence is n bits. The code rate **RC = k/n**. The parameter N is called the constraint length and indicates the number of input data bits that the current output is dependent upon. The constraint length determines how powerful and complex the code is.

Convolution codes are also linear codes. A convolution encoder consists of a shift register, which provides temporary storage, and shifting operation for the input bits and exclusive-OR logic circuits, which generate the coded output from the bits currently, held in the shift register.

A Trellis diagram shows all possible information and how the sequences are encoded in the Convolutional Encoder. In the decoding or receiver site a Viterbi Decoding Algorithm is used to decode the sequence encoded at the transmitter. Viterbi uses maximum likelihood decoding techniques, and estimates the bit sequence using Trellis Diagram. The information is decoded using hard or soft decision, it refers on the type of quantization used on the received bits.

Turbo Code [4] uses two identical recursive systematic convolutional codes (RSC) in parallel. The two convolutional encoders are separate by an Interleaver which is used to change the input bit stream using a certain rule. Figure 4 shows a Turbo Encoder.

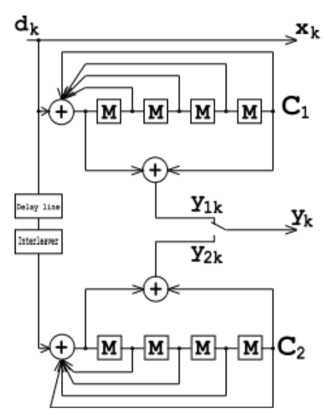

Fig. 4 Basic Turbo Encoder [7]

M is the memory registers, bits in the dk stream appear in different sequences after they pass through the delay line and the interleaver. After the first iteration the input stream appear at encoder outputs, xk and y1k or y2k, this is because the encoder systematic property. Encoders C1 and C2 are used in respectively in the n1 and n2 iterations, their rates are equal to [7]:

$$R_1 = \frac{n_1 + n_2}{2n_1 + n_2} \quad R_2 = \frac{n_1 + n_2}{2n_2 + n_1}$$

The Turbo decoder works in a similar way as the Encoder does, the difference is that two decoders are connected in serial (Fig. 5) to perform the decoding process as oppose to the parallel connection in the encoder.

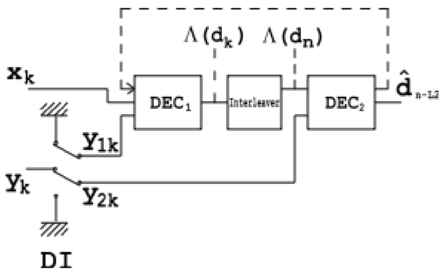

Fig. 5 Turbo Decoder [7]

Decoder 1 (DEC1) works on lower speed, DEC1 decodes streams from C1, from the same token DEC2 is used to decode C2. DEC1 uses a soft decision causing a delay L1 which is the same delay at the encoder. DEC2 in its decoding process causes delay L2.

The Interleaver is using to prevent burst errors coming from the output of DEC1. D1 acts as a switch injecting the yk stream to DEC1 and DEC2, one at a time.

DEC1 gives a soft decision and feeds it to DEC2. The logarithm of likelihood ratio is expressed by:

$$\Lambda(d_k) = \log \frac{p(d_k = 1)}{p(d_k = 0)}$$

The probability of detecting a 0 or 1 in the dk stream is A Posteriori Probability (APP), and is expressed as p(dk=i). Viterbi Algorithms are not capable to calculate APP, so the decoding algorithm in DEC1 is a modified BCJR algorithm; in the case of DEC2 a Viterbi Algorithm is used.

5. Simulation Results

In our simulation we combine the use of high order modulation techniques such as 8PSK and 16 QAM and Turbo Codes to show how with the combination of both we can achieve good results in the Underwater Acoustic Communication Channel.

We use a Bernoulli Binary Generator as our source of random bits, the output of this source is fed into a Turbo Encoder which uses two Convolutional Encoders in parallel separated by an Interleaver. The encoded stream is fed into a Modulator and then into a Multipath Rician Fading Channel, the factor K is set to 2 which is a reasonable representation of real life situations. The Doppler shift is 10 Hz and the delay is 10 ms. (Fig. 6)

At the receiver site the signal is first demodulated and then fed into the Turbo Decoder, which contains two Decoders connected serially and separated by an Interleaver. The output of the Turbo Decoder is compared with the output of the Bernoulli Binary Generator to calculate the Bit Error rate of the transmission chain. We use 7 iterations in the calculations. A general block diagram of the system is presented below, specifying each block and its functions.

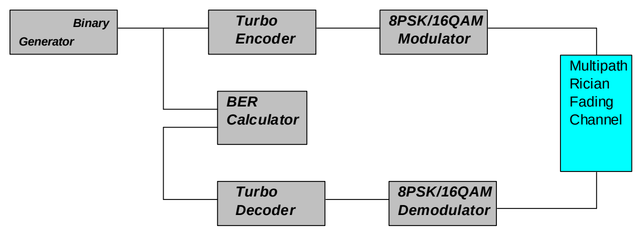

Fig. 6 System Block Diagram

The detailed system is show in the figure below:

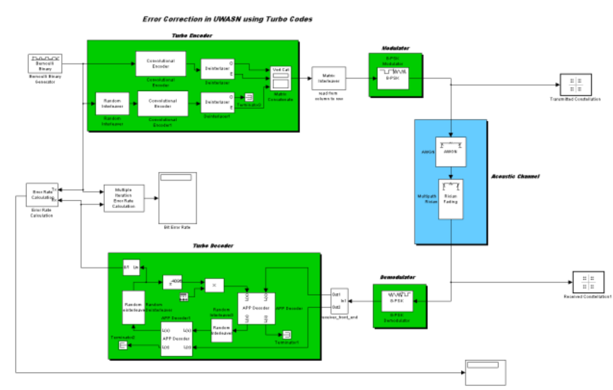

Fig. 7 Matlab Simulation Diagram

First we present the results using 8PSK Modulation; results are Eb/No values vs Bit Error Rate. Blocksize is 4096 and seven Iterations. During the simulation other values for the number of Iterations were used, but we decided to use seven for the final run since with a value higher than seven the BER does not improve dramatically, actually above three Iterations the improvement is not that good.

Table 1: Results of Eb/No vs BER. 8PSK Modulation

Eb/No (dB)	***BER***
5	.3
10	.15
15	.03027
20	.00732
25	.006212
30	.000562
35	.000554

From Table 1 we can see that the BER value improves when we increase the value of Eb/No. At Eb/No = 20 dB the BER is acceptable and data can be recovered at the receiver. Turbo Code performs well when combined with higher order 8PSK Modulation, the modulation constellation is well defined at Eb/No= 20 dB (Fig. 8). The phase noise is significant but the received signal is strong enough to allow us to recovery the initial bit stream.

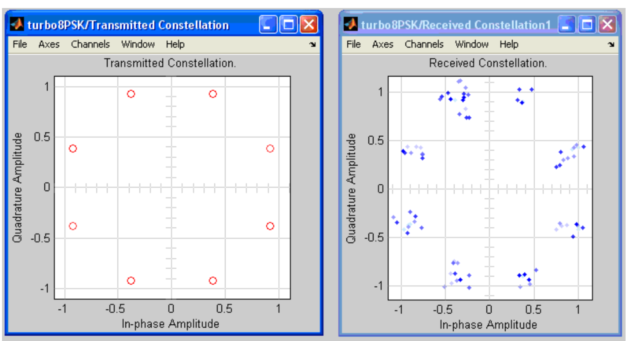

Fig. 8 Transmitted and Receive Constellations. 8PSK

Similarly the results of the 16QAM signal of Eb/No vs BER and Transmitted and Received signals are shown below:

Table 2: Results of Eb/No vs BER. 16QAM Modulation

Eb/No (dB)	***BER***
5	.42
10	.21
15	.05321
20	.012533
25	.0095412
30	.0025645
35	.00094512

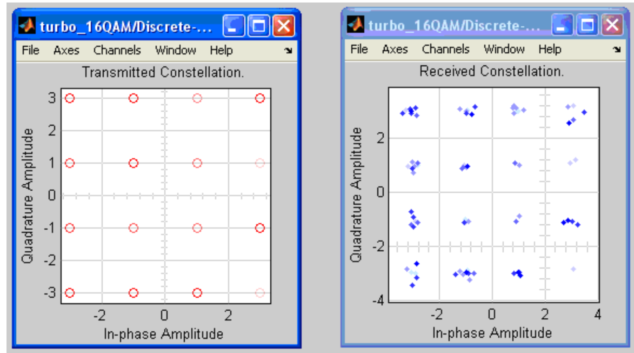

Fig. 9 Transmitted and Received Constellations. 16QAM

In the 16QAM (Fig. 9) case the phase noise will damage the signal more that in the 8PSK case, thus affecting the amount of correct bits we can recuperate from the received stream and after the Turbo Decoder. Although 16QAM generally give much worse results than MPSK signals, in our Simulation the results were satisfactory in terms of the amount of errors.

Comparing Tables 1 and 2 we can see that at Eb/No value above 10 dB we obtain a BER of 0.03027 for 8PSK and 0.05321 for 16QAM. This is consistent with the Constellation Diagrams of Fig. 8 and 9. In 8PSK the 8 phases of the received signal are more spaced than the 16 phases of the 16QAM signal (the distance between phases is bigger). The receiver will have more difficult in detecting the 16QAM phases than in detecting the 8 phases of 8PSK. That is why the BER of 8PSK is better. It is easier for the receiver to decide that a detected phase is the one that it is supposed to be and not the adjacent phase.

6. Conclusions and Future Work

The use of fully coherent modulation, such as the one that we used in our simulation 16QAM and 8PSK have been a reality in the past years due to advances in digital processing. The intersymbol interference (ISI) can be suppressed by using channel equalization techniques; also decision feedback equalizers (DFE) can track the varying channel response and give high throughput values if the channel varies slowly, if the channel is varying much faster it is necessary to combine DFE with Phase Locked Loop (PLL). PLL compensate the phase offset.

The Table below shows the development of modulation techniques through years of research [5].

Table 3: Modulation and Rates

Modulation	***Year***	***Rate [Kbs]***	***Band [KHz]***
FSK	1984	1.2	5
PSK	1989	500	125
FSK	1991	1.25	10
PSK	1993	0.5	0.3-1
PSK	1994	0.02	20
DPSK	1997	20	10
16QAM	2001	40	10

Coherent modulations had lower performance for long transmissions, this was resolved by using ISI compensation with DFE, but these filtering algorithms are not easy to implement and do not performed well in real time links.

One very promising solution for the Underwater Communication Channel is the use of Orthogonal Frequency Division Multiplexing, OFDM is vey efficient when the noise is spread across the bandwidth. OFDM transmit several carriers instead of a single modulated carrier, so it is referred as a Multicarrier Modulation, carrier with better SNR are allocated with higher data rate, and so lower data rate will be used in carriers with smaller SNR values. In OFDM systems give more spectral efficiency and perform robustly in Multipath Fading Channels

OFDM [6] eliminates the need of complex time domain equalizers thus making it very suitable for real time communications in the acoustic medium. Modulation and Demodulation of OFDM is achieved using fast Fourier transforms (FFT). OFDM is sensible to Doppler shift so new algorithms for wideband Doppler compensation are needed, once this algorithms are ready we expect that OFDM in the underwater channel can perform very robustly and make the data transmission as reliable as it is in the wireless networks that use OFDM. We expected that the combination of strong Forward Error Correction algorithms with fully develop OFDM techniques can be a reality in the next few years.

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