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MASTER THESIS

TITLE: Desenvolupament, proves de camp i anàlisi de resultats en una xarxa de sensors

MASTER DEGREE: Master in Science in Telecommunication Engineering & Management

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Resum

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Overview

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To Raja Jurdak, this Master Thesis would have been much more difficult without its help.

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INTRODUCTION

Currently, there has been a growing interest in monitoring underwater mediums for scientific exploration, commercial exploitation, and attack protection. A distributed underwater wireless sensor network is the ideal vehicle for this monitoring.

Underwater acoustic communications depend on path loss, noise, multi-path, Doppler spread, and high and variable propagation delay. All these aspects establish the temporal and spatial variability of the acoustic channel. Therefore, the available bandwidth of the underwater acoustic channel is severely limited and dependent on both range and frequency. Moreover, the communication range is reduced in comparison with the terrestrial radio channel. For these reasons we want to study and describe the problems of the underwater communications in an underwater sensor network. The experiments carried out try to characterize the communication in underwater environments in order to be able to develop underwater sensor networks

In the first chapter we describe the motivations, features of aquatic environment, the difficulties of underwater acoustic channels, and the open questions in mobile underwater sensor network design. In the second chapter we try to explain some underwater experiments, show the results and try to explain these results. And finally in the third chapter we explain the conclusions of this master thesis and the further works.

CHAPTER 1. THEORETICAL INTRODUCTION

1.1. Introduction

The earth is a water planet. Currently, there has been a growing interest in monitoring underwater mediums for scientific exploration, commercial exploitation, and attack protection. A distributed underwater wireless sensor network (UWSN) is the ideal vehicle for this monitoring. A scalable UWSN is a good solution for exploring the aquatic environments.

By deploying scalable wireless sensor networks in 3-dimensional underwater space, each underwater sensor can monitor and find environmental events. The aqueous systems are also dynamic and processes happen within the water mass as it disperses within the environment. In a mobile underwater sensor network, the sensor mobility has two major benefits:

- Mobile sensors injected in the current in relative large numbers can help to track changes in the water mass, thus provide 4D (space and time) environmental sampling.
- Floating sensors can help to form dynamic monitoring coverage and increase system reusability.

The self-organizing network of mobile sensors produces better supports in sensing, monitoring, surveillance, scheduling, underwater control, and failing tolerance. Mobile UWSNs have to use acoustic communications, since radio does not work well in underwater environments. Due to the unique features of large latency, low bandwidth, and high error rate, underwater acoustic channels bring much defiance to the protocol design. Furthermore, the best parts of underwater nodes are mobile due to water currents. This mobility is another problem to consider in the system design.

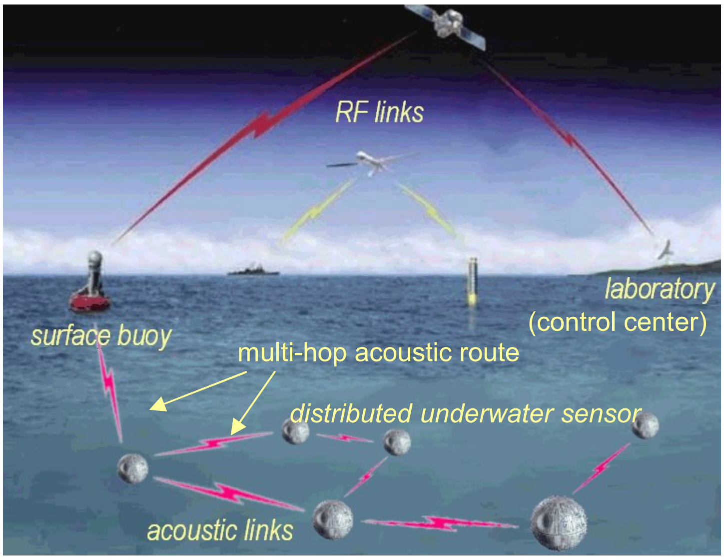


Fig. 1.1 Scenario of the mobile UWSN architecture

1.2. Characteristics of the environment

1.2.1. Basics of acoustic communications

Underwater acoustic communications depend on path loss, noise, multi-path, Doppler spread, and high and variable propagation delay. All these aspects establish the temporal and spatial variability of the acoustic channel. Therefore the available bandwidth of the underwater acoustic channel is severely limited and dependent on both range and frequency. In long-range systems and short-range system these factors lead to low bit rates. In addition, the communication range is reduced as compared to the terrestrial radio channel.

Underwater acoustic communication links can be classified depending on their range. Moreover, acoustic links are classified as vertical and horizontal, according to the direction of the sound ray. Their propagation attributes differ consistently, especially with respect to time dispersion, multi-path spreads, and delay variance. Now, we analyze the factors that influence acoustic communications in order to state the challenges posed by the underwater channels for underwater sensor networking.

Path loss:

- Attenuation: Is mainly caused by absorption due to conversion of acoustic energy into heat, which increases with distance and frequency. It is also caused by scattering and reverberation, refraction, and dispersion. Water depth is determinant in the attenuation.
- Geometric Spreading: This refers to the spreading of sound energy as a result of the expansion of the wave-fronts. It increases with the propagation distance and is independent of frequency. There are two types of geometric spreading: spherical (Omni-directional point source), and cylindrical (horizontal radiation only). The cylindrical spreading appears in water with depth less than 100m (shallow water) because acoustic signals propagate with a cylinder bounded by the surface and the sea floor. When sea is deep enough the propagation range is not bounded so that spherical spreading applies.

Noise:

- Man made noise: This is mainly caused by machinery noise and shipping activity.
- Ambient Noise: Is related to hydrodynamics, seismic and biological phenomena.

Multi-path:

- Multi-path propagation: This may be responsible for severe degradation of the acoustic communication signal, since it generates Inter-Symbol Interference.
- The multi-path geometry: It depends on the link configuration. Vertical channels are characterized by little time dispersion, while horizontal

channels may have extremely long multi-path spreads, whose value depend on the water depth.

High delay and delay variance:

- Delay: The propagation speed in the underwater acoustic channels is five orders of magnitude lower than in the radio channel. This large propagation delay can reduce the throughput of the system considerably.
- Delay variance: The very high delay variance is even more harmful for efficient protocol design, as it prevents from accurately estimating the round trip time, key measure for many common communication protocols.

Doppler spread:

- The Doppler frequency spread can be significant in underwater acoustic channels, causing degradation in the performance of digital communications. High data rate communications cause many adjacent symbols to interfere at the receiver, requiring sophisticated signal processing to deal with the generated ISI.

1.2.2. Underwater acoustic channels

Underwater acoustic channels are temporally spatially and variable due to the characteristics of the transmission medium and physical properties of the environments. The signal propagation speed in underwater acoustic channel is about 1.5×10^3 m/sec. The convenient bandwidth of underwater acoustic channels is limited and dramatically depends on both transmission range and frequency. The acoustic band under water is restricted due to absorption.

The bandwidth of underwater acoustic channels working over several kilometers is about several tens of kbps, whereas short-range systems over several tens of meters can reach at hundreds of kbps. The path loss, noise, multipath, and Doppler spread affect the underwater acoustic communication channels. All these factors generate high bit-error and delay variance.

1.2.3. Distinctions between mobile UWSNs and ground-based sensor networks

A mobile UWSN is very different from any ground-based sensor network in the following aspects:

- Communication Method: Electromagnetic waves cannot propagate over a long distance in underwater environments. Each underwater wireless link features large latency and low-bandwidth. Due to such distinct network dynamics, communication protocols used in ground-based sensor networks may not be appropriate in underwater sensor networks.

- Node Mobility: The sensor nodes in ground-based sensor networks are fixed, though it is possible to implement interactions between these static sensor nodes and a limit number of mobile nodes. However, the best part of underwater sensor nodes are with low or medium mobility due to water current and other underwater activities. From experimental observations, underwater objects may move at the speed of 3-6 kilometers per hour in a typical underwater condition.

1.2.4. Current underwater network systems

An underwater sensor network is a next step forward with respect to existing small-scale Underwater Acoustic Networks (UANs). UANs are associations of nodes that collect data using remote telemetry or assuming point-to-point communications. The different between UANs and underwater sensor networks are the following:

- Scalability: A mobile underwater sensor network is a scalable sensor network, which relies on localized sensing and coordinated networking among large numbers of sensors. In contrast, an existing underwater acoustic network is a small-scale network relying on data collecting strategies like remote telemetry or assuming that communication is point-to-point. In remote telemetry, long-range signals remotely collect data. In point-to-point communication, a multi-access technique is not necessary.

- Self-organization: Usually, in underwater acoustic networks nodes are fixed, while a mobile underwater sensor network is a self-organizing network. Underwater sensor nodes may be redistributed and moved by the aqueous processes of advection and dispersion. Thus, sensors should automatically adjust their buoyancy, moving up and down based on measured data density. In this way, sensors are mobile in order to track changes in the water mass rather than make observations at a fixed point.

- Localization: In underwater acoustic networks sensors localization is not desired because nodes are usually fixed. In mobile underwater sensor networks, localization is required because the majority of the sensors are mobile with the current. Determining the locations of mobile sensors in aquatic environments is very challenging. We need to face the limited communication capabilities of acoustic channels. Moreover, we need improving the localization accuracy.

1.3. Protocol stack for underwater acoustic channels

Now, we talk about the design of a protocol stack for underwater acoustic communication and we describe the problems and solutions of each layer. For more information you can read research works [1], [2], [3], [4] and [5].

1.3.1. Physical layer

The underwater modem development was based on non-coherent frequency shift keying (FSK) modulations. These techniques do not need phase tracking, which is a very difficult task in underwater environment.

Non-coherent modulation methods are distinguished by a high power efficiency and low bandwidth efficiency. This last feature is inappropriate in high data-rate multiuser networks. Therefore, coherent modulation techniques have been developed for long-range, high-throughput systems (such as phase shift keying (PSK) and quadrature amplitude (QAM)).

There are three limitations of conventional receivers. The first one is the time variability of the underwater channels. The second one is the multi-path phenomenon, which generates two problems, the delay spread, which causes ISI at the receiver side, and the phase shift of the signal envelope. Third is the Doppler spread. All these limitations together with the bandwidth limitation of acoustic links cause that the underwater acoustic channels are rate-limited and their performance is limited in comparison with the radio channels.

In order to solve all the problems we need to provide robustness at the physical layer to the system performance and to offer significantly higher data rate for underwater communication networks. A paradigm shift from current single-carrier transmissions and equalizations to multi-carrier modulation in the form of orthogonal frequency division multiplexing (OFDM) is envisioned as a viable approach, as OFDM has well-demonstrated success in broadband wireless radio systems. Another direction is to pursue multi-input multi-output (MIMO) techniques for substantial rate and performance improvement. Distributed MIMO is also possible if clustered single-antenna nodes could cooperate

1.3.2. Data link layer

Multiple access techniques are developed to allow devices to access a common medium, sharing the scarce available bandwidth in an efficient and fair way. Channel Access Control in an underwater sensor network poses additional challenges due to the peculiarities of the underwater channel, in particular limited bandwidth and high and variable delay.

Multiple access techniques can be roughly divided into two main categories: contention free, such as FDMA, TDMA, and CDMA; and the non-contention free, which are either based on random access (ALOHA, slotted-ALOHA), or on collision avoidance with handshaking access (MACA, MACAW).

The mail differences between these techniques are the following:

- Frequency division multiple access (FDMA) divides the available band into sub-bands. Afterwards, it assigns each sub-band to a device. Due to the narrow bandwidth in underwater acoustic channels and to the vulnerability of limited band systems to fading, FDMA is not suitable for UWSNs.

- Time division multiple access (TDMA) divides time into slots, providing time guards to limit packet collisions from adjacent time slots. These time guards are designed accounting for the propagation delay of the channel. Due to the characteristics of the underwater environment it is very challenging to realize a precise synchronization, with a common timing reference, which is required for a proper utilization of time slots in TDMA. Additionally, due to the high delay and delay variance of the underwater acoustic channel, TDMA efficiency is limited because of the high time guards required to implement it.

- Code division multiple access (CDMA) allows multiple devices to transmit simultaneously over the entire frequency band. Signals from different devices are distinguished by means of pseudo-noise codes that are used for spreading the user signal over the entire available band. This makes the signal resistant to frequency selective fading caused by multi-paths. In conclusion, although the high delay spread which characterizes the horizontal link in underwater channels makes it difficult to maintain synchronization among the stations, especially when orthogonal code techniques are used, CDMA is a promising multiple access technique for underwater acoustic networks.

- ALOHA is a class of MAC protocols that do not try to prevent packet collision, but detect collision and retransmit lost packets. In the underwater acoustic environment, ALOHA protocols are affected by low efficiency, mainly due to the slow propagation of the acoustic channel. Additionally, the need for retransmissions increases the power consumption of sensors, and ultimately reduces the network lifetime.

- Carrier sense multiple access (CSMA) protocols are aimed at reducing the packet retransmissions, by monitoring the channel state: if the channel is sensed busy, packet transmission is inhibited so as to prevent collisions with the ongoing transmission. If the channel is sensed free, transmission is enabled. However this approach, although it prevents collisions at the sender, does not avoid collisions at the receiver due to the hidden and exposed terminal problems.

- Contention based techniques that use handshaking mechanisms, such as RTS/CTS in shared medium access are impractical in underwater, due to three reasons. The large delays in the propagation of RTS/CTS control packets lead to low throughput. The high propagation delay of underwater channels impairs the carrier sense mechanism. And finally, the high variability of delay in handshaking packets makes it impractical to predict the start and finish time of the transmissions of other stations. Thus, collisions are highly likely to occur.

Many novel access techniques have been designed for terrestrial sensor networks, whose objectives are to maximize the network efficiency and prevent collisions in the access channel. These similarities would suggest to tune and apply those schemes in the underwater environment; on the other hand, the main focus in medium access control in underwater sensor network is trade-off between energy and latency.

1.3.3. Network layer

The network layer is in charge of determining how messages are routed within the network. In underwater acoustic sensor networks, this translates into determining which path should follow data packets from the source that samples the physical phenomenon to the onshore sink.

In the last few years there has been an intensive study in routing protocols for ad hoc wireless networks. However, due to the different nature of the underwater environment and applications, there are several drawbacks with respect to the suitability of the existing solutions for Underwater Acoustic Networks. The existing routing protocols are usually divided into three categories, namely proactive, reactive and geographical routing protocols:

- Proactive protocols: These protocols attempt to minimize the message latency induced by route discovery, by maintaining up-to-date routing information at all times from each node to every other node. This is obtained by broadcasting control packets that contain routing table information. These protocols provoke a large signaling overhead to establish routes for the first time and each time the network topology is modified because of mobility or node failures, since updated topology information has to be propagated to all the nodes in the network. This way, each node is able to establish a path to any other node in the network, which may not be needed in underwater acoustic sensor networks.

- Reactive protocols: A node initiates a route discovery process only when a route to a destination is required. Once a route has been established, it is maintained by a route maintenance procedure until it is no longer desired. These protocols are more suitable for dynamic environments but incur a higher latency and still require source-initiated flooding of control packets to establish paths. Thus, both proactive and reactive protocols incur excessive signaling overhead due to their extensive reliance on flooding. Reactive protocols are deemed to be unsuitable for UWSNs as they also cause a higher latency which may even be amplified by the slow propagation of acoustic signals in the underwater channel. Moreover the topology of UWSNs is unlikely to vary dynamically on a short time scale.

- Geographical Routing Protocols: These protocols establish source-destination paths by leveraging localization information. Each node selects its next hop based on the position of its neighbors and of the destination node. Although these techniques are very promising, it is still not clear how accurate localization information can be obtained in the underwater environment with limited energy expenditure.

Thus, routing schemes that jointly minimize the signaling overhead and the latency need to be developed. In the routing protocols developed for ad-hoc networks the routing function is performed for each single packet separately, in underwater acoustic sensor network virtual circuit routing techniques could be considered. In these techniques, paths are established a priori between each source and sink, and each packet follows the same path. This many require some form of centralized coordination but can lead to more efficient paths.

Furthermore, routing schemes that account for the 3D underwater environment need to be developed. Especially, in the 3D case the effect of currents should be taken into account, since the intensity and the direction of currents are dependent on the depth of the sensor node. Thus, underwater currents can modify the relative position of sensor devices and also cause connectivity holes, especially when ocean column monitoring is performed in deep waters.

1.3.4. Transport layer

In sensor networks reliable event detection at the sink should be based on collective information provided by source nodes and not on any individual report from each single source. For this reason, conventional end-to-end reliability definitions and solutions can be inapplicable in the underwater sensor field, and could lead to waste of scarce sensor resources. But the absence of a reliable transport mechanism altogether can seriously impair event detection due to underwater challenges. Thus, the underwater acoustic sensor network paradigm necessitates a new event transport reliability notion rather than the traditional end-to-end approaches.

A transport layer protocol is needed in underwater acoustic sensor networks not only to achieve reliable collective transport of event features, but also to perform flow control and congestion control. The primary objective is to save scarce sensor resources and in-crease network efficiency. A reliable transport protocol should guarantee that the applications are able to correctly identify event features estimated by the sensor network. Congestion control is needed to prevent the network from being congested by excessive data with respect to the network capacity, while flow control is needed to avoid that network devices with limited memory are overwhelmed with data transmissions.

Several solutions have been proposed to address the transport layer problems in Wireless Sensor Networks (WSN). For example, Event-to-Sink Reliable Transport (ESRT) protocol is proposed to achieve reliable event detection with minimum energy expenditure. However, the ESRT mechanism relies on spatial correlation among event flows which may not be easily leveraged in underwater acoustic sensor networks. Hence, further investigation is needed to develop efficient transport layer solutions.

1.4. Questions in mobile UWSN design

Now, we catalogue the design questions along the network protocol stack. We will see the critical problems of each layer.

1.4.1. Security, resilience and robustness

An underwater sensor network needs more protections than cryptography due to the limited energy, computation, and communication capabilities of sensor nodes. A critical security point is to defend against denial-of-service attack. This attacks could be in the form of depleting node's on-device resource and disrupting network collaboration. And due to the unique attributes of underwater acoustic channels, these attacks can interfere or even disable sensor networks independent of cryptographic protections.

In particular, wormhole attacks (in which an attacker records a packet at one location in the network, tunnels the data to another location, and replays the packet there) introduce serious threat to underwater acoustic communications. Many solutions that have been proposed to stop wormhole attack in radio networks are ineffective in underwater sensors networks. Thus, to protect against wormhole attacks in underwater sensors networks, we need new techniques.

Another problem in underwater sensor networks is intermittent partitioning due to water turbulence, currents, and ships. There may be circumstances where no connected path exists at any given time between source and destination. This discontinuous situation may be found through routing and by traffic observations. A new network model that deals with such disruptions was recently developed, namely Disruption Tolerant Networking DTN. DTN includes the use of intermediate store and forward proxies. If the data sink suspects the presence of such conditions, it can then take advantage of some of the DTN methods to reach the data sources.

1.4.2. Reliable and real-time data transfer

Reliable data transfer is very important. In sensor networks, path redundancy is exploited to get better reliability. In underwater sensor networks, due to the high error probability of acoustic channels, efficient erasure coding schemes could be utilized to help achieve high reliability and at the same time reduce data transfer time by suppressing retransmission.

There are normally two approaches for reliable data transfer: end-to-end or hop-by-hop. The most popular solution at the transport layer is TCP, which is an end-to-end technique. We expect TCP performance to be problematic because of the high error rates in the links. Under the water, we have an additional problem: propagation time is much larger than transmission time, setting the stage for the well known large bandwidth*delay product problem. Managing such unusually large windows with link error rates is a major challenge since TCP would time out and would never be able to maintain the maximum rate.

Another solution for reliable data transfer is hop-by-hop. The hop-to-hop approach is favored in error-prone and wireless networks. It is believe to be more convenient for sensor networks. In the hop-by-hop PSFQ (Pump Slowly and Fetch Quickly) protocol, a sender sends data packet to its near neighbors at very slow rate. When the receiver detects some packet losses, it has to get the lost packets quickly. Hop-by-hop, data packets are finally delivered to the data sink reliably. In PSFQ, ARQ is used for per-hop communication. However, due to the propagation delay of acoustic signals in underwater sensor networks, ARQ would cause very low channel utilization.

Real-time data transfer is necessary for short-term in the time-critical aquatic exploration applications. To dive time-constrained services is yet another tough research issue in the network community. UDP allows to transfer data as fast as possible but UDP does not work well with UWSNs because UDP does not guarantee reliability in the way that TCP does. Datagrams may arrive out of order, appear duplicated, or go missing without notice.

1.4.3. Traffic congestion control

Congestion control is an important while tough issue to study in many types of networks. In underwater sensor networks, high acoustic propagation delay makes congestion control even more difficult. Two methods for congestion control and avoidance have been proposed in CODA (Congestion Detection and Avoidance): (open-loop hop-by-hop backpressure and closed-loop multi-source regulation) for the congestion control problem in ground-based sensor networks. In the open-loop hop-by-hop backpressure mode, a node broadcasts a backpressure message as soon as it detects congestion. The backpressure message will be propagated upstream toward source nodes. In a densely deployed network, the backpressure message will be most likely to reach the source directly. In the closed-loop multi-source regulation, the source uses the ACKs from the sink to self-clock.

In view of the poor quality of acoustic channels, one aspect that needs more investigation is the distinction between loss due to congestion and loss due to external interference. Most schemes assume all loss is congestion related. The higher the loss, the lower becomes the source rate. This will generate problems in underwater systems where random errors and loss are more common. From received packet inter-arrival statistics and from other local measurement, the data sink may be able to infer random loss versus congestion and maintain the rate if loss is not congestion related.

1.4.4. Efficient multi-hop acoustic routing

Like in sensor networks, saving energy is a significant point in underwater sensor networks. Another open issue for data forwarding in underwater sensor networks is to handle node mobility. This requirement makes most energy efficient data forwarding protocols unsuitable for underwater sensor networks. There are many routing protocols propounded for ground-based sensor networks. They have been mainly designed for stationary networks and usually use query flooding as a powerful technique to discover data delivery paths. In underwater sensor networks, most sensor nodes are mobile, and the “network topology” changes dynamically even with small displacements due to multipath. Thus the routing algorithms designed for ground-based sensor networks are no longer workable in underwater sensor networks. Many routing protocols have been proposed.

These protocols generally fall into two categories: proactive routing and reactive routing.

- Proactive routing protocols, the cost of proactive neighbor detection could be very expensive.

- Reactive routing protocols: in on demand routing, routing operations is triggered by the communication demand at sources. In the phase of route discovery, the source looks for to establish a route towards the destination by flooding a route request message, which would be very costly.

With no proactive neighbor detection and with less flooding, it is a big challenge to provide multi-hop packet delivery service in underwater sensor networks with node mobility necessity. One possible direction is to utilize location information to do geo-routing, which seems to be very effective in handling mobility.

1.4.5. Distributed localization and time synchronization

It is very important for every underwater node to know its current position and the synchronized-time with respect to other coordinating nodes in aquatic applications. A Global Positioning System (GPS) system does not work well under the water due to quick absorption of high frequency radio wave. A low cost positioning and time synchronization system while with high precision like GPS for ground-based sensor nodes is not yet available to underwater sensor nodes. Thus, it is expected that underwater sensor networks must rely on distributed GPS-free localization or time synchronization scheme, which is referred to cooperative localization or time synchronization. To realize this type of approaches in a network with node mobility, the key problem is the range and direction measurement process. The common GPS-free approach used in many ground-based sensor networks of measuring the Time-Difference-of-Arrival (TDoA) between a RF and an acoustic or ultrasound signal is no longer feasible as the commonly available RF signal fails under the water. Receiver-signal-strength-index (RSSI) is vulnerable to acoustic interferences like near-shore tide noise, near-surface ship noise, multi-path, and Doppler frequency spread. Angle-of-Arrival (AoA) systems require directional transmission-reception devices, which could be explored, though they usually incur non-trivial extra cost.

Promising approaches may include acoustic-only Time-of-Arrival (ToA) approaches as well as deploying many surface-level radio anchor points. Moreover, the underwater environment with motion of water and variation in temperature and pressure also affects the speed of acoustic signal. Sophisticated signal processing will be needed to compensate for these sources of errors due to the water medium itself.

CHAPTER 2. EXPERIMENTS

More people have done more experiments about UWSN. For example, in [6], the authors tried to present the concept of a highly flexible acoustic modem called the reconfigurable Modem (rModem) that can be used for rapid testing and development of such networks. We can observe the software block diagram of rModem but they didn't show any results.

In [7], the authors tried to build an underwater digital acoustic communications system based on multicarrier modulation technique. This technique is relatively robust in a strong multipath fading environment and has been successfully tested in shallow water. The test system uses 48 carrier frequencies for transmitting 48 parallel hits of data in each packet. The data stream is encoded into a sequence of packets.

In the research works [8], [9], [10], [11] and [12] the authors tried to build an underwater digital acoustic communications system based on FSK modulation. We describe these research works in section 2.2 (Previous experiment) because their results were more interesting.

In [13], the authors developed low-cost and low-power acoustic modems for short-range communications. They presented one design a modem. They described their design rationale followed by details of both hardware and software development.

2.1. Fundamental calculations of underwater acoustics

Now, we try to explain how to calculate the loss in aquatics environments as it has been done in previous experiments and in this master thesis. The signal-to-noise ratio (SNR) equation of an emitted underwater signal at the receiver is governed by the following equation (passive sonar equation):

$$SNR_U = SL - TL_U - NL_U + DI \tag{2.1}$$

where SL is the source level, TL_u is the underwater transmission loss, NL_u is the noise level, and DI is the directivity index. All the quantities in Equation 2.1 are typically in *dB* relative to the power density level of the threshold of human hearing, that is 10^{-12} *W/m*^{2} *(dB re* 10^{-12}*)*. The directivity index DI for our network is zero because we assume omnidirectional hydrophones. Note that this is another conservative assumption, since the use of a directive hydrophone reduces power consumption.

- Source Level: Typically, the specifications of audio speakers indicate the speaker's maximum emitted signal power. As source level *SL* depends on the speaker's transmission power, we must obtain the signal intensity I_t at 1m from the speaker. Assuming that the signal spreading is cylindrical with a unity radius within a distance of 1m from the source I_t can be obtained through the following expression:

$$I_t = \frac{P_t}{2\pi \times 1m \times H} \tag{2.2}$$

where I_t is expressed in W/m^2, P_t is the transmission power in *W*, and *H* is the water depth in m. The following equation determines the source level SL relative to the threshold level of human hearing:

$$SL = 10\ log\left(\frac{I_t}{10^{-12}}\right) \tag{2.3}$$

- Transmission Loss: The transmission loss depends on the surroundings and the transmitted signal pattern has been modeled in various ways, ranging from a cylindrical pattern to a spherical one. When the water mass is deep enough (deep water or water with depth larger than 100 m) the propagation range is not bounded by the sea floor and the surface, so that spherical spreading applies. On the other hand, acoustic signals in shallow water (water with depth lower than 100m) propagate with a cylinder bounded by the surface and the sea floor: as a result cylindrical spreading appears. The following expression describes the signal transmission loss:

$$T\,L_u = 10 \times \mu \log d + \alpha d \times 10^{-3} \tag{2.4}$$

where *d* is the distance between source and receiver in meters, *α* is the frequency dependent medium absorption coefficient in *dB/km*, and TL_u is the transmission loss in *dB*. The variable *μ* depends on the signal spreading pattern. If the acoustic signal spreads in all directions from the sound source, then μ is equal to 2 (spherical spreading). If the acoustic signal spreads in a cylindrical pattern from the source (as is the case signals propagating along the surface or ocean floor), then *μ* equals to 1 (cylindrical spreading).

Equation (2.4) indicates that the transmitted acoustic signal loses energy as it travels through the underwater medium, mainly due to distance dependent attenuation and frequency dependent medium absorption. The authors in [11] conducted measurements of medium absorption in shallow seawater at temperatures of 4°C and 20°C. We derive the average of the two measurements in equation 2.5, which expresses the average medium absorption at temperatures between 4°C and 20°C:

$$\alpha = \begin{cases} 0.0601 \times f^{0.8552} & 1 \leq f \leq 6 \\ 9.7888 \times f^{1.7885} \times 10^{-3} & 7 \leq f \leq 20 \\ 0.3026 \times f - 3.7933 & 20 \leq f \leq 35 \\ 0.504 \times f - 11.2 & 35 \leq f \leq 50 \end{cases} \quad (2.5)$$

where f is frequency in *KHz*, and α is in *dB/Km*. Through Equation (2.5), we can compute medium absorption for any frequency range of interest. We use this value for determining the transmission loss at various inter-node distances through Equation (2.4) which enables us to compute the source level in Equation (2.3) and subsequently to compute the power needed at the transmitter.

- Noise Level: Factors contributing to the noise level NL_u in shallow water networks include waves, shipping traffic, wind level, biological noise, seaquakes, volcanic activity, and rain, and the impact of each of these factors on NL_u depends on the particular setting. Shipping activity may dominate noise figures in bays or ports, while water currents are the primary noise source in rivers.

2.2. Previous experiments

More people make more experiments about UWSN ([8], [9], [10], [11], [12], [13] and [14]). These experiments use different modulations (FSK, OFDM) and tried to find the best bit rate and maximum distance. To determine the best bit rate, they transmitted to use different bit rate and distance. Some experiments used PC speakers and PC microphone protected by elastics membranes. Others used underwater speakers and hydrophone.

One of previous experiments is very interesting because its results provide us with more information [11]. In this paper, the SNR, bit error and maximum transmission distance between transmitter and receptor. They tried to build an underwater digital acoustic communications system based on FSK modulation for to build underwater sensor network. Next we will explain the experiments carried out in [11] with more detail as the basis to later introduce our own experiments.

After reading the papers of Raja Jurdak we tried to contact him because he had done very interesting experiments in underwater environments and we wished to learn more about. He was very friendly and he offered me the opportunity to travel to Dublin to talk about his research work. So I visited him and he answered all my questions and explained me the functioning of his synchronization, encoder and decoder algorithm. What is more, he handed over his code.

The experiments carried out tried to compare aerial acoustic communication with underwater acoustics communication. First, they use PC speakers as the transmitter and a generic microphone as a receiver. Afterwards, they use Tmote Invent speakers to do aerial and underwater experiments. These speakers work well in aquatics environments. The favorable frequencies of operation for the hardware enable them to design a software FSK mode.

First they measure SNR of the channel for different frequencies and distances in their experiments to profile the aerial channel. They have limited their analysis to the frequency range from 400 Hz to 6700 Hz at 100 Hz increments and to the distance range from 1 m to 7 m at 2 m increments. In the picture Fig 2.1 we can see the SNR in aerial environments. We can observe the range frequency (1000 Hz to 2000 Hz) used in the experiments is more constant.

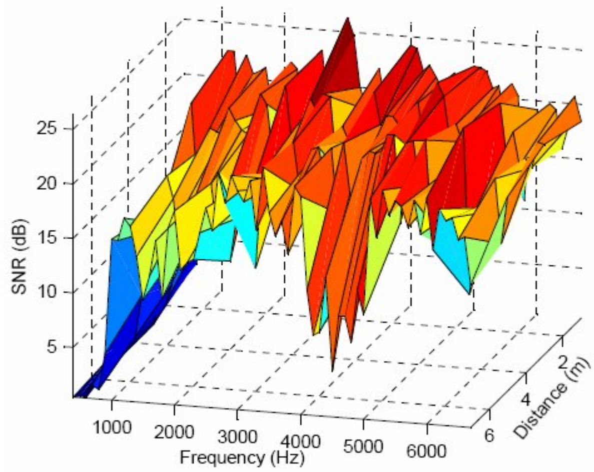


Fig. 2.1 Received signal quality of the PC speakers in the indoor aerial channel

The picture Fig. 2.2 shows two things. The solid plot represents the measured SNR and the transparent plot represents the expected SNR. The SNR in aquatic environments decreases more severely if we compare with aerial environments. But the range frequency (1000 Hz to 2000 Hz) used in the experiments is more constant.

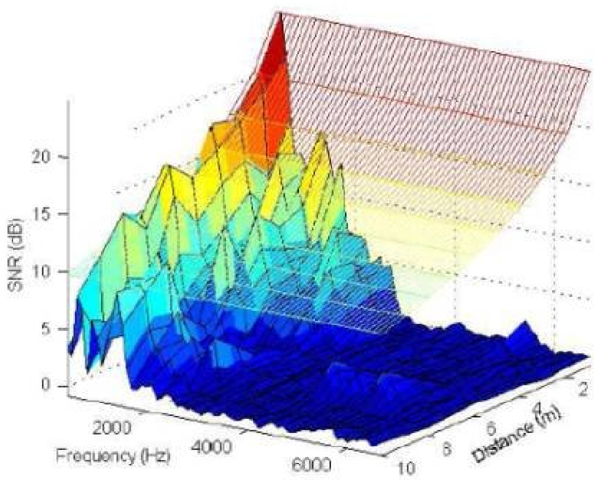


Fig. 2.2 Received signal quality of PC speakers in the aquatic environments

After these measures, they decided use eight frequencies (1000, 1200, 1300, 1500, 1600, 1700, 1800 and 2000 Hz). Then they calculated channel capacity using the Shannon-Hartley expression.

$$C = B\log_2(1 + SNR)\ bps \tag{2.6}$$

In the picture Fig. 2.3, we can see the channel capacity for each distance when the experiments have been carried out using generic PC speakers and a PC microphone in the aquatic environment. The dip that is observer at 6m and 7m is an artifact most likely due to the pool geometry or to the presence of a large drain near the components. Overall, the expected error-free channel capacity drops steadily from about 30 bps at 1m to 10 bps at 10m. Observing these results, we can conclude that PC speakers and microphone don't work well in aquatics environment.

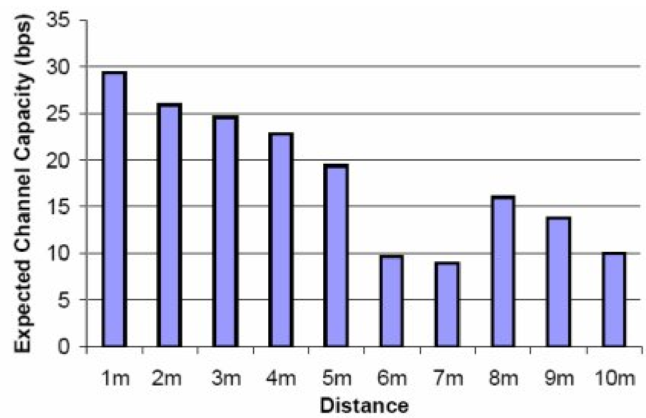


Fig. 2.3 Channel Capacity of PC speakers and microphones

With this information, they tried to transmit with de PC speakers. They used different bit rates and transmission distances. In Fig. 2.3 we can see the result. They concluded the best bit rates were 12 bps and 24 bps, because the percentage of symbols correctly with other rates was smaller than 50%.

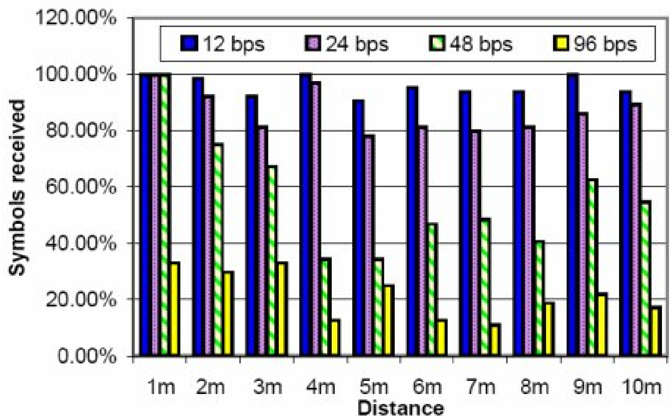


Fig. 2.4 Percentage of Symbols correctly received with PC speakers and microphone

The next step was to repeat the measures with Tmotes speakers. With the same bit rate the Tmote speakers the communications was possible for larger distances. The distance was increased from 10 m with generic PC hardware to 17m with Tmote speakers and the percentage of correctly received symbols with the same bit rate was increased with Tmote speakers, too. In picture Fig. 2.4., we can see the result of the measures.

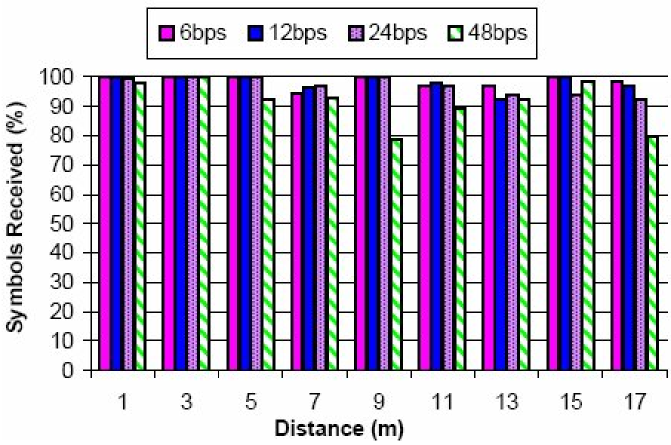


Fig. 2.5 Percentage of Symbols correctly received with Tmote speakers

We explain the algorithm used in the annex.

2.3. Our experiments

Our experiment entailed transmitting audio signals in underwater environment. We used FSK modulation with different bit rates. These signals were generated by Matlab that encoded symbols and saved result in wav files – one file for bit rate. These files were reproduced by aquatic speaker and they were recorded by hydrophone. The wav files were processed by Matlab that decoded the symbols.

For our experiments we have carefully searched for the well-suited hardware specifically designed to underwater networks because we wished to know if with specific underwater hardware it was possible to improve the results obtained in the previously described research works. Therefore we have decided to use a LAB-40 hydrophone [15] and a DRS-8 underwater speaker [16]. LAB-40 hydrophone is an omnidirectional hydrophone. In picture Fig. 2.6 shows the connection diagram.

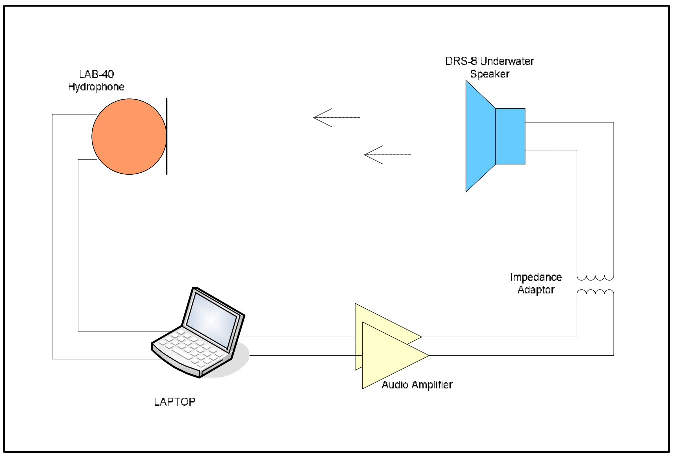


Fig. 2.6 Connection diagram

The LAB-40 hydrophone has a wide frequency range of 5 Hz to 85 kHz and it a frequency response is above and below human hearing (40 to 20,000 Hz). The LAB-40 can pick up the sounds of the Sperm or Blue Whale (5 to 15 Hz) to the Dolphin's (60,000 Hz and above), fish tags, acoustic transmitters and beacons (15 KHz to 80 KHz).

The DRS-8 underwater speaker is omnidirectional at 2 KHz, but it is directional at higher frequencies. It has a wide frequency range of 200 Hz to 32 KHz. Maxim input power rating is 200 Watts. Input impedance is 4000 Ohms and varies with frequency. But can be matched to 4 or 8 Ohms amplifier output tap with the included audio isolation impedance matching transformer assembly.

We didn’t plug directly the LAB-40 hydrophone into the microphone jack of a laptop. We used a 20 dB in-line boost amplifier included with the LAB-40 hydrophone. If we didn’t use it, we didn’t detect any signal. We needed an audio amplifier in order to amplify the signal from the speaker jack of a laptop before it enters the DRS-8 underwater speaker. We used the audio amplifier E-11 of CEBEK [17]. We needed a mobile supply for audio amplifier. Therefore, we used a battery of 12 Volts and 7.2 Ah.

The picture Fig. 2.7 shows the location of the testing site for these experiments. It is the lake of Mediterranean Technology Park (PMT) in Castelldefels.

Fig. 2.7 Location of the testing site for experiments

We wished to know if using our specific underwater hardware it was possible to improve the results obtained in the previously described research works [11]. Therefore we repeated of the experiments from previous research works. We wanted to measure the SNR for various frequencies and distances in our experiments to profile our underwater hardware for the underwater channel. We limited our analysis to the frequency range from 50 Hz to 6000Hz. Each tone is sent in a square signal for duration of 0.25 seconds, and the full signal consists of a sequence of the square signals separated by guard times of 0.25 seconds. The next picture shows the measured SNR in aquatic channel using the specific already described underwater hardware. We can observe that the received signal quality is increased slightly with increasing frequencies. The received signal quality increases slightly with increasing frequency from 500 Hz. For frequencies between 4 KHz and 6 KHz the signal quality is stable with minor variations among frequencies. From section 2.1 (Fundamental Calculation of Underwater Acoustics) it is expected that the SNR decreases with increasing distances because the passive sonar equation depends on *TL* and it increases with distance. On the other hand the SNR will experience minor variations with increasing frequency and what is more, the transmission power can be adjusted to achieve the desired SNR level. Therefore we can conclude that the results obtained are right because the received SNR in the underwater environment keep with distance and only suffers minor modifications with respect to changing frequencies.

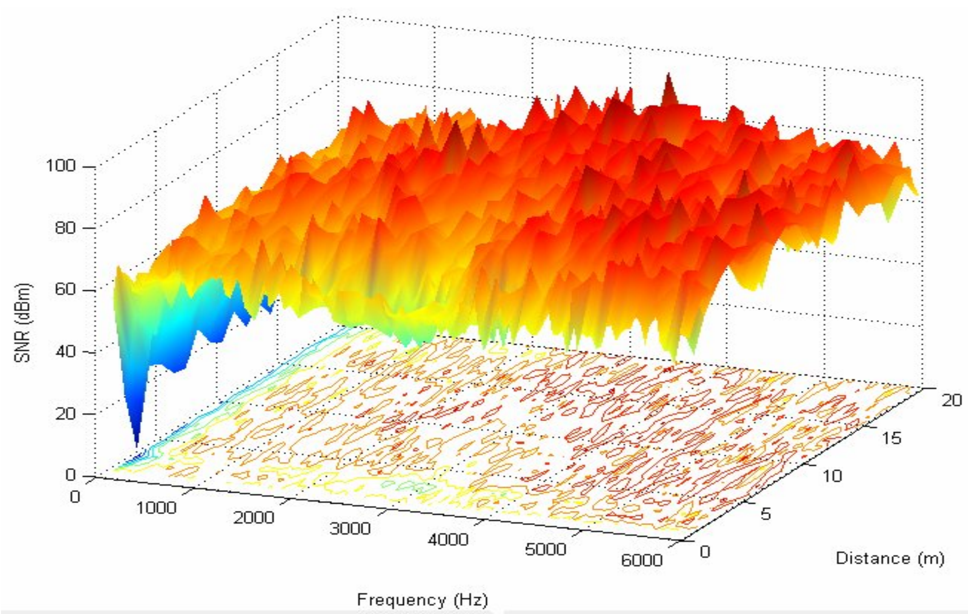


Fig. 2.8 Representation 3-D of SNR of DRS-8 underwater speaker and LAB-40 hydrophone in underwater channel

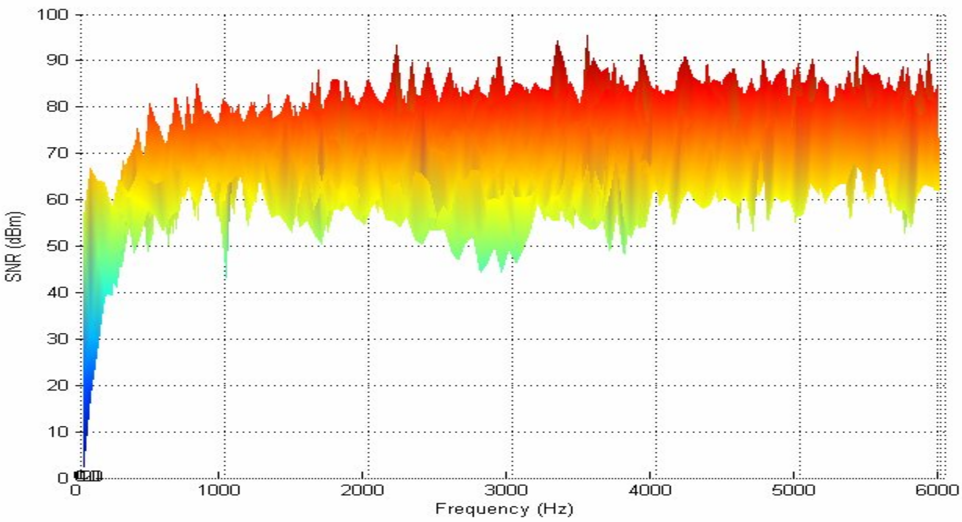


Fig. 2.9 Representation 2-D of SNR of DRS-8 underwater speaker and LAB-40 hydrophone in underwater channel

With the measures of SNR we can now calculate the channel capacity. We have used the Shannon-Hartley expression. We have used the same frequencies that in research work [11] (1000, 1200, 1300, 1500, 1600, 1700, 1800 and 2000 Hz). The next figure shows (see Fig. 2.8) the results obtained. We cans distinguish between two different curves. The first one shown the expected channel capacity computed doing the average of the resulting SNR for all frequencies- and second one is the expected channel capacity considering only the minimum measured SNR for all frequencies. We can observer that the values of the channel capacity are about 150 bps (using the minimum SNR for the calculations) and 200 bps (using the average SNR for the calculations).

If we compare our results with the results of experiments from [11], we can observe that the resulting SNR is much better in our experiments. The minimum SNR -in the frequencies that we used in the research work [11] and our experiments- is 45 dB, it is better than 6.95 dB of Tmote speakers from the paper [11]. This is predictable, because the DRS-8 underwater speaker and the LAB-40 hydrophone were built specifically build for aquatics environments, whereas the remaining speakers and microphones used in the paper [11] were built for aerial environments.

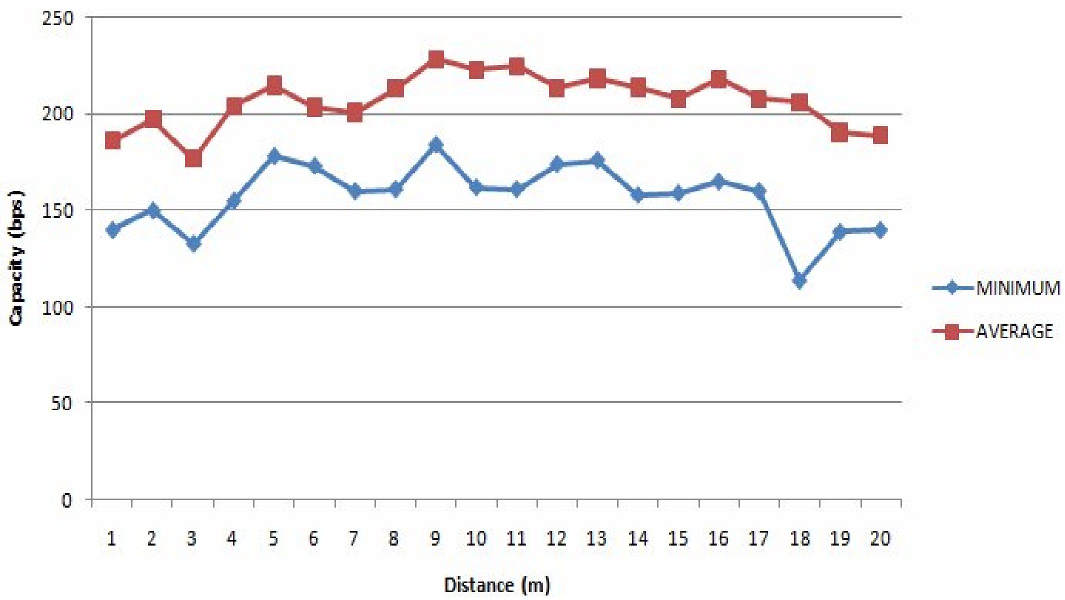


Fig. 2.10 Capacity of DRS-8 underwater speaker and LAB-40 hydrophone

After having computed the channel capacity, we were able to begin to transmission. We used the same algorithm from the experiments in [11]. We transmitted the same symbols than in the previous experiment (64 symbols) using FSK – in annexe B we explain this modulation. There is only one difference. We transmitted using more bit rates. The results are shown in Fig 2.11.

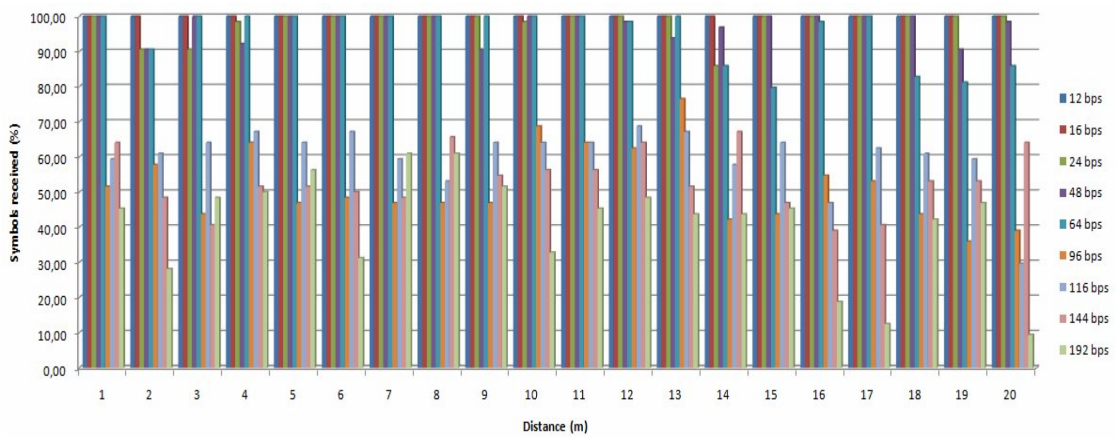


Fig. 2.11 Percentage of Symbols correctly received with the first transmission algorithm

This figure shows the percentage of symbols correctly received. In [11] the authors were able to decode at least 79% of the transmitted symbols using Tmote Invent speakers as hardware and with bit rates from 12 bps up to 48 bps. On the other hand, Fig. 2.11 shows that the percentage of symbols correctly received is increased for bit rates from 12 bps up to 48 bps; the lowest value is 85,95% of symbols correctly received with 24 bps bit rate and 14 m, however we can notice that with the bit rate of 48 bps the results are improved because the receiver decodes between 90,63 and 100% of the symbols. The percentage of symbols correctly received with range bit rate of 92 bps to 192 bps severely diminishes. Therefore we can conclude that the decoding capability of the receiver with the specific underwater hardware in our experiments is better than the Tmote Invent Speakers for aquatic environments. We are able to correctly decode the symbols received even if they have been transmitted at larger distances (up to 20 m) and with higher bit rates (up 192 bps). We are able to correctly transmit at 64 bps up to 20 meters. We can conclude that the results are right and as we expected they fit in with the obtained SNR.

Afterwards, after a careful analysis, we have decided to modify the transmission algorithm in order to improve the results. The already presented algorithm (first algorithm) tries to find an index vector for each frequency (symbol) and it rounds down the value of frequency because the index vector must be a whole number. After the algorithm has found the power of each symbol and it looks for the symbol with maximum value, it deduces that this is the transmitted symbol. The modifications introduced (second algorithm) try to find the index vector with the maximum power spectrum density vector. With this index vector we calculate the transmitted frequency nearest to the theoretical frequency. Therefore with the modified version of the algorithm the nearest frequency (symbol) to the measured frequency is found. In the annexe A the main modifications of the code are described in detail. We have specifically modified the symbol coder. When we finished the modifications, we repeated the experiments; we transmitted the same symbols at the same bit rates as in Fig.2.11 but now we used the new code to decode the symbols correctly received. Fig. 2.12 shows the results obtained.

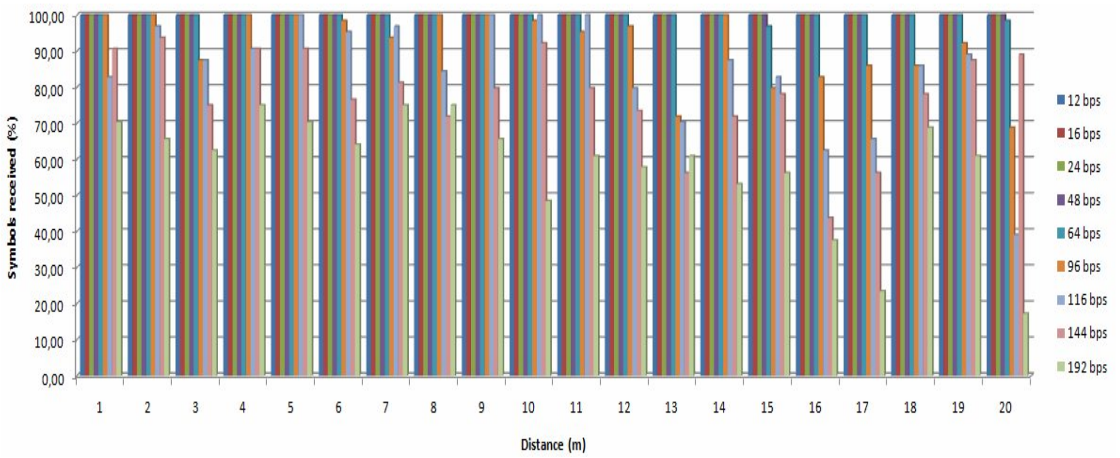


Fig. 2.12 Percentage of Symbols correctly received with second transmission algorithm

As we can notice, using the modified version of the symbol decoding algorithm, it is possible to transmit at higher bit rates with a much better percentage of symbols correctly received. We brought near capacity of Shannon-Hartley limit. We were able to correctly transmit at a bit rate of 96 bps up to 20 meters. The values in table 2.1 (percentage of symbols correctly received with the first algorithm) and 2.2 (percentage of symbols correctly received with the second algorithm) help us to better compare the results with our modification and to observe the communication improvements.

Distance	12 bps	16 bps	24 bps	48 bps	64 bps	96 bps	116 bps	144 bps	192 bps
1	100,00	100,00	100,00	100,00	100,00	51,56	59,38	64,06	45,31
2	100,00	100,00	90,63	90,63	90,63	57,81	60,94	48,44	28,13
3	100,00	100,00	90,63	100,00	100,00	43,75	64,06	40,63	48,44
4	100,00	100,00	98,44	92,19	100,00	64,06	67,19	51,56	50,00
5	100,00	100,00	100,00	100,00	100,00	46,88	64,06	51,56	56,25
6	100,00	100,00	100,00	100,00	100,00	48,44	67,19	50,00	31,25
7	100,00	100,00	100,00	100,00	100,00	46,88	59,38	48,44	60,94
8	100,00	100,00	100,00	100,00	100,00	46,88	53,13	65,63	60,94
9	100,00	100,00	100,00	90,63	100,00	46,88	64,06	54,69	51,56
10	100,00	100,00	98,44	100,00	100,00	68,75	64,06	56,25	32,81
11	100,00	100,00	100,00	100,00	100,00	64,06	64,06	56,25	45,31
12	100,00	100,00	100,00	98,44	98,44	62,50	68,75	64,06	48,44
13	100,00	100,00	100,00	93,75	100,00	76,56	67,19	51,56	43,75
14	100,00	100,00	85,94	96,88	85,94	42,19	57,81	67,19	43,75
15	100,00	100,00	100,00	100,00	79,69	43,75	64,06	46,88	45,31
16	100,00	100,00	100,00	100,00	98,44	54,69	46,88	39,06	18,75
17	100,00	100,00	100,00	100,00	100,00	53,13	62,50	40,63	12,50
18	100,00	100,00	100,00	100,00	82,81	43,75	60,94	53,13	42,19
19	100,00	100,00	100,00	90,63	81,25	35,94	59,38	53,13	46,88
20	100,00	100,00	100,00	98,44	85,94	39,06	29,69	64,06	9,38

Table 2.1 Percentage of symbols correctly received of first transmission algorithm

Both tables show that the second algorithm can transmit more symbols correctly in all frequencies and all bit rates.

Distance	12 bps	16 bps	24 bps	48 bps	64 bps	96 bps	116 bps	144 bps	192 bps
1	100,00	100,00	100,00	100,00	100,00	100,00	82,81	90,63	70,31
2	100,00	100,00	100,00	100,00	100,00	100,00	96,88	93,75	65,63
3	100,00	100,00	100,00	100,00	100,00	87,50	87,50	75,00	62,50
4	100,00	100,00	100,00	100,00	100,00	100,00	90,63	90,63	75,00
5	100,00	100,00	100,00	100,00	100,00	100,00	100,00	90,63	70,31
6	100,00	100,00	100,00	100,00	100,00	98,44	95,31	76,56	64,06
7	100,00	100,00	100,00	100,00	100,00	93,75	96,88	81,25	75,00
8	100,00	100,00	100,00	100,00	100,00	100,00	84,38	71,88	75,00
9	100,00	100,00	100,00	100,00	100,00	100,00	100,00	79,69	65,63
10	100,00	100,00	100,00	100,00	100,00	98,44	100,00	92,19	48,44
11	100,00	100,00	100,00	100,00	100,00	95,31	100,00	79,69	60,94
12	100,00	100,00	100,00	100,00	100,00	96,88	79,69	73,44	57,81
13	100,00	100,00	100,00	100,00	100,00	71,88	70,31	56,25	60,94
14	100,00	100,00	100,00	100,00	100,00	100,00	87,50	71,88	53,13
15	100,00	100,00	100,00	100,00	96,88	79,69	82,81	78,13	56,25
16	100,00	100,00	100,00	100,00	100,00	82,81	62,50	43,75	37,50
17	100,00	100,00	100,00	100,00	100,00	85,94	65,63	56,25	23,44
18	100,00	100,00	100,00	100,00	100,00	85,94	85,94	78,13	68,75
19	100,00	100,00	100,00	100,00	100,00	92,19	89,06	87,50	60,94
20	100,00	100,00	100,00	100,00	98,44	68,75	39,06	89,06	17,19

Table 2.2 Percentage of symbols correctly received of second transmission algorithm

CHAPTER 3. Environmental impact

Sound is very important to whales, dolphins and porpoises for navigation, communication and finding food. However, in the lower frequencies ship noise from propeller driven ships is now dominant. In addition, active sonar and seismic surveys are the loudest forms of underwater noise. Unlike terrestrial noise, marine noise is virtually unregulated. There is now strong evidence of fatal stranding of whales following military exercises and other sub-lethal effects including permanent tissue damage, temporary hearing loss and avoidance have been documented in a variety of whales and other species. For these reasons, it is important to considerer the environmental impact of UWSN’s.

Our experiments use the frequency range of 800 Hz at 2000 Hz (two reference frequencies and 8 frequencies associate to symbols). There are more cetacean species. Some examples of cetacean communication are the Sperm whales use frequencies range of 15 Hz to 15 KHz, the Blue Whales use frequencies range of 10 Hz to 60 Hz or the Dolphin’s use frequencies range of 100 to 150 KHz. In summary the frequencies range is larger, it goes from a few tens of Hz to tens of thousands of Hz, and our signal can interfere cetacean communications. But the signal power is very small and the transmission distance between devices is short too. We don’t need a very large distance because it is not necessary for the network that we want to build.

In summary, our system will probably not to interference cetacean communications because transmission distance is very short. There are other noise sources much more important that our system.

CHAPTER 4. CONCLUSIONS AND FURTHER WORKS

After these results, we think that there are more things to do in order to improve transmission distance and bit rate. One example is the modification of the decoder algorithm. With a small modification we were able to increase the percentage of symbols correctly received. Second, our algorithm calculates the index vector of the maximum power spectrum density and it finds the nearest theoretical frequency (symbol) to the measured frequency. On the contrary, the first algorithm calculates the index vector of each frequency (symbol). After, it has found the value to each frequency and it elects bigger. But the first decoder algorithm can be more severely affected by interference and noise because it doesn't look for the maximum value of power spectrum density, it look for the maximum value for a concrete index – the index associates at symbols.

However, we think that to modify the decoder algorithm can increase the bit rate and to approach to channel capacity. We can increase the bit rate using more frequencies, more symbols, or increasing the distance between frequencies and reducing the bit time. These improvements can be considered as further work.

What is more, we could try to use other modulations to improve the bit rate and transmission distance. More people do research with CDMA and OFDM modulations in order to improve the transmission of the underwater acoustic channel. These modulations can use spectrum more effective that FSK. But complexity is bigger and there would be an increase in the computational load. As future work we recommend to do the same experiments and modulate using OFDM.

Information about research using OFDM and some experiments related can be found in [14]. We include a brief summary of OFDM in annexe C. For more information we can read papers [18], [19], [20], [21], [22].

BIBLIOGRAPHY

[1] J. H. Cui, J. Kong, M. Gerla, S. Zhou, "Challenges: Building Scalable Mobile Underwater Wireless Sensor Networks for Aquatic Applications", Special issue of IEEE Network on Wireless Sensor Networking, May 2006.

[2] I. Akyildiz, D. Pompili, T. Melodia, "Challenges for efficient communication in underwater acoustic sensor networks", Special issue on embedded sensor networks and wireless computing of ACM SIGBED, July 2004.

[3] M. Chitre, L. Freitag, E. Sozer, S. Shahabudeen, M. Stojanovic, J. Potter, "An architecture for Underwater Networks", Proceedings of the IEEE Oceans'06 Asia Pacific Conference, Singapore, May 2006.

[4] J. Heidemann, W. Ye, J. Wills, A. Syed, Y. Li, "Research Challenges and Applications for Underwater Sensor Networking", Proceedings of the IEEE Wireless Communications and Networking Conference, Las Vegas, April 2006.

[5] B. Benson, G. Chang, D. Manov, B. Graham, R. Kastner, "Design of a Low-cost Acoustic Modem for Moored Oceanographic Applications", Proceedings of the 1st ACM international workshop on Underwater networks, Los Angeles, 2006.

[6] E. Mutlu, M. Stojanovic, "Reconfigurable Acoustic Modem for Underwater Sensor Networks", Proceedings of the 1st ACM international workshop on Underwater networks, Los Angeles, 2006.

[7] L. Yeung, R. Bradbeer, E. Law, A. Wu, B. Li, Z. Gu, "Underwater Acoustic Modem Using Multi-Carrier Modulation", Proceedings of the IEEE Oceans '03, Boston, September 2003.

[8] R. Jurdak, C. Videira, P. Aguiar, P. Baldi, "A comparative analysis and experimental study on wireless aerial and underwater acoustic communications", Proceedings of the IEEE International Conference on Digital Telecommunications (ICDT), Cap Esterel, August 2006.

[9] R. Jurdak, C. Videira, P. Baldi, "Software acoustic modems for Short Range Mote-based Underwater Sensor Networks", Proceedings of IEEE Oceans, Singapore, May 2006.

[10] R. Jurdak, A. Ruzzelli, G. O'Hare, C. Videira, "Reliable Symbol Synchronization in Software-Driven Acoustic Sensor Networks", Proceeding of IEEE GlobeCom, Washington D.C., 2007.

[11] R. Jurdak, P. Aguiar, P. Baldi, C. Videira, "Software Modems for Underwater Sensor Networks", Proceedings of IEEE/OES Oceans'07. Aberdeen, June 2007.

[12] R. Jurdak, A. Ruzzelli, G. O'Hare, C. Videira, "Design Considerations for Deploying Underwater Sensor Network", Proceedings of IEEE Conference on Sensor Communication, Valencia, October 2007.

[13] J. Wills, W. Ye, J. Heidemann, "Low-Power Acoustic Modem for Dense Underwater Sensor Networks", Proceedings of the 1st ACM international workshop on Underwater networks, Los Angeles, 2006.

[14] S. Mason, R. Anstett, N. Anicette, "A Broadband Underwater Acoustic Modem Implementation Using Coherent OFDM", The National Conference On Undergraduate Research, California, April 2007.

[15] http://www.whaleresearch.com/thecenter/OrcaSound.html

[16] http://www.oceanears.com/2001speaker1page.html

[17] http://www.cebek.com/Producte.aspx?referencia=428&article=etapa-de-potencia-20-w-roadpower-

[18] R. Chang, R. Gibby, "A Theoretical Study of Performance of an Orthogonal Multiplexing Data Transmission Scheme", IEEE Transactions on Communications, August 1968.

[19] B. Saltzberg, "Performance of an Efficient Parallel Data Transmission System", IEEE Transactions on Communications, December 1967.

[20] J.H. Stott, "The how and why of COFDM", EBU Technical Review, 1998.

[21] B. Li, S. Zhou, M. Stojanovic, L. Freitag, "Pilot-tone based ZP-OFDM Demodulation for an Underwater Acoustic Channel", Proceedings of MTS/IEEE OCEANS conference, Boston, September 2006.

[22] http://en.wikipedia.org/wiki/OFDM

[23] J. G. Proakis, D. G. Manolakis, "Tratamiento Digital de Señales. Principios, algoritmos y aplicaciones", Prentice Hall, Madrid, 1998.

[24] A. V. Oppenheim, R. W. Schafer, J. R. Buck, "Tratamiento de Señales en Tiempo Discreto", Prentice Hall Iberia, Madrid, 2000.

[25] C. S. Burrus, J. H. McClellan, A. V. Oppenheim, T. W. Parks, R. W. Schafer, H. W. Schuessler, "Ejercicios de Tratamiento de la Señal. Utilizando Matlab v.4", Prentice Hall, Madrid, 1998.

[26] S. S. Soliman, M. D. Srinath, "Señales y Sistemas continuos y discretos", Prentice Hall Iberia, Madrid, 1999.

[27] A. Gilat, "Matlab. Una Introducción con ejemplos prácticos", Editorial Reverté, Barcelona, 2006.

[28] C. Pérez, “Matlab y sus Aplicaciones en las Ciencias y la Ingeniería”, Pearson Educación, S. A., Madrid, 2002.

ANNEXE

A. Transmission Algorithms

Now we explain both transmission algorithms. The first algorithm was developed by Raja Jurdak and used in [11]. The second algorithm is our modification of the first one.

The previous algorithm has two parts. The first part consists of synchronization of software modems and the second part consists of the decoder.

They used the S^4 method of research work [10] for synchronizing the transmitter and receiver. This technique proposes to add a pre-amble and a post-amble between the transmitted data symbols. The pre-amble and post-amble are a signal signature that includes the two square waves of predetermined frequencies, the duration of the square wave signals and guard time between them are known. This signal shape allows the correlation function at the receiver to detect the surge at the beginning of data. The Fig. A.1 shows an example of a signal with two waves in the pre-amble and post-amble.

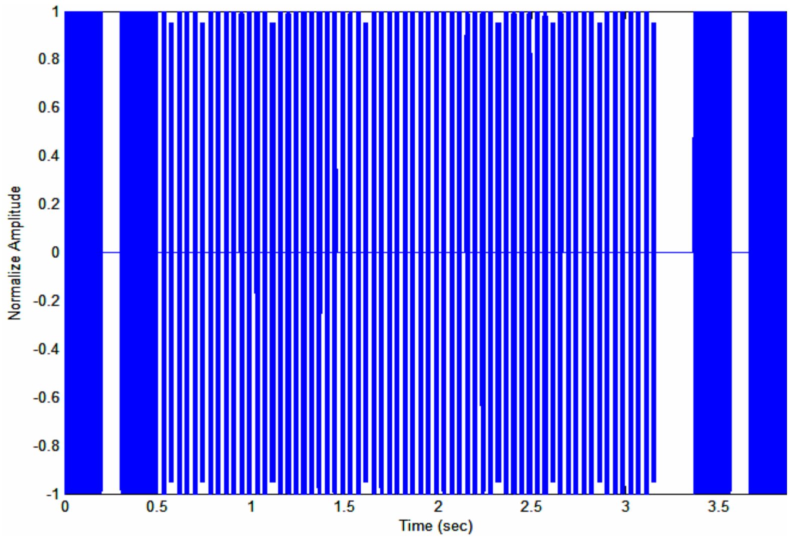
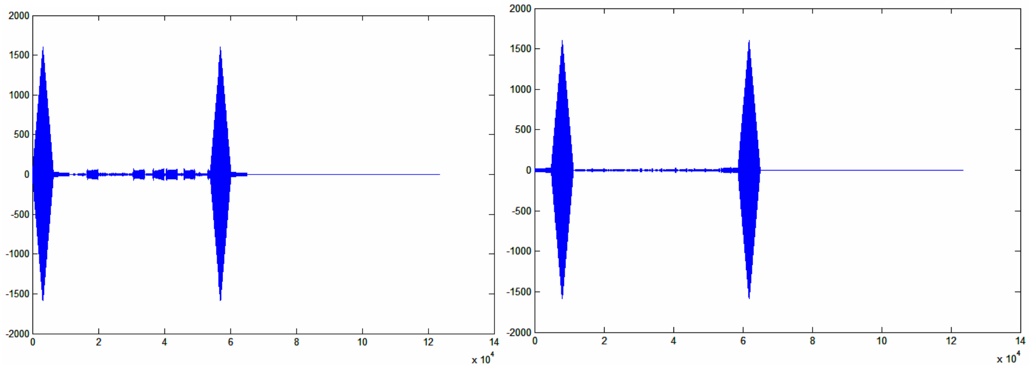


Fig. A.1 Signal example

The receiver can correlate both reference signals with the incoming. The Fig. A.2 shows the correlation outputs of the received signal and the two reference signals (first reference frequency is 800 Hz and second is 900 Hz).

Visual inspection of the 4 peaks shows the preamble in the first correlation output and the post-amble in the second correlation output have the sharper peaks. The algorithm then checks each peak's properties in order to determine the best peak to consider. The algorithm selects the peak with the highest peak quality and uses the peak sample to synchronize to the received signal. If best peak corresponds to the post-amble, the algorithm subtracts the fixed frame length to compute the start of the data symbols.



(a) (b)

Fig. A.2 Result of correlation of first reference signal with input signal (a). Result of correlation of second reference signal with input signal (b)

When the algorithm knows the location of first the symbol of the frame, it can filter the signal and decode the symbols. Algorithm filter signal because we only interests range frequency of 1000 Hz to 2000 Hz, because this range was that we used in FSK modulation.

Now we try to explain the decoder algorithm step to step (the algorithms steps have been summarized in Fig. A.3):

- Calculate Fast Fourier Transform (FFT) of filter signal.
- Calculate power spectrum density.
- For each symbol.
    - o Calculate the index for each symbol in power spectrum density vector from the frequency of each symbol (to round down because index vector whole number, therefore certain precision is lost).
    - o Find the power of each symbol from calculated index vector.
- The transmitted symbol is that with the maximum power value.

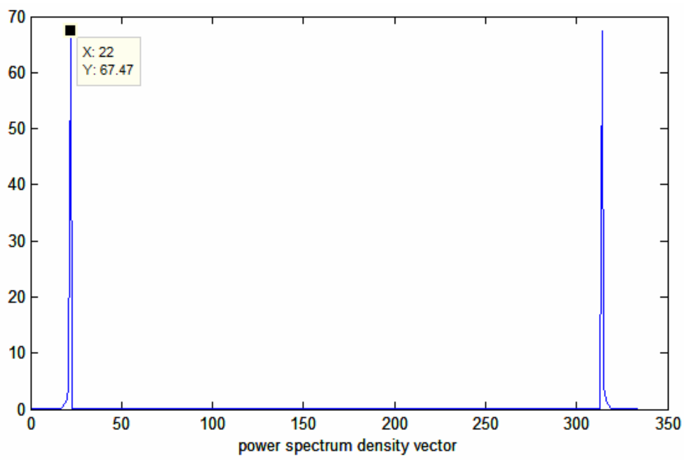


Fig. A.3 Representation power spectrum density vector

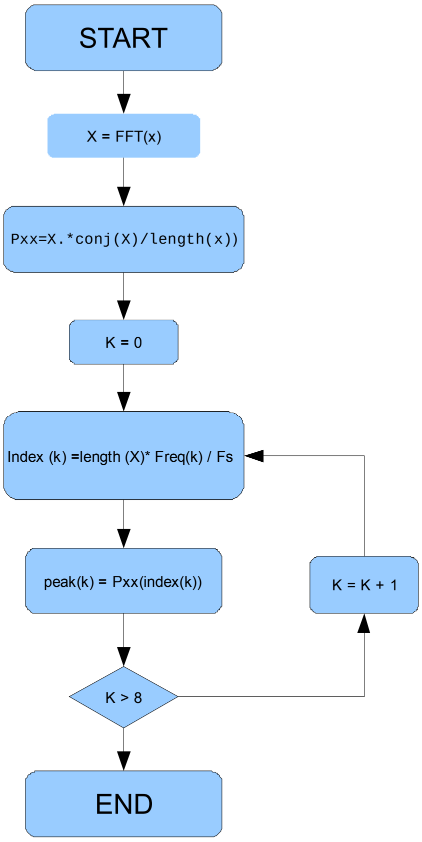


Fig. A.4 First Encoder Algorithm

Now we explain our algorithm. The only difference between both algorithms is the symbol decoder. Our algorithm uses the same synchronization technique, the S^4 method [10].

Now we try to explain our decoder algorithm step to step (the algorithms steps have been summarized in Fig. A.4):

- Calculate FFT of filter signal.
- Calculate power spectrum density.
- Find maximum of power spectrum density vector.
- Calculate frequency from index of maximum of power spectrum density vector.
- Calculate absolute value of the difference between the measured frequency and the theoretic frequency of each symbol.
- Find the nearest frequency to the measured frequency.
- The symbol associated to the nearest frequency is transmitted symbol.

For we made this algorithm, we needed to read the books [23], [24], [25], [26], [27] and [28]

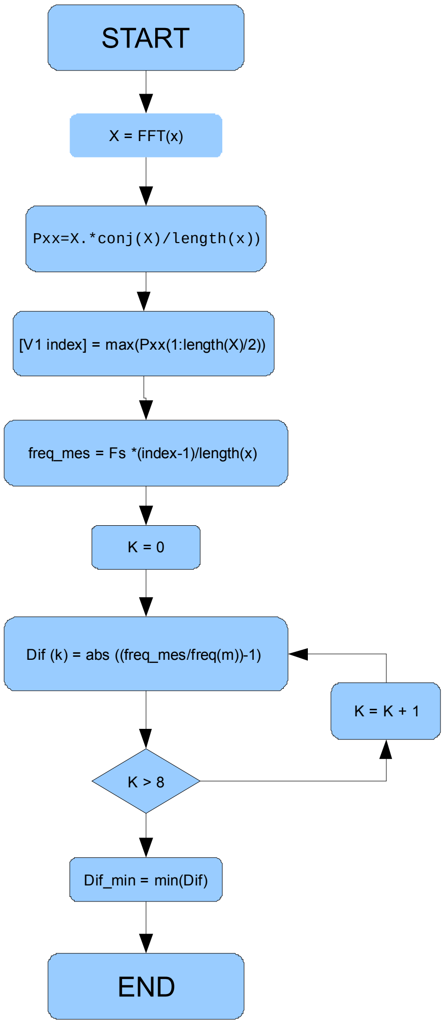


Fig. A.5 Second Encoder Algorithm

B. Frequency-Shift Keying (FSK)

Frequency-shift keying (FSK) is represented conceptually by Fig. B1, where the digital signal *x(t)* controls a switch that selects the modulated frequency from a bank of M oscillators. The modulated signal is discontinuous at every switching instant $t = kD$. Unless the amplitude, frequency, and phase of each oscillator has been carefully adjusted, the resultant output spectrum will contain relatively large sidelobes which don't carry any additional information and thus waste bandwidth.

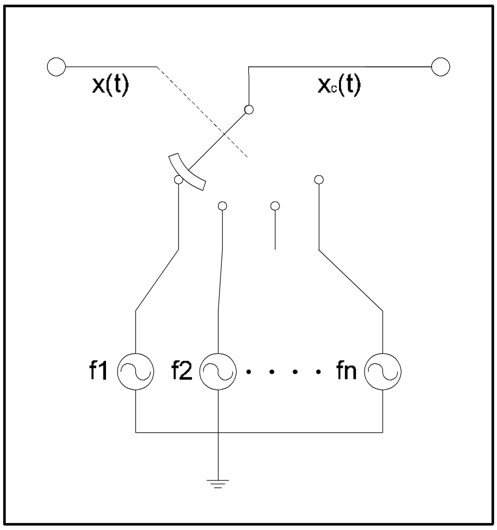


Fig. B.1 Digital frequency modulation FSK

First, consider M-ary FSK. Let all oscillators in Fig. B.1 have the same amplitude A_c and phase θ, and let their frequencies be related to a_k by

$$f_k = f_c + f_d a_k \qquad a_k = \pm 1, \pm 3, \ldots, \pm(M-1) \tag{B.1}$$

which assumes that M is even, then

$$x_c(t) = A_c \sum_k [\cos(\omega_c t] + \theta + \omega_d a_k t) p_D(t - kD) \tag{B.2}$$

where $\omega_d = 2\pi f_d$. The parameter f_d equals the frequency shift away from f_c, when $a_k = \pm 1$, and adjacent frequencies are spaced by $2f_d$. Continuity of $x_c(t)$ at $t = kD$ is assured if $2\omega_d D = 2\pi N$ where N is an integer.

The most common form of frequency shift keying is binary FSK. The binary FSK uses two discrete frequencies to transmit binary (0's and 1's) information. One interesting facet of the binary FSK modulation scheme is that when it is generated using a quadrature modulator, the baseband waveform consists of two sinusoids which use phase shifting to move from one symbol to the next. In this scenario, baseband I and Q signals adjust from 90 degrees to 270 degrees out of phase to mark each symbol of the FSK baseband waveform. For that reason, the binary FSK modulation scheme is sometimes referred to as "phase reversal" as well.

Minimum frequency-shift keying or minimum-shift keying (MSK) is a particularly spectrally efficient form of coherent frequency-shift keying (CFPSK). In MSK the difference between the higher and lower frequency is identical to half the bit rate. As a result, the waveforms used to represent a 0 and 1 bit differ by exactly half a carrier period. This is the smallest FSK modulation index that can be chosen such that the waveforms for 0 and 1 are orthogonal.

C. Orthogonal frequency division multiplexing (OFDM)

Now, we briefly try to explain OFDM modulation for more information read the references [18], [19], [20], [21] and [22].

C.1 Introduction to OFDM

Signal design techniques for transmitting data by frequency division orthogonal multiplexing have been studied by a number of authors. Orthogonal signals have been used which have widely spread frequency spectra. Consequently, when these signals are transmitted through an essentially band-limited transmission medium, certain portions of the signal spectrum will be cut off and interference will result. A system using such signals has the following properties.

- The maximum signaling rate for the given channel (Nyquist rate) can be approached without the use of sharp cutoff filters. This is probably the most important property in high-speed data transmission.

- With the entire frequency band divided into a number of narrow channels, the frequency division multiplexing system is less sensitive to wide-band impulse noise interference and channel distortion than is the ordinary time division multiplexing system.

- The parallel system provides maximum signal-to-noise ratio for analog data transmission and minimum probability of error for digital data transmission in the presence of band-limited white Gaussian noise under the constraint of no inter-symbol or inter-channel interference.

- This can be avoided in the parallel system by adaptively dropping out only the channels affected. The flexibility of the parallel system enables to adapt to a variety of noise environments.

OFDM involves modulating the data onto a large number of carriers using the FDM technique. The key features which make it work, in a manner that is so well suited to terrestrial channels, include orthogonality and the addition of a guard interval.

The use of multiple carriers follows from the presence of significant levels of multipath. Suppose we modulate a carrier with digital information. During each symbol, we transmit the carrier with a particular phase and amplitude which is chosen from the constellation in use. Each symbol conveys a number of bits of information, equal to the logarithm (to the base 2) of the number of different states in the constellation.

If this signal is received via two paths, with a relative delay between them. Taking transmitted symbol n as an example, the receiver will attempt to demodulate the data that was sent in this symbol by examining all the received information relating to symbol n –both the directly-received information and the delayed information.

When the relative delay is more than one symbol period, the signal received via the second path acts purely as interference, since it only carries information belonging to a previous symbol or symbols. Such inter-symbol interference (ISI) implies that only very small levels of the delayed signal can be.

When the relative delay is less than one symbol period, part of the signal received via the second path acts purely as interference, since it only carries information belonging to the previous symbol. The rest of it carries the information from the wanted symbol –but may add constructively or destructively to the main-path information.

This tells us that, if we are to cope with any appreciable level of delayed signals, the symbol rate must be reduced sufficiently so that the total delay spread is only a modest fraction of the symbol period. The information that can be carried by a single carrier is thus limited in the presence of multipath. If one carrier cannot then carry the information rate we require, this leads naturally to the idea of dividing the high-rate data into many low-rate parallel streams, each conveyed by its own carrier, of which there are a large number.

Even when the delay spread is less than one symbol period, a degree of ISI from the previous symbol remains. This could be eliminated if the period for which each symbol is transmitted were made longer than the period over which the receiver integrates the signal.

C.2 Orthogonality

The use of a very large number of carriers is a prospect which is practically daunting. We would need many modulators/demodulators and filters to accompany them. It would also appear that an increase of bandwidth would be required to accommodate them. Both these worries can fortunately be dispelled if we do one simple thing: we specify that the carriers are evenly spaced by precisely, where is the period over which the receiver integrates the demodulated signal. When we do this, the carriers form what mathematicians call an orthogonal set. The kth carrier (at baseband) can be written as:

$$\psi_k(t) = e^{jk\omega_u t} \qquad \textbf{(C.1)}$$

where $\omega_u = 2\pi / T_u$, and the orthogonality condition that the carriers satisfy is:

$$\int_{\tau}^{\tau + T_u} \psi_k(t)\psi_l^*(t)dt = 0, \quad k \neq 1$$
$$= T_u, \quad k = 1 \qquad \textbf{(C.2)}$$

More intuitively, what this represents is the common procedure of demodulating a carrier by means of multiplying it by a carrier of the same frequency (“beating it down to zero frequency”) and then integrating the result. Any other carriers will give rise to “beat tones” which are at integer multiples of ω_u. All of these unwanted “beat tones” therefore have an integer number of cycles during the integration period T_u, and thus integrate to zero.

Hence, without any “explicit” filter, we can separately demodulate all the carriers without any mutual cross-talk, just by our particular choice for the carrier spacing. Furthermore, we have not wasted any spectrum either. The carriers are closely packed so that they occupy the same spectrum in total as would a single carrier – if modulated with all the data and subject to ideal sharp-cut filtering.

C.3 Preserving the orthogonality

In practice, our carriers are modulated by complex numbers which change from symbol to symbol. If the integration period spans two symbols, not only will there be same-carrier ISI, but in addition there will be inter-carrier interference (ICI) as well. This happens because the beat tones from other carriers may no longer integrate to zero if they change in phase and/or amplitude during the period. We avoid this by adding a guard interval, which ensures that all the information integrated comes from the same symbol and appears constant during it.

The symbol period is extended so it exceeds the receiver integration period T_u. Since all the carriers are cyclic within T_u, so too is the whole modulated signal. Thus the segment added at the beginning of the symbol to form the guard interval is identical to the segment of the same length at the end of the symbol. As long as the delay of any path with respect to the main (shortest) path is less than the guard interval, all the signal components within the integration period come from the same symbol and the orthogonality criterion is satisfied. ICI and ISI will only occur when the relative delay exceeds the guard interval.

The guard interval length is chosen to match the level of multipath expected. It should not form too large a fraction of T_u, otherwise too much data capacity (and spectral efficiency) will be sacrificed. To tolerate very long delays, must therefore be made large, implying a large number of carriers – from hundreds to thousands.

In fact it is possible to show that the signal demodulated from a particular carrier is very similar to the transmitted signal, but is simply multiplied by the effective frequency response of the multipath channel at that same carrier frequency.

Many other things can cause a loss of orthogonality and hence also cause ICI. They include errors in the local-oscillator or sampling frequencies of the receiver, and phase-noise in the local oscillator. However, in practice, the effects of these can, with care, be held within acceptable limits.

C.4 Use of FFT

We need thousand of filters for implementing all the demodulating carriers, multipliers and integrators, but this is not possible.

In practice, we work with the received signal in sampled form (sampled above the Nyquist limit, of course). The process of integration then becomes one of summation, and the whole demodulation process takes on a form which is identical to the Discrete Fourier Transform (DFT). Fortunately, efficient Fast Fourier Transform (FFT) implementations of this already exist, so that we are able to build laboratory OFDM equipment reasonably easily. Common versions of the FFT operate on a group of time samples and deliver the same number of frequency coefficients. These correspond to the data demodulated from the many carriers. In practice, because we sample above the Nyquist limit, not all of the coefficients obtained correspond to active carriers that we have used.

The inverse FFT is similarly used in the transmitter to generate the OFDM signal from the input data.

C.5 Conclusions

In summary, parallel transmission provides a method of transmitting digital data at speeds very close to the Nyquist rate of band limited channels without using sharp cutoff filters. In addition, the use of a large number of narrow channels is effective in combating delay and amplitude distortion of the transmission medium.

The two end channels will have considerably less distortion since crosstalk from only one other channel can exist. This effect is helpful in using a larger part of the frequency range of transmission media such as voice telephone facilities, in which the degradations increase gradually at the band edges.

In the presence of delay distortion, it is essential that the receiver separately adjusts the demodulating carrier phase and the sampling time for each channel. In addition, these adjustments must be fairly precise and free of jitter due to the high sensitivity of the system to these errors. An efficient parallel data system will therefore inevitably be a costly one to implement.

The strategy of designing an efficient parallel system should concentrate more on reducing crosstalk between adjacent channels than on perfecting the individual channels themselves, since the distortions due to crosstalk tend to dominate.