IJCSI International Journal of Computer Science Issues, Vol. 11, Issue 3, No 2, May 2014
ISSN (Print): 1694-0814 | ISSN (Online): 1694-0784
www.IJCSI.org

Underwater Acoustic Communications Using Direct-Sequence Spread Spectrum

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Abstract

In this paper, it has considered Orthogonal Frequency Division Multiplexing (OFDM) as one of the most used technique in communications applied in a wide-band radio channel, it has been chosen as an air interface for the downlink in the framework of Long Term Evolution (LTE) standardization due to its flexibility and adaptively in the technical system design, also considered the use of undersea acoustic communications, to achieve more robust communications as well as high quality of service, it is necessary to develop a type of adaptive receiver for the acoustic link. This process involves estimating the channel scattering function by processing the received Direct Sequence Spread Spectrum (DSSS) acoustic signal. This information is used by an innovative receiver introduced for usage in the underwater environment.

Keywords: OFDM, LTE, DSSS, UWSN.

1. INTRODUCTION

The Orthogonal Frequency Division Multiplexing (OFDM) modulation technique and associated Orthogonal Frequency Division Multiple Access (OFDMA) channel access mechanism have become the primary technologies used by the latest cellular technologies being standardized, developed, and deployed. Wireless LAN technologies based on the IEEE 802.11a, 802.11g, and 802.11n standards all use OFDM. Wireless MAN technologies based on the IEEE 802.16d, 802.16e, and 802.16m standards all use OFDM. Cellular broadband technologies based on the 3GPP Long Term Evolution (LTE) all use OFDM. While OFDM is often selected because of its robust performance in multipath environments, and ability to cope with interference and noise, it is not ideal for environments where adversarial elements intentionally try to jam communications, such as tactical scenarios.

In the evolution of mobile communication systems, approximately a 10 years periodicity can be observed between consecutive system generations. Research work for the current 2nd generation of mobile communication systems (GSM) started in Europe in the early 1980s, and the complete system was ready for market in 1990. At that time, the first research activities had already been started for the 3rd generation (3G) of mobile communication systems (UMTS, IMT-2000) and the transition from second generation (GSM) to the new 3G systems was observed around 2002. Compared to today's GSM networks, these UMTS systems provide much higher data rates, typically in the range of 64 to 384 k bit/s, while the peak data rate for low mobility or indoor applications is 2 M bit/s. With the extension of High Speed Packet Access (HSPA), data rates of up to 7.2 M bit/s is available in the downlink. The current pace, which can be observed in the mobile communications market, already shows that the 3G systems will not be the ultimate system solution. Consequently, general requirements for a fourth generation (4G) system have been considered in the process of the "Long Term Evolution" (LTE) standardization. These requirements have mainly been derived from the types of service a user will require in future applications. Generally, it is expected that data services instead of pure voice services will play a predominant role, in particular due to a demand for mobile IP applications. Variable and especially high data rates (100 Mbps and more) will be requested, which should also be available at high mobility in general or high vehicle speeds in particular (see Fig. 1). Moreover, asymmetrical data services between up- and downlink are assumed and should be supported by LTE systems in such a scenario where the downlink carries most of the traffic and needs the higher data rate compared to the uplink.

To fulfill all these detailed system requirements, the OFDM transmission technique applied in a wide-band radio channel has been chosen as an air interface for the downlink in the framework of LTE standardization due to its flexibility and adaptively in the technical system design. From the above considerations, it already becomes apparent that a radio transmission system for LTE must provide a great flexibility and adaptively at different levels, ranging from the highest layer (requirements of the application) to the lowest layer (the transmission medium, the physical layer ,i.e., the radio channel) in the ISO-OSI stack. Today, the OFDM transmission technique is in a completely matured stage to be applied for wide-band communication systems integrated into a cellular mobile communications environment [1].

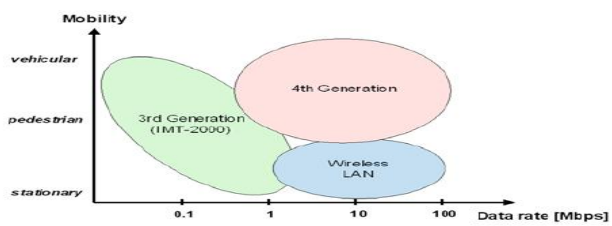

Fig. 1 General requirements for 4G Mobile Communication Systems

1.1 Radio Channel Behavior

Generally the mobile communication system design is in general always dominated by the radio channel behavior [2] [3]. In a typical radio channel situation, multi-path propagation occur Fig. (2), due to the reflections of the transmitted signal at several objects and obstacles inside the local environment and inside the observation area. The radio channel is analytically described unambiguously by a linear (quasi) time-invariant (LTI) system model and by the related channel impulse response $h(\tau)$ or, alternatively, by the channel transfer function $H(f)$. An example for these channel characteristics is shown in Fig. (3), where $h(\tau)$ and $H(f)$ of a so-called Wide-Sense Stationary, Uncorrelated Scattering-channel (*WSSUS*) are given.

Due to the mobility of the mobile terminals, the multipath propagation situation will be continuously but slowly changed over time which is described analytically by a time-variant channel impulse response $h(\tau, t)$ or alternatively by a frequency-selective and time-dependent radio channel transfer function $H(f, t)$ as it is shown in Fig. (4). All signals on the various propagation paths will be received in a superimposed form and are technically characterized by different delays and individual Doppler frequencies which finally lead to a frequency-selective behavior of the radio channel (see Fig. (5)).

The other two system functions, the Delay Doppler function, $\nu(\tau, f_D)$ and the Frequency Doppler function, $U(f, f_D)$ can be used as an alternative description of the radio channel behavior. The Delay Doppler function $\nu(\tau, f_D)$ describes the variation of the channel impulse response related to certain values of the Doppler frequency f_D. This means, the channel delays change due to alteration of the relative speed between a mobile terminal and the base station.

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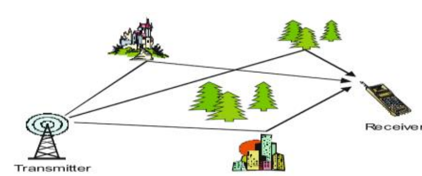

Fig. 2 Multipath Propagation Scenario

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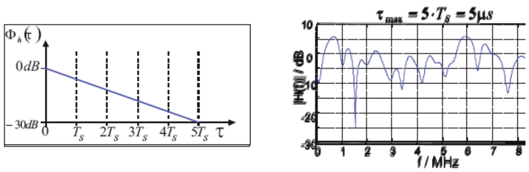

Fig. 3 Power delay (left) and channel transfer function of WSSUS channel (right)

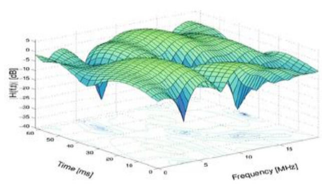

Fig. 4 Frequency-selective and time-variant radio channel transfer function

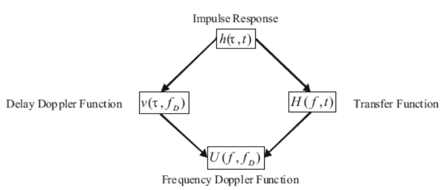

Fig. 5 Relationships between different system functions

The Frequency Doppler function $U(f, f_D)$ models the same effects for the channel behavior in the requency domain. The radio channel can roughly and briefly be characterized by two important system parameters: the maximum multipath delay τ max and the maximum Doppler frequency f_{Dmax} which are transferred into the coherence time T_C and the coherence bandwidth B_C of the radio channel:

$$T_c = \frac{1}{f_{Dmax}}, \; B_C = \frac{1}{\tau_{max}} \quad (1)$$

Over time intervals significantly shorter than T_C, the channel transfer function can be assumed to be almost stationary. Similarly, for frequency intervals significantly smaller than B_C, the channel transfer function can be considered as nearly constant.

Therefore, it is assumed in this paper that the coherence time T_C is much larger compared to a single OFDM symbol duration T_S and the coherence bandwidth B_C is much larger than the distance Δf between two adjacent subcarriers:

$$B_C \gg \Delta f, \Delta f = \frac{1}{T_C}, \;\; T_S \ll T_C \quad (2)$$

This condition should always be fulfilled in well-dimensioned OFDM systems and in realistic time-variant and frequency-selective radio channels. There are always technical alternatives possible in new system design phases. However, future mobile communication systems will in any case require extremely large data rates and therefore large system bandwidth. If conventional single carrier (SC) modulation schemes with the resulting very low symbol durations are applied in this system design, very strong intersymbol interference (ISI) is caused in wideband applications due to multi-path propagation situations. This means, for high data rate applications, the symbol duration in a classical SC transmission system is extremely small compared to the typical values of maximum multi-path delay τ max in the considered radio channel. In these strong ISI situations, a very powerful equalizer is necessary in each receiver, which needs high computation complexity in a wide-band system. These constraints should be taken into consideration in the system development phase for a new radio transmission scheme. The computational complexity for the necessary equalizer techniques to overcome all these strong ISI in a SC modulation scheme increases quadratically for a given radio channel with increasing system bandwidth and can be extremely large in wide-band applications. According to this alternative transmission techniques for broadband applications are of high interest.

Alternatively, OFDM can efficiently deal with all these ISI effects, which occur in multi-path propagation situations and in broadband radio channels. Simultaneously, the OFDM transmission technique needs much less computational complexity in the equalization process inside each receiver. The performance figures for an OFDM based new air interface and for next generation of mobile communications are very promising even in frequency-selective and time-variant radio channels.

1.2 Basics of the OFDM Transmission Technique

At the transmitter side, each subcarrier signal is modulated independently and individually by the complex valued modulation symbol $S_{n,k}$, where the subscript n refers to the time interval and k to the subcarrier signal number in the considered OFDM symbol. Thus, within the symbol duration time interval T the time continuous signal of the *n-th* OFDM symbol is formed by a superposition of all N simultaneously modulated subcarrier signals.

$$s_n = \sum_{k=0}^{N-1} S_{n,k}\, g_k(t-nT) \quad (3)$$

The total time-continuous transmit signal consisting of all OFDM symbols sequentially transmitted on the time axis is described by Eq. (4):

$$s(t)= \sum_{n=0}^{\infty} \sum_{k=0}^{N-1} S_{n,k} e^{j2\pi k\Delta f(t-nT)} rec\left(\frac{2(t-nT)+T_G-T_S}{2T}\right) \quad (4)$$

The analytical signal description shows that a rectangular pulse shaping is applied for each subcarrier signal and each OFDM symbol. Due to the rectangular pulse shaping, the spectra of all the considered subcarrier signals are sinc-functions which are equidistantly located on the frequency axis, e.g., for the k-th subcarrier signal, the spectrum is described in Eq. (5):

$$G_k(f) = T.sinc[\pi T(f-k\Delta f)] \quad (5)$$

The typical OFDM spectrum shown in Fig. (6), consists of N adjacent sincfunctions, which are shifted by Δf in the frequency direction [1].

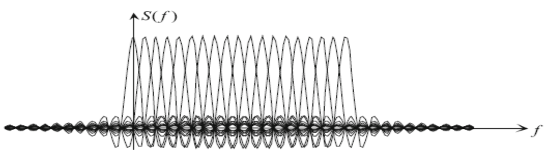

Fig. 6 OFDM spectrum which consists of *N* equidistant *sinc* functions

1.3 Under Sea Communications

Underwater acoustic communications have a wide variety of applications in undersea warfare. These applications include wireless networked sensor telemetry in littoral areas, controlling of minefields and networking of surface vessels, submarines, Unmanned Underwater Vehicles (UUVs) and divers. The underwater communications channel is impaired by its band-limited nature and multipath propagation. Direct-Sequence Spread-Spectrum (DSSS) has proven to be very effective in radio frequency (RF) wireless networks, which experience similar difficulties. If the same robust performance can be achieved in the underwater acoustic channel, then the future design and development of underwater networks can incorporate many of the techniques that have proven so successful in RF wireless networks [6].In last several years, underwater sensor network (UWSN) has found an increasing use in a wide range of applications, such as coastal surveillance systems, environmental research, autonomous underwater vehicle (AUV) operation, by deploying a distributed and scalable sensor network in a 3-dimensional underwater space, each underwater sensor can monitor and detect environmental parameters and events locally. Hence, compared with remote sensing, UWSNs provide a better sensing and surveillance technology to acquire better data to understand the spatial and temporal complexities of underwater environments. Clearly, *efficient underwater communication* among units or nodes in a UWSN is one of the most fundamental and critical issues in the whole network system design [5].

Present underwater communication systems involve the transmission of information in the form of sound, electromagnetic (EM), or optical waves. Each of these techniques has advantages and limitations. *Acoustic communication* is the most versatile and widely used technique in underwater environments due to the low attenuation (signal reduction) of sound in water. This is especially true in thermally stable, deep water settings. On the other hand, the use of acoustic waves in shallow water can be adversely affected by temperature gradients, surface ambient noise, and multipath propagation due to reflection and refraction. The much slower speed of acoustic propagation in water, about 1500 m/s (meters per second), compared with that of electromagnetic and optical waves, and is another limiting factor for efficient communication and networking. Nevertheless, the currently favorable technology for underwater communication is upon acoustics. Present underwater communication systems involve the transmission of information in the form of sound, electromagnetic (EM), or optical waves. Each of these techniques has advantages and limitations. *Acoustic communication* is the most versatile and widely used technique in underwater environments due to the low attenuation (signal reduction) of sound in water. This is especially true in thermally stable, deep water settings. On the other hand, the use of acoustic waves in shallow water can be adversely affected by temperature gradients, surface ambient noise, and multipath propagation due to reflection and refraction. The much slower speed of acoustic propagation in water, about 1500 m/s (meters per second), compared with that of electromagnetic and optical waves, and is another limiting factor for efficient communication and networking. Nevertheless, the currently favorable technology for underwater communication is upon acoustics.

On the front of using *electromagnetic (EM) waves* in radio frequencies, conventional radio does not work well in an underwater environment due to the conducting nature of the medium, especially in the case of seawater. However, if EM could be working underwater, even in a short distance, its much faster propagating speed is definitely a great advantage for faster and efficient communication among nodes.
Free-space optical (FSO) waves used as wireless communication carriers are generally limited to very short distances because the severe water absorption at the optical frequency band and strong backscatter from suspending particles. Even the clearest water has 1000 times the attenuation of clear air, and turbid water has more than 100 times the attenuation of the densest fog. Nevertheless, underwater FSO, especially in the blue-green wavelengths, offers a practical choice for high-bandwidth communication (10-150 Mbps, bits per second) over moderate ranges (10-100 meters). This communication range is much needed in harbor inspection, oil-rig maintenance, and linking submarines to land, just name a few of the demands on this front [4].

The research of Underwater Acoustic Networks (UANs) is attracting attention due to their important underwater applications for military and commercial purposes. More and more research interest and efforts are shifting to this area in recent years. The broad applications of UANs include but not limited to [5]:

1) Information exchange among nodes that are within the range of the network, or outside the network with the help of, e.g., a gateway, or a switch center. The primary design goal of communication networks is for exchanging information. In an UAN, exchanging information among nodes is one of its essential applications. An example is that underwater internet, in which users can share information without tether, will become realistic instead of just a dream, if UANs are deployed. Another important application is realtime communication with submarines and autonomous underwater vehicles in network configurations.

2) Environmental monitoring. Pollution in near-shore oceans is an urgent issue and needs close watch. UANS can perform different kinds of pollution monitoring and can also be used to monitor ocean currents and temperature change, e.g., the global warming effect to ocean.

3) Underwater explorations. Underwater explorations are difficult for human beings due to the high water pressure, unpredictable underwater activities and vast size of unknown area. UANs can help us explore the underwater world that we are not familiar with. Such kinds of activities include exploring minerals and oilfields, determining routines for laying undersea cables, etc.

4) Disaster prevention, by deploying Acoustic Sensor Networks (ASN) in remote locations to monitor undersea activities, ocean-related disaster like tsunami and seaquake can be warned to coastal areas in real time when it happens (but this is very risky).

Research direction: In the last two decades, only two fundamental changes were made in underwater acoustic physical layers. The first one is the introduction of digital communication techniques in early 1980's frequency-shift keying (FSK) and the second one is the implantation of coherent modulation in early 1990's (PSK and QAM). Equalization technology has also been introduced to compensate ISI; however, different numbers and different placement of the equalizer tap coefficients are needed for different ocean environments. Another way to remove ISI is to use passive-phase conjugation (PPC). A Decision Feedback Equalizer (DFE) using a fixed set of parameters with minimal user supervision has been introduced [5].

Where Peter, considered the feasibility and performance of using DSSS modulation for utility packet transmission in Seaweb, an underwater wireless acoustic communications network. The Seaweb network requires robust channel-tolerant utility packets having a low probability of detection (LPD) and allowing for multi-user access. The transmitter and receiver structures are simulated in MATLAB and the underwater environment are represented by a modeled channel impulse response. The specific modulation scheme implemented is direct-sequence, differentially encoded binary phase-shift keying (DS-DBPSK) with quadrature spreading. The performance of DSSS modulation is examined using Monte Carlo simulation. Bit error rates and packet error rates for various signal-to-noise ratios and channel conditions are presented and the use of a RAKE receiver and forward error-correction coding are examined for improving system performance.
The results show that DSSS modulation is well suited for communications in the underwater environment. Bit error rates below 10^{-5} are achievable using soft decision Viterbi decoding and a RAKE receiver that

adaptively places the RAKE taps to coincide with the multipath arrivals. The limitations in the modem's performance are associated with poor acquisition algorithm performance at low signal-to-noise ratios [6].

The underwater acoustic environment is a particularly challenging communications medium. In order to effectively design underwater communications systems, it is necessary to have a deep understanding of the nature of the physical barriers constraining performance in the underwater channel.

To achieve more robust communications, as well as high quality of service, it is necessary to develop a type of adaptive receiver for the acoustic link. This process involves estimating the channel scattering function by processing the received Direct Sequence Spread Spectrum (DSSS) acoustic signal. This information is used by an innovative receiver introduced for usage in the underwater environment.

This dissertation addresses the problem of designing a robust and reliable coherent receiver which will be able to operate in a multiuser environment. The proposed scheme is based on Direct Sequence Spread Spectrum (DSSS) modulation. The DSSS signal can also be used for channel identification, which is a vital task for estimating the time-varying characteristics of the communication medium and adjusting the receiver parameters for optimum detection. The developed receiver is based on an array of hydrophones, thus exploiting the advantages of spatial diversity. The signals received from the hydrophones are processed in such a way that the principal multipath arrival is amplified, whereas environmental noise and weaker multipath arrivals are suppressed. The spatial combiner is designed based on array processing theory and processes the received signals before equalization.

The receiver includes a dual input equalizer based on a Kalman filter estimator, which has a relatively low computational complexity. With multiple receivers, the transmitted signal can be viewed as the state of a dynamic system with the received signals as inputs and outputs. As a consequence, a Kalman estimator can be used. Furthermore, in order to ensure the reliability of the equalizer, a method was developed for choosing the multipath components to be used as inputs of the equalizer based on the channel consistency of those components. Additionally, an adaptive mechanism is developed on the equalizer in order to track the channel changes and ensure the filter's convergence. Incorporated into the equalizer is also an algorithm for dynamically tracking and correcting the Doppler shift on each of the multipath arrivals. Since the Doppler shift is strongly time-varying, dynamically tracking it and correcting it for best performance of the acoustic receiver is crucial.

Finally, an external interference and noise canceller mechanism was developed, intended for use in the multiuser underwater environment. The outcome of this dissertation is a robust receiver able to operate in a fast time-varying, multipath fading, and multiuser environment. Experimental results prove the effectiveness of the receiver [7].

2. Conclusion

As a summary of what is considered in this work (paper), a general view was taken about Orthogonal Frequency Division Multiplexing (OFDM) its definition, types and practical applications also considered principle of operation with derivation of mathematical parameters of operation and also considered its ability to overcome difficulties in underwater communications through use of DSSS and make use , low probability of intercept (LPI) and to achieve more robust communications, as well as high quality of service, it is necessary to develop a type of adaptive receiver for the acoustic link. This process involves estimating the channel scattering function by processing the received Direct Sequence Spread Spectrum (DSSS) acoustic signal. This information is used by an innovative receiver introduced for usage in the underwater environment. The output of transmitter considered for this system is simulated in simulink of mat-lab and gives output shown in Fig. 7-a, and Fig. 7-b, which show equally spaced orthogonal frequency components where N was considered equal 45.

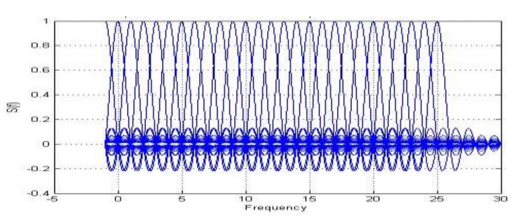

Fig. 7-a OFDM spectrum of N equidistant sinc functions simulated by Simulink

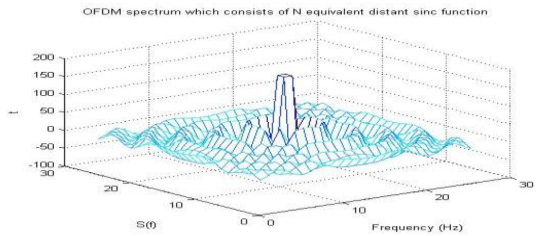

Fig. 7-b OFDM spectrum of N equidistant sinc functions simulated by Simulink

ACKNOWLEDGMENT

The authors would like to thank the University Malaysia Perlis (UniMAP) for their financial support for publishing this paper.

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