A Broadband Underwater Acoustic Modem Implementation Using Coherent OFDM

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***Abstract*— Multicarrier modulation in the form of orthogonal frequency division multiplexing (OFDM) has prevailed in recent broadband wireless systems over radio channels. In this senior design project, we have implemented an acoustic OFDM modem that transmits digital data through sound propagation. We have demonstrated OFDM transmission first in air, and then in water. We find that the underwater channel is much more complex than the air channel, and careful signal designs are needed for underwater transmissions. We have also handled another difficulty incurred by sampling rate mismatches at the transmitter and the receiver due to low-cost sampling devices. With two-way communication capabilities, this project provides a simple online chatting tool between two computers relying on acoustic links.**

***Index Terms*— OFDM, multicarrier transmission, underwater acoustic communication.**

I. Introduction

Recently, there has been a growing interest in monitoring aqueous environments (including oceans, rivers, lakes, ponds, and reservoirs, etc.) for scientific exploration, commercial exploitation, and protection from attacks. The ideal vehicle for this type of extensive monitoring is a networked underwater wireless sensor distributed system, referred to as the Underwater Wireless Sensor Network (UWSN) [1], [2].

Establishing effective communications among a distributed set of both stationary and mobile sensors is one key step toward UWSNs. Since electromagnetic waves do not propagate well in underwater environments, underwater communications have to rely on other physical means, such as sound, to transmit signals [3]. Unlike the rapid growth of wireless networks over radio channels, the development of underwater communication has been at a much slower pace [3]. The last two decades have witnessed only two fundamental advances in underwater acoustic communications. One is the introduction of digital communication techniques, namely, noncoherent frequency shift keying (FSK), in the early 1980s, and the other is the application of coherent modulation, including phase shift keying (PSK) and quadrature amplitude modulation (QAM) in early 1990s [4]. Existing underwater coherent communication has mainly relied on serial single-carrier transmission and equalization techniques over the challenging underwater acoustic media [4]. As the data rates increase, the symbol durations decrease, and thus the same physical underwater channel contains more channel taps in the baseband discrete-time model (easily on the order of several hundreds of taps). This poses great challenges for the channel equalizer. Receiver complexity will prevent any substantial rate improvement with existing approaches.

Due to its low equalization complexity in the presence of highly-dispersive channels, multicarrier modulation in the form of orthogonal frequency division multiplexing (OFDM) has prevailed in recent broadband wireless systems. Motivated by the success of OFDM in radio channels, there is a recent re-emergence of interest in applying OFDM in underwater acoustic channels [5]–[7].

The aim of this project is to demonstrate the OFDM technique for underwater acoustic communications. We have pursued this project in two phases.

- Phase 1: Creation and testing of a physical air-to-air acoustic communication link using OFDM.
- Phase 2: Creation and testing of a physical underwater acoustic communication link using OFDM.

Comparing the air and water experiments helps us to better understand underwater acoustic channel characteristics and the challenges they pose on OFDM transmissions.

II. Project Demonstration

We demonstrated our project at the ECE senior design day, Dec. 8, 2006, at the University of Connecticut. Our demonstration has two settings. In the first setting, we demonstrated multicarrier OFDM transmission and reception in air. The testbed is depicted in Fig. 1 (left), where a speaker together with a laptop serve as the transmitter, and a microphone together with another laptop serve as the receiver. A typical channel impulse response is shown in Fig. 1 (right), where we clearly see the dominant line-of-sight path. In the second setting, we demonstrated OFDM transmission and reception in water. The testbed is depicted in Fig. 2 (left). The underwater speaker and hydrophone are used as transmitting and receiving devices, respectively. A typical channel impulse response is shown in Fig. 2 (right), where we clearly see the reverberation effects. The last path is about 37 meters longer than the first path inside the water tank of 2 meters long, 0.5 meters wide and 0.5 meter deep. Hence, the transmitted signal has bounced back and fourth between the hard surfaces of the water tank a plenty of times. We can also see the amplitude attenuation associated with each bounce.

*More details can be found at the website of our senior design project: http://www.engr.uconn.edu/ece/SeniorDesign/projects/ecesd75/

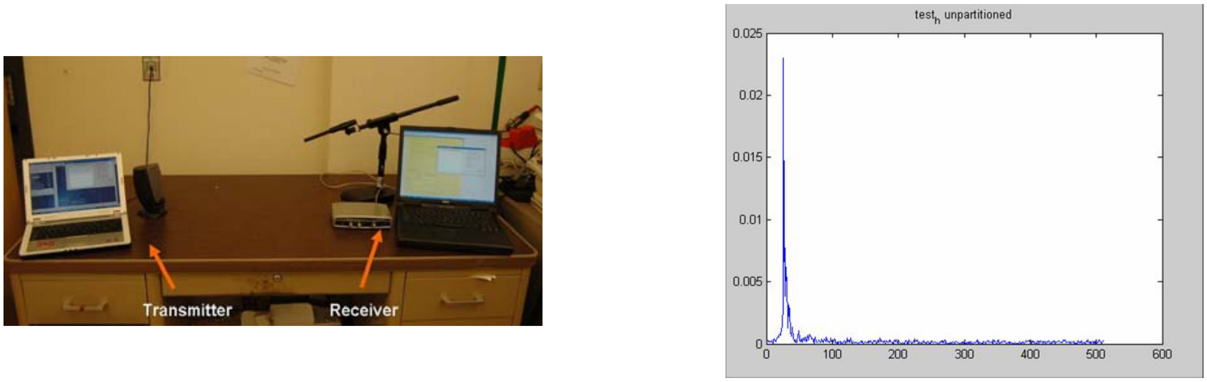

Fig. 1. OFDM testing in air: testbed (left) and typical channel impulse response (right)

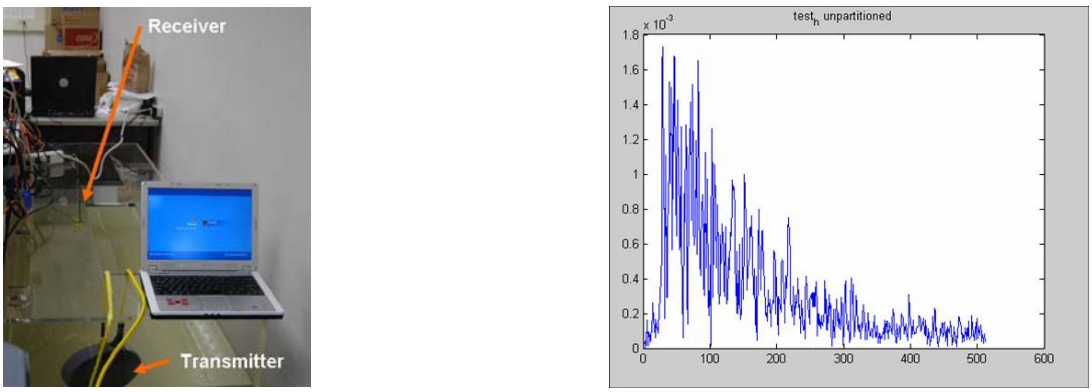

Fig. 2. OFDM testing in water: testbed (left) and typical channel impulse response (right)

III. Project Description

A. OFDM Basics

Broadband applications has imposed great challenges for conventional single carrier transmissions, as the channel exhibits strong frequency selectivity that prevents effective channel equalization to remove inter-symbol-interference (ISI) at affordable complexity. Multi-carrier techniques divide the available bandwidth into a large number of overlapping subbands, so that the symbol duration is long compared to the multipath spread of the channel. Consequently, ISI may be neglected in each subband, greatly simplifying the receiver complexity of channel equalization.

OFDM is an efficient multi-carrier implementation based on fast-Fourier-transform (FFT). Consider an OFDM transmission over a frequency selective channel, that is described by its discrete-time baseband impulse response vector $\mathbf{h} := [h(0), \ldots, h(L)]^T$, with L standing for the channel order. The channel impulse response includes the effects of transmit-receive filters and physical multipath. By implementation an inverse FFT at the transmitter and an FFT at the receiver, OFDM converts an ISI channel into parallel ISI-free subchannels with gains equal to the channel's frequency response values on the FFT grid. Specifically, let K denote the number of subcarriers in the OFDM system, $x(p)$ as the transmitted symbol on the pth subcarrier, $y(p)$ as the received symbol on the pth subcarrier, the equivalent channel input-output relationship can be described by:

$$y(p) = H(p)x(p) + v(p), \quad p = 0, 1, \ldots, K-1, \qquad (1)$$

where $v(p)$ stands for the additive white Gaussian noise, and $H(p)$ is the channel's frequency response on the pth subcarrier:

$$H(p) = \sum_{l=0}^{L} h(l) e^{-j\frac{2\pi}{N_c}pl}, \quad p = 0, \ldots, N_c - 1. \qquad (2)$$

Channel equalization amounts to scalar inversion:

$$\hat{s}(p) = y(p)/H(p) \qquad (3)$$

on each subcarrier. The equalization complexity thus does not depend on the channel length.

Precisely due to its low equalization complexity in the presence of highly-dispersive channels, OFDM has prevailed in recent broadband wireless systems. Those include digital audio/video broadcasting (DAB/DVB) standards in Europe, high-speed digital subscriber line (DSL) modems in the United States, digital cable television systems, and wireless local area networks. See e.g., [9], [10] for more details on OFDM.

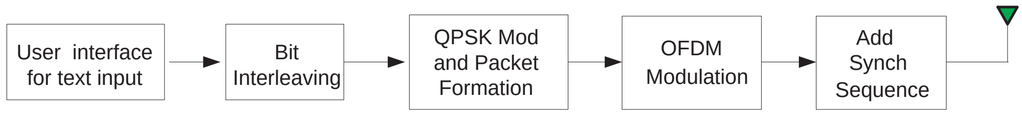

Fig. 3. Block diagram for the transmitter

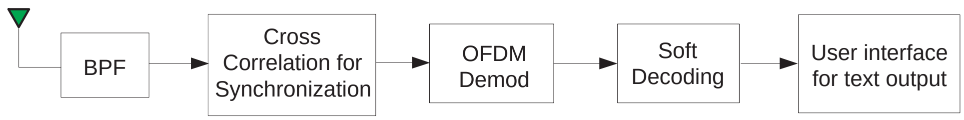

Fig. 4. Block diagram for the receiver

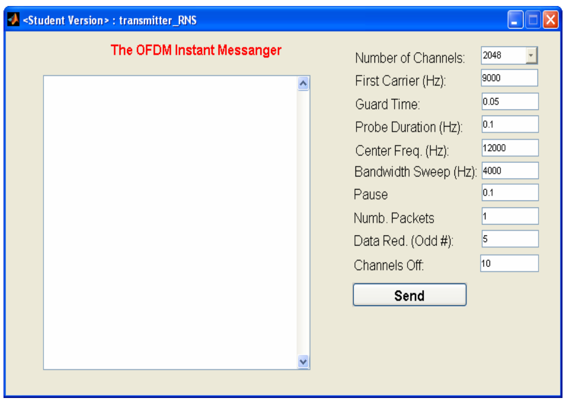

Fig. 5. The transmitter GUI

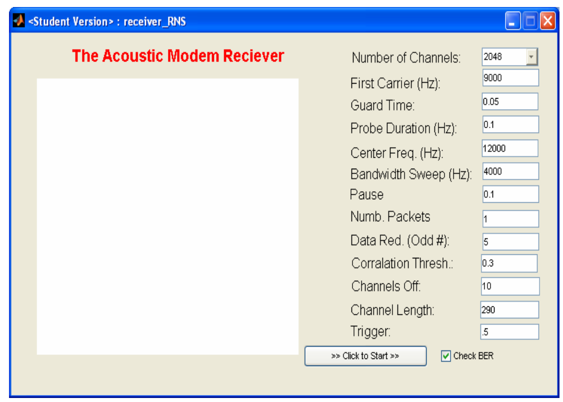

Fig. 6. The receiver GUI

B. Transceiver Design

The transmitter diagram is shown in Fig 3. We use a graphics user interface to input some text messages. The transmitter first converts the text data into a binary format. The binary data is then interleaved and coded by a simple (3,1) repetition code for error correction. The coded data is mapped to QPSK (quadrature phase-shift keying) symbols. The symbol stream is partitioned into blocks, and each block is OFDM modulated. During the OFDM modulation, we insert pilot symbols at every 4th subcarrier, which facilitates channel estimation at the receiver for coherent demodulation; more details can be found in [6]. The modulation is implemented at baseband and then up-shifted to passband for transmission. A synchronization sequence is inserted in front of the data packets during transmission.

The receiver diagram is shown in Fig. 4. The receiver first applies bandpass filtering on the incoming data stream, then it looks for where the useful data begins via correlating the received data with the synchronization sequence template. Once the receiver has found a useful data packet, it down-shifts the passband signal to baseband. We then estimate the carrier frequency offset (CFO) to correct any carrier mismatch between the transmitter and the receiver, as will be elaborated in Section IV-B. After CFO compensation, we rely on pilot tones to estimate the channel frequency response [6]. With coherent demodulation at each OFDM subcarrier, we decode the (3,1) repetition coding using maximum ratio combining. The receiver finally extracts the binary bits from the QPSK symbols and generates the text message.

C. Graphic User Interface

There are two graphical user interfaces (GUIs), one for the transmitter and one for the receiver. These interfaces allow for configuration of basic parameters such as

- Number of OFDM subcarriers
- First carrier frequency
- Guard time between data packets
- Synchronization sequence duration
- Center frequency for synchronization sequence
- Bandwidth of synchronization sequence
- Pause time between synchronization sequence and first data packet
- Number of packets per transmission
- Repetition coding rate
- Number of edge channels deactivated

The receiver has a few extra parameters including the correlation threshold, the amplitude trigger and the channel length. The receiver also gives the user the option of checking the bit error rate and plotting each channel estimation graph during the CFO estimation process.

D. Packet Formation

The typed text messages have different lengths, while the packet has a fixed length. Hence, different number of packets

Fig. 7. Full packet binary sequence

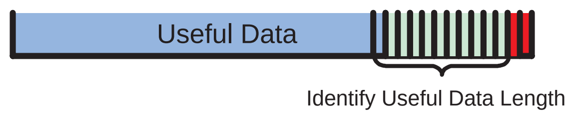

Fig. 8. Partial packet binary sequence

may be required for different messages. To create an automatic message transmission, we design the packet format as follows. The binary sequence created by the transmitter consists of the text data along with 2 metadata bits (administrative bits) at the end. These metadata bits are known as the *partial packet bit* and the *continuation bit*, respectively. The continuation bit will tell the receiver whether the next transmission contains a continuation of the current transmission. The partial packet bit alerts the receiver that the text data was not able to fill the packet and extra meaningless data have been inserted, which must be removed at the receiver end. In this case, eleven extra bits must be added before the partial packet bit to identify the length of the useful data (and where the meaningless data begins). The format of the binary data in a packet can be seen in Figs. 7 and 8.

On the transmitter end, the maximum allowable data length determines what the metadata bits are set to. If the length of the binary data is equal to the maximum allowable data length minus two, the transmitter will set both of the metadata bits to zero. If the binary data length is greater than the maximum allowable data length minus 2, the transmitter will break the data up into multiple transmissions, set the partial packet bit to zero, and the continuation bit to one. If the data length is less than the maximum data length minus thirteen, meaningless data is added to the end of the sequence. The eleven identifying bits are added to the end of the meaningless data and the partial packet bit is set to one while the continuation bit is set to zero. If the data length is between the maximum allowable data length minus two and the maximum allowable data length minus thirteen, the binary sequence is zero padded (to make the data length equal to the maximum allowable data length, zeros are added between the end of the useful data and the meta data) and both the partial packet and continuation bits are set to zero. These zeros will not be removed by the receiver but will not affect the received data. The point to placing meaningless data in the partial packets (as opposed to just zeros) is to ensure that the peak-to-average ratio of the OFDM symbol is randomized.

IV. Lessons Learned

We next present some useful experience accumulated during the project development.

A. Air vs. Water

Figs. 1 and 2 show that the underwater channel in the water tank has significant longer delay spread than the air channel. We truncate the air channel in Fig. 1 to have $L_1 = 60$ channel taps in the discrete-time baseband model, while we use $L_2 = 350$ channel taps for the underwater channel in Fig. 2. In this project, we set the signal bandwidth to be $B = 11.25$ kHz, occupying the band of $10 \sim 21.25$ kHz. The delay spread is thus $L_1/B = 5.3$ ms for the air channel, and $L_2/B = 31.1$ ms for the water channel.

The channel length has a significant impact on the signal design. For the OFDM signal with K subcarriers, we have used $K/4$ pilot tones for channel estimation. To ensure successful channel estimation, we have to let $K/4 > L$. In the underwater case, we have set $K = 2048$ so that the transmission can get through, while in air we have tried various configurations of $K = 256, 512, 1024, 2048$ and each case works well.

Therefore, when moving an existing OFDM design from air to water, one has to make sure that the signal design can accommodate a significant increase in channel delay spread.

B. Sampling Rate Mismatch

The transmitter and the receiver use low-cost external sound cards for digital-to-analog and analog-to-digital conversions. The mismatch between the sampling rates of the transmitter and the receiver might cause slight waveform compression or dilations. Waveform scaling from duration T to $(1+a)T$ cause a frequency shift af_0 at the frequency f_0. This is similar to the Doppler effect when there is a relative motion between the transmitter and the receiver.

We observe Doppler shifts within a range of one to five Hz in the signal band in our experiments. Suppose that the Doppler shift is 3 Hz at the frequency 15 kHz. This means $a = 3/15000 = 2 \cdot 10^{-4}$. The sound card is operating at a nominal sampling frequency of 44.1 kHz. The value of $a = 2 \cdot 10^{-4}$ could come from a rate mismatch of 8.82 Hz between the transmitter and the receiver. This looks very reasonable for the low-cost devices that we used.

With a bandwidth of 11.25 kHz, the subcarrier spacing is

$$\Delta f = B/K = \begin{cases} 43.1\text{Hz}, & K = 256 \\ 21.5\text{Hz}, & K = 512 \\ 10.8\text{Hz}, & K = 1024 \\ 5.4\text{Hz}, & K = 2048 \end{cases} \quad (4)$$

With the Doppler frequency in the range of 1 to 5 Hz, our air-to-air transmission is still successful when $K = 256$ and $K = 512$. But when $K = 1024$ and $K = 2048$, the Doppler frequency cannot be neglected compared to subcarrier spacing, and considerable intercarrier interference (ICI) leads to very poor reception performance.

When K is large, the Doppler shifts need to be explicitly compensated. To this end, we treat the Doppler shift as if it is due to a carrier frequency offset (CFO) between the transmitter and the receiver. The process of removing the CFO is an iterative one, using a guess-and-check method [6]. The receiver walks through a range of possible CFO values at a step size of 0.1 Hz. At each step, it pre-compensates the CFO before FFT processing, and then performs pilot-based channel estimation in the frequency domain. Since channel estimation is based on the Least-Square (LS) principle, we decide that the best CFO compensation is the one that leads to the smallest LS fitting error in the channel estimation step.

Due to the guess-and-check (one-dimensional search) procedure, the CFO estimation at the receiver might take a considerable amount of processing time. If speed is an issue, the CFO can be determined once for one set of machines and then disabled. The CFO will remain fairly constant between two machines and can be manually entered into the main receiver file.

Summarizing the discussions in Sections IV-A and IV-B, we have the practical guidelines as:

- For air transmission, sampling rate mismatch can be ignored if the subcarrier spacing is large (a small number of subcarriers), and needs to be considered if the subcarrier spacing is small (a large number of subcarriers).
- For underwater transmission, sampling rate mismatch shall be considered since a large number of subcarriers is needed due to the large delay spread of the underwater channels.

C. Continuous Receiver

It is desirable for the receiver to continuously monitor the incoming signals, so that the transmitter can transmit at any time. However, this is difficult with the `waverec` function in Matlab: when Matlab is recording, the recording buffer is not readable by the user. The receiver has to stop recording to analyze what it has picked up, hence missing the incoming signals during the processing time.

We solve this problem via using the Data Acquisition Toolbox (DAQ) for Matlab. The DAQ can continuously monitor the data. We set an amplitude trigger for the DAQ to input a portion of data in its buffer to Matlab. Once a trigger is activated, the receiver runs the synchronization algorithm to determine whether the incoming data contains some useful data. If yes, it activates the data demodulation modules. If not, it discards the incoming data and waits for the next trigger.

In short, with the data acquisition toolbox, our receiver runs in a continuous mode for data reception.

D. Two-Way Communication

The testbeds in Fig. 1 and 2 have only implemented one-way communication: there is only one pair of speaker and microphone. We have recently upgraded the testbed in Fig. 1 to include a two-way communication. As shown in Fig. 9, there are two pairs of speaker and microphone, one pair implementing the forward link and the other for the backward link. Both receivers run in the continuous mode, and can transmit at any time when a message is typed. Essentially, this testbed allows online chatting between two computers relying on the acoustic links.

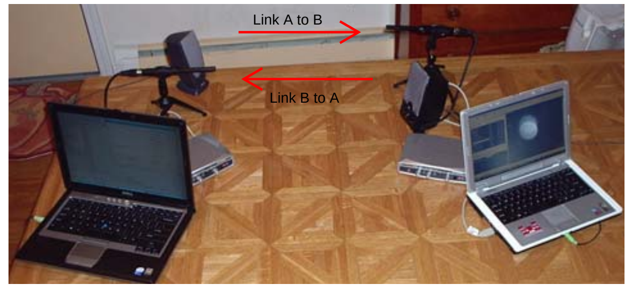

Fig. 9. The testbed with two-way communication

V. CONCLUSIONS

Our senior design project has implemented a coherent OFDM modem that can let two computers communicate via acoustic transmissions in air and in water. This paper has summarized our experience gained in the project development.

Our OFDM modem will be used by another senior design team to test networking protocols such as routing and medium access control based on acoustic links.

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