## SCALABLE OFDM DESIGN FOR UNDERWATER ACOUSTIC COMMUNICATIONS

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## ABSTRACT

Multicarrier modulation in the form of OFDM has been actively pursued for underwater acoustic communication recently. In this paper, we present a desirable property of OFDM that one signal design can be easily scaled to fit into different transmission bandwidths with negligible changes on the receiver. We have tested the proposed design with data collected from experiments at AUV Fest, Panama City, FL, June 2007, and at the Buzzards Bay, MA, Aug. 2007. With QPSK modulation, we have used different bandwidths from 3 kHz to 50 kHz, leading to data rates from 1.5 kbps to 25 kbps after rate 1/2 coding. With 16-QAM modulation, we have used different bandwidths from 12 kHz to 50 kHz, leading to data rates from 12 kbps to 50 kbps. Excellent BER performance has been achieved, which confirms the flexibility of OFDM under different system setups.

*Index Terms*— Underwater acoustic communication, multicarrier, OFDM, scalability

## 1. INTRODUCTION

Recently, multicarrier modulation in the form of orthogonal frequency division multiplexing (OFDM) has been actively pursued for underwater acoustic (UWA) communications, due to its unique ability to deal with high rate transmission over long dispersive channels [1–7].

Due to the frequency-dependent high attenuation, the available bandwidth of the UWA channel depends on both the distance and the frequency range [8]. When the system bandwidth and thus the symbol rate change, the same physical channel of a certain delay spread leads to discrete-time channels with different number of taps. For single carrier transmission and adaptive equalization approaches [9], the equalizer length and adaptation constants need to be carefully tuned when the channel length changes. On the contrary, channel equalization complexity in OFDM does not depend on the channel length in the time domain, and hence intuitively OFDM can effectively handle symbol rate changes.

This paper aims to illustrate how easily OFDM can be scaled to adapt to drastic changes on the system bandwidth. We first outline how to design a base OFDM system. Then simply changing the number of subcarriers, the OFDM design can fit into various bandwidths, while the changes on the receiver algorithms are minimal.

We tested the scalable OFDM design in two different geographic locations, one at Panama City, FL and the other at the Buzzards Bay, MA. The tested bandwidths are 3 kHz, 6 kHz, and 12 kHz in the first experiment, and are 25 kHz and 50 kHz in the second experiment. The achieved data rate varies from 1.5 kbps to 25 kbps after rate 1/2 coding and QPSK modulation. The 16-QAM modulation was also tested in the settings with bandwidths 12 kHz, 25 kHz, and 50 kHz, leading to data rates about 12 kbps, 25 kbps, and 50 kbps, respectively. All the cases use the same block-by-block receiver that we have developed in [5,6]. In each case, excellent bit-error-rate (BER) performance is achieved after channel decoding over one or multiple receivers. These results confirm the flexibility of OFDM under different system setups.

The rest of the paper is organized as follows. We present the signal design in Section 2, and discuss the receiver algorithm in Section 3. Performance results are reported in Section 4, while the conclusions are drawn in Section 5.

### 2. SIGNAL DESIGN

As in [4–7], we consider a zero-padded (ZP) OFDM transmission. The key steps at the transmitter are the following.

- 1. *Channel coding.* Let  $r_c$  denote the code rate. In this paper, we will use two different codes: (i) a 16-state rate 1/2 convolutional code with generator polynomial (23,35), and (ii) a rate 1/2 nonbinary regular low-density-parity-check (LDPC) cycle code [10].
- 2. *Bit-to-symbol mapping*. Using a constellation of size M, one information symbol can carry  $\log_2 M$  bits. In this paper, we will use both QPSK and 16-QAM.
- 3. *OFDM modulation*. An inverse FFT of size K is performed on a block of symbols to generate time-domain samples, where K is the number of subcarriers [5, 6]. However, not all K subcarriers carry information symbols. Specifically, there are  $K_n$  null subcarriers and  $K_p$  pilot subcarriers inserted to aid the block-by-block receiver processing, as detailed in [6]. The number of carriers that carry useful data is  $K_d = K K_p K_n$ .

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Table 1. One choice of subcarrier allocation

Number of subcarriers	$K_0 = 1024$
Number of data subcarriers	$K_{d0} = 672$
Number of pilot subcarriers	$K_{p0} = K_0/4 = 256$
Number of null subcarriers	$K_{n0} = 96$

- 4. *Carrier modulation*. The baseband signal is frequencytranslated to passband. Note that passband samples are usually generated at the sampling rate of the system.
- 5. Zero padding. For each OFDM block of duration T, a guard interval of length  $T_g$  is inserted to prevent the inter block interference.

Accounting for the various transmission overhead due to coding, null and pilot subcarriers, and guard time, the achieved data rate is

$$R = \frac{r_c K_d \log_2 M}{T + T_g}.$$
 (1)

The OFDM bandwidth is B = K/T, and the bandwidth utilization factor of the system is

$$\alpha = \frac{R}{B} = \frac{T}{T + T_g} \cdot \frac{K_d}{K} \cdot r_c \cdot \log_2 M \quad \text{bits/sec/Hz.} \quad (2)$$

#### 2.1. Design a base system

We first design a base system with bandwidth  $B_0$  and  $K_0$  subcarriers, based on which one can scale the bandwidth while keeping the bandwidth utilization factor unchanged.

We adhere to the following rules.

• *Rule 1:* The sampling rate  $f_s$  is an integer multiple of the bandwidth  $B_0^{-1}$ .

The baseband signal is generated at either the symbol rate  $B_0$  or the sampling rate  $f_s$ , while the passband signal is generated at the sampling rate. Keeping  $f_s/B_0$  an integer facilitates the upsampling operation at the transmitter, and the downsampling operation at the receiver for baseband signal processing.

• *Rule 2:* The FFT size  $K_0$  is a power of 2 for easy implementation.

After  $K_0$  is chosen, we divide  $K_0$  subcarriers into  $K_{d0}$  data subcarriers,  $K_{p0}$  pilot subcarriers, and  $K_{n0}$  null subcarriers. One example with  $K_0 = 1024$  is shown in Table 1.

• *Rule 3:* The OFDM symbol duration T is much larger than the guard time  $T_g$  to avoid significant bandwidth reduction, while is small enough so that channel variation within T seconds can be considered slow.

Table 2. Parameters for the test at AUV Fest, June 2007

Symbol duration	T = 85.33  ms		
Guard interval	$T_g = 25 \text{ ms}$		
Subcarrier spacing	$\Delta f = 11.72 \text{ Hz}$		
Bandwidth B	3 kHz	6 kHz	12 kHz
Number of subcarriers K	$K_{0}/4$	$K_{0}/2$	$K_0$

In our earlier designs [6, 7], we use  $T_g = 25$  ms, and choose T around 100 ms.

The choice of  $B_0$ ,  $K_0$  and T shall satisfy  $B_0 = K_0/T$ . e.g., if  $B_0 = 12$  kHz and  $K_0 = 1024$ , then T = 85.33 ms.

#### 2.2. Scale the bandwidth of the base system

To design a system with a different B, one could utilize the base system by keeping T and  $T_g$  constant and just changing the number of subcarriers.

Define the scale factor as

$$c = \frac{B}{B_0}.$$
 (3)

We then scale the number of subcarriers as

$$K = cK_0, \ K_d = cK_{d0}, \ K_p = cK_{p0}, \ K_n = cK_{n0}.$$
 (4)

This way, the bandwidth utilization factor is the same as the base system. The achieved data rate scales with the bandwidth as  $R = \alpha B$ . We next show two design examples.

#### 2.3. Signal design for the test at AUV Fest June 2007

Prior to the experiment, we learned that the sampling rate would be  $f_s = 96$  kHz and the usable bandwidth would be no more than 15 kHz. We first choose  $B_0 = 12$  kHz and  $K_0 = 1024$  to design a base system, and then scale the bandwidth *B* to three different values 3 kHz, 6 kHz, and 12 kHz, respectively. The parameters are shown in Table 2.

With rate 1/2 coding, the bandwidth utilization factor is

$$\alpha = \begin{cases} 0.5075 \text{ bits/sec/Hz}, & \text{for QPSK} \\ 1.0151 \text{ bits/sec/Hz}, & \text{for 16-QAM.} \end{cases}$$
(5)

#### 2.4. Signal design for the test at Buzzards Bay, Aug. 2007

Prior to the experiment, we learned that the sampling rate would be  $f_s = 400$  kHz and the usable bandwidth would be no more than 75 kHz. We choose  $B_0 = 12.5$  kHz and  $K_0 = 1024$  to design a base system, and then scale the bandwidth *B* to 25 kHz and 50 kHz, respectively. The signal parameters are listed in Table 3.

With rate 1/2 coding, the bandwidth utilization factor is

$$\alpha = \begin{cases} 0.5028 \text{ bits/sec/Hz}, & \text{for QPSK} \\ 1.0056 \text{ bits/sec/Hz}, & \text{for 16 QAM.} \end{cases}$$
(6)

<sup>&</sup>lt;sup>1</sup>We followed this rule used by Dr. M. Stojanovic in [4].

Symbol duration	T = 81	.92 ms
Guard interval	$T_g = 25 \text{ ms}$	
Subcarrier spacing	$\Delta f = 12.21  \mathrm{Hz}$	
Bandwidth B	25  kHz	50  kHz
Number of subcarriers K	$2K_0$	$4K_0$

Table 3. Parameters for the test at Buzzards Bay, August 2007

## **3. RECEIVER ALGORITHM**

Assume that the physical channel has a delay spread of  $\tau$ . The number of channel taps in discrete time is

$$L = B\tau = cB_0\tau = cL_0,\tag{7}$$

where  $L_0$  is the number of channel taps in the base system.

Clearly, when *B* changes, the number of taps changes. However, OFDM channel equalization does not depend on the number of taps. As we have reserved  $K_p = K/4$  subcarriers for channel estimation,  $K_p$  and *L* scale at the same rate. The channel estimation accuracy remains at the same level regardless of the bandwidth change.

For all different cases, the receivers processing are the same, as described in [6]. Since the transmitter and the receiver were stationary, no resampling operation of [6] is needed. Adjusting the data rate has minimal impact on the receiver design.

#### 4. EXPERIMENTAL RESULTS

#### 4.1. AUV Fest, June 07

This experiment was conducted in AUV Fest at Panama City Beach, FL, June 2007. The three sets of signals in Section 2.3 were used, where the center frequency was set at 32 kHz.

In this experiment, the receiving boat had an array in 65ftdepth water. The array depth is 30 ft to the top of cage. Array is 2 m in aperture with 16 hydrophones. A total of 8 channels were recorded (indexed as  $2, 4, 8, \dots$ ). In this paper, we show the results with a transmission distance of 500 m.

Fig. 1 illustrates some sample channel estimates for different settings, where the peak is normalized to 1 for visualization. The channel duration is around 18 ms for all the settings in this experiment.

#### 4.2. Buzzards Bay experiment, Aug. 2007

This experiment was conducted at the Buzzards Bay, MA, Aug. 2007. The two sets of signals in Section 2.4 were used, where the center frequency was set at  $f_c = 110$  kHz.

The transmitter gear was deployed to the depth of 20 ft to 25 ft with water depth 47 ft. Receiver array was deployed to the depth of 20 ft with water depth 47 ft. Array spacing is 0.2

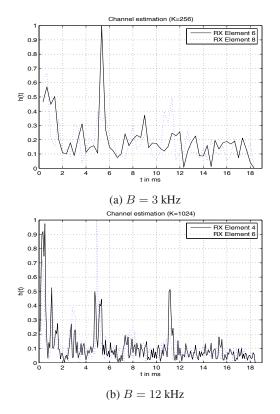


Fig. 1. Some estimated channels, AUV Fest, June 2007.

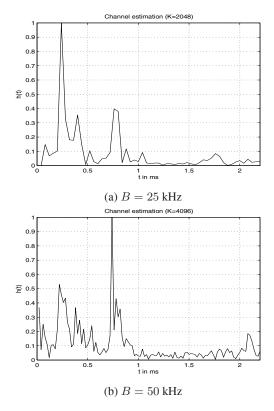


Fig. 2. Some estimated channels, Buzzards Bay, Aug. 07

Table 4. DER Results for CC with QI SR			
	1 receiver	2 receivers	
Bandwidth B	uncoded / coded	uncoded / coded	
AUV Fest, 3 kHz	0.1219 / <b>0.0403</b>	0.0395 / <b>0</b>	
AUV Fest, 6 kHz	0.0762 / <b>0.0063</b>	0.0218 / <b>0</b>	
AUV Fest, 12 kHz	0.0752 / <b>0.0048</b>	0.0185 / 0	
Bay Test, 25 kHz	0.0016 / 0	-	
Bay Test, 50 kHz	0.0834 / <b>0.0191</b>	0.0243 / <b>0</b>	

Table 4. BER Results for CC with QPSK

 Table 5. BER Results for LDPC with QPSK

	1 receiver	2 receivers
Bandwidth B	uncoded / coded	uncoded / coded
AUV Fest, 12 kHz	0.0613 / 0	-
Bay test, 25 kHz	0.0015 / <b>0</b>	-
Bay test, 50 kHz	0.1828 / <b>0.1851</b>	0.1102 / <b>0</b>

m. Tests were performed at 180 m and 300 m. We show only the results for the 180 m case.

Fig. 2 illustrates some sample channel estimates, with peak normalized to 1. The channel duration is about 2.5 ms in both settings.

## 4.3. BER performance for QPSK

BER results for convolutional coding (CC) with QPSK are collected in Table 4, and those for LDPC are in Table 5. A total of 43008 information bits were transmitted in each setting.

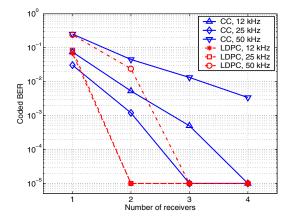
In some cases, there is no decoding error even with a single receiver. When two receivers are used, there is no error after channel decoding for all the cases tested.

## 4.4. BER performance for 16-QAM

The BERs after channel coding are plotted in Fig. 3, when 16-QAM is used. A total of 43008 information bits were transmitted in each setting. For the B = 12 kHz case, two receivers are needed for zero BER for LDPC, while three receivers are needed for zero BER for CC. For the B = 25 kHz case, two receivers are needed for zero BER for CC. For the B = 50 kHz case, three receivers are needed for zero BER for CC. For the B = 50 kHz case, three receivers are needed for zero BER for CC. For the receivers are needed for zero BER for CC. For the receivers are needed for zero BER for CC. For the receivers are needed for zero BER for CC. For the receivers are needed for zero BER for CC. For the receivers are needed for zero BER for LDPC, while for CC, a large BER still occurs with four receivers. This is because the used LDPC code has much better error-correction capability than the used convolutional code.

#### 5. CONCLUSIONS

In this paper we presented a scalable OFDM design that well adapts to a large range of transmission bandwidths. The receiver performed well in different geographic locations, with different coding and constellation choices. Via bandwidth scaling, high-rate transmission is readily available in short range underwater acoustic channels.



**Fig. 3**. The coded BER results with 16-QAM; zero BER is plotted as BER=1e-5 for presentation convenience.

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#### 6. REFERENCES

- B. C. Kim and I. T. Lu, "Parameter study of OFDM underwater communications system," in *Proc. of MTS/IEEE Oceans*, Providence, Rhode Island, Sept. 11-14, 2000.
- [2] M. Chitre, S. H. Ong, and J. Potter, "Performance of coded OFDM in very shallow water channels and snapping shrimp noise," in *Proc. of MTS/IEEE OCEANS*, 2005, pp. 996–1001.
- [3] P. J. Gendron, "Orthogonal frequency division multiplexing with on-off-keying: Noncoherent performance bounds, receiver design and experimental results," U.S. Navy Journal of Underwater Acoustics, vol. 56, no. 2, pp. 267–300, Apr. 2006.
- [4] M. Stojanovic, "Low complexity OFDM detector for underwater channels," in *Proc. of MTS/IEEE OCEANS*, Boston, MA, Sept. 18-21, 2006.
- [5] B. Li, S. Zhou, M. Stojanovic, and L. Freitag, "Pilot-tone based ZP-OFDM demodulation for an underwater acoustic channel," in *Proc. of MTS/IEEE OCEANS*, Boston, MA, Sept. 2006.
- [6] B. Li, S. Zhou, M. Stojanovic, L. Freitag, and P. Willett, "Non-uniform Doppler compensation for zero-padded OFDM over fast-varying underwater acoustic channels," in *Proc. of MTS/IEEE OCEANS*, Aberdeen, Scotland, June 2007.
- [7] B. Li, S. Zhou, M. Stojanovic, L. Freitag, J. Huang, and P. Willett, "MIMO-OFDM over an underwater acoustic channel," in *Proc. of MTS/OCEANS*, Vancouver, Canada, Oct. 2007.
- [8] M. Stojanovic, "On the relationship between capacity and distance in an underwater acoustic communication channel," *Proc.* of WUWNET, Los Angeles, CA, Sept. 2006.
- [9] M. Stojanovic, J. A. Catipovic, and J. G. Proakis, "Phasecoherent digital communications for underwater acoustic channels," *IEEE Journal of Oceanic Engineering*, Jan. 1994.
- [10] J. Huang, S. Zhou, and P. Willett, "Structure of non-binary regular LDPC cycle codes," in *Proc. of ICASSP*, Apr. 2008.