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Compressive Sensing Based Underwater Communication System.

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Abstract

Underwater acoustic communication is an important research area having significant industrial and marine applications. Ocean exploration and observation, command and control of Autonomous Underwater Vehicles, coastal surveillance etc. are a few among the many applications. Underwater acoustic channel is considered as one of the most difficult communication media used today. Limited low data rate due to the narrow bandwidth of the channel, extensive and time varying Inter Symbol Interference results from multipath propagation etc are the main obstacles in this area. This paper aims to develop an efficient and reliable underwater communication system that solves the above problems very much. The Compressed Sensing (CS) based underwater communication system uses Orthogonal Frequency Division Multiplexing (OFDM) as an alternative to single carrier broadband modulation to achieve high data rate transmission. Compressed Sensing is an attractive technique for transmission of signals at sub-Nyquist rate leading to faster acquisition and power conservation. The proposed system comprising OFDM and CS will be a viable solution for reducing signal degradation and achieving high data rate transmission over an underwater acoustic channel.

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Keywords: Underwater acoustic communication; Orthogonal Frequency Division Multiplexing (OFDM); Compressive Sensing (CS); Quadrature Phase Shift Keying (QPSK); Orthogonal Matching Pursuit algorithm (OMP).

1. Introduction

For the past decade, there has been a severe study in the area of underwater acoustic communications (UWC). The key idea behind this is the wide opportunities or applications offered by the underwater communication system. Since radio wave degrades rapidly in the water acoustic communication is considered as the most promising method for underwater applications [3]. But there is a network of problems that challenge the performance of underwater acoustic communication system. The performance of acoustic communication system mainly depends on the channel

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characteristics. Underwater acoustic channel is a time varying channel with Inter-Carrier Interference (ICI), Inter Symbol Interference (ISI), ambient noise, Doppler spread and fading. Due to the detrimental effect of time and frequency spreading, achieving high data rate in UWC is practically impossible.

The researchers are looking for an efficient system that offers high data rate in this particular fading channel along with high accuracy. One way to solve this issue is by the use of multicarrier modulation techniques for wireless communication systems. CDMA, MIMO, OFDM etc. form the basis technologies for 3G/4G wireless systems. Wi-Max, LTE, DVB, IEEE 802.11a etc. have adopted OFDM as a mean to increase future wireless communication requirements [4]. This paper combines OFDM and CS together in underwater communication scenario flowering in to a novel system capable of transmitting a few numbers of samples at a higher data rate for the exact and reliable recovery of the original signal with improved bit error rate as an extra advantage.

Unlike FDM, OFDM uses the sub carriers in an efficient way for transmission. The orthogonality between the carriers utilizes the available limited spectrum for high data rate. The traditional signal acquisition methods, leads to poor efficiency, due to large number of samples at Nyquist rate, higher time requirement and complex hardware requirement. Here we employ a parallel CS structure that exploits the sparsity of speech signal in the DCT domain as a mean to compress the signal at the transmitter side. The signal is transmitted in the underwater multipath environment and signal detection is carried out from the projections at sub Nyquist rate by l_0 minimization techniques. The proposed system simulations demonstrate that OFDM signals can be successfully reconstructed even in the presence of noisy multipath channels at sub-Nyquist sampling rates using CS.

2. Compressive Sensing background

CS is a revolutionary technique in signal processing. The basic difference between CS and Nyquist sampling lies in the way they achieve reconstructed signals. Nyquist theorem demands the sampling frequency to be greater than or equal to twice the maximum frequency of the original signal for signal recovery. But CS says, signal recovery is possible with much less number of samples. CS is based on two principles: Sparsity and Incoherence. A signal is sparse if it has compact representation in some basis $\boldsymbol{\psi}$. For example let a discrete signal, $\mathbf{f} \in C^N$ be sparse in some basis $\boldsymbol{\psi}$. Then we can express the signal $\mathbf{f}$ as [5],

$$\mathbf{f} = \boldsymbol{\psi}\, \mathbf{x} \tag{1}$$

where $\mathbf{x}$ is the K- sparse signal. A signal is K-sparse if it has at most K non-zero components; $K << N$. Since the signal has only less number of non- zero components in some basis $\boldsymbol{\psi}$, we can throw away a large coefficients without much perceptual loss. Let $\boldsymbol{\varphi}$ be the sensing matrix used to obtain some linear measurements, then the incoherence property says that $\boldsymbol{\varphi}$ and $\boldsymbol{\psi}$ must have less coherence value. The coherence between the sensing basis $\boldsymbol{\varphi}$ and the representation basis $\boldsymbol{\psi}$ can be obtained from [5] $\mu(\boldsymbol{\varphi}, \boldsymbol{\psi}) = \sqrt{N} \max|\boldsymbol{\varphi}_k, \boldsymbol{\psi}_j|$,where $1 \leq k, j \leq N$, and $\mu(\boldsymbol{\varphi},\boldsymbol{\psi}) \in [1, \sqrt{N}]$. Considering M linear measurements of the signal $\mathbf{f}$ which is given by [5],

$$\mathbf{b} = \boldsymbol{\varphi}\, \mathbf{f} + \mathbf{w} \tag{2}$$

Where $\boldsymbol{\varphi} \in C^{M\times N}$ represents the measurement system, $\mathbf{w} \in C^M$ represents additive measurement noise, and $\mathbf{b} \in C^M$ represents the measurement vector. Let $\boldsymbol{\psi} \in C^{N\times N}$ be the sparsifying basis, the measurement vector

$$\mathbf{b} = \boldsymbol{\varphi}\, \mathbf{f} = \boldsymbol{\varphi}\, \boldsymbol{\psi}\, \mathbf{x} = \mathbf{A}\mathbf{x} \tag{3}$$

where $\mathbf{A} = \boldsymbol{\varphi}\, \boldsymbol{\psi}$. The CS theory states that f can be recovered using [5],$M = KO(logN)$ non-adaptive linear projection measurements of the compressed array data vector $\mathbf{b}$, on to a random matrix namely $\boldsymbol{\varphi} \in C^{M\times N}$ sensing matrix.

3. Greedy Algorithms

Now our objective is to solve the problem $\mathbf{b} = \mathbf{Ax} + \mathbf{w}$. This is an underdetermined system of equation. i.e. Reconstructing $\mathbf{x}$ from a small number of measurements is an ill-posed problem and unique signal reconstruction is not possible, in general. However, for a K-sparse signal, unique signal reconstruction is possible by solving an optimization problem of the form [6], $min \parallel \boldsymbol{x} \parallel_0 \ subject\ to\ \parallel \boldsymbol{Ax} - \boldsymbol{b} \parallel_2 \leq \varepsilon$ provided spark($\mathbf{A}$) > $2K$. Solving this l_0 minimization require an extensive search through all possible sets of K columns of $\mathbf{A}$ and is an NP hard problem which is not feasible for practical applications. One of the popular alternatives is to use the closest convex optimization i.e. l_1 minimization [6],$min \parallel x \parallel_1 \ subject\ to\ \parallel \boldsymbol{Ax} - \boldsymbol{b} \parallel_2 \leq \varepsilon$, provided spark ($\mathbf{A}$) > $2K$.

However convex relaxation methods are computationally expensive leading to implementation issues. Hence an alternate strategy is to use greedy family of algorithms like Matching Pursuit, Orthogonal Matching Pursuit, Weak Matching Pursuit, Subspace Pursuit etc. The algorithm is called greedy because the optimal solution to each smaller instance will provide an immediate output. The algorithm doesn't consider the larger problem as a whole. Once a decision has been made it is never reconsidered. A formal description of Orthogonal Matching Pursuit strategy is given in the references [6].

4. CS based underwater communication system

This section describes the implementation of proposed compressed sensing based underwater communication system in detail. The implementation is done in mathematical programming language MATLAB. The block diagram of proposed transmitter section is shown in figure 1a.

4.1. Orthogonal Frequency Division Multiplexing (OFDM)

OFDM is an emerging technology used in wireless communication for high data rates. OFDM allows multiple carrier modulation as in Frequency Division Multiplexing (FDM). In OFDM the orthogonal subcarriers are selected for modulation which allows overlapping and hence the available limited channel bandwidth can be efficiently utilized. Orthogonality of subcarriers and guard band insertion in OFDM helps in eliminating ICI and ISI. Recent researches reveal application of OFDM for under water acoustic communication for high data rate with improved bit error rate performance. CS is a breakthrough in signal processing in which data acquisition is possible with less number of samples. i.e., it is a method for acquisition of sparse signals and reconstruction from compressed measurements. The transmission of information at Nyquist rate is quite unhealthy because of large and complex hardware, longer acquisition time and high power requirement. Compressed sensing technique in this context will simplifies this problem. This CS technique can be employed in OFDM based communication systems for data acquisition, and the sparse reconstruction can be done using greedy algorithms. In this paper, a compressive sensing based underwater OFDM communication system has been developed such that the system will provide good signal recovery using samples at sub Nyquist rate while maintaining high data rates. Since the underwater communication is inevitable for many applications, the proposed system will be relevant in this field.

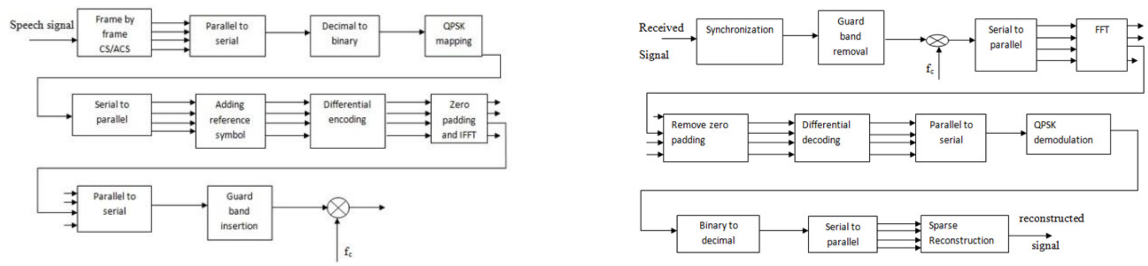


Fig. 1.(a) Proposed Transmitter Section;(b) Proposed receiver section

4.2. Transmitter section

A speech signal from NOIZEUS database sampled at a rate of 16 kHz is used as the information signal. The data is divided into a number of frames and compressed at Sub Nyquist rate. In order to perform compressed sensing we have to find the proper sparse basis of the signal and sensing matrix with great care. Each frame is represented as $\mathbf{f_1}$, $\mathbf{f_2}$, ...,$\mathbf{f_{t-1}}$, $\mathbf{f_t}$, $\mathbf{f_{t+1}}$, ... where $\mathbf{f_t}$ is the t^{th} frame. Considering M number of compressed measurements for each frame, where M << size of the frame, and taking M projections for each frame such that the projections are $\mathbf{y_t}$, $\mathbf{y_{t-1}}$, $\mathbf{y_{t-2}}$,.... where $\mathbf{y_t} = \boldsymbol{\varphi}\, \mathbf{f_t}$. $\boldsymbol{\varphi}$ is an $M{\times}N$ Gaussian random measurement matrix.

Now the compressed samples are in decimal format which must be converted as binary stream of data for further processing. Then the various OFDM parameters like bandwidth, sampling rate, symbol rate etc must be declared. Considering N orthogonal subcarriers for transmission and QPSK as the modulation technique, the number of symbols for transmission will be S=Number of bits for transmission/($2N$) , since 2 bits are considered at a time for QPSK mapping. Assuming f_s as sampling frequency, $T=1/f_s$ will be the elementary period. B is the bandwidth. T_s is taken as the symbol length and *delta* as the guard interval, so the total symbol period will be $Ts+delta$. Guard interval is $1/5^{th}$ of symbol duration. Sub carrier spacing is inverse of symbol duration. From these values rest of the variables can be calculated.

The binary values of the compressed data is then modulated into QPSK. The QPSK symbol mapping is shown in Table 1. Since QPSK maps 2 bits per subcarrier, the total data rate of the system will increase. The probability symbol error for QPSK over a channel with AWGN is given by,

$$P_e = \mathrm{erfc}(\sqrt{E_b}/N_0) \tag{4}$$

Where E_b is the symbol energy and $N_0/2$ is the noise power spectral density. The Bit Error Rate for QPSK symbols can be expressed as,

$$\mathrm{BER} = 0.5\ \mathrm{erfc}(\sqrt{E_b}/N_0) \tag{5}$$

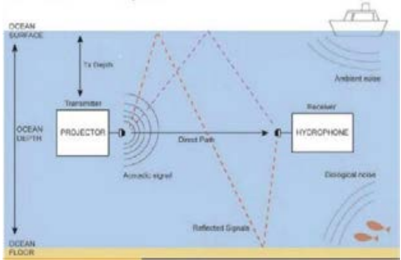
Table 1. QPSK symbol mapping

Bits	QPSK symbol	Phase shift
00	-1-i	1
01	-1+i	I
10	1-I	-I
11	1+i	-1

This implies a QPSK system can achieve the same average probability BER as that of BPSK system, where the QPSK system uses only half the channel bandwidth. Once the data bits are converted to required modulation format, they need to be superimposed on the required orthogonal subcarriers for transmission. Finally the data is transmitted over the underwater acoustic channel.

4.3. Underwater Acoustic Channel

The propagation of signal through the underwater acoustic channel might get affected by some of the physical properties and environmental conditions of the sea. It is important to study those factors to properly reconstruct the signal at the receiver end. This section discusses the channel factors which have a profound impact on the transmitted signal.



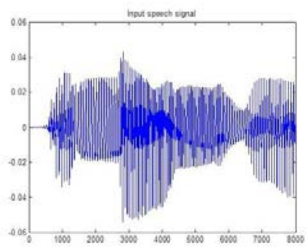


Fig. 2 (a) Multipath Propagation;(b) Input speech signal

4.3.1. Multipath Propagation

In underwater acoustics the signal from the projector may undergo surface and bottom reflections and thus results in multiple arrivals of the same signal. Figure 2a shows the multipath formation. Considering l_p as the length of p^{th} propagation path with $p = 0$ corresponds to the first arrival, the path delay can be calculated as l_p/c. The relative path delay is given by, $\tau_p = \frac{l_p}{c} - t_0$Where t_0 is the arrival time of the strongest path signal. The channel response can be expressed as

$$h(t) = \sum_p hp\ \delta(t - \tau_p) \tag{6}$$

Where $hp = \Gamma p \,/\, \sqrt{A_p}$ represents the gain of the p^{th} path, A_p is the propagation loss of the p^{th} path and Γp is the cumulative reflection co-efficient of the p^{th} path. This is for a usual radio channel. For underwater acoustic channel each frequency will experience different attenuation. The frequency response of the p^{th} path is given by [8],

$$Hp(f) = \sum_p \frac{\Gamma p}{\sqrt{A(l_p,f)}} \tag{7}$$

The overall channel response is given by [8],

$$H(f) = \sum_p Hp(f) e^{-j2\pi f\, \tau_p} \tag{8}$$

In underwater acoustics the multipath time delays and low propagation velocity of the acoustic wave leads to lengthening of received signal.

4.3.2. Doppler Effect

Doppler spread arises due to the relative displacement between the transmitter and the receiver. Due to this the frequency of the received signal might significantly differ from the frequency of the signal transmitted (typically by 1%).

4.3.3. Noise

Underwater acoustic channel contains ambient noise as well as site specific noise. Ambient noise exists everywhere in the sea whereas site specific noise exists only in certain places. Ice cracking in Polar Regions is an example of site specific noise. The ambient noise comes from sources such as turbulence, breaking waves, rain, and distant shipping. Ambient noise can be modelled as Gaussian but it is not white.

4.4. Receiver Section

The block diagram of proposed receiver section is shown in figure 1b .The OFDM receiver will do the reverse operation as that of the transmitter. The receiver side transducer will picks up the signal and down converted to base band frequency, Then the Guard Interval (GI) is removed. The signal is then demodulated which is same as the FFT operation and converted back to a binary stream using an appropriate symbol detector. From the transmitted and the received data we can find BER.

5. Simulation Results

From the figure 1a proposed system transmitter section takes a speech signal for transmission. The speech signal is taken from NOIZEUS database having a duration of 10 seconds. For this simulation we have considered only first 8000 samples of the speech signal as shown in figure 2b . The primary block in the transmitter section will divide the input signal into a number of frames such that the frames are not overlapping. For this simulation each frame consists of 1000 samples, resulting in a total of 8 frames. Since the input speech signal is sparse when represented in DCT basis, compressed sensing can be applied. The compressed samples are taken from each frame by using a sensing or projection matrix (here a random matrix is taken). 600 compressed samples from one frame are taken for conventional compressed sensing and for adaptive compressed sensing the number of projections/samples taken may vary adaptively. Since the data is transmitted as serial stream of bits, the compressed samples of each frame is transformed to a serial data consisting of 4800 samples (if 600 samples taken from all the 8 frames) shown in figure 3a.

Now it's the time to perform OFDM technique in the transmission section. OFDM parameters used for this particular simulation are shown in table 2. For that the samples must be converted in to binary format from the decimal format. For this simulation 1,53,600 bits are used. The bit stream is coded by a QPSK modulator which means that the two by two bits are given a specified QPSK symbol. Since di bits are considered at a time and are transformed to QPSK symbols, the length of array of QPSK symbols will be half of the original size that is 76,800 QPSK symbols. Then the serial stream of QPSK symbols is converted to parallel form. The number of parallel streams is determined by the number of subcarriers used for data transmission. Here 512 sub carriers are used with 150 symbols.

After the serial to parallel conversion comes the differential encoding. It is important that the differential coding is done on the parallel data; otherwise the differential coding will fail. In QPSK modulation the information is carried by the phase of the signal. But in DQPSK the data will be carried by the phase difference between the successive symbols. In the differential coding block a reference signal is created using *rand( )* function. This will be the first symbol to be sent and will be used by the receiver for synchronization. So here a total of 151 symbols transmitted. In order to perform differential coding first the QPSK symbols are mapped to complex phase according to table 1. Then the reference symbol is multiplied by the first phase mapped symbol and forms a new column. Now the information lies in the phase difference between new column and reference column. Then the new column will be multiplied by the second phase mapped column and create second DQPSK column. Similarly the process goes until all the information are mapped to the phase difference between consecutive columns of DQPSK matrix. The OFDM block consists of three operations, zero padding, an inverse Fourier transform and the insertion of guard interval.

The zero padding is done to achieve oversampling which will improve the system performance over underwater acoustic channels with large Doppler spread. The zeros are inserted at the middle of the parallel data. Then the data mapped onto the subcarriers by performing the IFFT operation using the built in *ifft( )* function. This creates a time domain symbol which is the sum of all the subcarriers. This operation also converts the data from parallel to serial again. Next guard bands are inserted between OFDM symbols to avoid inter symbol interference (ISI). Here a guard band of 50 ms is considered. Then the data is up converted from baseband frequency to carrier frequency, f_c=25kHz before it is transmitted over the channel (Figure 3b).

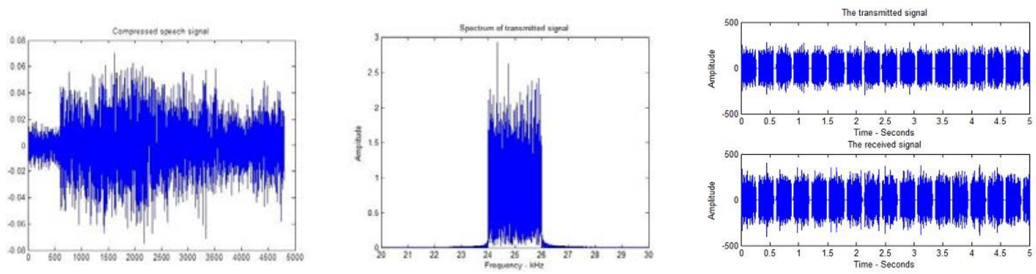


Fig.3. (a) Compressed signal; (b) Spectrum of transmitted signal;(c) Transmitted and received signal

In a real communication channel there are many sources that affect the transmitted signal. In this simulation we have implemented a number of factors that cause errors in the received signal. The first thing that comes into picture for a wireless communication channel is the effect of noise. Noise is simulated in the channel by adding Additive White Gaussian Noise (AWGN). Next considering the phase error that occurs while transmission. This is simulated by adding a linear increasing phase. That is multiplying the transmitted signal with the exponential function *exp(iθ),* where θ is the phase error. The phase error increases from 0 to 2π. Another important factor in underwater wireless channel is multipath propagation. Figure 4a shows the channel impulse response used for this simulation. This channel impulse response is convolved with the transmitted signal to get the effect of multipath. The resulting signal will be sums of five transmitted signals with four signals have a delay and weaker amplitude. These factors will change the shape of the received signal as shown in figure 3c.

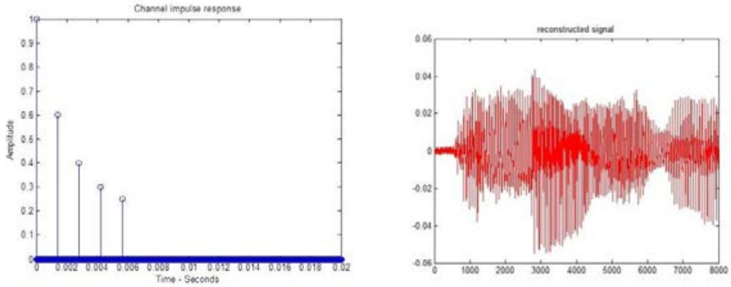


Fig.4. (a)) Channel impulse response; (b) Reconstructed speech signal at the receiver end

The block diagram of proposed receiver section is given in figure 1b. After receiving the signal it will undergo reverse operations as that of the transmitter section. The received signal may be some delayed version of the transmitted. So in order to perform the reverse operations the received signal has to be synchronized. The receiver side synchronization can be achieved by correlating the signal with the reference signal. By locating the maximum value of correlation function one can able to find the start of the signal. In Matlab correlation is obtained by using the inbuilt *xcorr( )* function. Matlab *max( )* function will find the location of maximum value. Thus by synchronizing the signal receiver side FFT operation can be applied to the correct window.

The OFDM block at the receiver is also made up of three operations. First the guard interval is discarded. Then the data is down converted to baseband frequency from the carrier frequency. After that the received signal is transformed back into the frequency domain by an FFT operation using the built in *fft( )* function. The zeros which were inserted at the middle of the data at the transmitter is then removed. The data must be converted back to the parallel format before doing all OFDM reverse operations. Since we now have the data in the phase difference of the received signal, the next step is to decode the differential coding. The differential decoder finds the phase difference between two succeeding symbols from the same subcarrier and the output is QPSK symbols. This is done by multiplying a symbol at a given sub-carrier by the complex conjugate of the previous symbol at the same sub-carrier and the phase difference is found.

Table 2. Channel Parameters and OFDM parameters

Channel Parameters	Values	OFDM Parameters	
Modulation	DQPSK	OFDM symbol length, T_s	256ms
Transmitter and Receiver elements	One transmitter one receiver	Guard Interval	50 ms
Sampling frequency, f_s	150 kHz	Subcarrier spacing	3.9 Hz
Carrier frequency, f_c	25 kHz	Number of Subcarriers	512
Bandwidth	2 kHz	Number of OFDM symbols	150

Practically the phase differences found by the differential decoder is not exactly equal to the values shown in the table 1. These values can be found by first comparing the absolute values of the imaginary and the real parts of the phase difference. If the imaginary part is the biggest then the phase difference is either i or -i or if the real part is the biggest then the phase difference is either 1 or -1. The sign is then decided by comparing the value to zero. If the imaginary part is less than zero then the phase difference is -i and if it is greater than zero the phase difference is i. Similarly if the real part is less than zero the phase difference is -1 else it is 1. When the phase differences are decided it is decoded into QPSK symbols as shown in table 1 .The parallel data consisting of QPSK symbols are converted into serial data before it is demodulated. The QPSK symbols are demodulated into di bits according to table 1. Ideally the bit stream given to the transmitter and the demodulated bit stream from the receiver are same. Since the underwater acoustic channel affects the transmitted information up to an extent, the bit stream demodulated at the receiver may contain some error. The Bit Error Rate (BER) can be calculated as the ratio of number of bit errors to the total number of bits transmitted.

The next step is to reconstruction of original signal from the demodulated bit stream. For that the binary data is converted to decimal format. Then the decimal values are rearranged in frames such that each frame contains 600 values since we are taken 600 compressed samples from a frame of 1000 samples at the transmitter section. For adaptive compressed sensing the number of compressed samples taken must be depend on the inter frame correlation. Now the data looks in matrix form with 600 rows and 8 columns, where 8 is the total number of frames available. Then by l_0 minimization sparse values of the speech signal are reconstructed. Since the speech signal is more sparse in DCT basis compared to all other basis, by performing *idct( )* operation the original signal can be found. Here orthogonal matching pursuit is employed for sparse reconstruction. Thus the speech signal with 8000 samples is successfully reconstructed at the receiver end (Figure 4b).

6. Conclusions

Wireless communication as always a complex field for researchers. Even though the area of underwater acoustics the level of complexity increases a bit more, it explores a wide variety of applications and advantages. The study of underwater acoustic channel reveals that the existing underwater communication experience several difficulties like low data rate, limited band width, multipath propagation, complex hardware, large power consumption, high bit error rate etc. The compressed sensing based underwater communication system is a novel method in the field of underwater acoustics. The new system utilizes OFDM with DQPSK modulation and compressed Sensing techniques together sweep away the channel difficulties and avoid channel equalization.

MATLAB simulation implies that the proposed system will be an asset to the field of underwater communication. Accurate recovery of large signal is possible with minimum data transmission. The simulation shows that, proper detection is possible for a speech signal of 8000 samples transmitted over an underwater acoustic channel with a limited bandwidth of 2kHz, symbol duration of 256 ms using 512 sub carriers. A total of 150 OFDM symbols are transmitted at a data rate 3.3 kbps where the carrier spacing is 3.9 Hz. Signal detection is even possible with a compression ratio of 0.5 resulting in a bit error rate only in the order of 10^{-4} .In short, the new Compressed Sensing based underwater communication system can reconstruct original large information signal from compressed signal transmitted at a high data rate over a limited bandwidth under water acoustic channel, while holding high bit error rate performance.

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