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The Solution of Multipath Fading Problem at Receiver in Underwater Acoustic Channel using Frequency Spectrum

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***Abstract*— The aim of the paper is to mitigate the effects of multipath fading at the receiver in underwater acoustic communication channel. Our approach leverages the frequency spectrum analysis of the composite received signal to identify the most reliable information-bearing frequency. By examining the magnitudes of frequencies present in the received signal, which may consist of multiple signals with varying degrees of orthogonality, we develop a technique to select a specific frequency as the true bit. This selection process is based on comparing the magnitudes of the frequencies within a given time interval. By choosing the frequency with either the highest or the lowest magnitude, we provide a solution that offers a probability of 1/2 for receiving accurate information.**

Keywords*— Multipath propagation, Multipath fading, Fast Fourier transform, frequency spectrum.

I. INTRODUCTION

Earth, the third planet from the sun, is an ellipsoid with a circumference of about 40,000 km. Maximum earth's surface is covered with ocean water, accounting for about 71% of earth's surface. Ocean underwater is a home to unique creatures from the dumbo octopus to the faceless cusk. It has become a significant research area for mining. Because, maximum deep-sea areas are yet to be discovered. Researchers, to explore deep sea areas, always prefer acoustic communication [1]. Because it provides long distance communication. However acoustic communication has some inconvenience, limited bandwidth, multipath propagation, multipath fading, limited battery, limited data capacity, delay in propagation, for example, are the limitations [2], that make it hard to explore deep-sea areas. One of the limitations is multipath propagation because multipath fading at the receiver [3]. Underwater acoustic sensor networks consist of a number of sensors, act as data transmitter, and beacon nodes, act as data receiver [4]. There are several techniques that can surpass multipath propagation problems. Normally, a multipath environment for multiple propagation is built up in shallow underwater channels [5]. Time difference of arrival (TDoA) is one of techniques [6], based on maximum or minimum value screening. In [7], they proposed that a single cluster process be formulated using a recursion Bayesian estimation framework to overcome multipath propagation problems. In [8], they experimented measurement to detect multipath propagation and proposed their work can be used to design a system using ultrasounds for transfer of information over multipath propagation. In [9], in order to make unambiguity multipath returns from direct-path arrivals, they normalized cross-correlations by an expected direct-path delay and geometrically averaged over multiple array orientations. Another experiment was conducted in the swimming pool [10], they explored the severe multipath propagation of acoustic signals in the swimming pool, and they attempted to create a model of the swimming pool response and finally they created an algorithm for object localization. A least-square estimation of the amplitude and time delay of each path is an algorithm, proposed to estimate multipath time-delay [11]. This algorithm is a function of a delta-dependent parameter and rules for choosing this parameter. Furthermore, multiuser acoustic communication can cause multipath propagation, resulting in distortion to high-rate digital signals [12]. In [13], they proposed an algorithm, named Kasami-based approach to the overhead using a cross-layer adaptive frame structure for the RTS/CTS hand-shaking protocol to bring solution of time and frequency variation due to multi-user communication. Apart from all these solutions, in this paper, a probabilistic solution has been proposed to solve the multipath fading problem at the receiver due to multipath propagation.

II. OCEAN AS TRANSMISSION MEDIUM OF SOUND SIGNAL

Speed of the wave totally depends on the medium through which the wave propagates. Ocean water is such a medium where sound waves can propagate more distances than in terrestrial. The average speed of sound waves in the air is approximately 331 meters per second. On the other hand, the sound wave travels, part second, between 1450 and 1570 meters in ocean water. It varies, depending on temperature, salinity, depth of water [14]. Different ocean vertical water columns contain different temperature, salinity. It, however, also varies, depending on the difference in latitude and longitude of earth. The average depth of the ocean is about 3688 meters, and it accounted for 71% of the earth's surface. As sound goes down, firstly, the speed of sound decreases then it increases (according to the calculation of speed of sound by using the dataset "vertical profile of temperature and salinity at SEATS and F2 [15]), provides depth, temperature, and salinity, by using the Mackenzie equation [16]. The speed of sound, therefore, is.

$$V = 1448.96 + 4.591T - 5.304\text{x}10^{2}T^{2} + 2.375\text{x}10^{-4}T^{3} + 1.340(S - 35) + 1.630\text{x}10^{-2}D + 1.675\text{x}10^{-7}D^{2} - 1.025\text{x}10^{-2}T(S - 35) - 7.139\text{x}10^{-13}TD^{3}$$

Where T, D and S are temperature, depth, and salinity. Fig 1. shows the average speed of sound in every vertical water column, based on the dataset "vertical profiles of temperature and salinity at SEATS and F2 [15]", where latitude and longitudes are (18.0110, 115.9790) and (22.0240, 1230250). For this calculation only the dataset from F2 station has been considered.

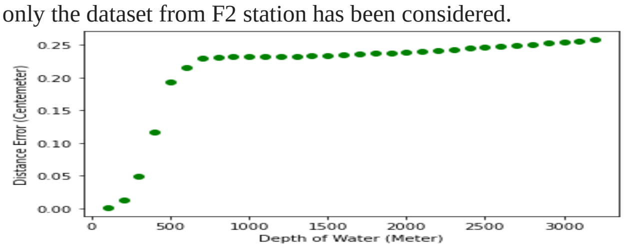

Fig 1.The average speed of sound in every vertical water column

Due to differences of speed of sound, sound refraction occurs, resulting in curved direction in sound propagation instead of straight direction. These are the bad properties of ocean water as a transmission medium of sound waves. But, in ocean water, a sound wave, is an acoustic wave, travels a longer distance than a radio wave. When travelling at minimum speed, sound waves lose little energy, leading to the Waves to propagate over distances of up to 25000 kilometers, whereas the range of travelling distance of radio waves is very low.

III. CHARACTERISTICS OF SOUND WAVE

There are two types of waves, longitudinal waves and transverse waves. Wave in which the particle of the medium vibrates 'to and fro' in the same direction in which the wave is moving, is called longitudinal wave. By contrast, in a transverse wave, particles vibrate up and down at right angles to the direction. Sound waves are longitudinal waves. A sound wave, in general, consists of 5 characteristics, wavelength, amplitude, frequency, time period and velocity. A sound wave as a representation of sine function of time domain is $y(t) = Asin(2\pi ft + \varphi = Asin(\omega t + \varphi)$, denote *A*, *f*, *ω* or $2\pi f$ and are amplitude, frequency, angular frequency and phase. Frequency is the most important characteristic to distinguish one sound wave from another. The number of sounds waves a sound produces per second refers to the frequency. The attitude is used to describe the size of the wave. You can consider it the height of the wave. There is a direct relationship between frequency and pitch and between amplitude and loudness. While sounds with a higher frequency produce high-pitched, shrill ear- splitting sounds, and sound with a lower frequency produces low-pitched, deeper sounds like bass sounds. Meanwhile, amplitude has a positive relation with loudness of sound means, the higher the amplitude the louder the sound and the lower the amplitude the milder the sound. It can be simplified by when we are higher the volume of any song in our mobile phone or laptop means, higher the amplitude of this song and vice versa. The distance of two repeats of sound waves is wavelength. Time period is the opposite of the frequency. Where, time period T = 1/f means, amount of time required to produce a single complete wave. And, lastly, velocity is the travelling distance of sound waves per second.

IV. LOSS OF SOUND'S AMPLITUDE

We know that the higher the amplitude, as sound travels through a medium, sound loses its energy. Sound energy is converted into heat when it passes through a medium. Basically, sound waves get absorbed by molecules within the medium. When molecules start vibrating by the energy of the sound wave, the molecules need energy because the molecules, before, are at rest. Molecules need to be powerful enough to overcome their interference to movement. The molecules, therefore, get energy from sound waves when they vibrate. Thus, a sound wave loses its energy as it travels through a medium. But in the ocean underwater, energy loss is more than in pure water or in air, due to the viscosity of the medium that sound waves must overcome. Furthermore, as sounds propagate through a medium it spreads out. Due to spread out sound waves also lose their energy [17]. The equation of intensity (I) of energy is,

$$I = P/A$$

Where, *P* is the power and *A* is the area. But we know that, in the ocean underwater, sound waves propagate spherically, so the area of a sphere is,

$$A = 4\pi r^2$$

Here, r is radius. If we consider sound wave spread out from r_1 position to r_2 position of the area, the energy will also spread out from r_1 position to r_2 position. Therefore, power within r_1 radius and power within r_2 radius will be the same. So,

$$P_1 = P_2$$

$$I_1 4\pi r_1^2 = I_2 4\pi r_2^2$$

$$I_2 = I_1 (\frac{r_1}{r_2})^2 \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(1)$$

If we consider, r_2 is n times greater than r_1. Then $r_2 = nr_1$, therefore from equation (1)

$$I_2 = I_1 (\frac{r_1}{nr_1})^2$$

$$I_2 = \frac{1}{n^2} I_1 \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(2)$$

From equation (2), we can understand that, as the sound wave travels through any medium, it decreases at rate $1/n^2$. Consequently. The amplitude decreases as sound waves travel distances.

V. MULTIPATH PROPAGATION OF SOUND SIGNAL IN OCEAN UNDERWATER

In shallow underwater, to communicate, sound signal provides higher travelling distance benefits than radio signal. But there are some problems that affect sound communication. Channel vibration, multipath propagation and doppler shift [18], are the problems that affect sound communication in shallow underwater. Multipath propagation is one of them that is controlled by two effects: 1. sound reflection at surface and bottom of ocean water, 2. refraction in ocean water occurs due to variation of sound speed in ocean water. Sound speed depends on ocean water's temperature, salinity, and depth, which is elaborately discussed in section 2. Due to multiple propagation of sound signal, multipath fading, and distortion in received signal occurs, causing significant degradation of the underwater sound communication signals [12]. Multipath fading includes construction and destruction interference and phase shifting of the signals [19]. Figure x shows the construction and destruction interference of the signal.

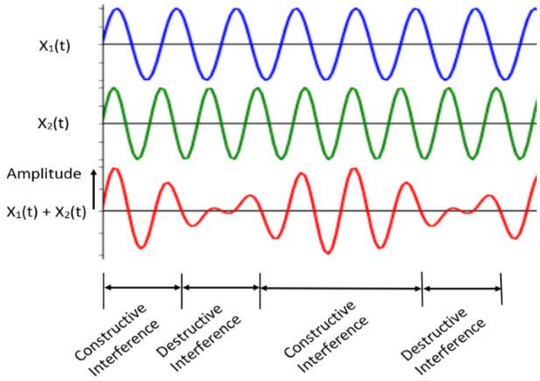

Fig 2. Constructive and destructive interference of sound signals

From Fig 2., we can see that the negative part of the wave form of X_2(t) is reduced by the positive part of the waveform of X_1(t) and vice versa. By contrast, the positive part and negative part of the waveform of X_2(t) are amplified by the positive part and negative part of the waveform of X_1(t) respectively. Where,

$$Y(t) = X_1(t) + X_2(t)$$

Therefore, the amplitude of a composite wave is,

$$|A_{Y(t)}| = \sqrt{A_{X_1(t)}^2 + A_{X_2(t)}^2}$$

In this paper, multipath fading and distortion has been addressed and a solution has been proposed to bring down this problem.

VI. ORTHOGONALITY OF SIGNAL

Two signals are orthogonal to each other; their wave functions do not overlap to each other. They are totally independent of each other. If we consider two periodic sound signals $X_1(t) = A_1 \sin(2\pi f_1 t + \varphi_1)$ and $X_2(t) = A_2 \sin(2\pi f_2 t + \varphi_2)$. Where A_1 and A_2 are amplitudes, φ_1 and φ_2 are phases and f_1 and f_2 are frequencies of these two signals. From the condition of orthogonality of signals, we know if $\int_{T_0}^{T} X_1(t).X_2(t)dt = 0$ over a time period then X_1(t) and X_2(t) signals are said to be orthogonal signals. And if $\int_{T_0}^{T} X_1(t).X_2(t)dt \neq 0$ over a time period then they are not orthogonal each other. Therefore,

$$\int_{T_0}^{T} X_1(t).X_2(t)dt$$

$$= \begin{cases} zero & if\ they\ are\ orthogonal\ to\ each\ other \\ non\ zero & if\ they\ are\ not\ orthogonal\ to\ each\ other \end{cases}$$

For sinusoidal wave, if the frequency of two signals (X_1(t) and X_2(t)) are different then,

$$\int_{T_0}^{T} X_1(t).X_2(t)dt = 0 \quad [where, f_1 \neq f_2]$$

And if their frequencies are same then,

$$\int_{T_0}^{T} X_1(t).X_2(t)dt \neq 0 \quad [where, f_1 = f_2]$$

Consequently,

$$X_1(t)\ and\ X_2(t)\ are \begin{cases} orthogonal & if\ f_1 \neq f_2 \\ not\ orthogonal & if\ f_1 = f_2 \end{cases}$$

If two different harmonic sound waves are orthogonal, in the frequency spectrum, two frequencies will be shown. On the other hand, if two same harmonic sound waves are not orthogonal, in the frequency spectrum, only one frequency will be shown.

VII. Frequency Modulation in Underwater Acoustic Communication

Underwater acoustic communication is considered as the most challenging wireless communication method [20]. However, there are other methods of Underwater acoustic communication such as sparse adaptive convolution cores [21][22][23], time-domain turbo equalization [24][25]. Due to its own characteristics, underwater acoustic communication is very difficult to convey. Such as low bandwidth, underwater noise, multipath fading, doppler shifting, limited data capacity, delay in propagation, prolonged time, serious inter-symbol interference [2][26]. Frequency modulation is such a technique that is used in underwater acoustic communication [27]. This technique uses sound frequencies to carry digital data. Frequencies represent digits, one certain frequency represents the digit-1, and another certain frequency represents the digit-0. Fig 3. shows the two different frequencies carry two digits and a signal carries digital data.

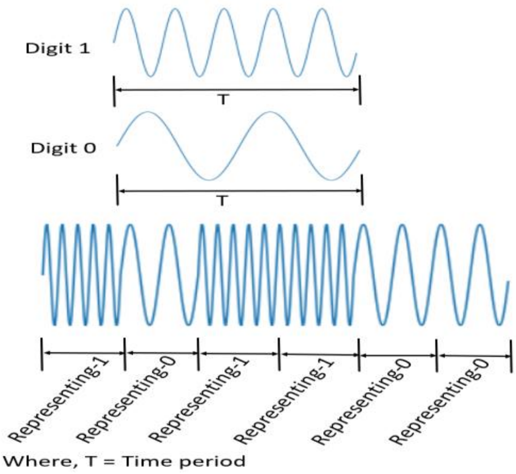

Fig 3. Frequency modulation technique

In this paper, frequency modulation method has been considered in acoustic communication and a solution has been shown to solve the multipath fading problem due to multipath propagation of sound waves.

VIII. Fast Fourier Transform (FFT) on Sound Signal

First of all, to know the fast Fourier transform (FFT), we need sufficient knowledge about Fourier transform (FT) and discrete Fourier transform (DFT). Fourier transform is a mathematical algorithm that transforms the time domain function to frequency domain function [28][29]. It is a generalization of the complex Fourier series in time limit from -∞ to ∞. The equation of Fourier transform X(f) in frequency domain (f) is,

$$X(f) = \int_{-\infty}^{\infty} X(t)e^{-2\pi ift}dt$$

Where, i is a complex number that refers $\sqrt{-1}$ and $e^{i\Theta} = cos\Theta + isin\Theta$. But it's a continuous Fourier transformation. In real life applications, it is impossible to apply. Then discrete Fourier transform (DFT) is invented [30]. Discrete Fourier transform is usable to any real-life application. It takes samples equally spaced in a finite sequence instead of the infinite limit of a time domain function and converts them into a frequency domain function. So, the discrete Fourier transform (DFT) is.

$$X(f) = \sum_{n=0}^{N-1} X_n e^{-2\pi if^{n}/_{N}} \qquad [for, 0 \leq f \leq N-1]$$

The complexity of a discrete Fourier transform algorithm is $O(n^2)$ which is troublesome for large data samples. Then came the fast Fourier transform [31][32] with complexity O(NlogN), which almost acts as a linear function. It, basically, computes the discrete Fourier transform of a sequence. Fast Fourier transform creates frequency spectrum that shows every signal, together creating composite signals, in frequency domain. In this paper, fast Fourier transform has been used to create frequency spectrum to distinguish two different frequencies of sound signals that come from sender in multiple different paths, cause multipath fading problem at receiver and a solution of multipath fading problem has been brought out.

IX. Problem Domain

Due to multipath propagation, at the receiver, phase shifting of different signals occurs, causing multipath fading. Consequently, the composite signal, consisting of multiple signals, is not able to carry the true information. The amount of time shifting of a signal totally depends on additional travelling distance due to propagation in different paths. Sound signals, in ocean underwater communication, carry 1 or 0, using frequency modulation, Fig 3. shows the modulation technique. Because, in digital communication, everything is being transferred into sequence of 1 and 0 digits. If we consider high frequency signal in a certain time length T represents 1 and low frequency signal in same time length T represents 0. Fig X shows the high frequency and low frequency signals in time length T, representing digit 1 and 0 respectively. If we consider digital data, for example 101100. The signal representation of this data in time domain X(t) is shown in Fig X. If phase shifting occurs in $X_2(t)$ in a positive direction with respect to phase of $X_1(t)$, and

$$X_1(t) = A_1\sin(2\pi f_1 t + \varphi_1)$$
$$X_2(t) = A_2\sin(2\pi f_2 t + \varphi_2)$$

$$\text{Then } \varphi_2 = \varphi_1 + d\varphi \qquad [Where, d\varphi = [0, \infty]]$$

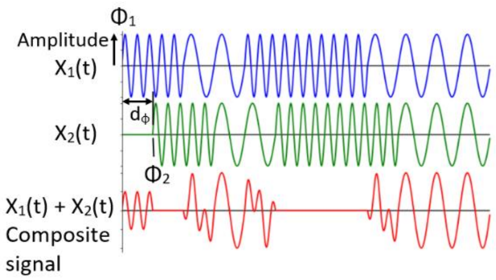

Fig 4. A composition of a composite signal from two signals, where their frequencies same but phases are different.

The composite signal will be carrying false information. In fig 4, two signals have been shown to generate composite signals. But, in real situations, it can be more than two signals. Phase shifting can be in different situations. Fig 5. Shows variations in phase shifting.

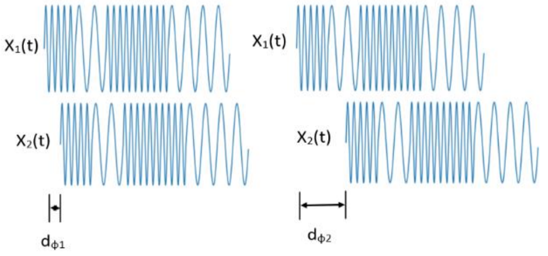

Fig 5. Variations of phase shifting

Due to multipath propagation, two or more signals, at the receiver, create a composite signal. If we convert this composite signal from time domain to frequency domain, two sceneries can be created-

1. The magnitude of high-frequency (representing 1) is higher than the magnitude of low-frequency (representing 0)

2. The magnitude of low-frequency (representing 0) is higher than the magnitude of high-frequency (representing 1)

It depends on how many high frequencies and low frequencies overlap with each other. However, other things can change the magnitude of frequencies, elaborately discussed in section 4. If the number of overlapping high frequencies is more than the number of overlapping low frequencies, the magnitude of high frequency will be more than the magnitude of low frequency and vice versa. Fig 6. shows the relation between the number of frequencies in a certain time length, overlapping with each other, and their magnitude in frequency spectrum.

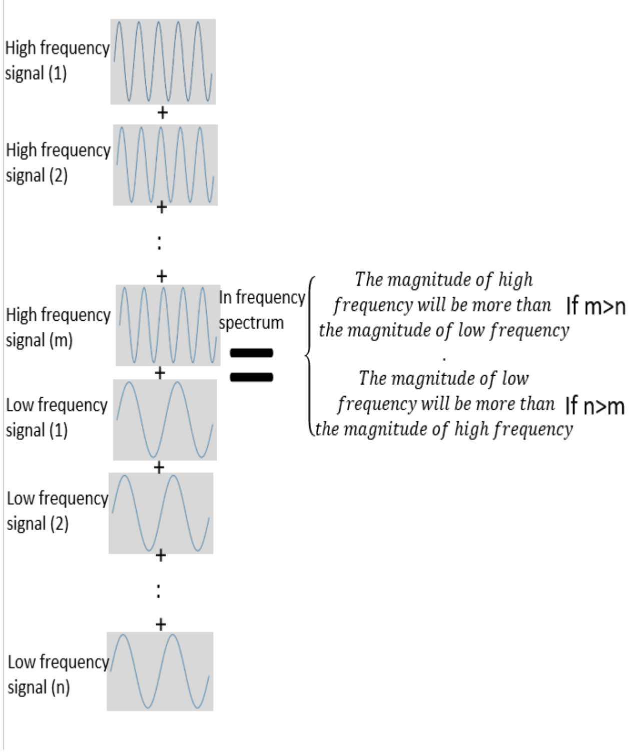

Fig 6. Relation between number of frequencies in a certain time lenth, overlaping each other, and their magnitude in frequency spectrum

In this paper, a solution has been proposed to deal with the multiple fading problem.

X. Proposed Method

If we look at fig 6 and consider frequency as the true bit, whose magnitude is more than other in the time length (T), then

1. The magnitude of high-frequency (representing 1) is more than the magnitude of low-frequency (representing 0) ∩ 1 is true bit in the time length(T) = True (1)

2. The magnitude of high-frequency (representing 1) is more than the magnitude of low-frequency (representing 0) ∩ 0 is true bit in the time length(T) = False (0)

3. The magnitude of low-frequency (representing 0) is more than the magnitude of high-frequency (representing 1) ∩ 1 is true bit in the time length(T) = False (0)

4. The magnitude of low-frequency (representing 0) is more than the magnitude of high-frequency (representing 1) ∩ 0 is true bit in the time length(T) = True (1)

So, the probability of true bit, receiver receiving in the time length (T), =

$$\frac{True + False + False + True}{4}$$
$$\frac{1+0+0+1}{4}$$
$$\frac{1}{2}$$

Thus, if we consider low frequency in frequency spectrum as true bit, the probability of true bit, receiver receiving in the time length (T), is $1/_2$. Therefore, when a receiver receives a composite signal, if we consider high or low frequency as true bit in the time length (T), the probability of receiving, at receiver, true information from this composite signal is $1/_2$. Because the probability of true bit, receiver receiving, in the rest of the time will be $1/_2$, the rest will be maintaining the same probability if sender and receiver remain static. So, if we conclude this, the receiver, either, should receive high frequency from the frequency spectrum or should receive low frequency from the frequency spectrum to get true information from this composite signal that is caused by multipath fading due to multipath propagation.

XI. Result and Analysis

To simulate this proposed model, I have used python. First of all I created an acoustic signal 8 seconds long consisting of 200 Hz and 800 Hz of 2 seconds long for each frequency consecutively. Fig 8 shows the data I used in my simulation

Fig 8. Data for simulation

I represented this acoustic signal as 0101, where 0 represents a 200 Hz of 2 seconds long and 1 represents a 800 Hz of 2 seconds long. Then I created another acoustic signal, the same as before. Afterward, I assembled the second signal with the first signal after 0, 0.5, 1, 1.5, and 2 seconds later. Then run the proposed model to find out the actual bit for every 2 seconds.

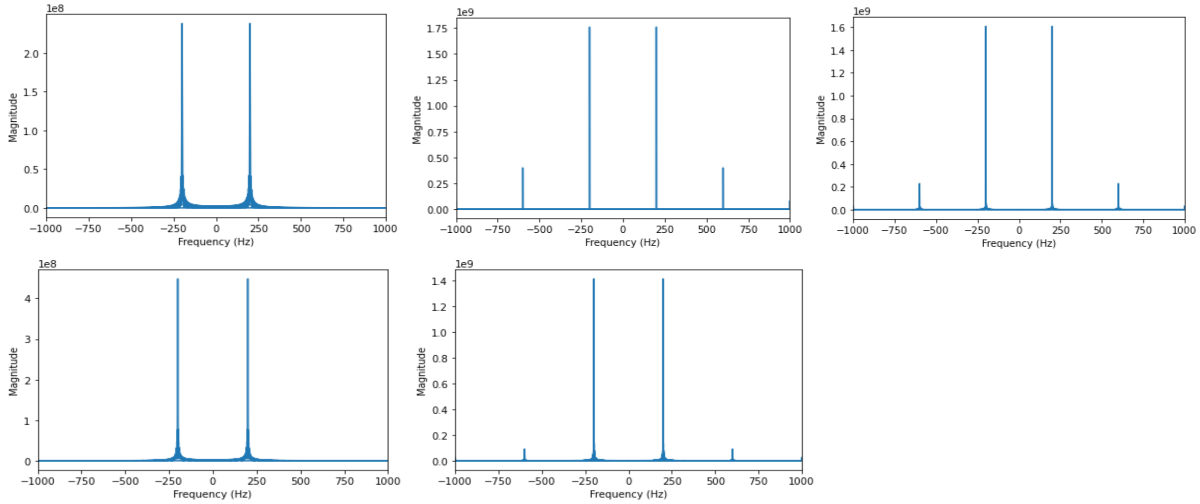

Fig 8. Frequency spectrum from 1st 2 seconds of composite signal with phase difference 0, 0.5, 1, 1.5, and 2 seconds of the two identical signals, where magnitude of 200 Hz is greater than the magnitude of 800 Hz.

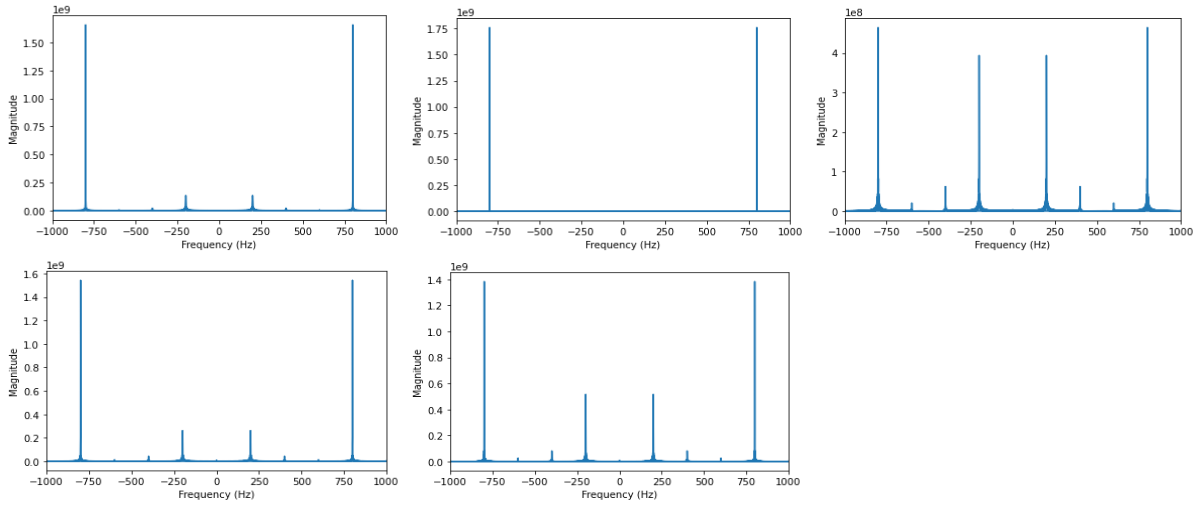

Fig 9. Frequency spectrum from 2nd 2 seconds of composite signal with phase difference 0, 0.5, 1, 1.5, and 2 seconds of the two identical signals, where magnitude of 800 Hz is greater than the magnitude of 200 Hz.

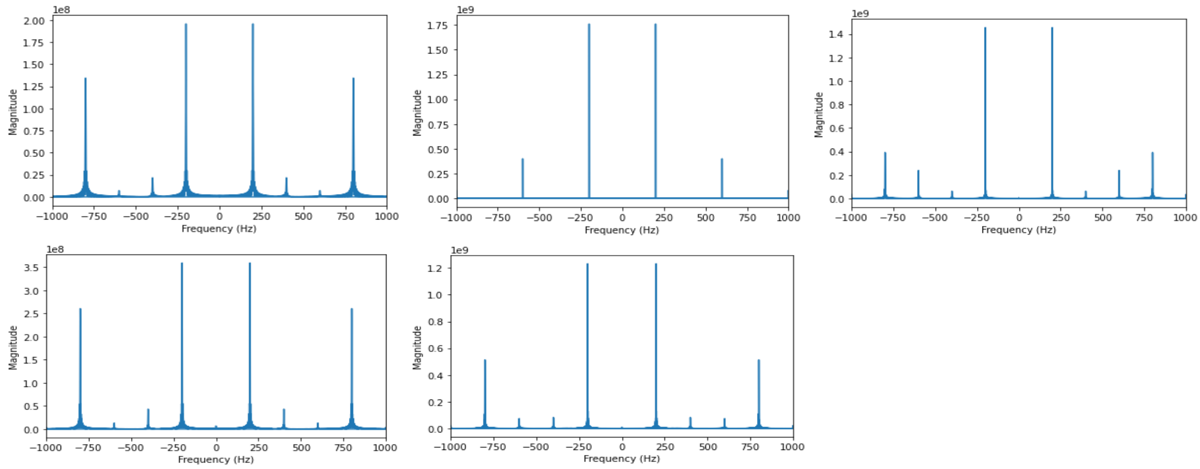

Fig 10. Frequency spectrum from 3rd 2 seconds of composite signal with phase difference 0, 0.5, 1, 1.5, and 2 seconds of the two identical signals, where magnitude of 200 Hz is greater than the magnitude of 800 Hz.

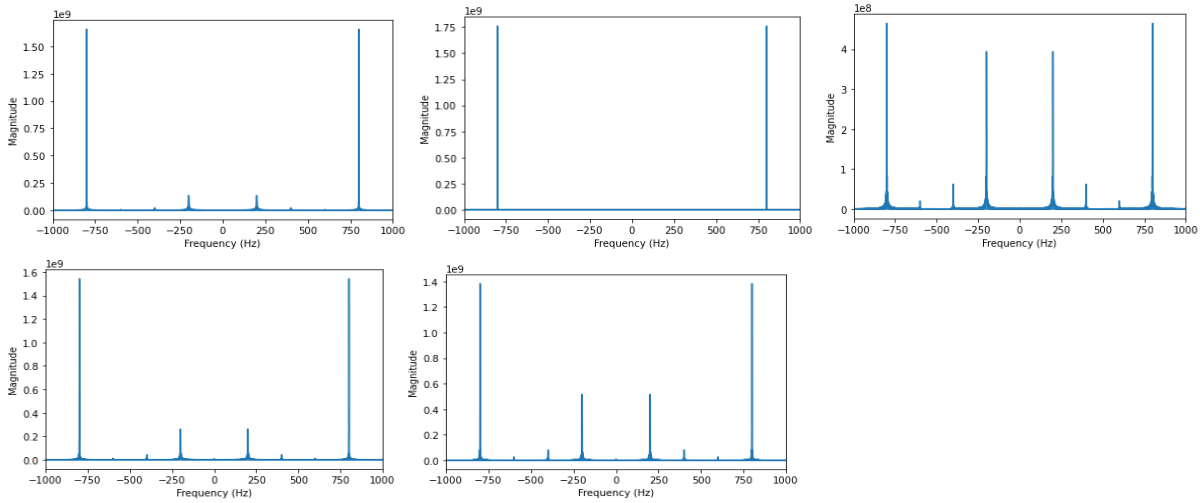

Fig 11. Frequency spectrum from 4th 2 seconds of composite signal with phase difference 0, 0.5, 1, 1.5, and 2 seconds of the two identical signals, where magnitude of 800 Hz is greater than the magnitude of 200 Hz.

In fig 8, and 10, all frequency spectrums show a magnitude of 200 Hz is greater than magnitude of 800 Hz. If I consider the frequency of higher magnitude as the true bit, it will give us the actual bit. And fig 9, and 11, all frequency spectrums show magnitude of 800 Hz is greater than magnitude of 200 Hz. Again, if I consider the frequency of higher magnitude as the true bit, it will give us the actual bit. Here I considered the frequency of higher magnitude as the true bit, it gives us the signal as 0101. If I had considered the lower magnitude of frequency as the true bit, then it would have given us the signal as 1010 which is the wrong signal. So, if we consider the frequency of either higher or lower frequency as the true bit, the probability of providing us the actual signal is ½. In this simulation, I have used 2 acoustic signals. This proposed also will be working for more than 2 signals. The result depends on which frequency you select as true bit from frequency spectrum. Sometimes it could be lower frequency is true bit.

XII. CONCLUTION

In this work, an approach has been presented to beat multipath fading problem at receiver in underwater acoustic communication. In shallow underwater channels, multipath propagation is a common problem that causes multipath fading. This approach can be a good solution in shallow underwater channels to get accurate information. Using fast Fourier transform on received composite signal provides a solution based on the magnitude of frequencies (by selecting either higher magnitude of frequency as a true bit or lower magnitude of frequency as a true bit) on frequency spectrum and is not very hard to implement in real life. Where data transmission is very necessary, but multipath fading impedes the data transmission, this approach can be implemented.

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