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Implementation of Adaptive Filter Algorithm for Underwater Acoustic System

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Abstract— **This paper describes development of adaptive filter algorithm for an Underwater Acoustic Telemetry (UWAT) system for communication between the off - shore underwater device (UD) and on-shore communication system. In this system, FSK modulation and de-modulation techniques are used with centre frequency of 38 KHz with baud rate of 300Hz. Adaptive filter algorithm has been developed to compensate the distortion introduced due to multipath in under water communication. It has been shown experimentally that received signal has been perfectly demodulated after compensating it by adaptive filter algorithm. This demodulated signal is useful for identifying the commands correctly from the transmitter. The applications of this system can be extended to command and control the activities of underwater data acquisition devices.**

Index Term—**Acoustic telemetry, remote control, modem, adaptive filter , Modulation and Demodulation.**

I. INTRODUCTION

There is a growing demand for underwater wireless acoustic communication with the rapid expansion in marine environmental monitoring activities like Autonomous underwater vehicles and remotely operated vehicles, ocean oil explorations applications. Electromagnetic waves do not propagate over long distance at extremely low frequencies. The disadvantage with under water wire communication is that the cables are easily damaged, by trawlers and other underwater activity [1-2]. The underwater acoustic telemetry system has the advantages against these conventional techniques. Several authors were reported in open literature about the underwater acoustic communication system [3-4]. The underwater acoustic telemetry channel is reverberant which posses many obstacles to reliable high speed digital communication [5]. It is also band limited. The three distinguishing feature of underwater acoustic communication are frequency dependent propagation loss, severe multi path and low speed of sound propagation.

Present system employed FSK modulation of digitally encoded data which are robust to time and frequency spreading of channel. Filtering is a process by which the frequency spectrum of signal can be modified, reshaped or manipulated according to some desired specifications. Digital filters are digital systems that can be used to filter discrete time signals. A advantages of digital filters over conventional filters are accuracy is high, reliability is high and filter parameters can be changed in order to change filter characteristics and it can design adaptive filters. The use of adaptive filters permits some level of compensation for the channel reverberation [6]. This paper presents the acoustic modem based on FSK modulation and implementation of adaptive filter to compensate channel reverberation. The Present system developed is cost effective and efficient.

II. UNDERWATER ACOUSTIC COMMUNICATION SYSTEM

The typical command/control/data monitoring system for an underwater device is shown in Fig. 1. The remote UW Device shall carryout relevant functions as desired by control commands from the COMOS control console, sends back acknowledgement, and desired data back to control centre. All the necessary software for command, control and monitoring is built in to Single Board Computer (SBC) of the COMOS control centre. Similarly the software required for transmission from underwater device is built into the UD SBC. The single board computer of under water device transmits data from sensors to control console based on commands from control console.

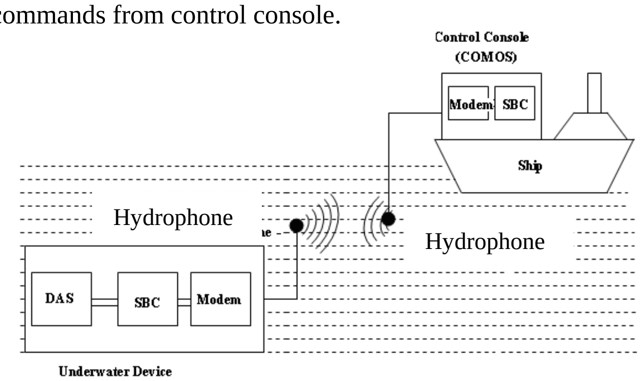


Figure 1. Basic Block Diagram of UWAT System

III. DESCRIPTION OF MODEM

Modems used at the monitoring station and the underwater device (UD) are identical in design as shown in Fig. 2. Frequency Shift Keying (FSK) modulation is used for data transmission of data. Frequency shift keying is a technique in which the frequency of carrier is switched or hopped between the two possible values corresponding to binary symbols 0 and 1, with frequency limits set by the channel, baud rate distance. The practical FSK transmitter is implemented using a voltage control oscillator. The FSK MODEM used in the system provides output of sub-carriers 37 KHz for 'mark' and 39 KHz for 'space'. The outputs from FSK modulator is fed to a power amplifier. The received signals are amplified and passed through adaptive filter for removing the channel noise. The outputs of filter FSK tones at 37 KHz and 39 KHz. These are fed to the demodulator to recover the data.

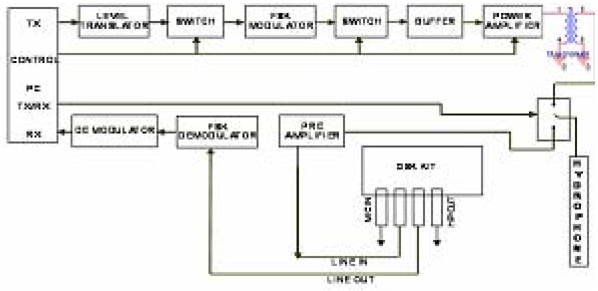


Figure 2. Block diagram of MOFDEM

IV. UNDERWATER CHANNEL

Information bearing signal is often found in practice have uncertainty created by additive noise. There is additional uncertainty created by the randomness of signal parameter. The usual cause of this randomness is the distortion of the transmission channel. Thus it is possible that even if addition noise could be removed, there will still remain an uncertainty about which signal was transmitted. The signal parameters of particular interest will be amplitude, phase, frequency and time of arrival. In wireless underwater acoustic channel, the signal from transmitter may arrive at receiver through different paths. As a result signal pick up by the receiver is a composite signal consists of multi path signals. The multi path arrives at receiver at slightly different delays and has different amplitudes. An underwater acoustic channel is characterized as multi path channel due to signal reflections from the surface and the bottom of the sea. Because of wave motion, the signal multipath component under goes time varying, propagation delays, that result in signal fading. This results in a composite signal which can vary widely and rapidly in amplitude and phase. This phenomenon is called fading. Multi path also causes inter symbol interference for digital signals.

V. ADAPTIVE NOISE CANCELLOR

The adaptive systems can automatically adapt in the face of changing environments and changing system requirements. They can be trained to perform specific filtering and decision-making tasks.. When the input signal characteristics are unknown or time varying they offer increased system performance than non adaptive systems [3]. The adaptive noise canceller behaves as an adaptive notch filter whose null point is determined by the angular frequency w_0 of the sinusoidal interference. Hence the canceller is tunable and tuning frequency moves with w_0. Thus unlike the situation with ordinary notch filter, we have control over the frequency response of the adaptive noise canceller. The primary input is

$$d(n) = s(n) + A_0 \cos(w_0 n + Ø_0) \qquad (1)$$

Where s(n) is information bearing signal, A_0 is the amplitude of the sinusoidal interference, w_0 is the normalized angular frequency and $Ø_0$ is the phase. The reference input $u(n)=A\cos(w_0 n+Ø)$, Where the amplitude A and Phase Ø different from primary input and angular frequency w_0 is the same. The block diagram of adaptive noise canceller is shown in Fig. 3.

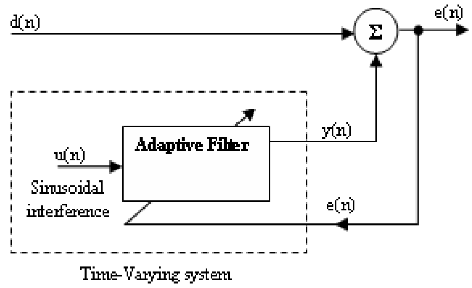


Figure 3. Representation of adaptive noise canceller

The adaptive system with input e(n) and output y(n) varies with time. The transfer function $H(z)=\frac{E(z)}{D(z)}$ is given

$$H(z) \approx \frac{z^2 - 2z\cos w_0 + 1}{z^2 - 2(1-\mu MA^2/4)z\cos w_0 + (1-\mu MA^2/2)} \qquad (2)$$

This is the transfer function of a second order digital notch filter with notch at normalized angular frequency w_0. For a small value of stepsize parameter μ such that $\frac{\mu MA^2(z)}{4} << 1$,

Poles of H(Z) are located at $Z \approx (1-\frac{\mu MA^2}{4}) + jw_0$

Two poles lies inside the unit circle means that adaptive noise canceller is stable, and can be used for practical use for real time. By its vary nature adaptive systems must be time varying and nonlinear. Their characteristics depend, among other things, on their input signals. The adaptive linear combiner is fundamental to adaptive signal processing. In this combiner there is an input signal vector with elements X_0, X_1, ……X_L. A corresponding set of adjustable weights A_0, A_1, …A_L, a summing unit and a single output signal y.

VI. IMPLEMENTATION OF ADAPTIVE FILTER

A procedure for adjusting or adapting the weights is called a "weight adjustment", "gain adjustment" or "adaptation" procedure. Since the weights are being adjusted, they are a function of the input components and the output is a nonlinear function of the input. Thus the operation of this combiner is nonlinear. If the adaptive system is an adaptive linear combiner, and if the input vector X_K and the desired response d_k are available at each iteration the adaptive Algorithm is generally the best choice because of its ease computation and simplicity.

$$€_k = d_k - X_k^T A_k \qquad (3)$$

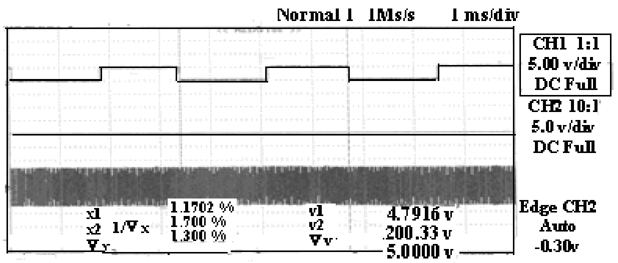
Where X_K is the vector of input samples in ether of the above two cases.

$$A_{k+1} = A_k + 2\,\mu\, €_k\, X_k \qquad (4)$$

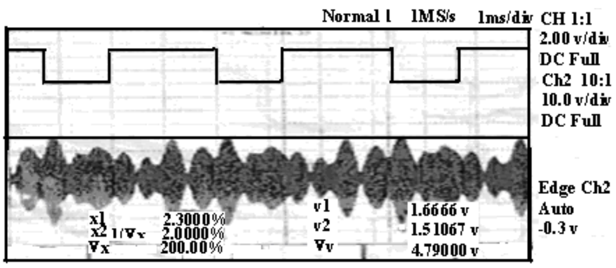
Where μ is a constant controlling the convergence of the weight vector having units inverse to that of the power of the signal. If the signal input is within -1 to +1 then its power also is within this range. For convergence the step size μ is selected in the range 0 < μ <1/M, where M is the order of the filter.

VII. DISCUSSION ON EXPERMENTAL RESULTS

The experimental test has conducted in acoustic tank. Square wave with frequency of 300Hz is FSK modulated send over the channel Pre amplifier output and FSK demodulator output are shown in the Fig. 4a for Line of sight transmission communication. It is observed that demodulated wave is a perfect square wave for line of sight communication. For multipath transmission, FSK demodulated wave is not perfect square wave shown in Fig. 4b. It is observed that on time and off time of square wave is not equal this shows effect of channel distortion on demodulation. To compensate the distortion, adaptive filter algorithm is applied. The input (bottom) and output (top) of adaptive filter are shown in Fig. 5. The distorted signal is input to adaptive filter. Adaptive filter algorithm removes this distortions. The output of adaptive filter is shown in Fig. 5 at top. Now, this signal is suitable for demodulation. From this demodulated received signal commands can be identified perfectly from transmitter. Experiments also has been conducted by transmitting ASCII data from modem and received the demodulated signal (blue color) perfectly as shown in Fig. 6.



(a) Line of Sight Transmission



(b) Multipath transmission

Figure 4. Experimental Results of Underwater Acoustic tank

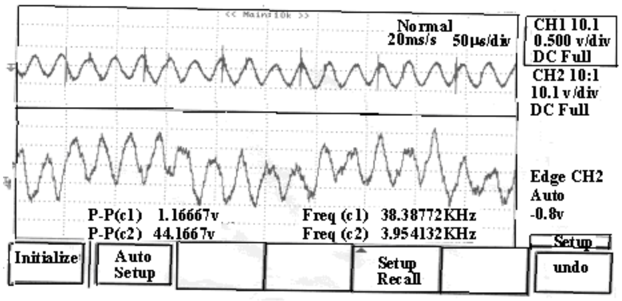


Figure 5. Output and Input of adaptive filter

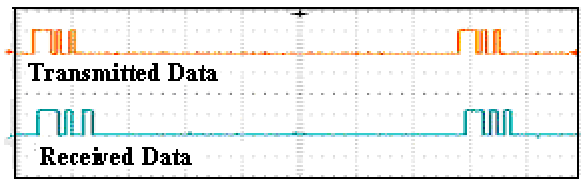


Figure 6. Input and Output Wave forms of Acoustic MODEM

VIII. CONCLUSION

FSK modem with adaptive filter algorithm was developed and tested in acoustic tank with baud rate of 300Hz. Adaptive filter algorithm is implemented for compensate the distortions. The received signal is highly suitable for identifying the commands. FSK modem with adaptive filter algorithm developed for underwater acoustic telemetry system is cost effective with center frequency of 38 KHz. The application of modem is limited to short range around 1000 meter and low data rate of communication. The application of the system is monitoring the activities of under water devices

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