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A RELIABLE UNDERWATER ACOUSTIC DATA LINK EMPLOYING AN ADAPTIVE RECEIVING ARRAY

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ABSTRACT

One of the main problems encountered when trying to produce a reliable underwater acoustic data link is that of multipath reflections from the sea surface and seabed. The most common method of overcoming this multipath problem is to use transmitting and receiving transducers with very narrow beamwidths. However, this solution makes alignment of the beampatterns critical, and any mis-alignment or motion of the transmitter or receiver may destroy the acoustic link completely.

In this paper, a reliable underwater data link which employs an adaptive receiving array is described. The purpose of the array is to provide automatic mainlobe tracking of the transmitter and to suppress directional mulitpaths and interferences. This adaptive array therefore solves the problems of beampatterns mis-alignment whilst also providing a certain degree of beampattern nulling to suppress directional multipaths and interferences.

The adaptive receiving array requires some prior knowledge of the desired incoming signal in order to distinguish it from the multipath reflections. This is provided by using a PN coded DPSK modulation scheme, where the PN coding is known at both the transmitter and receiver.

1. INTRODUCTION

High data rate acoustic telemetry for underwater communications has been pursued ever since it was recognised that the ocean could support signal transmission. In the past, most of the applications for underwater communications have arisen from military needs. However, as a result of oil and gas exploration work there are now an increasing number of offshore commercial applications (e.g. well-head monitoring and control, untethered submersibles etc). In all of these applications the major requirement is for a reliable, high data rate acoustic link.

The ocean is an extremely difficult medium in which to achieve the high data rates required for many applications. The ability of the ocean to support reliable high data rate transmissions is limited by the available transmission bandwidth and the detrimental effects of multipath propagation caused by boundary reflections and volume scattering. Also the maximum range of propagation in the ocean is limited by the increased absorption of acoustic energy at higher frequencies.

The complex properties of the acoustic channel therefore require a careful choice of modulation scheme and the use of multipath reduction techniques to achieve a reliable high data rate acoustic link. The modulation scheme must be chosen to make the best possible use of the available bandwidth. After considering the most common digital modulation techniques (i.e. OOK, FSK and PSK), a PSK type modulation scheme was chosen for its ability to provide low error probability for a given signal-to-noise ratio (SNR) and allow high data rate to be achieved in the available bandwidth [1]. Unfortunately, this PSK type modulation scheme, in common with the other basic binary modulation techniques, is unable to overcome the effects of multipath by itself. The most common method of overcoming this multipath problem is to use directional transducers with narrow beamwidths at both the transmitter and receiver. However, this solution makes alignment of the beampatterns critical, and any mis-alignment or motion of the transmitter or receiver may destroy the acoustic link completely. In order to reduce the effects of multipath an adaptive receiving array was used to provide automatic mainlobe tracking of the transmitter and to suppress multipaths and any other directional interferences. The adaptive receiving array therefore solves the problem of beampattern mis-alignment and is also able to operate in environments where the transmitter or receiver may be in motion.

2. THE UNDERWATER ACOUSTIC DATA LINK

The block diagram of the proposed underwater acoustic data link is shown in Fig. 1.

The transmitter employs a PN coded DPSK modulation scheme and operates at a centre frequency of 50kHz. The transmitting array provides a useable bandwidth of 20kHz and has a conical beampattern of 16^{o} beamwidth.

The adaptive receiving array is controlled by a least-mean-square (LMS) algorithm which determines the beampattern and frequency response of the array [2]. This adaptive array requires some prior knowledge of the desired incoming signal in the form of a reference signal. The reference signal must be correlated (spacially and temporally) with the desired incoming signal and uncorrelated with any multipaths or interferences. To provide such a reference signal, a PN coded DPSK modulation scheme was used with the PN code known at both the transmitter and receiver. At the receiver this PN code is used to distinguish the desired incoming signal from any multipath reflections or interferences. This combination of the PN coded DPSK modulation scheme and the LMS adaptive array provides a high level of multipath and interference protection.

2.1 THE MODULATOR - DEMODULATOR

Refering to the basic block diagram of the transmitter shown in Fig. 1a, the incoming data stream (rate Rd) is exclusive -ORed with a higher rate maximal length linear PN code (rate Rc, where Rc/Rd is an integer) whose properties[1,3-5] are:

i The value of the PN code's autocorrelation function is high at zero shift and very low elsewhere (to aid synchronisation)

ii The PN code has good statistical properties and equal numbers of 0's and 1's (to suppress the generation of an unwanted carrier frequency component) and

iii The length of the PN code is long enough to ensure that the code is not repeated during the transmission of a message (to prevent synchronisation problems)

The PN coded data stream is then used to DPSK modulate (PSK modulation requires additional receiver circuitry to resolve its phase ambiquity [3]) the chosen 50kHz carrier frequency, so that a data '1' produces a phase change and a data '0' causes the phase to remain the same as for the previous data value. The resultant PN coded DPSK signal has a spectrum whose null to null bandwidth (containing 90% of the total signal power and all the necessary phase information) is Rc/Rd times larger than the bandwidth of the incoming data stream (2Rd) [4]. The ratio of PN code to incoming data rate Rc/Rd, known as the spreading ratio (SR), therefore indicates that the resultant PN coded DPSK signal has a bandwidth of 2Rd x SR. As this bandwidth is necessary for the successful transmission of the PN coded DPSK signal through the underwater medium, the desire for a high data rate to be realised in a fixed (20kHz) transmission bandwidth indicates that the value of SR must be small and Rd = (10kbit/s)/SR. Thus, values of SR = 4 and SR = 8 (switchable) are chosen for this proposed data link, so that data rates of 2.5 and 1.25kbit/s respectively can be achieved.

At the output of the adaptive array shown in Fig. 1b, the received signal (PN coded DPSK + noise + interference) is coherently demodulated by extracting a reference carrier signal (directly from the received signal) with a suppressed carrier tracking loop, and using the reference in an optimum DPSK demodulator [3]. (The use of an auxillary channel for synchronisation or reference purposes is not only wasteful of transmission bandwidth, but is of little benefit in the underwater channel, as the channel is temporally and spatially variant as well as being frequency dependent). The resultant baseband signal containing contributions from the PN coded stream, interference (multipath and/or CW) and noise, is then multiplied by a synchronised version of the PN coded signal is then despread to the original data bandwidth (2Rd). All signal contributions not synchronised with the PN code (i.e. CW interference, multipath and noise) are spread to a larger bandwidth (>2Rc) with a resultant reduction in their spectral densities and correlation with the data stream in the despread data bandwidth. This reduction in the unwanted signal level causes an increase in the SNR observed in the data bandwidth (known as the process gain Gp, where Gp = 10log(SR) dB [4]) and indicates that by spreading and despreading the incoming data spectrum, the detrimental effects of the multipath may be reduced. The level of this improvement in performance is however small, as the chosen values of SR (4 or 8) restrict the process gain to either 6 or 9 dB. Finally, the decision on the value of each data bit is performed by taking a majority decision over the SR 1/Rc time slots of each despread data bit, after bit synchronisation (i.e. the location of the start/end of each data bit) has been performed with a data transition tracking loop [3,5]. As the majority decision performs as an error correction, the resultant error probability is further decreased.

In order for the despreading operation to be performed properly at the receiver, it is essential that the receivers' PN code is synchronised with the identical version present in the received signal. The task of obtaining and maintaining the required is synchronisation complex [3-5], but can be summarised as follows:

i Initial Synchronisation (Acquisition)

After a cold start (e.g. power-up) when there is no prior knowledge of the timing difference between the received and the receiver's PN code, the first task of the synchroniser is to rapidly obtain synchronisation with ±1 bit. This can be achieved by rapidly correlating the received PN code (at this point data is not usually sent) with all the receiver's PN code phases until a correlation peak (PN property (1)) is detected. In practice, this operation can take a considerable time, and it is common practice to transmit a known section of the PN code (preamble) to reduce the required search range

ii Tracking

Once the PN codes have been aligned to within ±1 bit, it is necessary to fine tune and maintain the alignment to within ±0.1 bit. This is achieved by using a closed loop configuration to adjust the phase of the receiver's PN code by small amounts in order to maximise the correlation (at zero shift) between the two codes.

2.2 THE ADAPTIVE RECEIVING ARRAY

A block diagram of the adaptive receiving array is shown in Fig. 1(b). This diagram can be divided into two parts: the adaptive array and the reference signal generation loop.
The adaptive array uses a rectangular transducer consisting of 96 elements arranged as 16 horizontal stores of 6 elements, with an interelement spacing of $\lambda/2$. In the system that was tested only 7 staves of this transducer were used, with either a $\lambda/2$ or λ spacing between staves. This gives an azimuth beamwidth of 17° and elevation beamwidths of either 14° ($\lambda/2$ spacing) or 7° (λ spacing). Since the multipath propagation is mainly caused by reflections from the sea surface and sea bed, the adaptive array was used to perform beamforming in elevation only. The input signals from the 7 array stores are adjusted in phase and amplitude by a set of finite impulse response (FIR) filters and then summed to produce the adaptive array output y. The array output y is then subtracted from the reference signal d to produce an error signal e for the LMS algorithm. The LMS algorithm is used to control the weights of the FIR filters and therefore determines the beampattern and frequency response of the adaptive array. The LMS alogrithm is a correlation loop which adjusts the FIR filter weights in order to minimise the mean-square-error between the reference signal d and the array output y. Therefore, the LMS algorithm adjusts the array beampattern and frequency response such that the array output y is an estimate of the reference signal d. The reference input can therefore be chosen to make the adaptive array track a

desired input signal. For the LMS adaptive array the reference signal must be highly correlated with the desired signal at the array output and uncorrelated with any interference components at the array output [6]. If the reference signal satisfies this condition then the adaptive array's beampattern will track the desired signal and null any directional interferences (i.e. multipaths). The generation of reference signal d for the adaptive array will now be considered.

The reference signal generation loop shown in Fig. 1(b) is used to generate a reference signal d for the adaptive array from the adaptive arrays output y. Consider that a desired signal s and a directional interference i (i.e. multipath) are present at the receiving array. Therefore the array output y consists of a desired signal component and an interference component. The desired signal is demodulated by the DPSK demodulator and then despread (bandwidth compressed) by the synchronised PN code generator to the data bandwidth. The interference is also present after DPSK demodulation, but since it is not synchronised with the PN code generator its spectrum is spread to at least the PN code bandwidth. Therefore, the data signal is available at the receiver output for data bit detection and the interference power is reduced, since only a small part of the interference spectrum is present at the data bandwidth. The data signal is then passed through a limiter which controls the amplitude of the reference signal. Finally, the limited data is PN coded and DSPK modulated to produce a constant amplitude of the reference signal s at the reference signal input d of the adaptive array. The remaining interference at the receiver output also passes through the limiter and its spectrum is spread again by the PN code generator. This interference is then DPSK modulated and appears at the reference signal input d of the adaptive array. However, this interference i' is now decorrellated with the original interference i at the output of the adaptive array. Therefore, the desired signal s at the array output passes through the reference signal generation loop virtually unaltered, except for a constant amplitude and possibly a small delay. Whereas the interference i at the array output is decorrelated by the loop as denoted by i' at the reference signal input.

The reference signal input to the adaptive array s + i' is therefore highly correlated with the desired incoming signal s at the array output and uncorrelated with the interference i at the array output. This satisfies the necessary requirements for a reference signal for the LMS adaptive array. Therefore the LMS algorithm adjusts the FIR filter weights form a beampattern mainlobe in the direction of the desired signal s and a null in the direction of the interference i (i.e. multipath). After the LMS algorithm has fully converged the array output will contain only the desired signal component s which passes through the loop to appear at the reference signal input. The adaptive array then has an ideal reference signal and the mean-square-error approaches zero.

After the adaptive array has converged then the multipath protection for the desired signal consists of the array gain phases the multipath nulling plus the protection offorded by the process gain of the PN code. In the previous example it was assumed that perfect synchronisation had already been required. In practice, the adaptive array beamforming and the synchronisation are performed simultaneously.

2.3 RESULTS

An all digital implementation of the PN coded DPSK modulator - demodulator described in Section 2.1 was realised and tested in the presence of Gaussian noise, without the generated signals being transmitted through the underwater medium or the adaptive array being used [5]. Typical performance results (probability of error against SNR) are shown in Fig. 2 for DPSK and PN coded DPSK (SR=4 and 8), and indicate that the modulator - demodulator is able to achieve the required reliability (low error probability). For example, Fig. 2 shows that the PN coded DPSK modulator - demodulator is able to attain an error probability of 10^{-5} (suitable for submersible control commands) when SNR (at the input to the demodulator) is 2.06 dB and -2.74 dB for SR= 4 and 8 respectively.

The adaptive array was implemented digitally using the bit-slice technology to achieve the high sampling frequencies required for operation at the carrier frequency [7]. To assess the performance of the adaptive array it was first tested by itself, without the reference signal generation loop. A few of the results of these tests are presented here. A 127-bit long PN code sequence was PSK modulated and then transmitted over a path length of 2 metres to the adaptive receiving array. The separation distance was kept short because of the small water tank that was used for these tests. An ideal reference signal was provided for the adaptive array by means of a "cheat-wire" from the transmitter. This reference signal was a time delayed version of the transmitted PN coded PSK modulated signal with the time delay corresponding to the direct path delay through the water. Therefore the reference signal for the adaptive array was an ideal direct path signal. The transmitted and received waveforms for this test are shown in Fig. 3. The PN code is shown in Fig. 3(a) and the transmitted PN coded PSK modulated signal is shown in Fig. 3(b). The adaptive array output waveform after convergence is shown in Fig. 3(c). This waveform is a delayed version of the transmitted signal and although it has some distortion due to bandlimiting, the phase changes are clearly visible. The weights of the FIR filters were originally set-up to give a mainlobe beamwidth of over 100^o. After the adaptive array had converged, the filter weights were frozen and a beampattern was taken. This beampattern is shown in Fig. 4. It can be seen that a mainlobe of approximately 14^o beamwidth has been formed at the 0^o direct path signal. This therefore shows that the adaptive array has tracked the direct path signal.

3. CONCLUSIONS

In this paper, a brief description of a proposed reliable high data rate underwater acoustic data link has been given. The main sections of the link consisting of a PN coded DPSK modulator - demodulator and an adaptive receiving array have been shown to provide the required high data rate and multipath reduction properties respectively. In addition, it has been shown that the PN coded DPSK modulation scheme chosen to provide a reference for the adaptive array, is able to produce the desired low error probability at attainable SNR's.

REFERENCES

[1] F.G. Stremmler, "Introduction to Communication Systems", Addison-Wesley, 1982

[2] B. Widrow, P.E. Mantey, J.J. Griffiths and B.B. Goode, "Adaptive Antenna Systems", Proceedings IEEE, No. 55, 1967, pp2143-2159

[3] W.C, Lindsey and M.K. Simon, "Telecommunication Systems Engineering", Prentice-Hall, 1973

[4] R.C. Dixon, "Spread Spectrum Systems", Wiley-Interscience, 1976

[5] H.D. Davidson, "A Reliable Data Channel for Underwater Communications using Phase Shift Keying", PhD Thesis, University of Newcastle, to be published.

[6] R.A. Monzingo and T.W. Miller, "Introduction to Adaptive Arrays", Wiley-Interscience, 1980

[7] P.J. Mulholland, "Adaptive Filters and their Application to an Adaptive Receiving Array for an Underwater Acoustic Data Link", PhD Thesis, University of Newcastle, to be published.

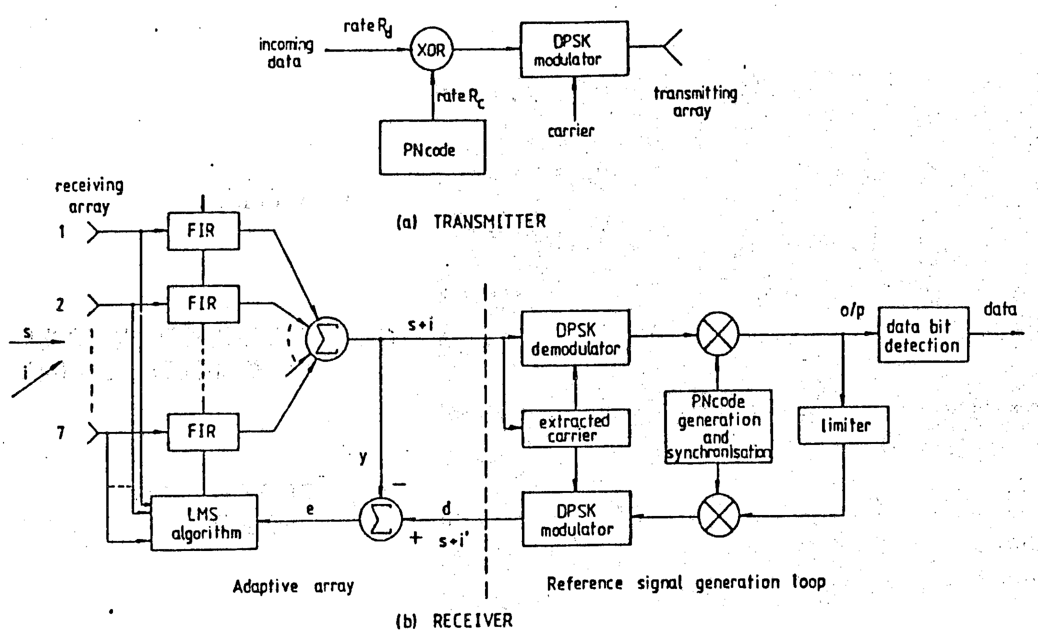


FIG.1 THE PROPOSED UNDERWATER ACOUSTIC DATA LINK

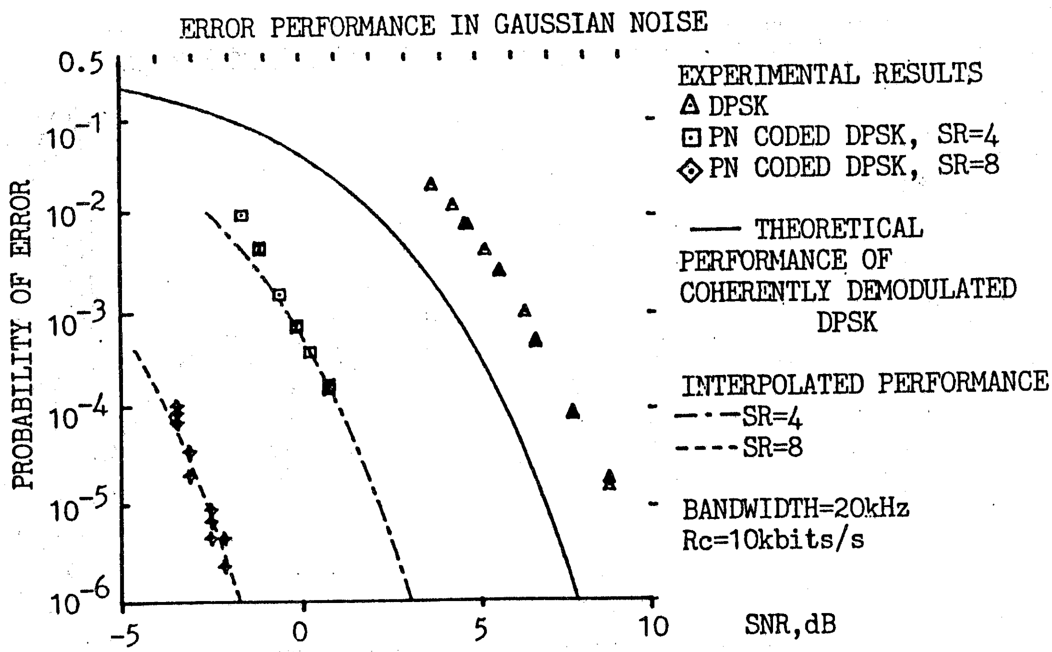
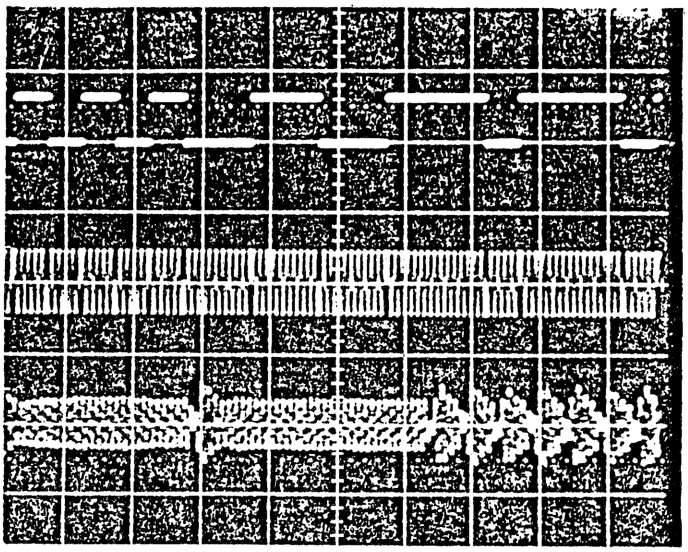


Fig. 2



(a) PN CODE

(b) TRANSMITTED PN CODED PSK MODULATED SIGNAL

(c) ADAPTIVE ARRAY OUTPUT

Fig. 3

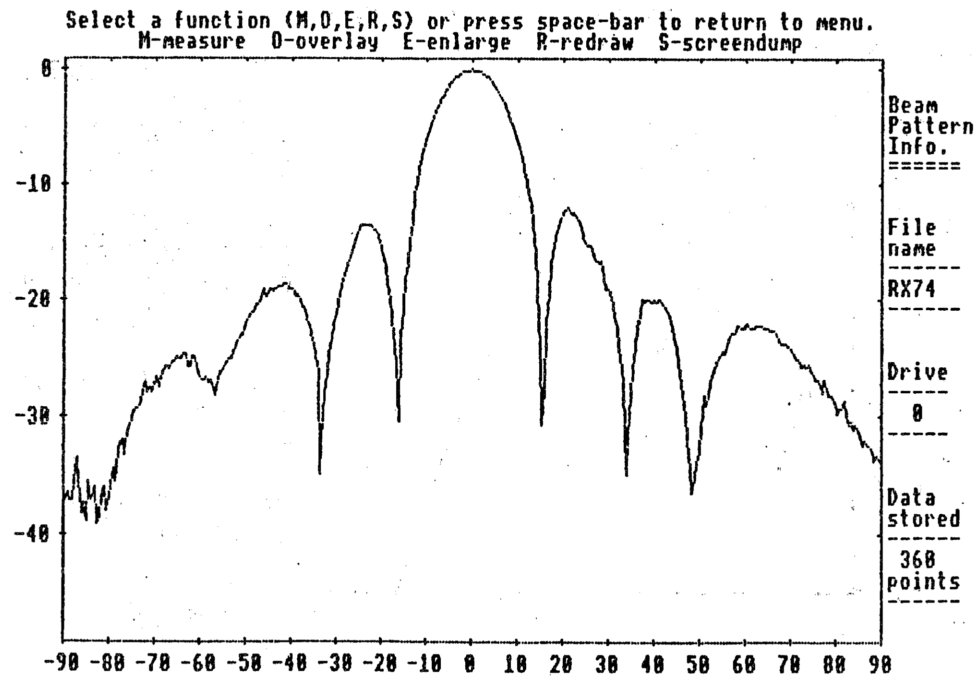


Fig. 4