Optimal Step Size and Performance Analysis of Adaptive Equalizer In Underwater Acoustic Channel

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Abstract - Ocean environment is highly characterised by spatial and temporal variation. For effective communication in the ocean, proper modelling of the channel is essential. In our paper, tap channel modelling is done and impulse response of the channel was calculated. An optimum step size for an adaptive equalizer is calculated and applied in the channel to study the effective transmission of signals and also Bit Error rate of the channel using Linear and Decision Feedback Equalizer are studied.

Keywords - Decision Feedback Equalizer, Linear Equalizer, Inter-symbol Interference, Bit Error Rate, Eigen Path, Orthogonal Frequency Division Multiplexing.

I. INTRODUCTION

The most challenging communication channel in use today [1,2] is Underwater Acoustic Channels (UAC). The acoustic signal propagates in one or more distinct curved paths between the transmitter and receiver. Each curved path (Eigen path) consists of a dominant and stable components (sub eigenpath components). The surface and bottom of an ocean causes the reflection of the signals which results in new sub-eigen paths as similar to direct path. The interaction of signals from all curved paths undergoes variation in amplitude and phase of the received signals which result in multipath propagation. In addition to these, movement of transmitter and receiver introduces Doppler and fading effects.

Basis Expansion Model (BEM) [3], [4], Radial Basis Function (RBF) model [5] and polynomial amplitude – variation channel model [6] are available to model a time varying channel. In this paper, for an effective channel estimator and equalizer a tap based channel model is used to describe the input – output relationship of the channel. UAC exhibits long delay spread and frequency selectivity [7, 8]. Orthogonal Frequency Division Multiplexing (OFDM) is a promising alternative to a single carrier system. Because of sensitivity to frequency offset that arises due to motion result in major problems non-uniform frequency shift across the signal bandwidth and Inter-carrier interference (ICI) [9, 10].

Adaptive signalling techniques have been extensively studied for radio channels [11]. For better result we have applied adaptive equalizer to study the performance of the channel. The paper is organised as follows, in second section, channel modelling and calculation of impulse response are described, the effects of adaptive equalizer in underwater channel is elaborated in section III. We then proceed to result and conclusions.

II. CHANNEL DESIGN

For proper communication in underwater, accurate modelling of the channel is very much essential, since ocean is highly characterised by spatial and temporal variabilities. Sound propagation is profoundly affected by the conditions of the surface and bottom boundaries of the ocean as well as by the vertical and horizontal distribution of sound speed within the ocean volume.

As acoustic signal propagates through the ocean, the effects of spreading and attenuation also minimize its intensity. Spreading loss includes spherical and cylindrical spreading losses in addition to focusing effects. Attenuation loss includes losses due to absorption, scattering, diffraction and leakage out of ducts.

The channel capacity depends on the distance, and may be extremely limited. Because acoustic propagation is best supported at low frequencies, although the total available bandwidth may be low, an acoustic communication system is inherently wideband in the sense that the bandwidth is not negligible with respect to its center frequency. The channel can have a sparse impulse response, where each physical path acts as a time-varying low-pass filter, and motion introduces additional Doppler spreading and shifting. Surface waves, internal turbulence, fluctuations in the sound speed, and other small-scale phenomena contribute to random signal variations.

For the simulation purpose, tap channel modelling techniques is applied and impulse responses of the channel are calculated. We have taken four channels with different condition. Channel1 is 25 kHz Sampling frequency with 0.001 Doppler Shift, channel2 with 25 kHz sampling frequency with .1 Doppler shift, channel3 and 4 are with 0.001 and 0.1 Doppler Shift with sampling frequency of 10 KHz. The magnitude of impulse response of four channels is shown in the Fig. 1. The complex gain value of these four channels is listed in Table I.

TABLE I. Complex gain value of the channel for four different condition

Channel	Complex Gain Value
I	-0.06-0.39i, -0.06+0.16i, 0.09+0.08i, 0.03-0.05i
II	-0.69+0.95i, -0.20+0.47i, 0.05+0.12i, 0.02-0.04i
III	0.71-0.46i, 0.01-0.07i, 0.1-0.04i, 0.02-0.05i, -0.002+0.004i
IV	-1.57-0.34i, -0.15+0.12i, 0.18+0.04i, 0.006+0.001i, 0.01+0.007i

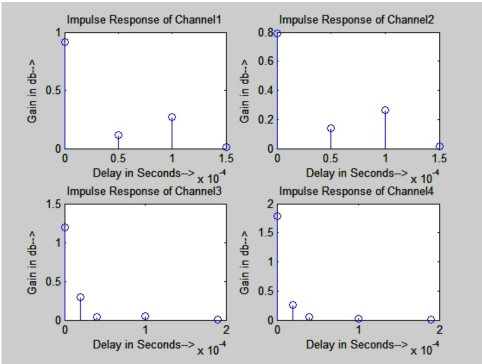

Fig. 1. Impulse response of channels

III. ADAPTIVE FILTER IN UAC

Adaptive equalizers assume channel is time varying channel and try to design equalizer filter whose filter coefficients are varying in time according to the change of channel, and try to eliminate ISI and additive noise at each time. The stochastic gradient algorithm (linear adaptive equalizer) is to minimize the mean square error between the output of the equalizer, and the transmitted signal. The iterative formulas of steepest descent method based on least mean square algorithm (LMS algorithm) are defined as follows:

$$y(n) = w(n)^T x(n) = x(n)^T w(n) ...(1)$$

Where, x (n) is the filter input; y (n) is filter output and w (n) is filter weights.

$$e(n) = d(n) - y(n) ...(2)$$

Where, d(n) is a reference signal and e(n) is the error between d(n) an y(n).

$$w(n+1) = w(n) + 2\mu e(n)x(n) ...(3)$$

Where μ is the step size.

For LMS algorithm, Convergence condition is limited to $0 < \mu < 1/\lambda_{max}$. λ_{max} is the largest eigenvalue of the autocorrelation matrix for input signal. The convergence speed is faster with greater μ causes oscillation. Smaller μ value can reduce steady state noise, improve the accuracy. Here step size is varied from 0 to 1 for difference algorithm and optimal step range is shown in Table II. It is evident from the table that the optimum range for LMS algorithms is <0.05 to that of Signed LMS is 0.05 – 0.1 and 0.02 – 1 and 0.05 – 1 for the Normalized LMS and RLS algorithms.

Table II. Optimal step size for channels: Linear Equalizer

Channel	Optimal Step size range			
	LMS	Signed LMS	Norm LMS	RLS
I	<0.2	<0.1	0.02 – 1	0.05 – 1
II	<0.28	<0.1	All range	All range
III	<0.05	<0.05	All range	All range
IV	<0.05	0.05 – 0.1	<0.1	<0.1

Fig. 2 – 5 shows the bit error rate of linear equalizer for four different channels. It is evident from the figures that, we can obtain the BER of zero if the Signal to Noise ratio is more than 12 for channel 3 and 1, whereas for the channel 2 and 4 it is more than 8.

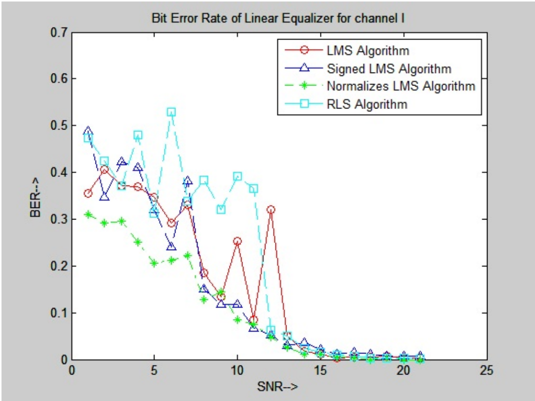

Fig. 2. Bit Error Rate of Linear Equalizer for Channel1

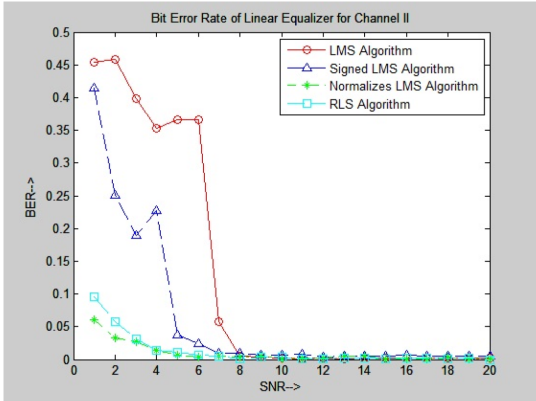

Fig. 3. Bit Error Rate of Linear Equalizer for Channel2

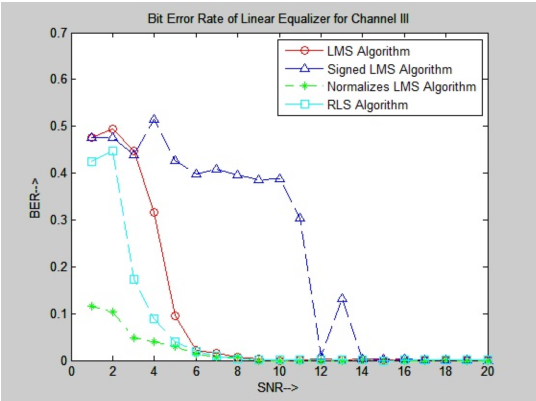

Fig. 4. Bit Error Rate of Linear Equalizer for Channel3

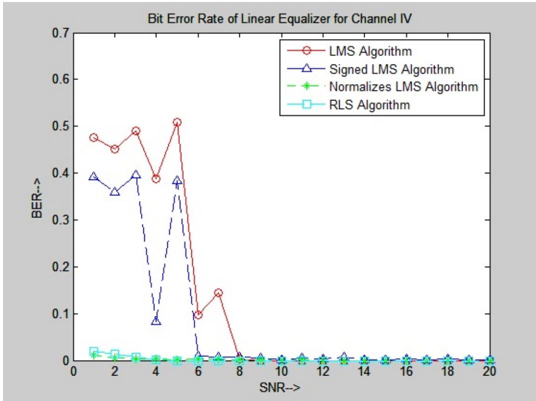

Fig. 5. Bit Error Rate of Linear Equalizer for Channel4

Decision feedback equaliser (DFE) incorporates a Feedforward filter (FFF) operating on the received signal to suppress precursor ISI, and a feedback filter (FBF) operating on previously detected symbols to suppress postcursor ISI.

A DFE uses a nonlinear decision device at the output, and the output represents a noise-free replica of the transmitted symbol assuming that the probability of decision error is small. It has been widely used in digital communications to suppress Inter-Symbol Interference (ISI) for over several decades.

Table III shows the optimal step size range DFE of the channels for different algorithms. It shows that the LMS and Signed LMS step size ranges from 0.1-0.45 and >0.1 respectively, whereas for Normalised LMS and RLS the optimal values are >0.35 and >0.05 respectively.

It is shows from the Fig. 6 to Fig. 9 that, for channel1 and 2 the bit error rate becomes zero for SNR is more than 22 and 14 respectively and it is more than 12 and 10 for channel3 and 4 respectively.

TABLE III. Optimal step size for channels: DFE

Channel	**Optimal step size range**			
	LMS	**Signed LMS**	**Norm LMS**	**RLS**
I	0.1-0.5	0.1-0.2	0.2-1	<0.3
II	<0.3	<0.2	All	all
III	<0.2	<0.2	All	all
IV	0.1-0.45	>0.1	>0.35	>0.05

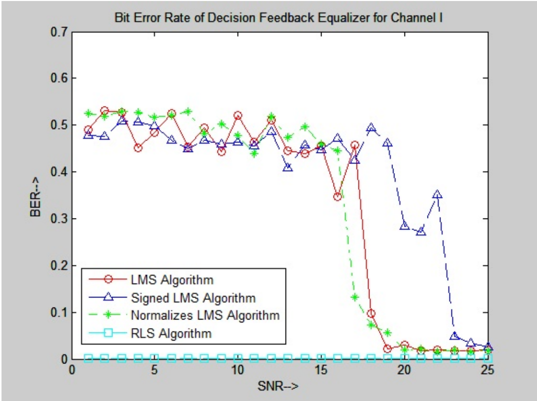

Fig. 6. Bit Error Rate of DFE for Channel1

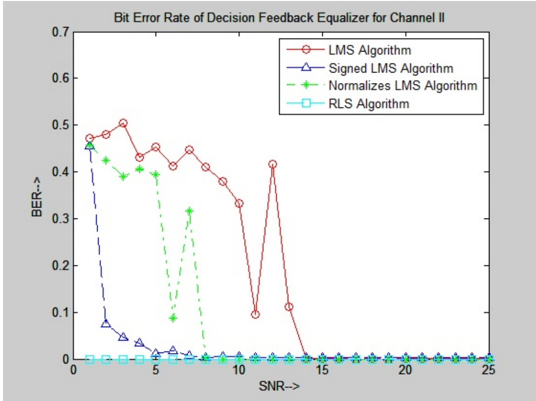

Fig. 7. Bit Error Rate of DFE for Channel2

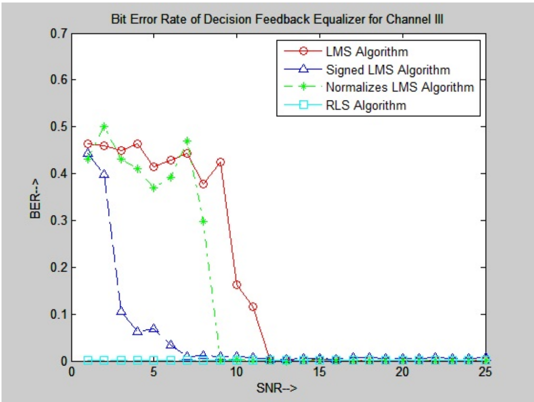

Fig. 7. Bit Error Rate of DFE for Channel3

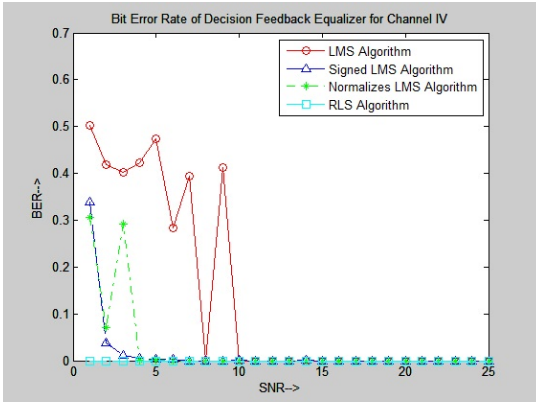

Fig. 8. Bit Error Rate of DFE for Channel4

IV. RESULT AND DISCUSSION

For Linear Equalizer, It is evident from the Table II that, channel 1 and 2 are having sampling frequency of 25 KHz for which step size range higher compared to the channel 3 and 4 whose frequency are 10 KHz. Therefore sampling frequency and step size are inversely proportional. It is clear from the Table 2 that Doppler shift and optimal step size are directly proportional and BER is almost zero for SNR of less than 12.

From Table III channel 2 and 3 is having optimal step size of less than 0.2 and channel 1 and 4 is greater than 0.2 which shows that for DFE, optimal step size is minimal, for channel having either higher Doppler shift or lower sampling frequency. Is also shows that BER is almost zero for SNR less than 24.

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