PERFORMANCE EVALUATION FOR DECISION-FEEDBACK EQUALIZER WITH PARAMETER SELECTION ON UNDERWATER ACOUSTIC COMMUNICATION

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ABSTRACT

This paper investigates the effect of parameter selection for the decision feedback equalization (DFE) on communication performance through a dispersive underwater acoustic wireless channel (UAWC). A DFE based on minimum mean-square error (MMSE-DFE) criterion has been employed in the implementation for evaluation purposes. The output from the MMSE-DFE is input to the decoder to estimate the transmitted bit sequence. The main goal of this experimental simulation is to determine the best selection, such that the reduction in the computational overload is achieved without altering the performance of the system, where the computational complexity can be reduced by selecting an equalizer with a proper length. The system performance is tested for BPSK, QPSK, 8PSK and 16QAM modulation and a simulation for the system is carried out for Proakis channel A and real underwater wireless acoustic channel estimated during SPACE08 measurements to verify the selection.

KEY WORDS

Decision feedback equalizer(DFE), minimum mean-square error (MMSE).

INTRODUCTION

In the last few decades underwater acoustic communications has become important in many application that utilize remotely controlled equipment and vehicles for sensing and exchanging information between devices located under the surface in the oceans such as military and commercial fishing and oil exploration [1]. Acoustic waves are used in underwater wireless communication instead of electromagnetic waves employed in radio frequency (RF) communication due to the rapid attenuation of electro-magnetic signals through salt water. Transmission over a frequency selective channel which is characterized by its long spread time is a challenging, especially when the channel suffers, in addition to multipath degradation, from Doppler shift which cause a rapid change in the channel impulse response (CIR). Due to the multipath, the receiver receives multiple copies of the same signal which causes the inter-symbol interference (ISI) distortion. To deal with the Doppler effect, a channel estimation within the shorter block of symbols is necessary and on the other hand, coding and equalization can help with the symbol detection and to mitigate the ISI distortion caused by the channel. Minimum mean square error decision feedback equalizer (MMSE-DFE) is a very common equalization algorithm used to alleviate the problem of the ISI. An important step in the implementation is how to select the parameters of the equalizer based on the characteristics of the channel. The channel taps are classified into postcursor (causal) and the precursor (anti-causal) parts are important characteristics needed to be estimated during design of the equalizer. Usually the equalization process implies designing an equalizer with large number of coefficients such that the amount of distortion is minimized. This introduces a higher computational complexity to the design since it involves manipulating with large matrices.

To design a finite length MMSE-DFE, the main parameters that need to be determined are the coefficients of the feedforward filter (FFF), the feedback filter (FBF) and the decision delay. The length of the feedforward filters (K = K_1+K_2+1) where K_1 and K_2 represent the precursor and the post-cursor length respectively, and the length of the feedback filter (K_3) needs to be carefully selected in addition to the decision delay Δ. Some of the previous design methods are based on ad-hoc selection while some others use exhaustive search among these three parameters. Paul *et al.* [2] investigated the effect of the decision delay on the performance of the finite length DFE. An optimum solution for the FFF and FBF and the decision delay has been proposed for RF channels, but for Underwater wireless acoustic channel this solution might be too complicated to be implemented due to the large matrix manipulation, and the result could be suboptimal. Naofal *et al.*[3] proposed a simple order recursive algorithm that reduces the exhaustive search to one dimensional, instead of three dimensional, to select the parameters for the MMSE-DFE. Although the result is promising, it is not clear how to relate this parameter selection to the channel characteristics. William J *et al.* [4] mentioned that the length of the FBF (K_3) needs to be on the order of the postcursor part of the channel and length of the FFF should be greater than the channel length and the decision delay should be longer than the channel span, but there is still an issue of how to select these parameters. Yu Gong *et al.* [5] proposed an adaptive method that optimizes the structure based on the characteristics of the channel. It shows that the decision delay should be in the range between 0 and $K_1 - 1$.

The performance will not necessarily improve by selecting a large number of taps for the feedforward filter and feedback filter while keeping the decision delay constant. This could lead to increased noise, in addition to increasing the complexity. The system performance can be improved further by incorporating channel coding where the equalized hard/soft output sequence from the HD-DFE/SD-DFE is fed to the decoder to estimate the most likely sequence. This paper seeks to study the relationship between the channel length L and the length of the equalizer need to be used. A Proakis Channel A and an estimated SPACE08 channel have been used in the evaluation to discover the best design for a single input single output (SISO) equalizer in term of removing the distortion with minimum number of computations. On average, the estimated SPACE08 channels,

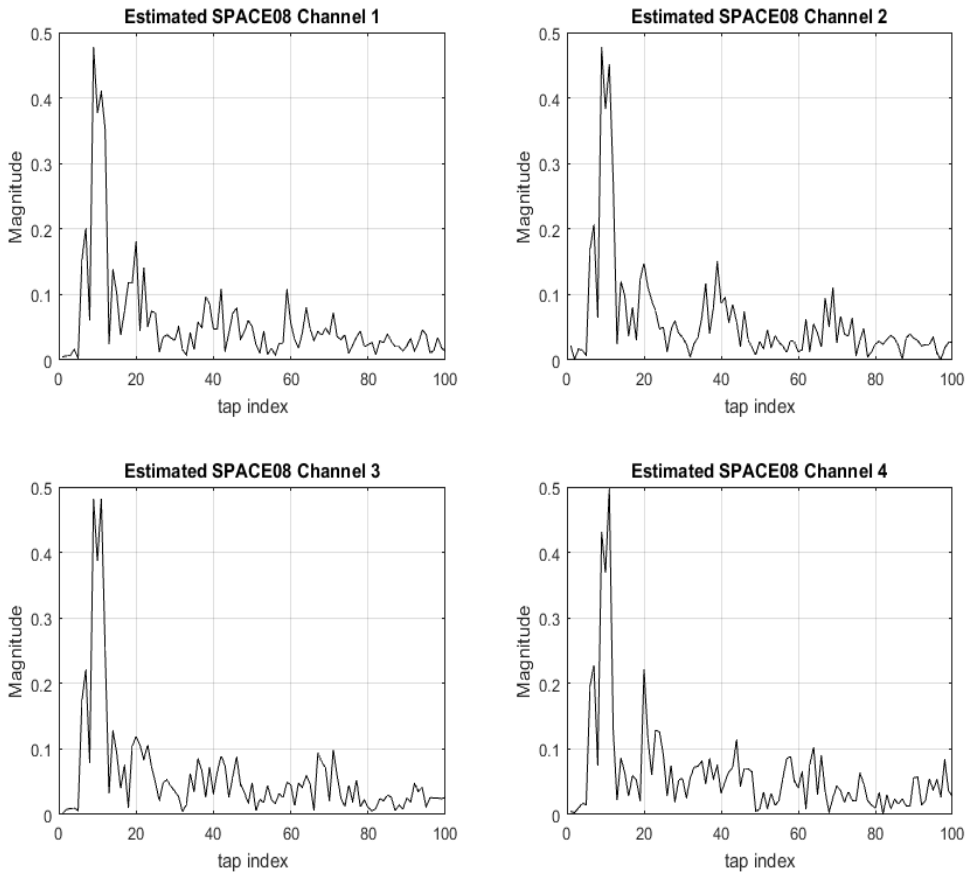

Figure 1: Tap Structure of the Magnitude of the Estimated SPACE08 Channel.

as shown in Figure 1, have around a 100 taps and none of them come from direct path (line of sight path). The length of the causal part for these channels L_2 is usually longer than the anti-causal part L_1 and in general these underwater acoustic channels are considered longer than the conventional RF channels. For these kinds of channels, minimizing the length of the equalizer is very important, especially when the channel is changing very fast, since channel estimation and equalization need to be repeated many times for each short block of the received data.

SYSTEM MODEL AND DATA FORMAT

Consider the single-input single-output (SISO) communication system in Figure 2, where the information bit stream is encoded and mapped such that the output of the encoder is a sequence of encoded bits stream $\mathbf{c} = [\mathbf{c}_1\mathbf{c}_2\cdots\mathbf{c}_\mathrm{N}]$ where $\mathbf{c}_\mathrm{k}$ represents $[c_{k,1}c_{k,2}\cdots c_{k,q}]$ with bits $c_{k,j} \in \{0,1\}$, and N is the number of transmitted codewords and q is the number of bits per symbol. The mapper uses a Gray code to map each bit vector $\mathbf{c}_\mathrm{k}$ to a symbol x_k from the 2^q array constellation set $S = \{\alpha_1\alpha_2\cdots\alpha_{2^q}\}$, where α_i corresponds to the deterministic bit pattern $\mathbf{s}_{i=}$ $[s_{k,1}s_{k,2}\cdots s_{k,q}]$ with

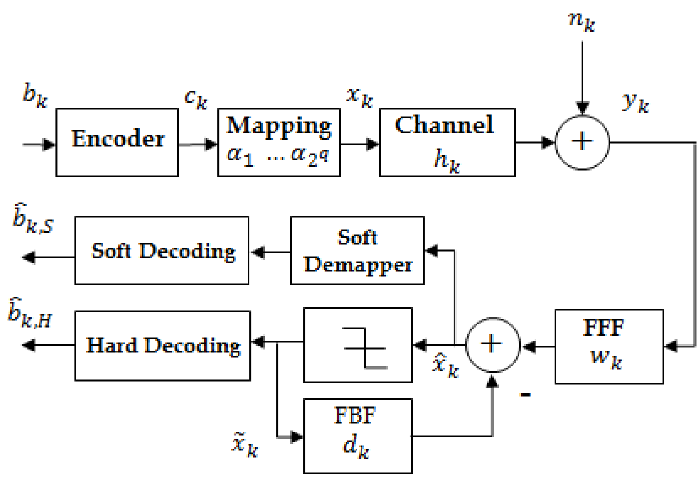

Figure 2: Simplified communication system.

$s_{i,j} \in \{0,1\}$ which specifies the mapping between the encoded bits and the elements of the constellation as shown in Table I in [6]. Next, the generated symbols are modulated with a carrier and transmitted through the channel h_k. At the receiver, the received baseband signal can be written as

$$y_k = \sum_{l=-L_2}^{l=L_1} h_l x_{k-l} + n_k \tag{1}$$

here h_l is the l^{th} tap of the channel and L_1 and L_2 are the lengths of the causal and anti-causal part of the channel respectively as shown in Figure 3, and L= L_1+L_2+1 is the total length of the channel. In addition, n_k represents an additive Gaussian white noise with zero mean and variance σ_n^2. Arranging equation (1) into a matrix form we have

$$\mathbf{y}_k = \mathbf{H}\mathbf{x}_k + \mathbf{n}_k \tag{2}$$
$$\mathbf{y}_k = [y_{k-K_1}\ y_{k-K_1+1} \cdots y_{k+K_2}]^T$$
$$\mathbf{x}_k = [x_{k-K_1-L_1}\ x_{k-K_1-L_1+1} \cdots x_{k+K_2+L2}]^T$$
$$\mathbf{n}_k = [n_{k-K_1}\ n_{k-K_1+1} \cdots n_{k+K_2}]^T$$
$$\mathbf{H} = \begin{bmatrix} h_{L_1} & \cdots & h_{-L_2} & \cdots & 0 \\ \vdots & \ddots & \ddots & \ddots & \vdots \\ 0 & \cdots & h_{L_1} & \cdots & h_{-L_2} \end{bmatrix}.$$

The bold letters represent a vectors and bold capital letters for a matrix. The variable k is used for the time index and $(\cdot)^*$, $(\cdot)^{-1}$, $(\cdot)^T$ and $(\cdot)^H$ are the conjugate, inverse, transpose and hermitian operators respectively. The signal $\mathbf{y}_k$ is input to the feedforward filter and the input of the feedback filter comes from the previously detected symbols. The output of the equalizer can be represented with the following equation.

$$\hat{x}_k = \sum_{j=-K_2}^{K_1} w_j y_{k-j} + \sum_{j=1}^{K_3} d_j \tilde{x}_{k-j}, \tag{3}$$

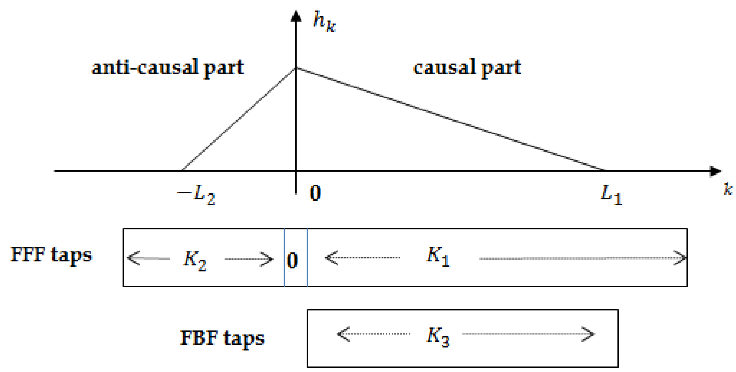

Figure 3: Parameter in relation to the channel.

where $\hat{x}_k$ and $\tilde{x}_{k-1}$ represent the estimated symbol and the previously detected symbol at time instance k respectively. $\{w_j\}$ and $\{d_j\}$ represents the the tap coefficients of the feedforward and feedback filter respectively. Assuming the detection of the previous symbol $\tilde{x}_{k-1}$was correct and equals to the previously transmitted symbol x_{k-1}, equation (3) can be rewritten at time instant k into a matrix form as:

$$\begin{aligned} \hat{x}_k &= \mathbf{w}\mathbf{y}_k + \mathbf{d}\mathbf{x}_k^d \\ \mathbf{x}_k^d &= [x_{k-K_3}\ x_{k-K_3+1} \cdots x_{k-1}]^T \\ \mathbf{w} &= [w_{K_2}\ w_{K_2-1} \cdots w_{-K_1}] \\ \mathbf{d} &= [d_{K_3}\ d_{K_3-1} \cdots d_1] \end{aligned} \tag{4}$$

where $\mathbf{x}_k^d$ is a vector contains the past K_3 decided symbols and $\mathbf{w}$ and $\mathbf{d}$ are feedforward and feedback coefficients at the time instance k respectively. Using the em orthogonality principle the values of $\mathbf{w}$ and $\mathbf{d}$ can be calculated as

$$\mathbf{w} = \mathbf{m}\mathbf{H}^h[\mathbf{H}\mathbf{H}^h + \sigma_n^2\mathbf{I}_{K\times K} - \mathbf{H}\mathbf{M}\mathbf{M}^h\mathbf{H}^h]^{-1} \tag{5}$$

$$\mathbf{d} = -\mathbf{w}\mathbf{H}\mathbf{M} \tag{6}$$

where $\mathbf{I}_{K\times K}$ represents the identity matrix of dimension K $\times$ K, and $\mathbf{m}$ and $\mathbf{M}$ are defined as

$$\mathbf{m} = [\mathbf{0}_{1\times(K_2+L-1)} \quad 1 \quad \mathbf{0}_{1\times K_1}]$$

$$\mathbf{M} = \begin{bmatrix} \mathbf{0}_{(K_2+L-K3-1)\times K_3} \\ \mathbf{I}_{K_3\times K_3} \\ \mathbf{0}_{(K_1+1)\times K_3} \end{bmatrix}.$$

The equalized symbols are now ready for decoding. For hard decision decoding (HDD), a decision based on the euclidean distance from the received symbol and the all points in the signal constellation needs to be made. The coded bits can then be determined using the regular mapping. In contrast, for soft decision decoding (SDD) soft bits are needed, and that is the job of the soft

Decoder. For BPSK, it is easy to extract the soft information, but for higher modulation level a specific technique is needed. Jianing Su *et al.*[7] provided an efficient way to extract the soft information from the equalized symbols for QPSK and 8PSK which can be generalized for 16QAM or any other constellation.

SIMULATION AND EXPERIMENTAL RESULTS

The simulation uses Proakis channel A with channel taps $h = [$ 0.04, -0.05, 0.07, -0.21, -0.5, 0.72, 0.36, 0, 0.21, 0.03, 0.07$]$ and underwater acoustic channel estimated from the SPACE08 experiment. In the simulation, the binary bits are encoded by 1/2 coding rate with generator polynomial G=[7,5]. The encoded bits then were mapped into BPSK, QPSK, 8PSK, and 16QAM and sent through the channel. As can be seen from Figure 1, the channel impulse response for the SPACE08 channel is very long and has common characteristics such as the length of the causal and anti-causal parts. During the equalization process, and for each received block of data, the lengths of equalizer parameters K_1, K_2 and K_3 have been selected within the range proportional to L_1 and L_2 of the estimated channel. Figure 4 and Figure 5 show the effect of K_1 on system performance for Proakis channel A while fixing the other parameters. As shown in Figure 4, the

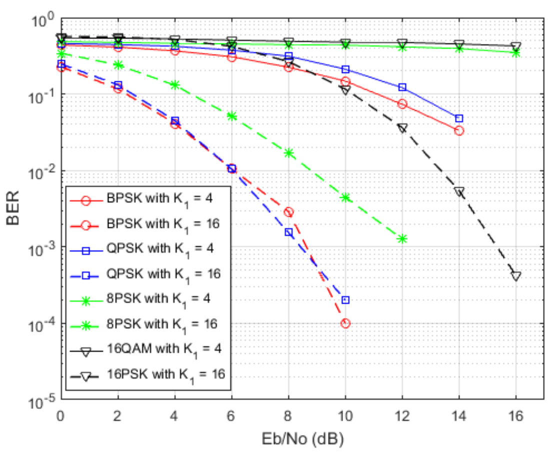

Figure 4: Impact of K_1 on system Performance for Proakis Channel A with K_2=4 and K_3=3.

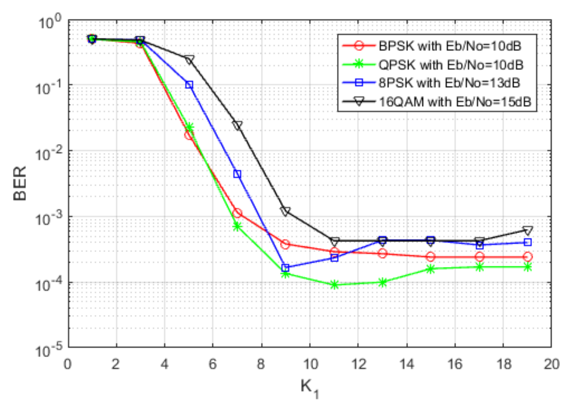

Figure 5: Changes of BER versus K_1 with K_2=4, K_3=3 for Proakis Channel A when Eb/No is fixed.

system performance can be substantially improved by proper selection of K_1. For example, at Eb/No of 16 dB the bit error rate (BER) can be reduced from around 0.4 to less than 10^{-3} by increasing K_1 from 4 to 16. On the other hand, and at 10 dB Eb/No, Figure 5 shows that increasing K_1 will not improve the system performance if it is increased beyond 10, which is about the length of the channel, where BER becomes almost constant. In such situations, increasing K_1 will add complexity to the implementation without significantly improving system performance. Moreover, it is clear form Figure 4 the amount of improvement in system performance depends on the Eb/No, the modulation order, and the values of the other two parameters K_2 and K_3 as well.

For the SPACE08 channel, and as shown in Figure 6 and Figure 7, the system performance at certain Eb/No increased by increasing K_1, but by an amount much smaller than what we had in the case of Proakis channel A at the same Eb/No. This is because of the length of the SPACE08 channel compare to the Proakis channel A. Also, the improvement of the system performance is proportional to the value of K_1 and when K_1 approaches length of the channel the performance becomes almost fixed. This is similar to Proakis channel A, where increasing K_1 will increase complexity to the implementation without any signficant improvement in performance. Figure 8

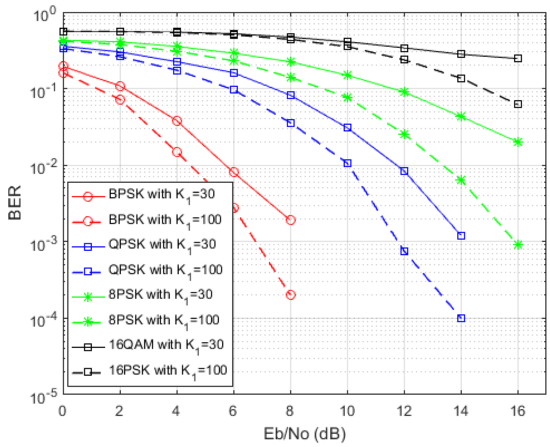

Figure 6: Impact of K_1 on system Performance for SPACE08 Channel with K_2=40 and K_3=40.

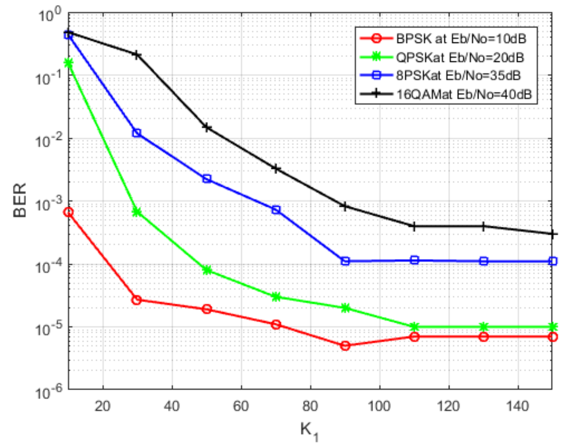

Figure 7: Changes of BER versus K_1 with K_2=30, K_3=30 for SPACE08 Channel when Eb/No is fixed.

and Figure 9 show the effect of K_2 while other parameters are fixed. As shown in Figure 8, at Eb/No less than 16 dB, changing k_2 will not provide improvement in performance for this short channel, because the job at this level of Eb/No can be done by K_1 and K_3 if they have been selected properly and this is clear also from Figure 9. For the SPACE08 channel and as shown in Figure 10

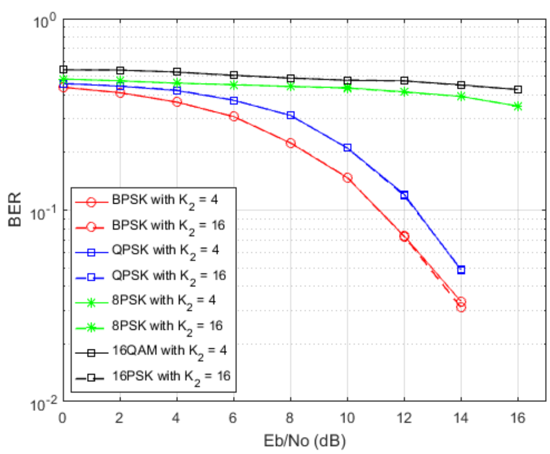

Figure 8: Impact of K_2 on system Performance for Proakis Channel A with K_1=4 and K_3=3.

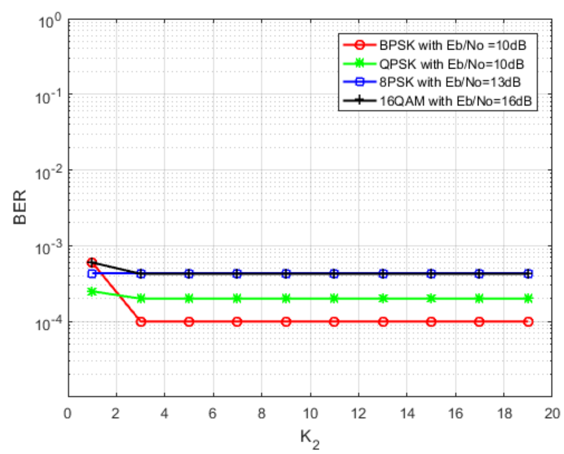

Figure 9: Changes of BER versus K_2 with K_1=4, K_3=3 for Proakis Channel A when Eb/No is fixed.

and Figure 11, K_2 still has an important effect on system performance and that is clear form Figure

10 where, for BPSK as an example, around 2 dB can be gained by increasing K_2 from 30 to 160. Such an improvement needs to be carefully evaluated because 160 in length of K_2 may add huge computational complexity. This result indicates that the selection of this parameter depends also on the length of the channel, and for such a long channel K_2 is important parameter and it could help improve the performance. The effect of the feedback section of the DFE is shown in Figure 12

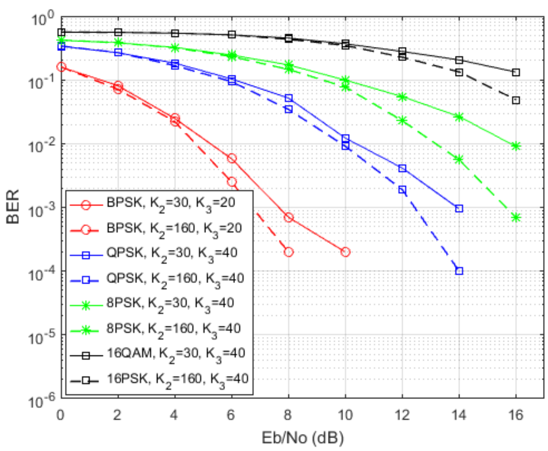

Figure 10: Impact of K_2 on system Performance for SPACE08 Channel with K_1=100.

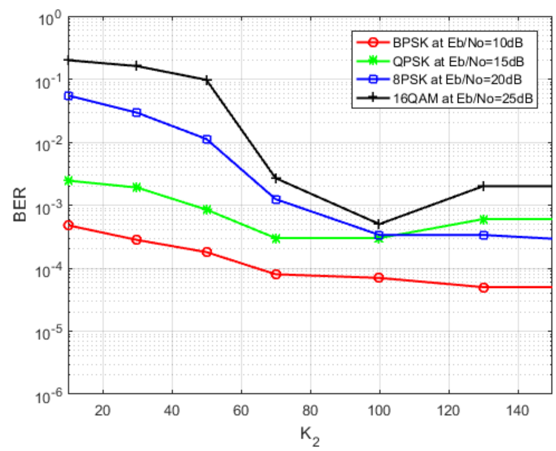

Figure 11: Changes of BER versus K_2 with K_1=50, K_3=30 for SPACE08 Channel when Eb/No is fixed.

through Figure 15 for both Proakis channel A and SPACE08 channel, where K_3 has been changed while the other parameters are held fixed. Similar to the previous results, this effect depends on the Eb/No and the values of other parameters and on the modulation order. For Proakis channel A and when K_1 is kept fixed, increasing K_3 will provide a large improvement in performance. This improvement will stop at a certain value of K_3 which is approximately the length of the causal part of the channel. In addition, the impact of changing K_3 on system performance is much larger in case of SPACE08 channel compare to Proakis channel A. This could be an another indication that for long channels, all parameters contribute to the improvement of the system when they selected properly. It is important to not increase the length of the parameter beyond the effective range, because at that moment we are adding only complexity to the system as in Figure 13 and Figure 15.

CONCLUSION

During this experimental simulation, the communication through Proakis channel A and SPSCE08 channel has been evaluated in relation to the equalizer parameter. For both channels, the equalizer parameters K_1 and K_2 are varied over a wide range around the length of channels L. The length of the feedback filter K_3 has been selected within a range around L_1 of each channel. As shown in the results, the parameter K_1 has a large impact on the system performance and it should be selected in the range of L_1 of the channel. For both channels, there will be minimal improvement in the BER when K_1 is chosen to be larger than L_1. Regarding the parameter K_2, the results showed that this parameter has a minor effect on the system performance, especially in Proakis channel A, while

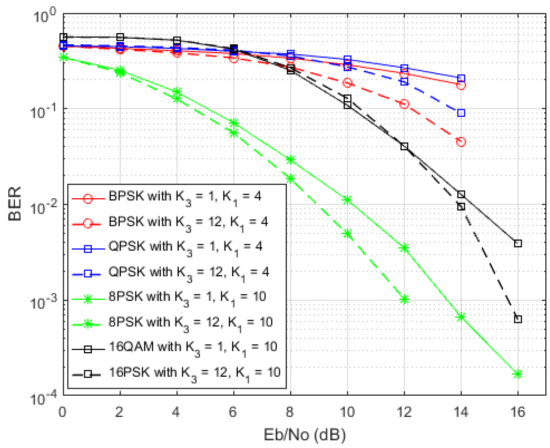

Figure 12: Impact of K_3 on system Performance for Proakis Channel A with K_2=4.

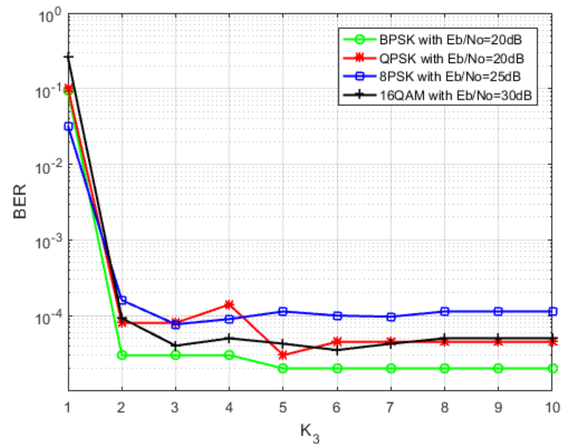

Figure 13: Changes of BER versus K_3 with K_1=4, K_2=4 for Proakis Channel A when Eb/No is fixed.

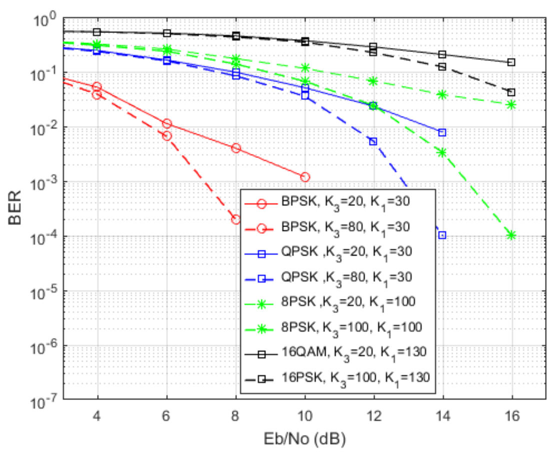

Figure 14: Impact of K_3 on system Performance for SPACE08 Channel with K_2=50.

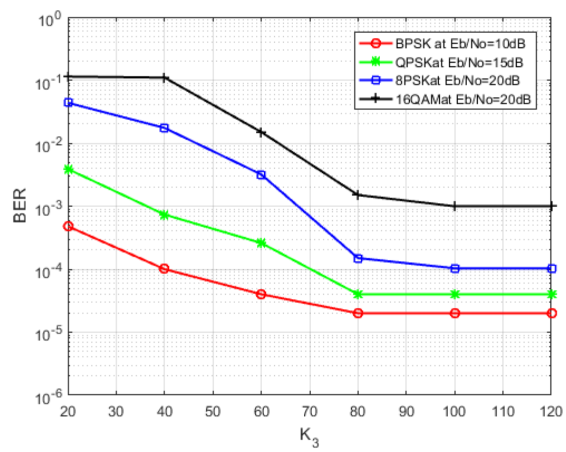

Figure 15: Changes of BER versus K_3 with K_1=40, K_3=40 for SPACE08 Channel when Eb/No is fixed.

it has limited impact in the case of the SPACE08 channel. On the other hand, K_3 is an important parameter and this makes DFE is preferable over Linear Equalizer.

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