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An Improved SIC based Turbo Equalizer

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Abstract: **--**This paper investigates a new soft interference canceller (SIC) based turbo equalizer. The SIC is configured as a decision feedback (DF) equalizer that uses two linear transversal filters. Both the filters update their taps as per the least mean square (LMS) algorithm. This receiver is different from the receivers discussed in literature in the sense that it uses the soft outputs of both the equalizer and the decoder in order to provide an improved estimate of the post cursors. It also uses the variance of the soft data bit estimates in its computation of the log-likelihood ratios (LLR). Probability distribution function for the soft data bit estimate is derived. This SIC based turbo equalizer is shown to result in a faster convergence and improved error floor at the decoder output. This study is carried out for a two typical highly frequency selective channels in order to establish the better performance of the proposed receiver.

Key-Words: - SIC, LLR, SCS, LMS, Soft Output

1 Introduction

The turbo equalizer is a base band digital receiver that carries out equalization of data transmitted over multipath time-dispersive channels and decoding of the equalized data in order to take care of additive white Gaussian noise (AWGN) in an iterative manner. The approaches to turbo equalization are broadly classified as belonging to three techniques, namely, the LMS based SIC [1], the minimum mean square error (MMSE) Wiener filter [2-3] and trellis based structures for the equalizer [4]. The second approach requires a matrix inversion of size equal to the filter taps while the third technique needs computation of certain probabilities and thus requires an exponentially large computational effort. Thus, the SIC type is considered to be more suitable than the other two approaches for implementing a turbo equalizer in this paper. In [5], an LMS filter and full decoder output were used where the decoder was of a Max-Log-MAP type. An LMS based SIC to serve as the equalizer that uses an improved estimate of the post cursors has been introduced in this paper. This may be considered to some extent, in line with the receiver proposed in [5]. A low complexity SIC has been discussed in [6] for possible application in a high data rate wireless application that uses a variable threshold technique. The issue of real time implementation of the SIC based turbo equalizer has been considered in [7]. An analysis of the SIC TEQ through BER transfer and extrinsic information transfer (EXIT) charts has been attempted in [8]. An optimization of equalization and interleaving tasks is carried out in [9]. The suitability of an interleaver for SIC based turbo equalization of frequency selective and time selective channels for higher order modulations has been investigated in [10].

However, in this paper, the soft outputs from the equalizer and the decoder sum for a nonlinear operation that produces the post cursors. The precursors are computed by applying the same nonlinear function but only to the decoder outputs.

2 Problem Formulation

The system model is a serial concatenated system (SCS) that consists of an outer forward error correcting (FEC) encoder of given parameters and an inner encoder represented by the intersymbol interference (ISI) channel. These two encoders are connected to each other by means of an interleaver of suitable type and size. The ISI channel is the discrete time equivalent of the modulation channel which consists of the transmit pulse shaping filter, the actual physical channel, the receive matched filter and the sampler. The ISI channel, hence, is represented as a finite impulse response (FIR) filter that has an equivalent finite state machine (FSM) representation.

The baud-rate sampled output of the matched filter provides a set of sufficient statistics for detection. The k-th time domain matched filter output samples are expressed as

$$z_k = \sum_{l=0}^{L-1} h_k x_{k-l} + w_k,\ k = 0,1,...N-1 \tag{1}$$

The x_k's are the interleaved encoded bits, the L channel taps are represented as $h_l, l = 0,...L-1$ and w_k is an i.i.d. AWGN sample with distribution $w_k \sim N\left(0,\sigma_w^2\right)$.The z_k s are input to the SIC that is also served by the soft outputs coming from the channel decoder. The SIC produces soft output on all N bits, where N is the total number of encoded bits and it is also the size of the interleaver. The soft output corresponding to the equalizer is computed by assuming the SIC output to be Gaussian distributed. The log likelihood ratio (LLR) for all the bits is computed from this approximation. The receivers, in general use the soft output of only of the decoder to compute the estimates.

The SIC is a combination of a linear forward filter and a DF filter. The coefficients of the forward filter are found out by optimizing the mean square error ε between x_k and its estimate $\hat{x}_k$ as

$$\varepsilon = E\left\{\left|x_k - \hat{x}_k\right|^2\right\} \tag{2}$$

where $E[.]$ is the ensemble average. In [4-5], the DF is used as interference cancellation filter. The transfer function for the forward filter corresponding to the optimum conditions is

$$P(f)_{opt} = \beta \frac{H^*(f)}{hh_0} \tag{3}$$

and the backward filter that is designed to cancel the effect of the interfering symbols has a transfer function

$$Q(f)_{opt} = \beta\left(\frac{\left|H(f)\right|^2}{hh_0} - 1\right) \tag{4}$$

This is due to the fact that, the central coefficient of the backward filter is $q_0 = 0$, so as not to cancel the desired symbol from the forward filter output. The weighting coefficient β is defined as

$$\beta = \frac{\sigma_d^2 hh_0}{\left(\sigma_d^2 hh_0 + \sigma_w^2\right)} \tag{5}$$

where σ_d^2 is the symbol energy, which for a constellation of unit energy and normalized channel taps becomes

$$\beta = \frac{1}{1+\sigma_w^2} \tag{6}$$

The SIC output at a given time instant is expressed as

$$s_k = \mathbf{P}_k^t \mathbf{z}_k - \mathbf{Q}_k^t \overline{\mathbf{x}}_k \tag{7}$$

where t denotes transpose of a vector. The forward filter coefficients are given as

$$\mathbf{P}_n = \left[p_{-L_1}(n)...p_0(n)...p_{L_1}(n)\right]^t \tag{8}$$

The boldface denotes a vector. The input to the feedforward filter comes from the matched filter outputs that is expressed as

$$\mathbf{z}_k = \left[z_{k+L_1}...z_k...z_{k-L_1}\right]^t \tag{9}$$

It is obvious that, the forward filter has a length of $2L_1+1$ with the central tap corresponding to the desired bit. Similarly, the DF filter coefficients are given as

$$\mathbf{Q}_k = \left[q_{-L_2}(k)...0...q_{L_2}(k)\right]^t$$

The input to the DF filter is a vector of soft bit estimates that is expressed as

$$\overline{\mathbf{x}}_k = \left[\overline{x}_{k+L_2}...\overline{x}_k...\overline{x}_{k-L_2}\right]^t \tag{10}$$

Thus, the length of the backward filter is $2L_2+1$.Each term in the RHS of (10) is computed by applying a nonlinear functional mapping to the extrinsic information of the decoder which becomes

$$\overline{x}_k = \tanh\left(\frac{\lambda_{dec,k}}{2}\right) \tag{11}$$

This is assumed to be an approximate value for $\hat{x}_k$ due to the $\tanh(.)$ function. The term soft comes from the fact that, each element in RHS of

(10) is obtained as a floating point number as shown by (11) instead of taking a hard decision. Here, $\lambda_{dec,k}$ is the extrinsic information produced by the soft-in soft-out (SISO) decoder for the k th bit. We note that, in literature, (11) is used for estimating the precursors as well as the postcursors, while the equalizer has updated the soft output for the previous bits corresponding to the current bit in the next iteration.

The receiver operation begins with the training of the forward filter coefficients. This is done by sending a pseudo-noise sequence (PNS). The taps of the filters are updated by the LMS algorithm as

$$\mathbf{P}_k = \mathbf{P}_{k-1} - \mu \mathbf{z}_{k-1}^* \left(s_{k-1} - \overline{x}_{k-1} \right) \tag{12}$$

and $\mathbf{Q}_n = \mathbf{Q}_{n-1} + \mu \overline{\mathbf{x}}_{n-1}^* \left(s_{n-1} - \overline{x}_{n-1} \right)$ (13)

where $0 < \mu < 1$ is the step size.

The soft output of the equalizer is generated by approximating the equalizer output as Gaussian distributed and it is expressed as

$$\lambda_{ex}\left(x_k \right) = \frac{2 s_k}{\sigma_w^2 + \sigma_{res}^2}, k = 0,1,...N-1 \tag{14}$$

where σ_{res}^2 is the variance of the residual ISI. We may note that, its value decreases as more confident reliability measures are available from the decoder and asymptotically it approaches a value of zero. It is defined as

$$\sigma_{res}^2 = \frac{1}{L-1} \sum_{i=1}^{L-1} v_i \tag{15}$$

The variance of the data estimates is

$$v_i = 1 - \left(\overline{x}_i \right)^2 \tag{16}$$

It is interesting to derive the asymptotic SIC output from (7), which for perfect interference cancellation becomes

$$s_k = \beta \left(x_k + \frac{1}{hh_0} \sum_{l=0}^{L-1} h_l^* w_{k+l} \right) \tag{17}$$

It is noted from (17) that, this SIC receiver is capable of attaining the matched filter bound. This is possible due to the improved data estimates coming from the decoder at each iteration. The process of turbo equalization yields an equivalent AWGN channel at the input of the decoder. We further note that, an identical expression has been obtained for the trellis based equalizers corresponding to this asymptotic case in [11] for a turbo equalizer. This brings out the immediate advantage of an SIC over the trellis based equalizers in the sense that, the former is computationally less demanding than the latter.

The SISO decoder accepts these $\lambda_{ex}\left(x_k \right)$ after suitable deinterleaving and computes a set of updated LLR values as

$$\Lambda_{Dec}\left(x_n \right) = \log \frac{p\left(x_n = 1 | \lambda_{ex,0} : \lambda_{ex,N-1} \right)}{p\left(x_n = 0 | \lambda_{ex,0} : \lambda_{ex,N-1} \right)}, \forall n \tag{18}$$

where $\lambda_{ex,0} : \lambda_{ex,N-1}$ denotes an $1 \times N$ dimensional vector of extrinsic information at the equalizer output. The decoder computes (18) by starting with the computation of the branch metrics. The branch metric for the decoder designed for a rate $1/n_0$ code is computed as

$$\begin{aligned} \gamma_n &= \prod\nolimits_{i=1}^{n_0} \exp\left(\lambda_{ex}\left(x_{n,i} \right) \right) x_{n,i} \\ &= \exp\left(\lambda_{ex}\left(x_{n,i} \right) \right) x_{n,i} \prod\nolimits_{j \neq i} \exp\left(\lambda_{ex}\left(x_{n,j} \right) \right) x_{n,j} \end{aligned} \tag{19}$$

It has been shown in [12] that, a soft output Viterbi algorithm (SOVA) [13] based turbo equalizer is more suitable from the implementation point of view with an acceptable performance. Hence, SOVA has been used as the SISO decoding algorithm in this work. As the SOVA decoder works in the log domain, (18) becomes equal to

$$\ln \gamma_n = \sum_{i=1}^{n_0} \lambda_{ex}\left(x_{n,i} \right) x_{n,i} = \lambda_{ex}\left(x_{n,i} \right) x_{n,i} + \sum_{j \neq i} \lambda_{ex}\left(x_{n,j} \right) x_{n,j} \tag{20}$$

The soft output that is further passed on to the SIC is the extrinsic message. This becomes, for the decoder,

$$\lambda_{dec}\left(x_n \right) = \Lambda_{Dec}\left(x_n \right) - \lambda_{ex}\left(x_n \right) \forall n \tag{21}$$

The soft data estimate as defined in (11) depends on the extrinsic information of the decoder output. The

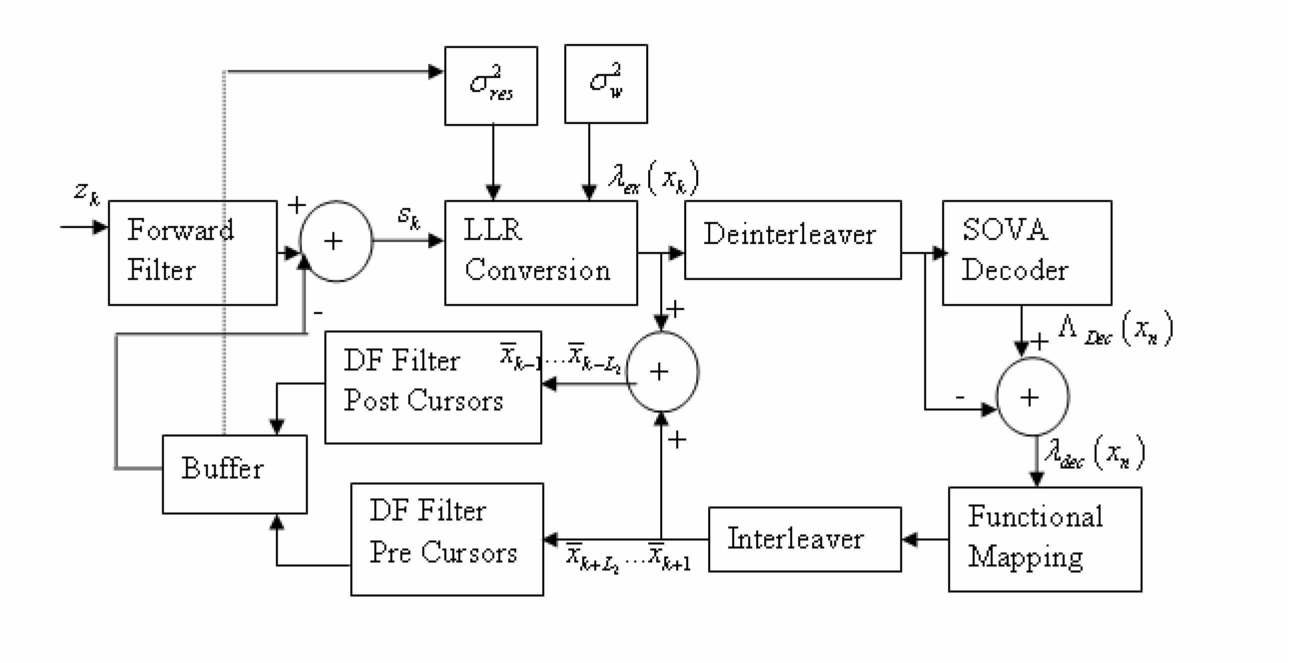


Fig.1 Block Schematic of Proposed SIC

asymptotic values of the soft bit estimates approach ± 1. It is noted from the soft bit expression that, the same expression is used for estimating the precursors as well as the postcursors.Both depend on the extrinsic information of the decoder. However, as the equalizer produces an updated extrinsic information in the next iteration, this can be used along with the decoder's extrinsic information to provide a better post cursor estimate. This is because, the main tap is considered to be the central tap of the backward filter and hence the improved post cursor estimates feed the taps of the backward filter corresponding to these postcursors. For a particular data bit, we have an improved postcursor estimate by summing the decoder and equalizer outputs.

It is noted from (11) that, if both terms in the argument of $\tanh(.)$ are of the same sign, then the soft bit estimates would approach their true binary values faster compared to when we consider only one argument for this function.

2.1.1 SIC Receiver Structure

The block schematic of the proposed receiver is illustrated in Fig.1. The equalizer produces an updated extrinsic information in the next iteration. The new post cursor estimate, namely the part of the DF filter corresponding to $\bar{x}_{n-1},\ldots\bar{x}_{n-L_2}$ for the n th bit becomes

$$\bar{x}_k = \tanh\left(\frac{\lambda_{dec,k-i}}{2} + \frac{\lambda_{ex,k-i}}{2}\right), 1 \le k-i \le k-L_2 \quad (22)$$

This makes the $\tanh(.)$ function approach ± 1 faster as compared to if only one variable is used as the argument as (11). It is noted from (22) that, if both terms in the argument of $\tanh(.)$ are of the same sign, then the soft bit estimates would approach their true binary values faster compared to when we consider only one argument for this function. If turbo equalization is converging, then the LLR produced by the equalizer is higher than that of the equalizer corresponding to a given bit and the summation as in (22) helps in achieving faster convergence. If the arguments are of opposite sign, then the particular bit would have a little contribution on the output of the DF filter and it would not affect the rate of convergence.

It may be noted that, for channels with a long delay spread, the proposed scheme produces significant improvements as it has to take care of a longer string of post cursors. We assume the decoder's extrinsic information to be Gaussian distributed defined as follows

$$y \sim N\left(m_y, \sigma_y^2\right) \tag{23}$$

where $$y = \frac{\lambda_{dec,k-i}}{2} + \frac{\lambda_{ex,k-i}}{2} \tag{24}$$

It may be noted that, this assumption has been used in literature [14]. We are interested in finding the distribution of the soft bit estimate. This is done by writing (23) as

$$\overline{x}_k = \tanh\left(y\right) = \frac{e^{2y}-1}{e^{2y}+1} \tag{25}$$

Doing a polynomial division, we obtain for (24),

$$\overline{x}_k = 1 - 2e^{-2y} + 2e^{-4y} - 2e^{-6y} + \dots \tag{26}$$

It is observed that, (26) represents a converging series and hence, by the central limit theorem, the statistics of the soft bit is observed to follow a Gaussian distribution. We note from (26) that, the higher order terms rapidly decay for higher values of y and we may approximate (26) by the first three terms. The resulting distribution is shown to have an approximated mean of

$$m_{\overline{x}_n} = 1 - e^{2m_y} + e^{12m_y} \tag{27}$$

and a variance of

$$\sigma_{\overline{x}_n}^2 = 1 + \left(e^{8m_y} - 1\right)e^{10m_y} + \left(e^{32m_y} - 1\right)e^{24m_y} + \left(e^{72m_y} - 1\right)e^{60m_y} \tag{28}$$

This has been derived by assuming the symmetry property for each exponential of (26) for the sake of convenience. For large values of y, the soft bit saturates to ± 1.

The symmetry property of the Gaussian distribution of the extrinsic information gives the SNR associated with the extrinsic information as

$$SNR_{ex,Eq} = \frac{m_{ex,Eq}^2}{\sigma_{ex,Eq}^2} = \frac{m_{ex,Eq}^2}{2m_{ex,Eq}} = \frac{m_{ex,Eq}}{2} \tag{29}$$

The mean of the Gaussian distribution is given by (27) when the soft bit is considered as (11). However, when we consider (22), we note that, if the equalizer is converging, then the two arguments of the $\tanh(.)$ are of same sign. As both are Gaussian random variables, the resulting mean is the sum of the two means and similar argument is valid for the variances. Thus, similar to (29), we express, the SNR corresponding to (24) as

$$SNR_{ex,SIC,Eq} = \frac{m_{ex,SIC,Eq}}{2} \tag{30}$$

As $m_{ex,SIC,Eq} > m_{ex,Eq}$ because the means of two independent Gaussian distributions add. This is true due to the presence of the interleaver. Thus, we define the figure of merit for the SIC receiver as SIC gain which is expressed as

$$SIC_{gain} = \frac{m_{ex,SIC,Eq}}{m_{ex,Eq}} > 1 \tag{31}$$

This shows that, the proposed receiver has a better SNR than the receiver proposed in literature and hence, the probability of average bit error is less for this receiver. This follows from the fact that, a better SNR at the equalizer output yields a better performance at the decoder output when the receiver is converging.

2.1.2 Soft Bit Estimation Error Behaviour for SIC Receiver

The soft bit formed as per (22) has been shown to be a Gaussian random variable. The average value of (22) is computed by averaging it over a Gaussian statistics with the symmetry property. For the sake of convenience, the soft outputs have been modeled as Gaussian distributed with a variance that is twice its mean. Under the assumption of all-zero transmission and the representation of zero by +1, we note that, the error in data estimate becomes

$$e = x_k - \overline{x}_k = 1 - \tanh\left(\frac{y}{2}\right) \tag{32}$$

We note that (32) is a random variable whose average is obtained by averaging it over the statistics of y. This is regarded as the error in the estimation of x_k.We, thus obtain, for the average value of this error

$$\bar{e} = 1 - \int_R \frac{1}{\sqrt{4\pi m}} \exp\left(-\frac{(y-m)^2}{4m}\right) \tanh\left(\frac{y}{2}\right) dy, m > 0 \tag{33}$$

Using a result of [15], after some algebraic manipulations, we obtain the following bounds on the average value of the error

$$\sqrt{\frac{\pi}{m}} \exp\left(-\frac{m}{4}\right)\left(1-\frac{3}{m}\right) < \bar{e} < \sqrt{\frac{\pi}{m}} \exp\left(-\frac{m}{4}\right)\left(1+\frac{1}{7m}\right), m > 0 \tag{34}$$

This gives the upper and lower bounds on the estimation error in a SIC based turbo equalizer. Next, we compare the BER of the two receiver schemes as given by (11) and (22). We have established that, the soft bit in either case is represented by a Gaussian random variable. As for a standard Gaussian random variable, the average probability of error is expressed as

$$P_e^b = 0.5 \int_{-\infty}^{0} p(y'-1)\,dy' + \int_{0}^{\infty} p(y'+1)\,dy' \tag{35}$$

where $z' = \lambda_{dec}$ for the receiver corresponding to (11).

Similarly,

$$P_e^b = 0.5\left[\int_{-\infty}^{0} p(y-1)\,dy + \int_{0}^{\infty} p(y+1)\,dy\right] \tag{36}$$

And from the property of Gaussian random variable, $p(y') < p(y)$ as $|y'| > |y|$ when the receiver is converging. Thus, the difference in (35) and (36) is a positive number that establishes that, the average probability of error of the receiver corresponding to (22) is smaller than that obtained by the use of (11).

It is noted from Fig. 2 that, the average value of the error in the data bit estimation tends to zero for increasing values of the mean of the extrinsic information. As shown by [16], this mean increases with respect to the iterations and convergence of the turbo equalizer is achieved when the ensemble average of the random component of estimation tends to zero with increase in the number of iterations. It is also noted from Fig.2 that, the estimation error attains a value approximately equal to zero when the mean of the extrinsic information approaches a value of 7.

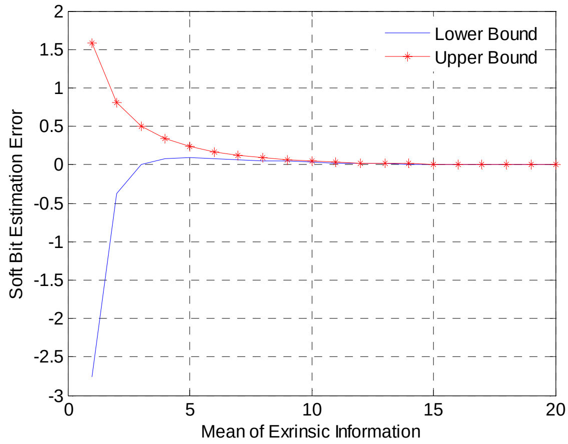
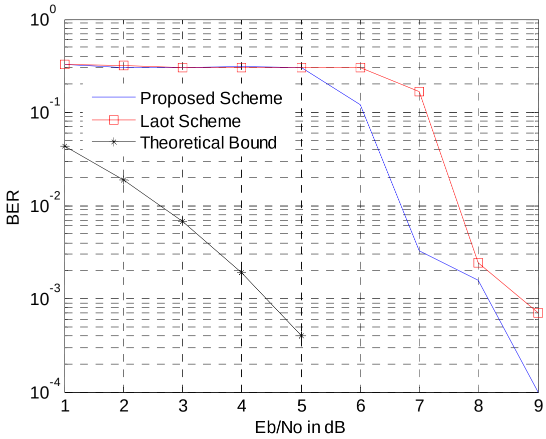


Fig.2 Analytical Bounds on the Soft Bit Estimation Error

This is true whether we consider the lower or the upper bound. It may be interpreted that, when the receiver is iterating, the mean of the extrinsic information should be approximately 7 for convergence to the true data values.

3 Problem Solution

We evaluate the performance of our SIC by considering the following simulation parameters. The outer FEC is a rate ½ recursive systematic convolutional (RSC) code with generator polynomials [7,5] for constraint length 3 and the ISI channel considered is the 5-tap Proakis-C channel expressed as [0.227 0.460 0.688 0.460 0.227]. A random interleaver of size 20000 bits has been used for each simulation run. The feedforward filter has 21 taps and initialized to zeros except the central tap being equal to 1. The feedback filter has similarly 21 taps with all the taps initialized to zero. The filters are trained by a training sequence of a 255 length pseudo noise sequence (PNS) at the beginning of receiver operation. The central tap of the feedback filter is forced to zero after every iteration of adaptation algorithm. The step size for the forwrad filter is 0.008 and its value is 0.0005 for the backward filter. This may be attributed to the fact that, the feedback filter is made insured to converge, however at a slower rate than that of the feedforward filter. It may be noted that a similar loop gain combination has been obtained in literature. The decoder is the traceback SOVA decoder that uses a traceback depth of 30 bits in order to compute the LLR on each coded bit. The result as obtained for 10 iterations is shown in Fig.3. The left most curves represent the theoretical lower bound plotted by using (16).



1: Proposed Scheme, 2: Laot Scheme 3: Theoretical Performance

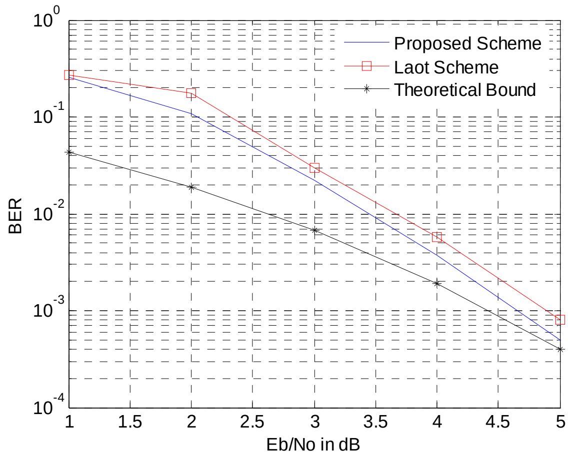
Fig.3 Performance of 21 tap SIC in Proakis- C Channel

In almost all cases, the performance gain is not significant upto 6 dB SNR. The SIC receiver performs better giving a gain of around 0.5 dB SNR from 7 to 9 dB SNR. The receiver performance is better than the reported performance by about 0.1-0.5 dB. It is observed that, the performance improvement becomes sharper for SNRs higher than 7 dB.

The performance of this receiver has been evaluated for the Proakis-B channel [0.40 0.815 0.407] also and is illustrated in Fig.4.

The proposed receiver seems to offer an SNR improvement of 0.2-0.4 dB over the reported result. It has been observed that, the receiver performance is better at 3.0 dB over the scheme reported in literature.

A comparison of Fig.3 and Fig.4 reveal that, for a channel with higher SNR loss (Proakis-C), the gap of the proposed receiver from the theoretical performance is more pronounced while the gap is about 0.5 dB for a channel with a comparatively small SNR loss.



1: Proposed Scheme, 2: Laot Scheme 3: Theoretical Performance

Fig.4 Performance of 11 tap SIC in Proakis-B Channel

4 Conclusion

A receiver structure proposed here provides improved estimates of the post cursors by summing up the extrinsic information of the equalizer at each iteration along with the decoder corresponding to the previous iteration. An analytical result on the estimation error shows the convergence of the receiver to the true data values when the mean of the extrinsic information is above a threshold.

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