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Efficient DFE Initialization for High Speed Wireless LAN

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Abstract

A very efficient procedure is introduced for computing the DFE coefficients for wireless packet type modems. The technique is especially suited for hardware VLSI implementation. After the channel impulse response is estimated using a short preamble, the Gauss-Siedel iterative algorithm is efficiently used to compute the DFE coefficients. The algorithm is derived and results shown for both for QPSK and for GMSK modulations on Indoor Multipath channels.

1. Introduction

In traditional applications, like voice-band modems, equalizers were designed to work adaptively, and data is continuous. Recently new demand has arisen for modems which operates in a bursty mode, and need equalization. The best example are the wireless LAN modems. The demand for ever increasing bit rate creates a severe multipath even in indoor applications. In such application, a receiver receives asynchronous packets which start with known pattern called preamble for start of packet acquisition and for equalizer initialization before data start. It is necessary to reduce the length of the preamble to reduce the accompanying overhead. A popular choice for the equalizer is the DFE, which has high performance in severe multipath conditions, yet has relatively low complexity (compared for example to Viterbi equalizer). Traditionally an adaptation algorithm is used for the computation of the DFE taps. Suitable algorithms are the LMS and the RLS. The RLS converges very fast, but is very complex to implement, especially when one like to use high speed hardware. The LMS on the other hand is easy to implement, but needs a long training sequence to converge, occasionally more than 500bits [1].

An alternative approach, better suitable for this application, is to estimate the channel impulse response by using appropriate preamble sequence, which has good auto-correlation properties. The sequence length may be as short as 50-100 bits. Then, the equalizer taps can be computed out of the estimated impulse response. Direct computation of DFE taps requires the solution of a set of linear equations or matrix inversion. Cioffi [2] has recognized certain structural properties in the matrices and was able to reduce the computation effort. The proposed method is still very complex, and not suitable enough for hardware implementation.

A novel algorithm is proposed for computing the feed forward filter and the feed back filter from the channel impulse response. The method is based on the Gauss-Siedel linear equation solving

algorithm. This algorithm is iterative and uses a simple to compute recursion formula. Boroujeny [3] have also proposed to use the Gauss-Siedel iterative technique for converting the channel impulse response to DFE coefficients. The result is an efficient algorithm, but still more complex than the algorithm proposed in this paper, and less naturally lends itself to hardware implementation as the proposed one. Though the derivation is different, it can be shown that the result of both algorithms is equivalent if D is the diagonal of Ω, and in this case convergence to MMSE solution is guaranteed. Simulation results show that up to 5 iterations are needed for negligible performance loss in the DFE for rms delay spread of 150ns and 50Mbps transmission. Unlike most of the work in this subject, our algorithm is also derived for the case of GMSK or OQPSK modulations.

2. The DFE Equations

Let us denote the channel impulse response by h, the feed forward filter of length N_f (FFF) by $\mathbf{w}$ and the feedback filter (FBF) of length N_b by $\mathbf{b}$. Recently Smee and Beaulieu [4] have developed a compact formulation for the equation needed to be solved to find $\mathbf{w}$ for QAM transmission for minimum mean square error (MMSE). We assume symbol-spaced equalizer, but the equations are easily extended to fractionally spaced equalizers. The equation is

$$\mathbf{w} = \Omega^{-1} \cdot \mathbf{p} \tag{1}$$

where $\Omega_{i,j} = \sum_{n=0}^{N_f-1} \mathbf{h}_{n-i}^* \mathbf{h}_{n-j} + R_v(i-j)$, $R_v(i)$ is the additive noise autocorrelation (in the simulation part of this paper we assume that the noise is white), and $\mathbf{p}_i = \mathbf{h}_{N_f-1-i}^*$. After $\mathbf{w}$ is found, $\mathbf{b}$ is computed by convoluting $\mathbf{w}$ with $\mathbf{h}$ and taking only the causal result.

2.1 Offset QPSK Case

We have derived the DFE equation for the case of offset QPSK or equivalently GMSK which is very popular in wireless modems. It is well known that OQPSK is equivalent to BPSK if there is a multiplier by j^{-n} in the receiver before the equalizer. In this case the channel seen by the BPSK modem is $\mathbf{h}_i = \mathbf{h}_i j^{-i}$. The difference between the BPSK (or PAM) case and the QAM case is that the MSE is now minimized in the real dimension only. This results in improved performance relative to equalizer that tries to needlessly minimize the imaginary MSE as well. The results obtained for BPSK is as follows. Let $\tilde{\mathbf{w}}_{2k} = \mathrm{Re}(\mathbf{w}_k)$ and $\tilde{\mathbf{w}}_{2k+1} = \mathrm{Im}(\mathbf{w}_k)$. Then

$$\tilde{\mathbf{w}} = \tilde{\Omega}^{-1} \tilde{\mathbf{p}}, \tag{2}$$

where

$$\Omega_{2i,2j} = \sum_{n=0}^{N_f-1} \mathrm{Re}(\mathbf{h}_{n-i})\,\mathrm{Re}(\mathbf{h}_{n-j}) + \tfrac{1}{2}\mathrm{Re}(R_v(i-j)), \tag{3}$$

$$\Omega_{2i,2j+1} = \sum_{n=0}^{N_f-1} -\mathrm{Re}(\mathbf{h}_{n-i})\,\mathrm{Im}(\mathbf{h}_{n-j}) - \tfrac{1}{2}\mathrm{Im}(R_v(i-j)),$$

$$\Omega_{2i+1,2j} = \sum_{n=0}^{N_f-1} -\mathrm{Im}(\mathbf{h}_{n-i})\,\mathrm{Re}(\mathbf{h}_{n-j}) + \tfrac{1}{2}\mathrm{Im}(R_v(i-j)),$$

$$\Omega_{2i+1,2j+1} = \sum_{n=0}^{N_f-1} \text{Im}(\mathbf{h}_{n-i})\,\text{Im}(\mathbf{h}_{n-j}) + \tfrac{1}{2}\text{Re}(R_v(i-j)),$$

and

$$\tilde{\mathbf{p}}_{2k} = \text{Re}(\mathbf{p}), \qquad \tilde{\mathbf{p}}_{2k+1} = \text{Im}(\mathbf{p}). \tag{4}$$

3. The New Equalizer Initialization Procedure.

One can realize that the main difficulty in performing the DFE initialization is the solution of the linear equation. We have successfully implemented the Gauss-Siedel iterative technique for solving this equation. The procedure is described by

$$w_r := \left(D^{-1} \cdot \left(p - \left(\Omega - D \right) \cdot w \right) \right)_r \tag{5}$$

where D is a matrix containing the diagonal of Ω, and $:=$ denotes assignment to a vector. The index r goes through the values $0 \dots N_f - 1$ in arbitrary order once in each iteration. It is easy to show that at each operation of (5), $\mathbf{w}_r$ is set to the value which minimizes the quadratic form $\mathbf{w}^+\Omega\mathbf{w} - 2\,\text{Re}\{\mathbf{w}^+\mathbf{p}\} + 1$ (complex case) which is the expression for the MSE as a function of $\mathbf{w}$ (assuming that $\mathbf{b}$ is also a function of $\mathbf{w}$, by convolution with $\mathbf{h}$). Hence, the algorithm monotonically converges to minimum MSE, a very desirable property.

At init $\mathbf{w}$ is set arbitrarily to all zeros. The implementation of the algorithm is very similar to matrix-vector product plus divisions by the diagonal elements of D. The matrix-vector product hardware is essentially the same needed to perform the feed-forward filter of the DFE, which is not needed during this initialization. The matrix Ω is also easy to compute by the same hardware. Thus, the hardware expense of this approach is minimal. The following modifications makes it even more attractive for implementation.

Claim: D can be chosen arbitrarily instead of being the diagonal of Ω.

Proof: The equation

$$w = \Omega^{-1} \cdot p \tag{6}$$

after change of variables can be modified to

$$M \cdot w = \left(\Omega \cdot M^{-1} \right)^{-1} \cdot p \tag{7}$$

where M is an arbitrary matrix. Then the iteration equation becomes

$$w_r := \left(M^{-1} \cdot D^{-1} \cdot \left(p - \left(\Omega \cdot M^{-1} - D \right) \cdot \left(M \cdot w \right) \right) \right)_r \tag{8}$$

or after simplification

$$w_r := \left(\left(D \cdot M \right)^{-1} \cdot \left(p - \left(\Omega - D \cdot M \right) \cdot w \right) \right)_r \tag{9}$$

As a result, we can write the equation as in the original form, and choose arbitrary D (instead of the arbitrary matrix DM). For simplicity D stays diagonal.

Note that the recursion equation requires one division per iteration. In hardware implementation it is desired to avoid division. Let us use the above claim for a modification in D to simplify the hardware. Let us define the elements of D as

$$\mathbf{D}_{i,i} = 2^{\text{round}(\log_2(\Omega_{i,i}))} \tag{10}$$

Then, the required division operation is only a shift by a barrel shifter. The speed of convergence is dependent on the eigenvalues spread of Ω. When the SNR is high the spread is increased and convergence is difficult. We found out a very effective way to speed up the convergence time of the algorithm. During the initial round we add certain values to the diagonal of Ω. This is equivalent to a change in the SNR to a lower SNR. Thus, during these rounds the algorithm converges toward the DFE solution of lower SNR than the actual one. In later iterations the modifier is removed gradually, and the last iterations converge toward the solution for the correct SNR. The modifier values has been determined experimentally. To summarize, the equalizer init is composed from the following tasks:

1. Correlate the received sync sequence with its template to obtain the channel impulse response **h**.
2. Calculate the DFE correlation matrix Ω.
3. Solve the linear equation for the **w** using the Gauss-Siedel matrix inversion algorithm.
4. compute the feed-back filter from the feed-forward filter by convolution of w with **h**.

4. <u>Simulation on Indoor Environment</u>.

We have used the popular exponential power-delay profile with ensemble-average rms delay spread of 150ns which is found in tough Indoor environment [5]. The data rate was 25Mbps, each tap in the channel was Rayleigh distributed and the taps were spaced evenly at 5ns spacing. We have simulated two cases. QPSK with transmitted pulse shape was square root raised cosine with roll off factor 0.4, and GMSK with BT=0.3. In both cases the receiver front end was square root raised cosine with roll off factor 0.4. In the QPSK case this front end filter is matched to the transmitter pulse shape and the equalizer was symbol spaced, i.e. the input is sampled at 12.5MHz. In the GMSK case, the sampling rate was the same as the bit rate, i.e. 25MHz, and the receiver front end is used as an anti-aliasing filter, producing white noise samples. Note that this filter is wide enough to pass the GMSK signal almost undistorted. When representing the GMSK system as BPSK, the resulting equalizer is symbol spaced, but due to the narrow signal this case behave like fractionally spaced equalizer, and this explains the lower error rate relative to the QPSK case. The results are shown in Figures 1 and 2, were $N_f = 10$, $N_b = 23$. The bit error rate was computed analytically with the assumption of correct decisions fed back. Actual bit error rate is few times higher. One can observe that the convergence speed depends on the SNR, but for useful values of average bit error rate a negligible degradation is obtained after only 6 iterations, i.e. $24N_f^2$ real multiplications.

Conclusions

A very efficient procedure is introduced for computing the DFE coefficients for wireless packet type modems. The technique is especially suited for hardware VLSI implementation. Moreover a large part of the hardware is shared with the existing feed forward hardware. Very fast initialization is obtained, and combined with the relatively short preamble needed for channel estimation result in low overhead even for short packets.

Acknowledgements

Bibliography

[1] J. Tellado-Mourelo, E. K. Wesel, J. M. Cioffi, "Adaptive DFE for GMSK in Indoor Radio Channels", IEEE J. of Sel. Area, vol. 14, no. 3, pp.492-501, April 1996.

[2] I. Lee and J.M. Cioffi, "A fast computation algorithm for the decision feedback equalizer", IEEE Trans. Comm., vol. 43, no. 11, pp. 2742-2749, Nov. 1995.

[3] B. F. Boroujeny "Channel Equalization via Channel Identification: Algorithms and Simulation Results for Rapidly Fading HF Channels", IEEE Trans. Comm., vol. 44, no. 11, Nov. 1996.

[4] J.E. Smee and N. C. Beaulieu, "On the Equivalence of the Simultaneous and Separate MMSE Optimizations of a DFE FFF and FBF", IEEE Trans. on Comm., vol. 45, no. 2,pp. 156-158, Feb. 1997.

[5] C. Tellambura, Y.J. Guo, S.K. Barton, "Equaliser Performance for HIPERLAN in Indoor Channels", Wireless Personal Comm., 3, 397-410, 1996.
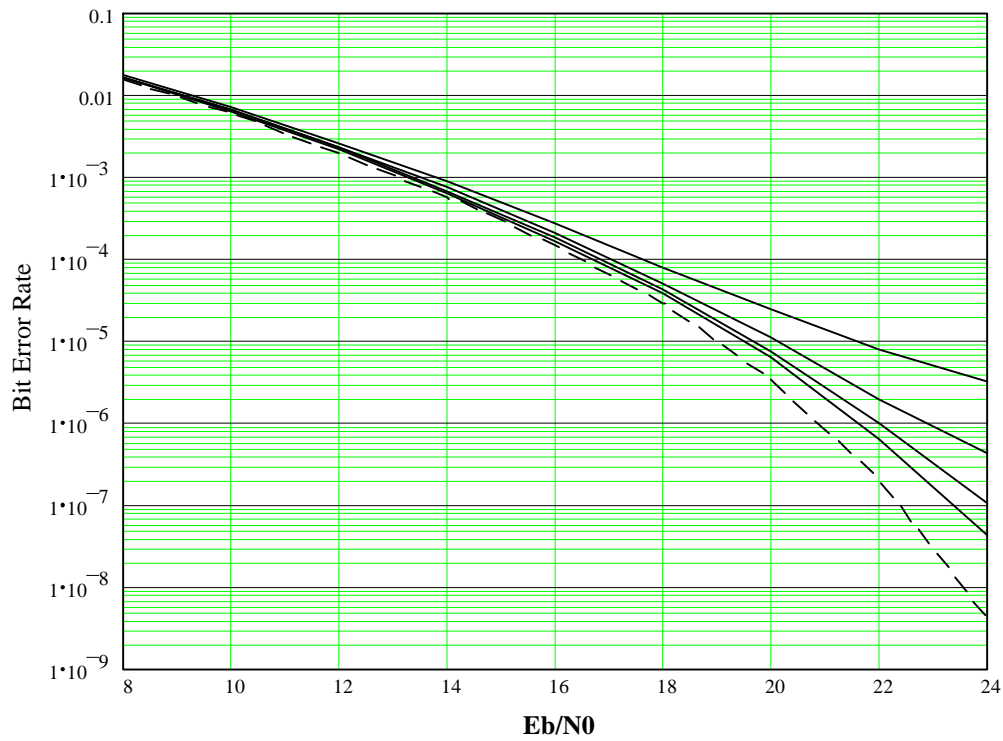
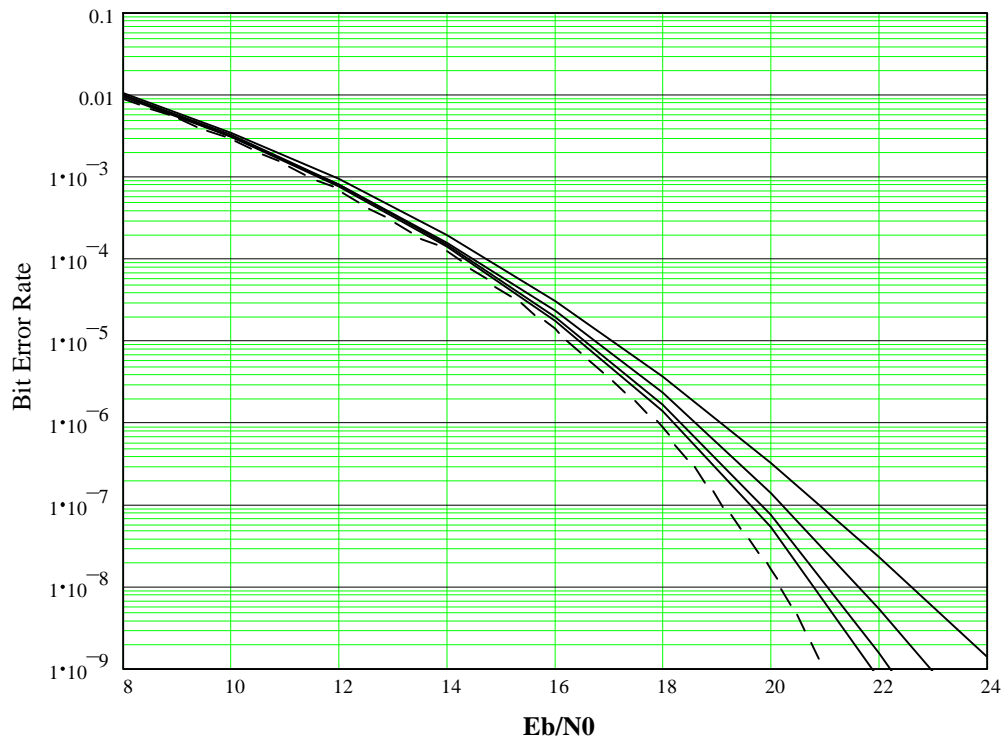Fig. 1: Results for QPSK system for 3,4,5,6 iterations (smooth line) and the optimal solution (dashed line).

Fig. 2: Results for GMSK system for 3,4,5,6 iterations (smooth line) and the optimal solution (dashed line).