AN ADAPTIVE FRACTIONAL T/2 EQUALIZER FOR HIGH-EFFICIENT TELEPHONE CHANNEL DATA MODEM

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ABSTRACT - This paper presents the T/2 fractional passband equalizer we developed for modem following V.32 and V.33 CCITT recommendations. The BLOCK LMS algorithm controls all adaptive equalizer parameters. This algorithm attains the same performance as a standard LMS does, but it is computationally less complex thanks to a more efficient use of signal processor capabilities. In addition, the important feature of this equalizer is the adaptive carrier phase tracking system designed to suppress phase jitter.

1 Introduction

The adaptive passband LMS equalizer shown in Fig.1 is very spread in high-speed voice-band modems because of a good performance and its computational simplicity. It is a fractional equalizer with a tap spacing less than symbol interval T and with a decision-directed phase lock loop for carrier phase tracking of two-dimensional modulated signals [1]. The adaptive equalizer presented in this paper is based on the structure depicted in Fig. 1, where the BLOCK LMS algorithm and the adaptive phase lock loop (APLL) are used instead of a standard LMS algorithm and the second-order phase lock loop with fixed coefficients (PLL).

The BLOCK LMS algorithm has the same convergence characteristics as a standard LMS algorithm, but it is computationally less complex. Namely, as it is known, the original block LMS algorithm realized in a frequency domain uses the computational efficiency of the fast Fourier transform (FFT) [2]. On the other side, our BLOCK LMS algorithm is hardware oriented and it is realized in a time domain using a signal processor instruction for a fast multiplication with accumulation.

The PLL with fixed coefficients suffer from a high level residual phase jitter that results in a noise tolerance reduction in high-density signal constellations (32QAM, 64QAM, 128QAM). That was our motivation to develop an adaptive phase lock loop which efficiently compensates phase impairments. The APLL is realized as a cascade of the PLL and the adaptive phase jitter predictor [3].

In Section 2 the relation between our BLOCK LMS equalizer realized in a time domain and the original block LMS adaptive filter [2] based on the fast Fourier transform (FFT) is discussed. The structure and the algorithm of the APLL are introduced in Section 3. In Section 4 results generated by the program simulator are shown. They provide parallel convergence behaviors of LMS and BLOCK LMS equalizers for different levels of linearly distorted telephone channels and illustrate efficiency of the APLL in the presence of a power-line harmonics phase jitter. Conclusions are given in Section 5.

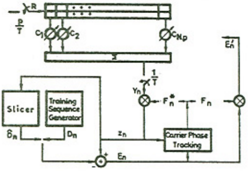


Fig.1. Adaptive linear equalizer structure

2 The BLOCK LMS algorithm

The block LMS adaptive filter [2] executes a serial-to-parallel conversion of input signal samples in blocks of length L and then performs all calculations in a frequency domain. In the case of a finite impulse response digital filter of order N-1 with complex-valued input samples the block LMS algorithm can be written as

$$\underline{Y}_j = \underline{R}_j \underline{C}_j \qquad (1.a)$$

$$\underline{E}_j = \underline{Y}_j - \underline{D}_j \qquad (1.b)$$

$$\underline{C}_{j+1} = \underline{C}_j - \beta_L/L \; \underline{R}_j^h \underline{E}_j$$

$$= \underline{C}_j - \beta_L/L \sum_{k=(j-1)L+1}^{jL} \underline{E}_k \underline{R}_k^* \qquad (1.c)$$

where j is a block index, $\underline{Y}_j$, $\underline{D}_j$ and $\underline{E}_j$ are, respectively, the filter output, the desired output and the output error, all vektors length L. $\underline{R}_j$ is the LxN matrix of input vectors $\underline{R}_k$, $\underline{C}_j$ is the Nx1 weight vector and β_L is the convergence constant. In (1) h and asterisk indicate Hermitian transformation and a complex-conjugate value. For this block filter it is important to note that an adaptive algorithm must allow a whole block of outputs to be calculated without modifying the filter parameters. It means the weights are adjusted once per block of data that is opposite to the standard filter which updates parameters once per data sample. The block data technique just described makes sense for long filter lengths (N is of the order of hundred), where the use of the FFT can speed up calculation of the convolution $\underline{R}_j\underline{C}_j$ and the correlation $\underline{R}^h_j\underline{E}_j$.

The block method in a frequency domain is not practical for voiceband equalizers whose length commonly does not exceed 32T in symbol intervals. In that case a time domain realization is much more convenient. This algorithm, we have denoted as BLOCK LMS, can gain a computational complexity advantage compared to the standard LMS algorithm when it is implemented on signal processors such as TMS320 family. The BLOCK LMS algorithm is derived directly from (1.c). The equation (1.c) for the k-th weight is

$$C^k_{j+1} = C^k_j - \beta_L/L \sum_{i=(j-1)L+1}^{jL} E_i R^*_{i-k+1} \quad k=1,.N \quad (2)$$

If we introduce the index n that indicates a time in symbol intervals, instead of the block index j, (2) can be written as

$$C^k_{n+L} = C^k_n - \beta_L/L \sum_{m=1}^{L} E_{n+L-m} R^*_{n+L-m-k+1} \quad (3)$$

The above equation is the BLOCK LMS algorithm for T equalizer (p=1, Fig.1.). As we can see the k-th weight is adjusted once per block samples of length L and the weight updating term is an average of L corresponding LMS terms. Generally, a block length is not restricted but the case L=N is probably the most convenient [2]. When L equals one the BLOCK LMS algorithm becomes the standard LMS algorithm. The BLOCK LMS algorithm for the T/2 equalizer implemented in our modem (p=2, Fig. 1) is given with

$$C^k_{n+L} = (1-\gamma\beta_L)C^k_n - \beta_L/L \sum_{m=1}^{L} E'_{n+L-m} R^*_{2(n+L-m)-k+1} \quad k=1,..2N \quad (4)$$

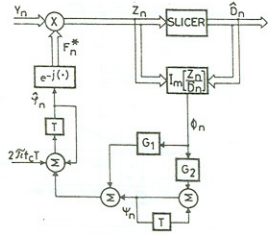
where E'_n is the error E_n translated in the passband of the received signal. The term $\gamma\beta_s C^k_n$, that emerges in (4), belongs to an algorithm for the stable operation of a digitally implemented fractional equalizer where γ is a small positive constant [4].

The equalizer has 2N coefficients T/2 spaced apart. The equalizer is fed in by two samples per T and each weight is adjusted once in every L output samples. The calculation of the updating term is implemented using the instruction MAC (MACD) [5] which indicates multiplication with accumulation.

3 The adaptive phase lock loop

The conventional PLL, which operates as a low-pass filter shows a good performance with respect to phase and frequency offset, Fig. 2a. Its coefficients can be optimized so that for the chosen bandwidth PLL yields a shortest phase error convergence time [6]. On the other side, the PLL is not an optimal solution for carrier phase tracking in the presence of a phase jitter. As it is known, the PLL achieves a better phase jitter suppression for the wider-band and noise effects are less for the narrow-band PLL [1,6]. Generally, coefficients of the PLL must be chosen to provide a compromise between phase jitter compression and noise enhancement.

Taking into account the above discussion and the fact that for high-efficient modems (5-7 bit/s/Hz) the carrier phase tracking must be reliable and precise, it is clear that just adaptive systems can synthesize phase jitter signals with a minimum noise enhancement. The APLL we have developed includes good features of conventional PLL and adaptive predictor. It is the cascade combination of the second-order decision-directed phase lock loop and the BLOCK LMS phase jitter predictor, Fig 2b. This APLL has emerged after reduction of redundant operations in the given cascade [7]. As can be seen, the APLL acts as the PLL when the predictor is turned-off. In fact, this corresponds to real conditions when a transmission path does not introduce a phase jitter so that predictor weights wander roundabout a zero value.



(a) Conventional second-order PLL

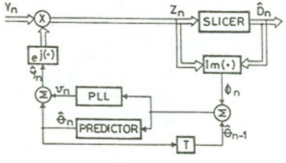


Fig.2. Decision-directed phase lock loop: (b) Adaptive PLL

4 Simulation results

The goals of simulations presented in this paper are: fist, to check the relation between convergence behaviors of LMS and BLOCK LMS equalizers and, second, to illustrate efficiency of the APLL with respect to phase jitter. The program simulator performs the main functions of the modem V.32 and synthesizes telephone channels shown in Fig 3. The input data, coming at the rate 9600 bit/s, are mapped into the 32QAM signal constellation and transmitted at the 2400 Hz symbol rate. The telephone channel introduces the white Gaussian noise and phase impairments. The first simulation tests the LMS equalizer with 64 weights for two different channels with signal-noise ratio 35dB. The mean square error is averaged over ten independent runs for $\beta_1 = 2^{-12}$. The same simulation is repeated for the BLOCK LMS equalizer. Its 64 weights are separated into two blocks 32 each and adjusted in the block manner where L=32. In this special case one period of weights adjustment is 32T long. The adaptation constant $\beta_L = 2^{-7}$ is chosento satisfy relation

$$\beta_L = L\ \beta_1 \qquad (5)$$

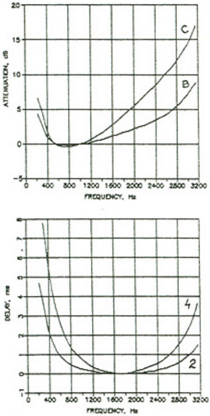


Fig.3. Amplitude and group-delay characteristics

Figure 4 shows that corresponding behaviors for LMS and BLOCK LMS equalizers are equal when the relation (5) holds.

The carrier phase tracking system is examined for ideal channel with SNR=35 dB and multi-harmonics phase jitter which has components of 50, 100 an 150 Hz and peak-to-peak deviations of 12, 8 and 4 deg respectively. Figure 5a shows the signal constellation at the equalizer output for the second-order PLL with bandwidth of 100 Hz. The output is sampled in steady-state operation and the time slot of 500 symbol intervals. We can see that constellation points suffer from a large residual phase jitter. The phase error power spectral density of the PLL is depicted in Fig. 5b. The same test is repeated for the APLL having widthband 20 Hz and 48 BLOCK LMS adaptive weights. An adaptation of weights starts after PLL learning. The APLL operation is illustrated in Fig. 6 which shows a phase jitter is entirely suppressed.

5 Conclusion

The BLOCK LMS algorithm has the same performance as the standard LMS algorithm and it is very practical for signal processor implementation. Using the BLOCK LMS algorithm for an adaptive equalization we succeeded in developing the receiver of the modem V.32, 32 with one processor TMS320C25. In addition, the important feature of the equalizer is an adaptive phase lock loop. The APLL includes good features of the conventional second-order PLL and the adaptive predictor of a phase jitter.

References:

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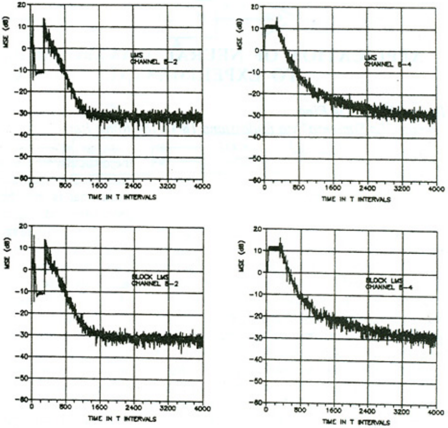
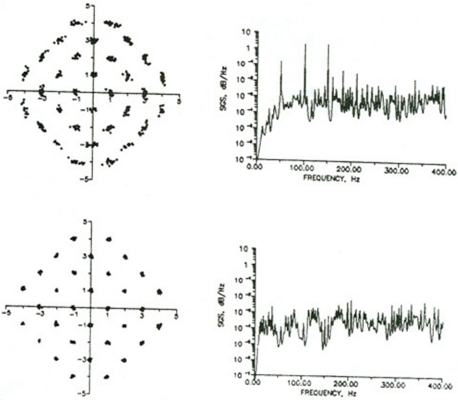
Fig.4. Comparison of the convergence behaviors of LMS and BLOCK LMS equalizers: (a) LMS



(b) BLOCK LMS

Fig.5. Signal constellation at the equalizer output and phase error power spectral density:
(a) PLL



(b) APLL