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THE PERFORMANCE ANALYSIS OF THE ADAPTIVE TWO–STAGE DFB NARROWBAND INTERFERENCE SUPPRESSION FILTER IN DSSS RECEIVER

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In this paper the performance of the direct sequence spread spectrum (DSSS) receiver containing the adaptive two stage decision feedback (DFB) filter in the presence of the nonstationary narrowband autoregressive interference is presented. The adaptation algorithm considered is Widrow–Hoff least-mean square (LMS). The receiver performance is expressed by the Signal/(NBI+Noise) ratio and the bit error rate. The results obtained show that the overall good receiver performance is not degraded by the variation of the filter coefficients caused by the adaptation process in the presence of the nonstationary electromagnetic environment.

1. INTRODUCTION

The frequency band used by the DSSS communication system is often occupied by one or more strong narrowband signals. These narrowband interference (NBI) signals, originating from the conventional communication systems or being purposely generated as jamming signals, are often so strong and nonstationary that even the inherent DSSS processing gain cannot provide the acceptable transmission quality. In that case some additional means of an adaptive NBI suppression has to be implemented, [2, 3].

The process of NBI suppression using transversal filters is based on the virtual absence of correlation between the desired signal chips. This means that by using such filter, the desired signal cannot be suppressed. However, there is the high correlation between the adjacent NBI signal samples at the transversal filter taps. Due to this it is possible to form the reference signal that can be subtracted from the input signal, thus reducing the interference level substantially.

The basic problem in this method of the NBI suppression is how to extract the reference signal that contains only NBI from the input signal. One possible idea is to use the DFB filter as the desired reference signal source, and then to perform the NBI suppression in the two-sided transversal filter. In paper [2], the DSSS receiver containing this particular filter combination, named DFB+2-TF is presented and its performance in the presence of the single tone interference is analyzed. In paper [3] the analysis of DFB+2-TF receiver performance is significantly extended to the more realistic model of the narrowband autoregressive (AR) interference at the receiver input, and the comparative analysis of the proposed receiver and the receivers containing the decision feedback (DFB) or two-sided transversal filter (2-TF), described in [5], is made.

In this paper the results of the DFB+2-TF adaptation process in the presence of the nonstationary electromagnetic (EM) environment is given. This environment is modeled as the second- and fourth-order AR process, [6], that models the cochannel and the adjacent channel interference, respectively. The DFB+2-TF coefficients are adapted to this EM environment by the Widrow–Hoff least mean square (LMS) algorithm, [4]. The results obtained show that the adaptive DFB+2-TF receiver can respond to the abrupt change in the nonstationary EM environment, and can achieve the performance comparable to theoretical values for stationary EM environment with the appropriate choice of the LMS adaptation parameter μ.

There is a number of different adaptation algorithms that can be efficiently used for the filter adaptation instead of LMS, namely Recursive Least-Squares (RLS) or Kalman filter, among the others. The authors, however, have adopted LMS for its well-known simplicity and widespread popularity. It is known that LMS has the drawback of its slow convergence, but this slowness is compensated by the fact that the algorithm consists of several simple calculations. In the situation where the signal processing has to be done fast, since the available processing time is determined by the pseudonoise sequence (PN) chip rate, the simple algorithms are preferred.

The paper is organized as follows. The proposed receiver is described in Section 2. Signal analysis at the receiver input is performed in Section 3. In Section 4, the description of the performance criteria is given and the adaptive DFB+2-TF receiver performance is analyzed. In the last section some concluding remarks are made.

2. PROPOSED RECEIVER DESCRIPTION

Block diagram of the proposed receiver is given in Figure 1.

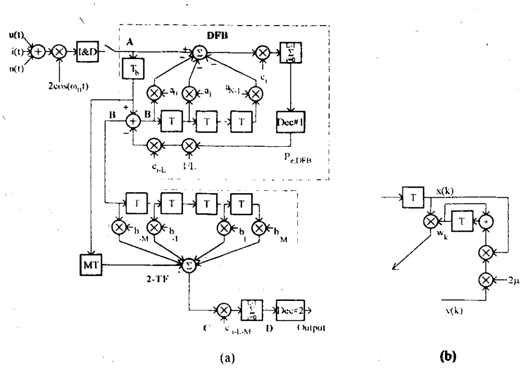

Fig. 1. (a) Block diagram of the proposed DSSS receiver with DFB+2-TF. I&D is the integrate and dump circuit, Dec# 1, Dec# 2 are the decision circuits. (b) The coefficient ($w_k = a_k, b_k$) LMS updating detail; $y(k)$ is the appropriate filter output.

The upper half of the receiver shown in Figure 1 (a) contains the DFB filter. The main purpose of this filter is to produce the reference interference signal without the desired signal, which can be done only if no decision error propagation occurs in DFB. This reference signal is fed to 2-TF, and based on the 2-TF output samples the received symbol detection is made. By taking the reference signal from the DFB filter output it is possible to avoid the intersymbol interference (ISI) that would be normally introduced by the 2-TF and to improve the overall receiver performance, [2].

Signal at the receiver input is demodulated and integrated at the PN chip duration interval, T. The order of the DFB filter is N, and 2-TF order is $2M$. Message bit duration is $T_b = LT$, where L stands for the processing gain.

3. THEORETICAL ANALYSIS

Input signal consists of three components. First one is the desired DSSS signal, given by,

$$u(t) = Ud(t)\,PN(t)\cos(\omega_0 t), \tag{1}$$

where U and ω_0 represent the DSSS carrier amplitude and angular frequency. Random message $d(t)$ is given by,

$$d(t) = \sum_i d_i \Pi_b(t - iT_b), \quad d_i = \in \{-1, 1\}, \quad \Pi_b(t) = \begin{cases} 1, & |t| \le T_b/2, \\ 0, & |t| > T_b/2. \end{cases} \tag{2}$$

At the sampler output, point A in the receiver, the desired signal is represented by its samples $u(k) \in \{-UT, +UT\}$ of power P_u.

Pseudonoise sequence is given by,

$$PN(t) = \sum_i c_i \Pi_c(t - iT), \quad c_i = \in \{-1, 1\}, \quad \Pi_c(t) = \begin{cases} 1, & |t| \le T/2, \\ 0, & |t| > T/2. \end{cases} \tag{3}$$

Second component of the received signal is the nonstationary NBI. This NBI is modeled as the narrowband autoregressive process of the fourth order, [3, 6], with two double poles, $z_{12} = 0.99 \exp(\pm 2\pi f_d T)$, where $f_d T$ stands for the normalized NBI carrier to DSSS carrier frequency offset. For $f_d T = 0$, NBI reduces to the second order AR interference at the DSSS carrier with double pole $z_0 = 0.99$.

NBI autocorrelation function is given by,

$$R_i(k) = P_{i,in}\left[1 - \left(1 - \frac{1.98}{(1 + 0.9801)z_0}\right)k\right] z_0^k \cos(k 2\pi f_d T), \tag{4}$$

where $P_{i,in}$ stands for the NBI average power at the DFB+2-TF input (Fig. 1 (a), point A), while $z_0 = 0.99$.

Third component of the received signal is the white Gaussian noise:

$$n(t) = n_c(t)\cos(\omega_0 t) + n_s(t)\sin(\omega_0 t). \tag{5}$$

The in-phase and quadrature components of this noise, $n_c(t)$ and $n_s(t)$ are statistically independent. One-sided power spectral density of this noise is η, and its power at the DFB+2-TF filter input (Fig. 1 (a), point A) is $P_{n,in}$.

All three components of the received signal are mutually independent and wide-sense stationary.

Signal at the 2-TF output, Figure 1 (a), point C, is given as the sum of the four following components:

— desired DSSS signal $s_C(k)$,

$$s_C(k) = d(k)\, c(k); \tag{6}$$

— intersymbol interference $e_C(k)$,

$$e_C(k) = -\sum_{i-1}^{M} b_i[d(k-i)\, c(k-i) + d(k+i)\, c(k+i)] + e_{DFB}(k); \tag{7}$$

where $e_{DFB}(k)$ stands for the ISI resulting from the error propagation in the DFB filter part:

$$e_{DFB}(k) = \begin{cases} 0, & \text{with no error in DFB,} \\ -2\sum_{i=0}^{N-1} a_i d(k-i-L)\, c(k-i-L), & \text{with error in DFB;} \end{cases} \tag{8}$$

— colored Gaussian noise $n_C(k)$,

$$n_C(k) = n(k) - \sum_{\substack{i=-M \\ i\neq 0}}^{M} b_i\, n(k-i); \tag{9}$$

— residual NBI $i_C(k)$,

$$i_C(k) = i(k) - \sum_{\substack{i=-M \\ i\neq 0}}^{M} b_i\, i(k-i); \tag{10}$$

In eq. $(7-10)$ $a_0 \ldots a_{N-1}$ stand for the DFB filter coefficients, and $b_1 \ldots b_M$ stand for the 2-TF coefficients in Figure 1 (a).

Tap weights of the DFB+2-TF receiver, Figure 1 (a), are adapted to the non-stationary AR NBI by simple and robust Widrow–Hoff least mean square (LMS) algorithm. This algorithm is described by, [4],

$$w_i(k-1) = w_i(k) + 2\mu y(k)\, x_i^*(k). \tag{11}$$

In this expression $w_i(k)$ represents the ith filter coefficient, a_i or b_i, at $t = kT$, $x_i^*(k)$ stands for the complex conjugate sample at the ith filter delay cell, $y(k)$ is the appropriate filter output signal, and μ is the algorithm convergence parameter.

4. NBI SUPPRESSION RESULTS

Adaptive DFB+2-TF filter performance is tested in the presence of nonstationary AR NBI by the computer simulation. Block diagram of the modeled receiver is presented in Figure 1 (a), and input signals are described in Section 3. The initial values of all filter coefficients are set to zero. Parameter values are: processing gain $L = 7$; Signal/Noise ratio at the DFB+2-TF input $A_n = 12\,\text{dB}$; Signal/NBI ratio at the DFB+2-TF input, $\Gamma = -40\ldots0\,\text{dB}$; DFB filter order $N = 4$; 2-TF filter order $2M = 4$. The receiver performance is expressed by the Signal/(NBI+Noise) improvement and bit error rate.

The reasons for the choice of these parameter values are twofold. First, they are accepted accordingly to $[1-3, 5]$, in order to enable the comparison of results obtained for different receiver structures and interference models. Second, the application of the interference suppression filter in the DSSS receiver is helpful in the cases where the interference at the receiver input is much stronger than the desired signal and/or the processing gain is small, so the narrowband interference suppression due to it is not sufficient.

Signal/(NBI+Noise) improvement

This measure is defined as the ratio of Signal/(NBI+Noise) at the DFB+2-TF output and input. On the condition of equal desired signal power at the DFB+2-TF input and output, this ratio becomes,

$$G = \frac{P_{i,in} + P_{n,in}}{P_{i,out} + P_{n,out}}. \tag{12}$$

where $P_{i,in}$ and $P_{n,in}$ stand for the NBI and noise power at the DFB+2-TF input (Fig. 1 (a), point A), and $P_{i,out}$ and $P_{n,out}$ stand for the NBI and noise power at the DFB+2-TF output (Fig. 1 (a), point C), respectively.

In Figure 2 the process of DFB+2-TF filter adaptation is presented. This figure gives the Signal/(NBI+Noise) ratio improvement as the function of the number of LMS algorithm iterations, with μ as a parameter. Algorithm performs one iteration per input signal sample, and sampling period is equal to the PN chip duration, T. Signal, NBI and noise power are calculated as an average of ten consecutive samples. Starting values of DFB+2-TF coefficients are all set to zero.

It can be seen from Figure 2 that smaller values of parameter μ result in slower algorithm convergence, but also perform better regarding the obtainable Signal/(NBI+Noise) improvement. Value of $\mu = 0.01$ can be regarded as optimal, considering the adaptation speed and achievable Signal/(NBI+Noise) improvement. This value of μ enables the DFB+2-TF coefficients to reach the steady state in approximately 100 algorithm iterations.

In Figure 3 the Signal/(NBI+Noise) improvement during the adaptation process is presented with the Signal/NBI ratio Γ at DFB+2-TF input as parameter.

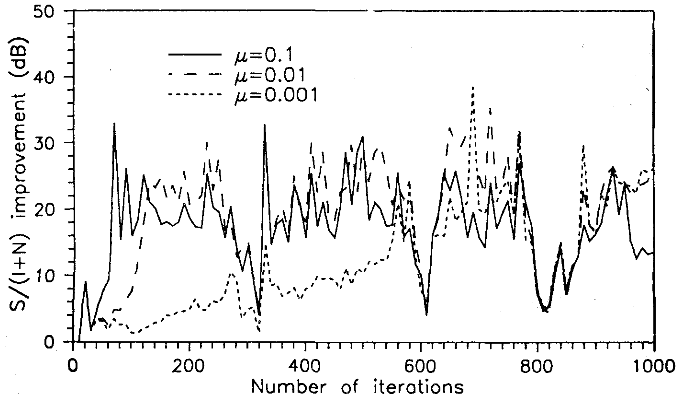

Fig. 2. Signal/(NBI+Noise) improvement as a function of the number of LMS algorithm iterations, with μ as parameter. Interference is second-order AR on the DSSS carrier frequency. $N = 2M = 4$, $\Gamma = -20\,\mathrm{dB}$, $A_n = 12\,\mathrm{dB}$, $L = 7$.

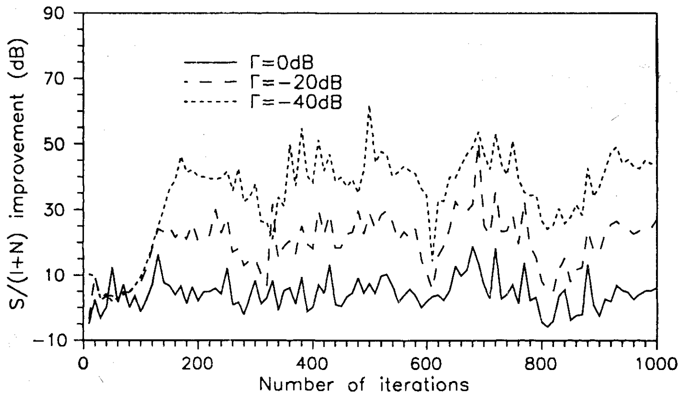

Fig. 3. Signal/(NBI+Noise) improvement as a function of the number of LMS iterations, with the Signal/NBI ratio at the DFB+2-TF input as the parameter. Interference is the second order AR on the DSSS carrier frequency. $N = 2$, $M = 4$, $A_n = 12\,\mathrm{dB}$, $\mu = 0.01$, $L = 7$.

It can be seen from Figure 3 that the Signal/(NBI+Noise) improvement is at first rising almost to the obtainable limit for the given input Signal/NBI ratio, and than is oscillating bellow that maximal value. The slope at the beginning of the adaptation process is independent of the input Signal/NBI ratio, so more time is needed by DFB+2-TF to obtain the steady state in the case of worst input Signal/NBI ratios.

The Signal(NBI+Noise) improvement achieved by the DFB+2-TF in the steady state is comparable to the inverse of the Signal/NBI ratio at the receiver input. This means that the signal and NBI at the DFB+2-TF output are of similar power, and the additional Signal/NBI improvement can be obtained through the DSSS processing gain.

In Figure 4 the influence of the fourth order AR NBI frequency offset on the DFB+2-TF adaptation process is presented. Normalized NBI frequency offset is parameter. It can be seen from Figure 4 that the NBI frequency offset does not have a noticeable influence on the DFB+2-TF adaptation speed.

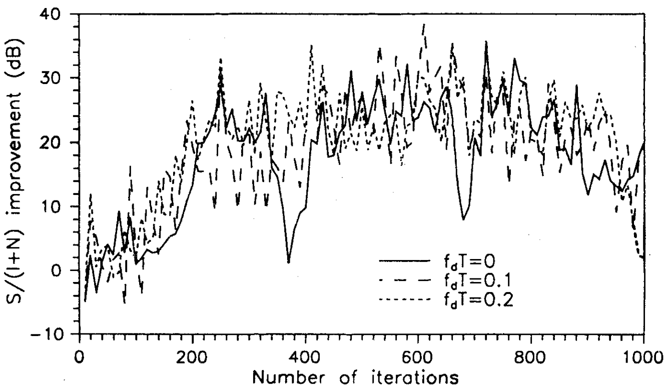

Fig. 4. Signal/(NBI+Noise) improvement as a function of the number of LMS algorithm iterations, with the fourth order AR NBI carrier frequency to DSSS carrier frequency offset as parameter. $N = 2M = 4$, $A_n = 12\,\text{dB}$, $\mu = 0.01$, $\Gamma = -20\,\text{dB}$, $L = 7$.

Bit error rate

Having in mind the receiver configuration, Figure 1 (a), and the error propagation at the Dec# 1 output, bit error rate at the receiver output can be calculated as the function of the particular PN sequence. The conditional error probability for the accepted PN sequence can be expressed as,

$$P_{e,PN} = \frac{1}{2}\,erfc\left(\frac{E\{S(T_b)\}}{\sqrt{2\,Var\{S(T_b)\}}}\right). \tag{13}$$

where $S(T_b)$ stands for the sample at the decision circuit input (Fig. 1 (a), point D):

$$S(T_b) = \sum_{k-1}^{L} c(k)\,[s_C(k) + e_C(k) + i_C(k) + n_C(k)]. \tag{14}$$

Symbols $E\{\}$ and $Var\{\}$ are the expectation and variance operators for the known PN sequence, and *erfc*( ) is the complementary error function. The probability of the error propagation in the DFB part of the receiver is also contained in the expectation operator in (13).

Total error probability is then calculated as the average of the conditional error probability for all possible PN sequences,

$$P_e = \frac{1}{K}\sum P_{e,PN}. \tag{15}$$

where K is the number of all possible combinations of PN chip contained in the delay lines of DFB and 2-TF relevant for the sample at the decision circuit input.

In Figure 5 the bit error rate is presented as a function of the number of iterations, for two different values of the convergence parameter μ. Initial values of filter coefficients are set to zero. The theoretical value of BER is presented in Figure 5 by the dashed bottom line.

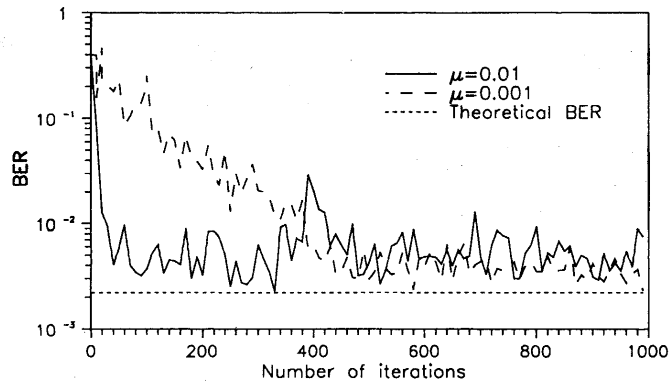

Fig. 5. Bit error rate as a function of the number of LMS algorithm iterations, with μ as parameter. Interference is the second-order AR on the DSSS carrier frequency. $N = 2M = 4$, $\Gamma = -20\,\text{dB}$, $A_n = 12\,\text{dB}$, $L = 7$.

It can be seen from Figure 5 that the value of the convergence parameter μ must be chosen carefully not only regarding the LMS stability. It is also possible to trade off between the convergence speed and the average value of BER obtained after the algorithm converge by the proper choice of μ. For $\mu = 0.01$ the algorithm converges to the steady state after 50 iterations, and the r.m.s. difference between the obtained and theoretical BER equals 5.44*10-3. On the other side, for $\mu = 0.001$ the algorithm converges much slower and reaches the steady state after 500 iterations, but the r.m.s. difference between the obtained and theoretical BER equals only 1.9*10-3. From this it follows that the proper choice of μ is of great importance in the nonstationary environment.

In Figure 6 the influence of the input Signal/NBI ratio on the algorithm convergence process is presented.

It can be seen from Figure 6 that, similar to the receiver performance regarding the Signal/(NBI+ Noise) improvement, presented in Figure 3, the input Signal/NBI ratio Γ has no significant influence on the convergence speed. Constant slope of the BER improvement in Figure 6 results in more iterations for better input Signal/NBI ratios before DFB+2-TF reaches the steady state.

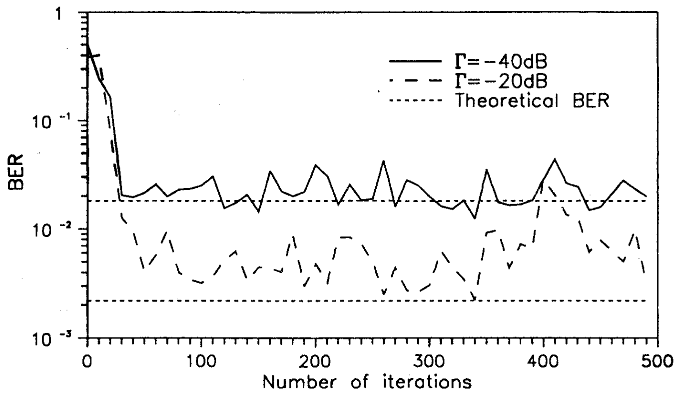

Fig. 6. BER as a function of the number of LMS iterations, with the Signal/NBI ratio at the DFB+2-TF input as parameter. Interference is the second-order AR on the DSSS carrier frequency. $N = 2M = 4$, $A_n = 12\,\text{dB}$, $\mu = 0.01$, $L = 7$.

In Figure 7 the influence of the frequency offset on the BER in the process of filter adaptation is presented.

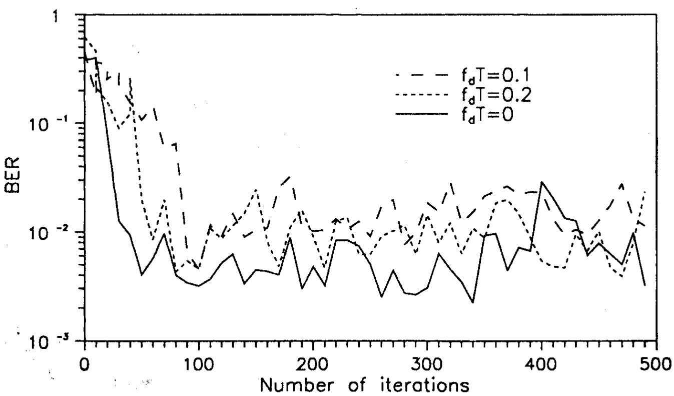

Fig. 7. BER as a function of the number of LMS algorithm iterations, with the fourth order AR NBI carrier frequency to DSSS carrier frequency offset as a parameter. $N = 2M = 4$, $A_n = 12\,\text{dB}$, $\mu = 0.01$, $\Gamma = -20\,\text{dB}$, $L = 7$.

It can be concluded that the AR NBI frequency offset has an influence on the DFB+2-TF adaptation process. The slowest adaptation is achieved for $f_d T = 0.1$. After a sufficient number of iterations, however, DFB+2-TF reaches the steady state regarding BER. For the accepted values, the transient time is less then 100 LMS iterations.

5. CONCLUSION

In this paper the performance of the adaptive DSSS receiver with the DFB+2-TF interference suppression filter in the nonstationary electromagnetic environment is analyzed. The DFB+2-TF receiver and the signals at its input are described. The nonstationary EM environment is modeled as a narrowband AR process of second and fourth order. The DFB+2-TF coefficients are adapted based on the LMS algorithm. The receiver performance is expressed in terms of the Signal/(NBI+Noise) improvement ratio and the bit error rate, as a function of the signal, interference, and receiver parameters. The analysis is oriented towards the systems characterized by the strong narrowband interference presented at the receiver input and/or the small processing gain, that is not sufficient to attain the desired system quality. Considering the results, it can be concluded that the overall good DFB+2-TF receiver performance in the stationary environment is not degraded by the variation of the filter coefficients caused by their adaptation in the nonstationary case, providing that the LMS convergence parameter μ is adequately chosen to satisfy the mutually opposite requirements of the fast convergence and the small variance in the steady state. It is possible to achieve the steady state in 100 chips with the acceptable reduction of the bit error rate.

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