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Adaptive Filter Approaches for Interference Suppression in CDMA Systems

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Abstract: The adaptive Least Mean Square algorithm is widely used in Code-Division multiple access detectors due to its simplicity. A new variant of the Adaptive Least Mean Square (LMS) filter is proposed in this study to counter the interference in CDMA system. Here the merits of the normalized LMS (NLMS) which has the faster convergence and the merits of the Variable Step Size LMS (VSSLMS) which is the less Mean Square Error are combined to give a Variable Step Size Normalized LMS (VSSNLMS). Also a comparison is made between the BER obtained when the NLMS, VSSLMS and VSSNLMS applied to the CDMA downlink. We show that the proposed VSSNLMS algorithm has the least BER and also faster convergence. Also the error is less at each iteration, compared to NLMS and VSSLMS. We have analyzed the behaviour of different adaptive algorithms for the problem of multiuser detection in Synchronous CDMA environment.

Key words: Code Division Multiple Access (CDMA), adaptive filters, Least Mean Square (LMS), Normalized LMS (NLMS), Variable Step Size LMS (VSSLMS)

INTRODUCTION

Code Division Multiple Access using the Direct Sequence Spread Spectrum technique (DSCDMA) has many advantages like the increased channel usage, less jamming capability. In multiuser CDMA, multiple access interference (MAI) is the main source of performance degradation. The MAI is caused by other users in the channel. The adaptive techniques have been successfully used to equalize the channel and thus reduce the MAI Interference in DSCDMA. Reduction of MAI is our prime concern which ultimately leads to capacity and system improvement. Two types of adaptive detectors are generally used. One is the blind adaptive detector which is used when the spreading code and the channel parameters are known or can be estimated. The other is the trained based implementation which is used in case of absence of this information (Egiazarian *et al*., 2003). In the case of the trained based systems, a known training sequence is transmitted which is used to tune the coefficients of the adaptive filter before the actual data is send. Usually an LMS algorithm is used for varying the weights of the coefficients adaptively. The LMS algorithm is simple and easy to implement. But it converges slowly. To overcome this drawback, many variants of LMS algorithm are proposed. The two important variants of the LMS algorithm are the Normalized LMS and the Variable Step Size LMS. The NLMS converges much more quickly than LMS at very little extra cost. In training process, the error between the output and the desired signal can be decreased by choosing a small step size. But the convergence becomes slow by doing so. So the step size is kept large initially and later decreased. i.e., we go for a variable step size. Both the variants can be combined to get faster convergence and also better accuracy.

For highly correlated inputs the LMS algorithm has a slow convergence which requires long training sequences and therefore low transmission speeds. Another drawback of the LMS is the trade-off between convergence speed and the steady state error since both are controlled by the same parameter, the step size. In order to eliminate these drawbacks the class of Variable Step Size LMS algorithm was introduced (Egiazarian *et al*., 2003). And finally we have shown that the proposed VSSNLMS algorithm has the least BER and also faster convergence. Also the error becomes less at each iteration, compared to NLMS and VSSLMS. Interference suppression which is the main objective in our work is achieved reasonably good by means of iterative process. These algorithms can also be tested for multiuser, multipath environment by employing Rake receiver which will be a new attempt.

SYSTEM BASIC MODEL

The basic CDMA model, the data is channel coded and then it is spreaded by multiplying it with the orthogonal codes. Now it is modulated on a carrier and transmitted along with the other modulated data. At the receiver side, we demodulate the data and it is despreaded by multiplying with the same orthogonal code and then decoded to get the message data. Here if the channel is noiseless, then the original message is obtained only if

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the spreading codes used are highly orthogonal i.e., the cross correlation between any two codes is zero. Here the transmission of the message data is done at all times and frequencies (both TDMA and FDMA combined).

For the sake of simplicity we consider a synchronous CDMA system in which a number of 2^N users transmit over a single-path time-invariant channel. The processing gain is denoted by N, the attenuation of each user data are denoted by a_K and the data symbols transmitted by all the users are aligned in time. The received symbol sampled at chip rate can be written in vector form as follows:

$$r(n) = SAd(n)-v(n)$$

Where the j^{th} column of d (n) represents the received spreading code of the j^{th} user. The vector contains the data symbols transmitted by all the users at the time instant n. The N X 1 vector r is the sampled channel noise and the 2^N X 2^N matrix A is given by

$$A = diag(a_1,a_2,a_3......a_{2N})$$

NORMALIZED VSSLMS ALGORITHM

The weight up gradation vector is given by

$$W(n+1) = W(n)+2\mu e(n)X(n) \qquad (1)$$

Here X(n) is the input signal, W(n) is the coefficient vector of the adaptive filter, $\mu(n)$ is the variable step size and e(n) is the error vector at time n, as shown in Fig. 2.

$$W(n) = [w(1), w(2),...., w(N)],$$

$$X(n) = [x(n),x(n-1),....,x(n-N+1)]$$

and

$$e(n) = d(n)-X^T(n)W(n)$$

To increase the speed of convergence, we go for normalizing the step size in the weight upgradation vector i.e.,

$$W(n+1) = W(n) + 2\frac{\mu}{\|X(n)\|^2}e(n)X(n) \qquad (2)$$

By varying the step size based on the iteration and by normalizing the step size, we have the weight upgradation

$$W(n+1) = W(n) + 2\frac{\mu(n)}{\|X(n)\|^2}e(n)X(n) \qquad (3)$$

Here

$$\mu(n) = \mu(n-1)+\rho e(n)e(n-1)X^T(n-1)X(n)$$

Here ρ is a small positive constant that controls the adaptive behavior of the step size sequence $\mu(n)$. To assure a convergence of the mean square error, the condition shown below is the calculation as per theoretical. The condition thus is

$$0 < \mu(n) < \frac{2}{(3+\frac{1}{M_{adj}})tr(R)} \qquad (4)$$

Here R is the input correlation matrix, trace of R is the sum of diagonal elements of R which is said to be equal to the signal power and M_{adj} is the mis-adjustment level for the fastest convergence, defined by

$$M_{adj} = \frac{E_{ex}}{\zeta_{min}} \qquad (5)$$

and

E_{ex} is the final misadjustment requirement as defined in (Egiazarian *et al*., 2003).

Algorithm steps

```
for known input X
{
Spread X
Tx_data = bpskmod (Spreaded X)
Rx data = Tx_data + Noise
Assume Initial Weights
for each iteration
{
find output at that instant
find error;
Upgrade Weight
Upgrade
}
}
for all the unknown input X
{
Spread X
Tx_data = bpskmod(Spreaded X)
Rx_data = Tx_data + Noise
for each iteration
{
Output = conv(weights,Rx_data)
}
find no of: errors and BER
}
```

SIMULATION PARAMETERS

The simulation parameters for the basic CDMA model (Fig. 1) are assumed that a suitable coding is done on the input data and the output of the channel coding as a random sequence of intergers. Then this has to be multiplied with the PN sequence. This sequence is generated by using Walsh-Hadamard Code Generator. The sequence generated in such a way is found to be orthogonal. Now the output of the channel encoder is multiplied with the orthogonal code generator. The output is then given to BPSK modulator. The output is then transmitted via the channel. These are the simulation parameters assumed at the transmitter side.

The channel is assumed to be having Rayleigh fading characteristics and Additive White Gaussian Noise is also assumed to be added in the channel. These are the simulation parameters assumed at the channel side.

Finally at the receiver, the received data is then given to the BPSK modulator. Now the output of the demodulator is then despreaded. It is done by multiplying the output of the BPSK demodulator with the same PN sequences as at the input. It is assumed that the PN sequence used at the transmitter is reconstructed at the receiver. These are the simulation parameters assumed at the receiver side.

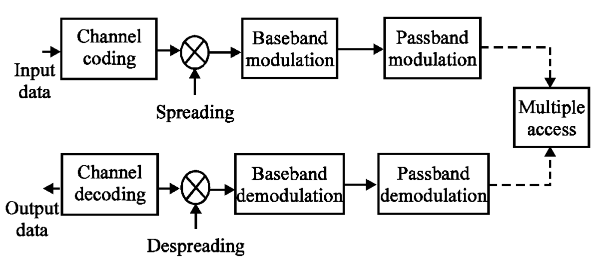

Fig. 1: Basic CDMA model

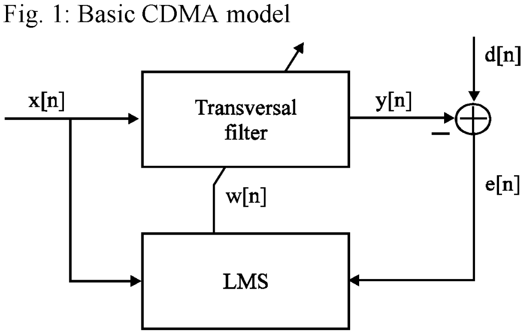

Fig. 2: A simple adaptive filter

SIMULATED RESULTS

This experiment was conducted at Computer Simulation Laboratory, Department of ECE, PSG College

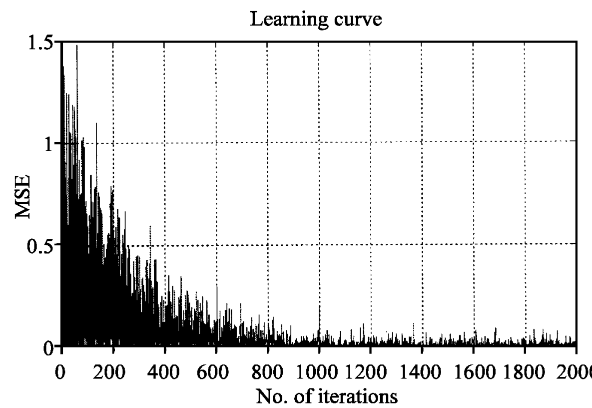

Fig. 3: LMS algorithm

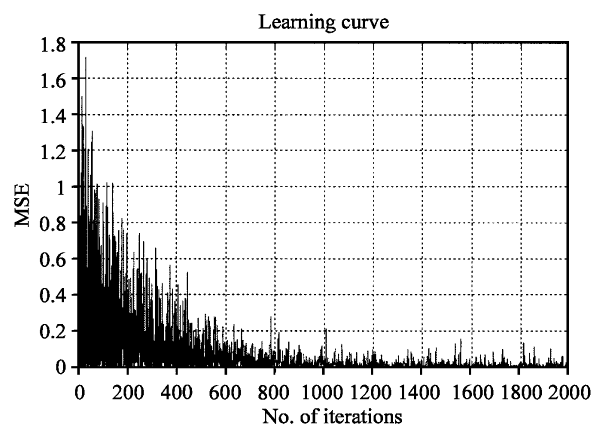

Fig. 4: Normalized LMS algorithm

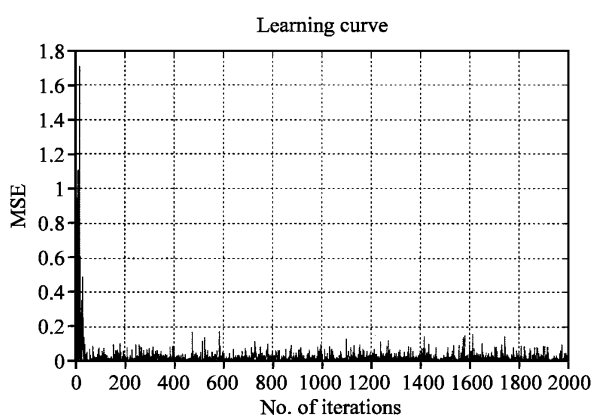

Fig. 5: VSS LMS algorithm

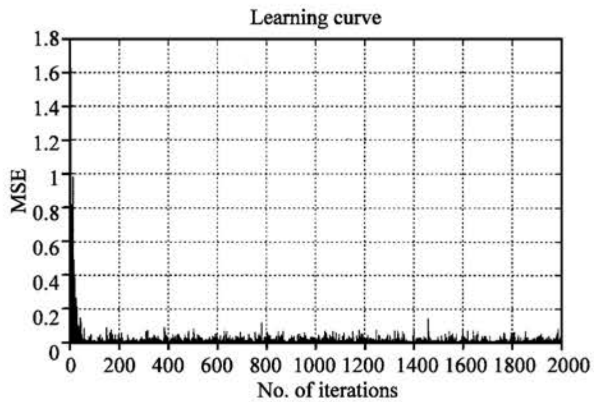

Fig. 6: Normalized VSS LMS algorithm

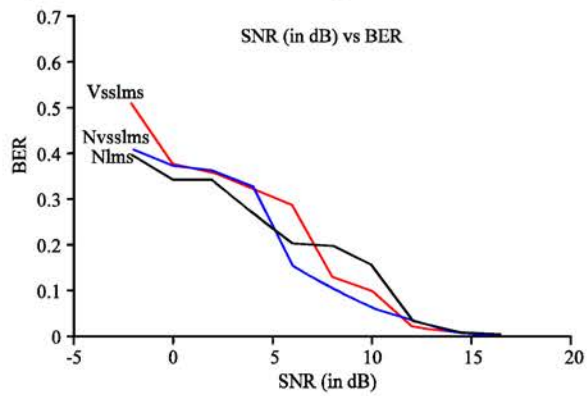

Fig. 7: SNR vs BER for NLMS, VSSLMS and NVSSLMS

SIMULATED RESULTS

LMS algorithm, NLMS, VSSLMS, NVSS-LMS in the adaptive filter, we have the learning curve as shown in Fig. 3-6. The plot of signal to noise ratio vs Bit Error Ratio is shown in Fig. 7 for all the three variants of LMS. And it is quiet evident from the plot that the proposed VSSNLMS has better performance with least Bit Error Rate (BER) and also faster convergence w.r.t all other algorithms tested. By observing, we can say that the NLMS converges faster than the LMS and a keen observation tells that the NVSS-LMS has the superior performance and bit faster in convergence than the VSSLMS.

CONCLUSIONS

From the results it is clear, that the NLMS has faster rate of convergence than the LMS. But by varying the step size, we can decrease the mean square error as compared to the existing technique. Here better results are obtained if the NVSSLMS is used in channel equalization. Still better results can be obtained if the NVSSLMS algorithm is used in the Rake Receiver of the CDMA model. With the introduction of rake along with the adaptive technique still the performance of the system as well as the convergence can be made much faster with the same algorithms discussed in this study. Hybrid adaptive algorithms utilized for analysis here is a novel attempt and the results are compared with the standard LMS technique. This leads to further research taking into account the multipath effects, Signal-to-interference ratio, Channel estimation, diversity problems and also channel equalization could be considered.

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