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 multiser 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 sent. 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
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 $a_i$ 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 $N \times 1$ vector $r$ is the sampled channel noise
and the $2^n \times 2^n$ matrix $A$ is given by

$$A = \text{diag}(a_0, a_n, a_1, \ldots, a_{2^n})$$

**NORMALIZED VSSLMS ALGORITHM**

The weight upgradation vector is given by

$$W(n+1) = W(n)+2\mu e(n)X(n)$$  \hspace{1cm} (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.

$$W(n) = [w(1), w(2), \ldots, w(N)]$$
$$X(n) = [x(n), x(n-1), \ldots, x(n-N+1)]$$
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.,

$$W(n + 1) = W(n) + 2 \frac{\mu(n)}{[X(n)]^T} e(n)X(n)$$  \hspace{1cm} (2)

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

$$W(n + 1) = W(n) + 2 \frac{\mu(n)}{[X(n)]^T} e(n)X(n)$$  \hspace{1cm} (3)

Here

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

Here $\rho$ 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

$$0 < \mu(n) < \frac{2}{(3 + \frac{1}{M_{\text{adj}}})\text{tr}(R)}$$  \hspace{1cm} (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 $M_{\text{adj}}$ is the mis-adjustment level for
the fastest convergence, defined by

$$M_{\text{adj}} = \frac{E_m}{\bar{r}_{\text{min}}}$$  \hspace{1cm} (5)
and
$E_m$ is the final misadjustment requirement as defined in
(Egiazarian et al., 2003).

**Algorithm steps**

for known input $X$

{ Spread X
  $Tx\_data = \text{bpskmod}(\text{Spreaded } X)$
  $Rx\_data = Tx\_data + \text{Noise}$
  Assume Initial Weights
  for each iteration
  { find output at that instant
    find error,
    Upgrade Weight
    Upgrade
  }
  for all the unknown input $X$
  { Spread X
    $Tx\_data = \text{bpskmod}(\text{Spreaded } X)$
    $Rx\_data = Tx\_data + \text{Noise}$
    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 integers. 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 despread. 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.

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. 1: Basic CDMA model

Fig. 2: A simple adaptive filter
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.

REFERENCES