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A SELF ADAPTIVE EQUALIZER WITH RAPID ACQUISITION

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ABSTRACT

Conventional adaptive equalization of data transmission channels requires an initial training period during which a known data sequence is transmitted and properly synchronized at the receiver. Transmission of the training sequence is not convenient for multipoint data communication networks. Recently, research is directed towards equalization schemes that operate without initial training sequence. Such schemes are called self-adaptive equalizers(SAE). Most of SAE schemes available in literature need a long time to converge to the optimum equalizer.

In this paper we propose an equalization scheme that shortens this time significantly. The proposed scheme is a cascade of two adaptive equalizers. The first one is short and thus it converges rapidly. The partial equalization done by this equalizer is coarse and will not enable reliable recovery of data. The second equalizer is long and it starts adaptation after the convergence of the short one. Due to the partial equalization performed by this latter the convergence time decreases. The length of the long equalizer is determined by the desired precision of equalization. This scheme is simulated on a digital computer. Simulation results exhibit superiority of the proposed scheme over published ones.

I. INTRODUCTION

A great deal of research has been performed in the last twenty years for equalizing data transmission channels [e.g. 1-3]. Equalization is done by passing samples of received signal through an adaptive equalizer consisting of tapped delay line having adjustable coefficients (tap-weights).The equalizer tap-weights are adjusted by starting the transmission with a short training sequence of digital data which is known in advance by

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the receiver. This mode is called the training mode. At the end
of this mode the equalizer coefficients reach their optimum
values. Thenceforward the equalizer will use the detected data
instead of actual data to continue the adaptation process. This
latter mode is called the decision-directed mode, and it is
intended for tracking possible changes in channel characteris-
tics during transmission.

In modern multipoint data-communication netweorks, several
receivers may be hearing the same transmitter at the same time.
The problem pecular to multipoint networks is that of retraining a
tributary receiver which, because of drastic changes in channel
characteristics or simply because it was not powered-on during
initial network synchronization, is not able to recognize data
and polling messages. Since lines are shared, the control modem
has to interrupt data transmission and initiate a new synchro-
nization procedure generally causing all tributaries to retrain.

It is clear that, particularly for large or heavily loaded
multipoint networks, data throughput is increased and network
monitoring is made easier by giving tributary receivers the
capability to achieve complete adaptation without disrupting
normal data transmission to other terminals. The equalizers used
to make the receivers operating as explained above are called
self adaptive equalizers (SAE).

A common feature of previous SAE algorithms [3-7] is that they need
a relatively long time to converge to the optimum setting. This
drawback may be not tolerable in cases of rapidly varying channels.
Also,the information lost during this time will be large. In
this paper, we propose an SAE algorithm that has rapid converge-
nce and low steady state error. The proposed scheme is a cascade
of two adaptive equalizers. The first one is short and thus it
converges rapidly. The second equalizer is long and it starts
adaptation after the convergence of the short one. Due to the
partial equalization performed by this latter, the convergence
time of the long equalizer is reduced.

The paper is orgainzed as follows : section II gives a review on
the previous work in the domain of SAE. The proposed SAE is
given in section III, and its simulation results are given in
section IV. Finally section V is devoted to discussion of resu-
lts and conclusions.

II. REVIEW ON PREVIOUS WORK

The block diagram of data transmission system is given in Fig.1,
in which the "channel " includes the effects of the transmitter
filter, the modulator, the transmission medium, and the demodu-
lator. A symbol $a_k$which is one of L discrete amplitude levels,
is the transmitted data at instant kT, where T seconds is the
signaling inlerval. Let the vector X_k represent the N samples
measurments that are available at the channel output and have

some relationship with a_k.

The equalizer consists of a tapped-delay line having N adjustable coefficients (tap weights). Let H denote the vector composed of the equalizer coefficients. The output of the equalizer $y_k = H^T X_k$, H^T is the transpose of H, is a linear estimate of a_k on the basis of X_k. The optimum equalization vector H_* is the one that minimizes the difference between y_k and a_k according to some criterion. According to the popular minimum mean-square error criterion, H_* is the vector that minimizes $E((a_k-y_k)^2)$. This value is the solution of the Wiener- Hoph equation,

$$R H_* = P \qquad\qquad (1.a).$$

where;

$$R = E (X_k X_k^T) \qquad , \qquad P = E (a_k X_k) \qquad (1.b).$$

Direct solution of this equation implies matrix inversion which is a complicated process that is usually avoided in signal processing. In practice, this solution is searched iteratively with the help of the least-mean square (LMS) algorithm of Widrow,

$$H_{k+1} = H_k + \mu (a_k - H_k^T X_k) X_k \qquad\qquad (2).$$

where μ , called the step size, is a small positive constant that controls the convergence of the algorithm (2). This algorithm is used in the training mode. At the end of this mode, the equalizer coefficients reach their optimum values, H_*. Then the adaptation is continued by recovered data $\hat{a}_k$ instead of a_k to track any, possible, variations in channel characteristics. Thus the adaptation is continued according to the decision directed algorithm,

$$H_{k+1} = H_k + \mu (\hat{a}_k - H_k^T X_k) X_k \qquad\qquad (3)$$

where;

$$\hat{a}_k = dec(y_k) \qquad\qquad (4)$$

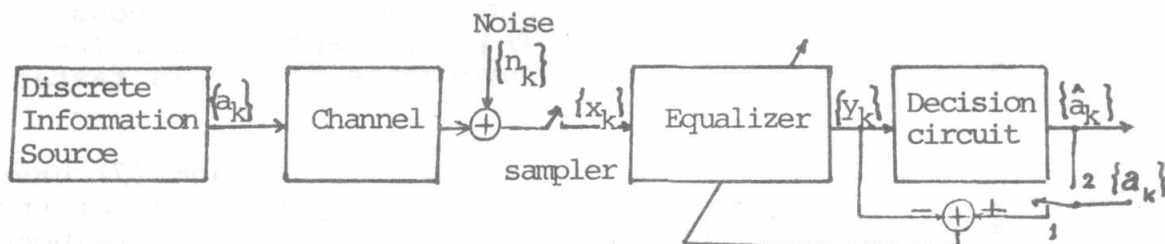$\hat{a}_k$ is obtained by quantizing y_k to the nearest data level.

Fig.1. Block diagram of a data
transmission system .

SAE schemes do not have a training mode. The direct use of the algorithm (3), from the start of communication, succeeds in reaching near to H_* for the cases where the eye pattern is initially open. This is the case of channels with mild distortion. If the eye pattern is initially closed, the algorithm (3) may diverge as has been observed from simulation and experimental results of this algorithm.

To overcome this problem, Sato[3] presented the following SAE algorithm,

$$H_{k+1} = H_k + \mu (\gamma \, \text{sign} \, (H_k^T X_k) - H_k^T X_k) X_k, \quad \gamma = \frac{E(a_k^2)}{E(|a_k|)} \qquad (5)$$

The probability of an erroneous sign-decision is smaller than that of the direct multilevel decision (3),(4). This explains the superiority of Sato algorithm (5) to the decision directed one (3), (4).

The disadvantage of Sato algorithm is that its equalization is more coarse than that of the decision directed algorithm . This problem is overcome by using Sato algorithm to open the eye pattern and then adaptation is switched to the decision directed algorithm (3),(4).

In [4] two other SAE algorithms, for quadrature amplitude modulation data communication systems, are presented. In such systems a_k, X_k and H_k are complex quantities. Denoting by X^* the complex conjugate of X_k, those algorithms are :

$$H_{k+1} = H_k + \mu \, y_k \, (\frac{\alpha}{|y_k|} - 1) X_k^* ; \quad \alpha = \frac{E(|a_k|^2)}{E(|a_k|)} \qquad (6)$$

and

$$H_{k+1} = H_k + \mu \, y_k \, (\beta - |y_k|^2) X_k^* ; \quad \beta = \frac{E(|a_k|^4)}{E(|a_k|^2)} \qquad (7)$$

The algorithms (6) and (7) are intended to minimize the coast functions, $E[(|y_k|-\alpha)^2]$ and $E[(|y_k|^2-\beta)^2]$, respectively. The results given in [4] show that the convergence of (7) is faster than that of (6).

In [5-7] , the convergence of SAE algorithms (5), (6) and (7) have been analyzed. According to those papers, the above SAE algorithms have numerous stationary points, some of those points correspond to good equalizers, while others correspond to bad ones. The points to which the algorithm converges depend on the initial setting of the equalizer. Thus, in general, one does not know if the above SAE algorithms will converge to a good equalizer or to a bad one. However, simulation results, given in [4] , show that

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in cases of channels with mild distortion, the above algorithms will converge to a good equalizer if one of the tap-weights is initially set to a large value, while the remaining ones are set to small (or zero) values.

III. PROPOSED SAE SCHEME

The proposed equalization scheme,Fig.2, is a cascade of two transversal filters. The first filter is short, i.e. it has small number of taps N_1, while the second one is long, it has N_2 taps. The principle of operation of this scheme relies on the following facts known from the theory of adaptive equalization [9] .
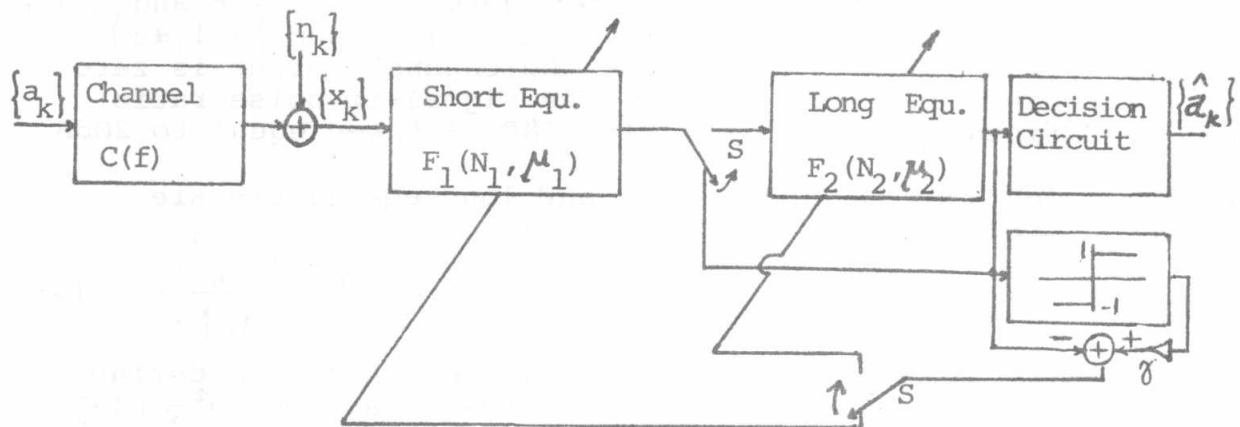
Fig. 2. SAE using two stages transversal filters.

As the equalizer length N increases, equalization becomes more precise. However, the allowable step size μ decreases and, consequently, the convergence speed decreases. The greater is the level of noise in the control error signal ($a_k - y_k$ in (3) and γ sign (y_k)$-y_k$ in (5)) the smaller should be the step size; thus the slower will be the convergence. Finally for a given adaptation algorithm (i.e. given N and μ), the farer is the initial setting from H_*, the longer will be the convergence time. The above arguments imply that precise SAE needs a long time to converge.

Now, in the proposed scheme, the short equalizer ensures rapid convergence. The partial equalization offered by this equalizer makes the equivalent channel in front of the long equalizer a mild one. This improves both the speed and accuracy of convergence of the long equalizer.

The adaptation of the proposed equalizer will be done in the following way. On the start of the adaptation process, only the coefficients of the short equalizer are adaptive. After the convergence of the short equalizer has been settled (say after i_c samples) the adaptation of the long equalizer is started while the adaptation of the short equalizer is terminated. The reason for this termination is to protect the long equalizer from the adaptation noise of the short one.

IV. SIMULATION RESULTS

Intersymbol interference equalization can be carried out for all pulse-modulation systems, including pulse amplitude medulation (PAM), frequency shift-keying, phase shift-keying and quadrature amplitude modulation.

In this paper only PAM systems are considered in the simulations. Conclusions, however, can be easily extended to other types of modulation. Impulse response and sampled impulse response of a typical telephone channel are given in figures 3 and 4 [5]. The data to be transmitted are equally distributed over L possible values. We have considered two cases. Those are : L = 8 and L = 4 and the data alphabets are $\{\pm 1, \pm 3, \pm 5, \pm 7\}$ and $\{\pm 1, \pm 3\}$ respectively. The additive noise at the channel output is zero mean white Gaussion with power σ'^2. The signal-to-noise ratio (SNR) is defined as $SNR = E(a_k^2)/\sigma'^2$. SNR is taken equal to 20dB.

The coefficients of both the short and long equalizers are apdated according to the algorithm,

$$H_{k+1} = H_k + \mu (\gamma \, sign \, (H_k^T X_k) - H_k^T X_k) \, X_k , \quad \gamma = \frac{E(a_k^2)}{E(|a_k|)} \qquad (5)$$

Before we give simulation results, it is convenient to define the following quantities. The first is the data power $a^2 = E(a_k^2)$. The second is the instantenous mean-square error $E((a_k-y_k)^2)$ at the output of the equalizer. Its estimate ϵ_n is calculated over the interval $[(n-1)M+1 , nM]$ according to

$$\epsilon_n = \frac{\sum_{i=(n-1)M+1}^{nM} (a_i-y_i)^2}{M} , \quad n = 1, 2, \ldots$$

where; M is few tens of samples. (M = 100 samples). The third is the error rate P_n in the interval $[(n-1)M+1 , nM]$. It is calculated according to

$$P_n = \frac{\sum_{i=(n-1)M+1}^{nM} \gamma_i}{M} , \quad n = 1, 2, \ldots$$

where ,

$$\gamma_i = \begin{cases} 1 & for \quad \hat{a}_i \neq a_i \\ 0 & for \quad \hat{a}_i = a_i \end{cases}$$

The performance of an equalizer is evaluated by the help of the convergence time T_c and the residual mean square error E. T_c is considered as the time after which ϵ_n fluctuates around

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a constant value and E is the average value of ϵ_n after convergence has been settled.

The figures 5-7 give the results of the proposed SAE with $N_1 = 9$, $N_2 = 21$, and $\mu_1 = 5.10^{-4}$ in the case of L=8. Fig. 5 shows the dependance of P_n on n. Without equalization, P_n is 70 percent (curve ①). The value of P_n when using conventional SAE (F_2 only) is shown in curve ② while that of the proposed SAE scheme ($F_1 + F_2$) is shown in curve ③ . It is clear that the convergence of the proposed SAE observed by curve ③ is faster than that of the conventional SAE (curve ②).

Relation between ϵ_n and n for $\mu_2 = 8.10^{-5}$ is shown in Fig. 6. The values of E and T_c for different values of μ_2 are given in Fig.7. From this figure it is obvious that the performance of the proposed scheme is better than that of the conventional one.

In the case of four level data (L=4), the results are given in the figures 8,9 and 10; by the same procedure explained in the previous case. It is concluded that the performance of the proposed SAE is also better than that of the conventional one.

Several trials have been done to reach the optimum values of i_c in the cases of L=8 and L=4, those values are 600 and 100 samples respectively.

In practical realization of the proposed equalization scheme, there should be a unite for estimating the values of i_c . The implementation of such a unite is now under our study.
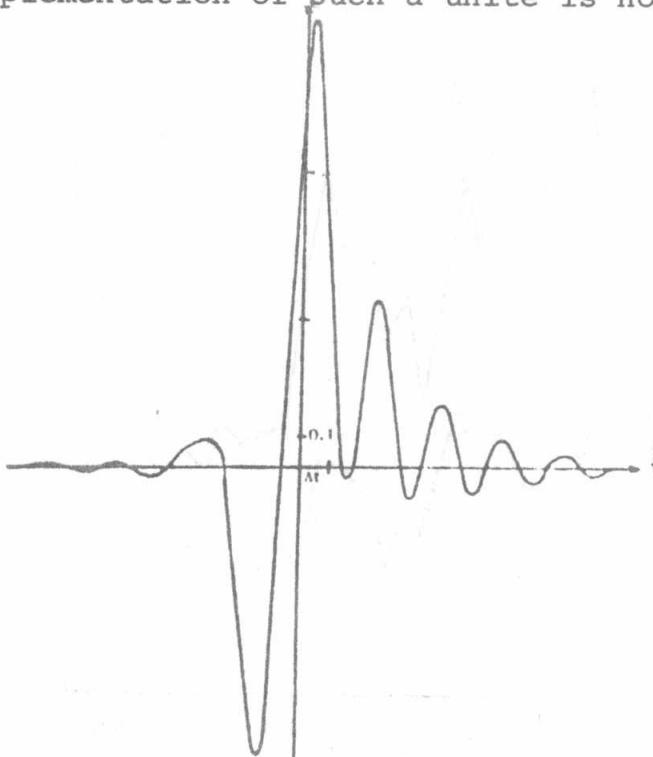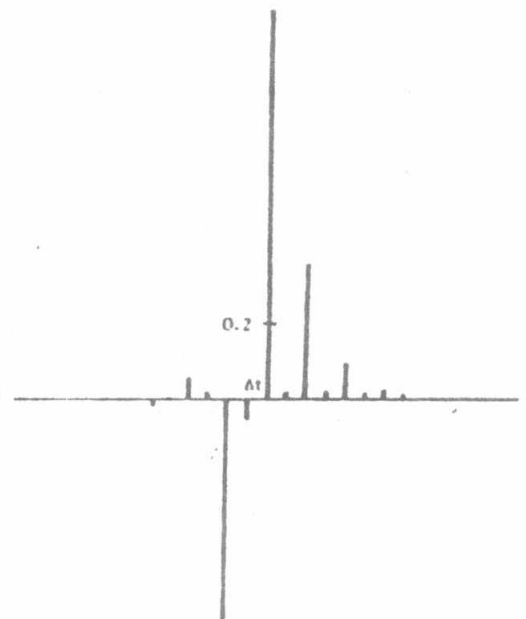

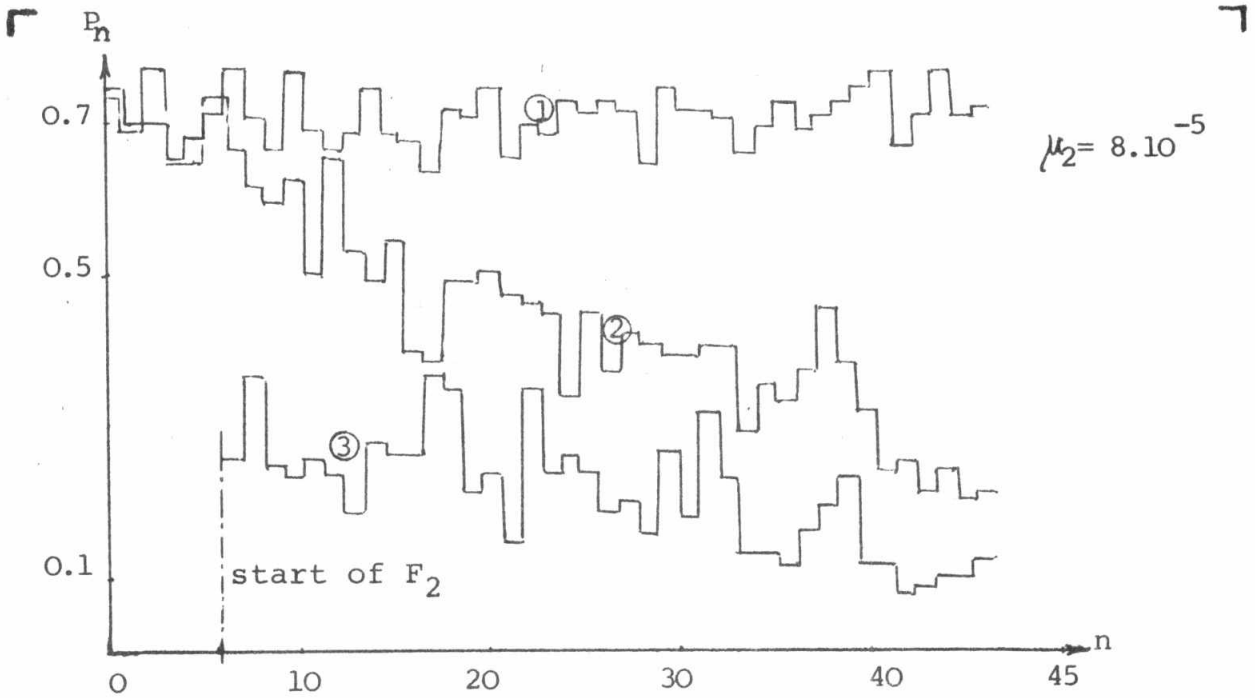Fig. 3. Impulse response of the channel

Fig.4. Sampled impulse response

Fig. 5. Evolution of the error rate. ① output of the
channel. ② output of the conventional equalizer
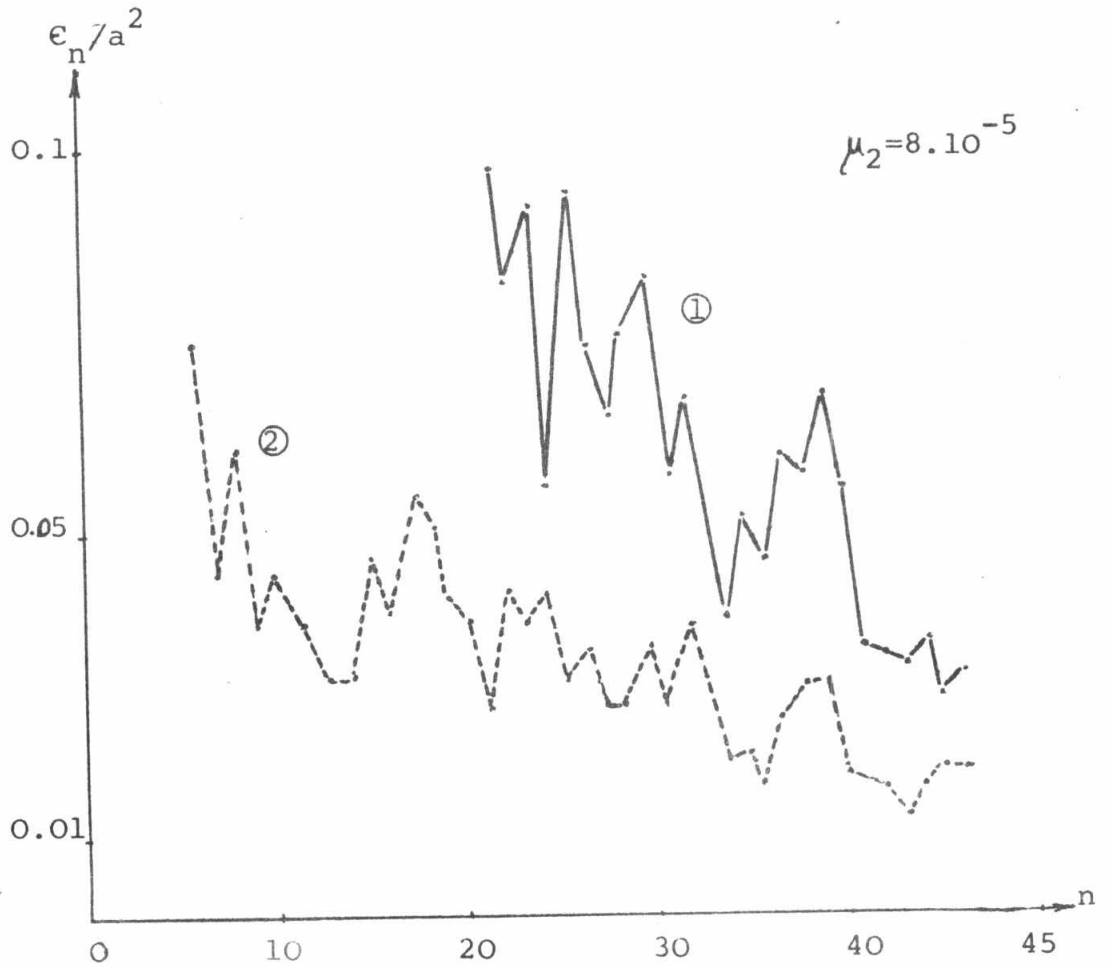(F_2 only). ③ output of the proposed scheme.(L=8).

Fig.6. Evolution of mean square error. ① output of F_2
only. ② output of the proposed scheme (L=8).

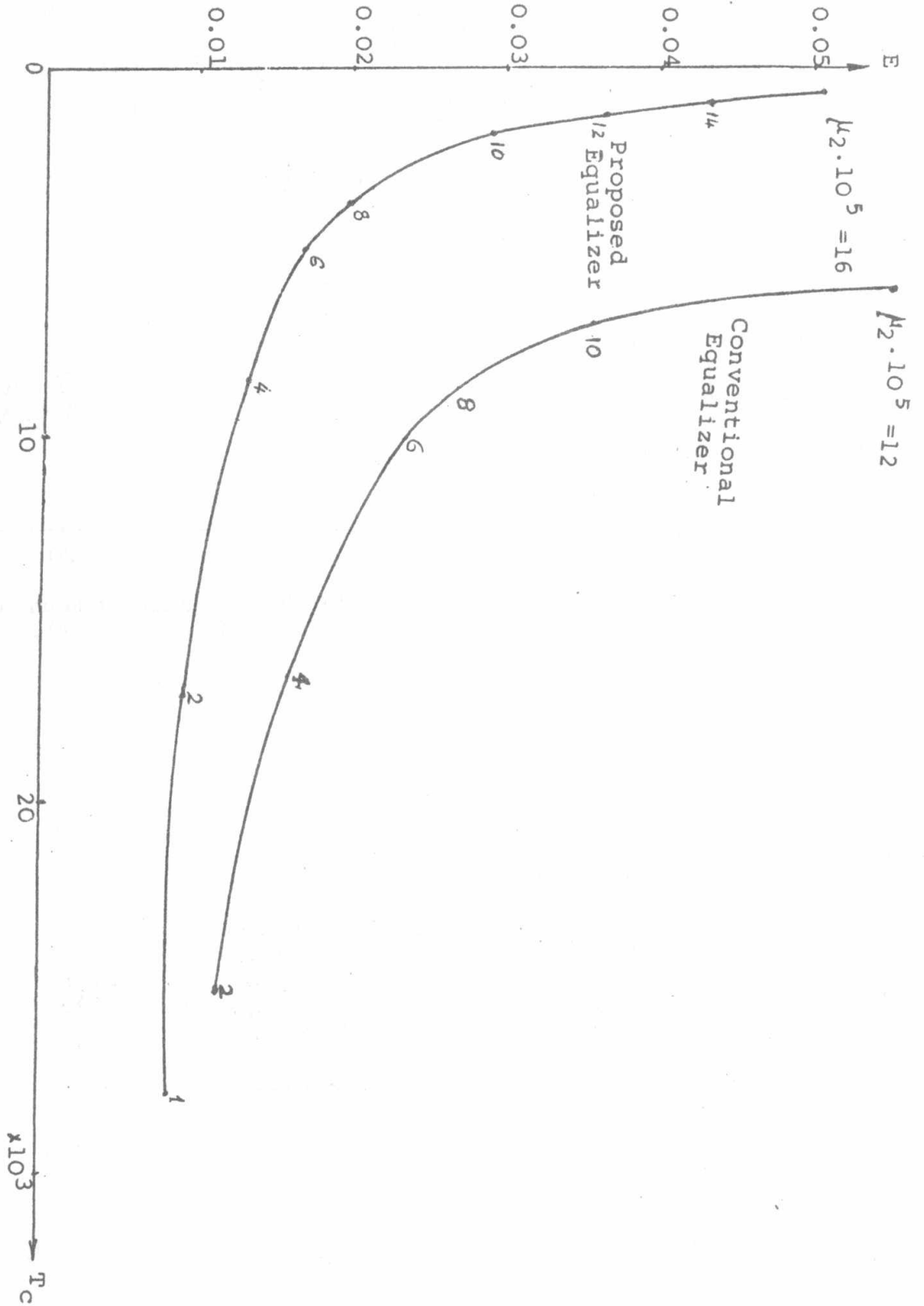Fig.7. Residual mean square error and convergence time at different values of μ_2. (L=8).
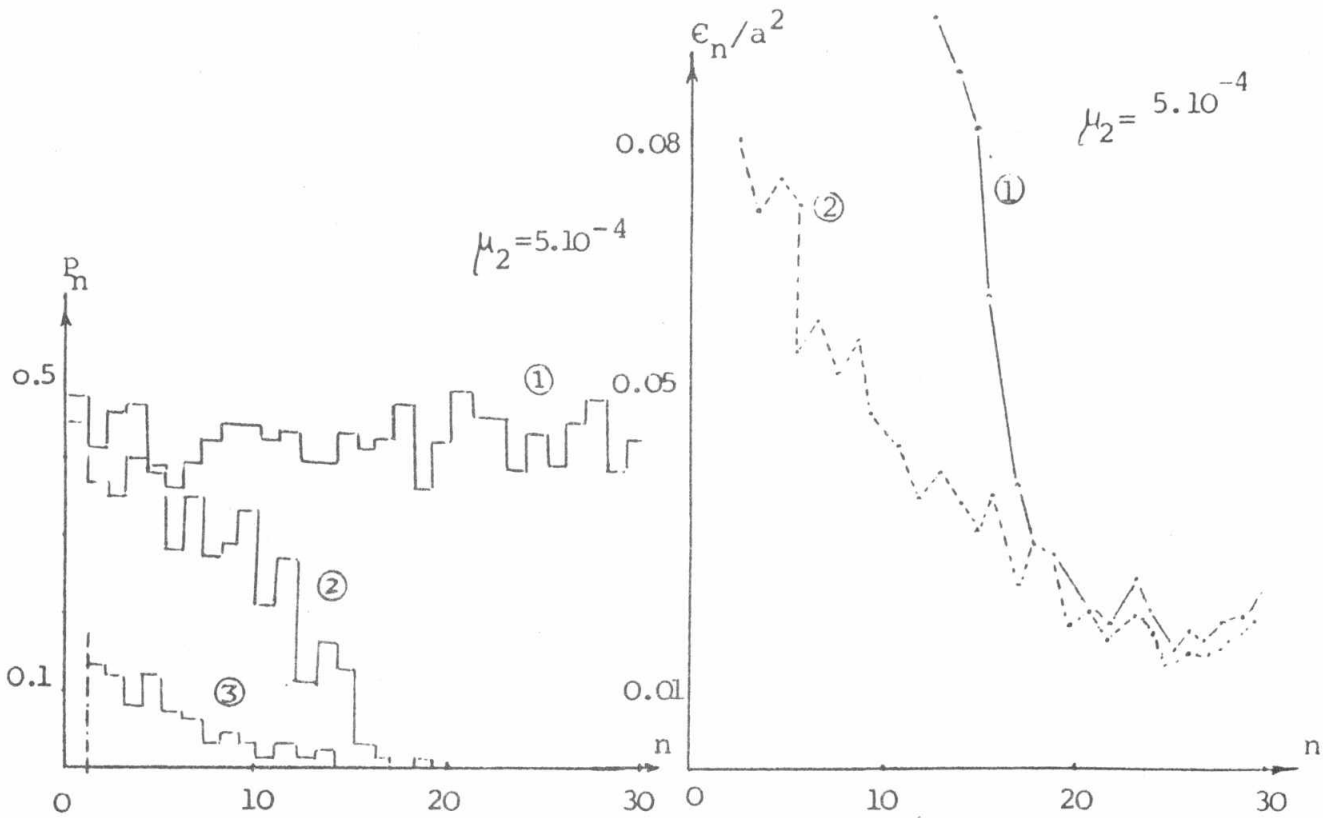
Fig. 8. Evolution of error rate. (L = 4).
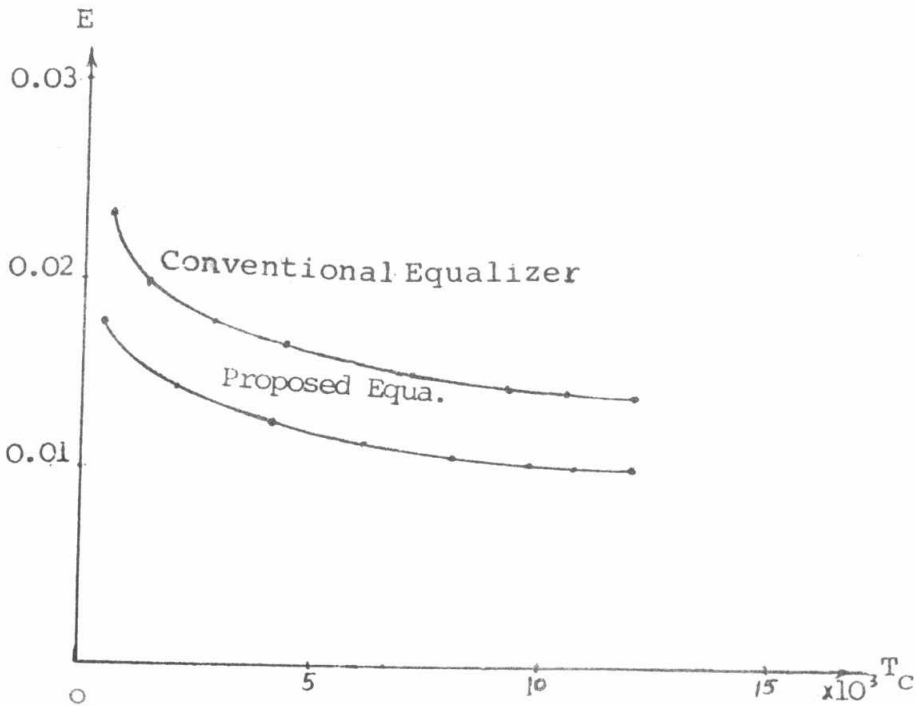

Fig.9. Evolution of mean square error . (L = 4).

Fig .10. Residual mean-square error and convergence time at different values of μ_2 . (L = 4).

V. CONLCUSIONS

In this paper we have introduced a self adaptive equalizer with
rapid acquisition. The results show that this scheme has faster
speed of convergence and smaller residual mean square error than
the conventional self adaptive equalizer. For example (see Fig.7),
the proposed self adaptive equalizer ensures a steady state mean
square output error of the value Q.012 with a convergence time of
2800 baud intervals. The conventional self adaptive equalizer
that ensures the same steady state error needs 11000 baud inter-
vals to converge.

The proposed scheme does not require more computing power than
the conventional self adaptive equalizers. That is due to the
fact that the two adaptive filters, of which the proposed self
adaptive equalizer consists, are updated sequentially.

REFERENCES

1. J.G.Proakis "Advances in equalization for intersymbol
   interference" in Advances in Communication Systems, Vol.4,
   New York : Academic Press, 1975, pp. 124-194.
2. B. Widrow and J.M. McCool "A comparision of adaptive
   algorithms based on the methods of steepest descent and
   random search,"IEEE Trans. on Antennas and Propagation,
   Vol. Ap-24, No. 5, Sept. 1976.
3. Y. Sato, "A method of self-recovering equalization for
   multilevel amplitude modulation systems", IEEE Trans.
   Comm., Vol. COM-23, Johe 1975, pp. 679-682.
4. D.N. Godard "Self-recovering equalization and carrier
   tracking in two-dimensional data communication systems",
   IEEE Trans. Comm., Vol. COM-28, Nov. 1980, pp. 1967-1975.
5. A. Benvensites,M. Goursat, and G.Ruget, "Robust identifi-
   cation of a nonminimum phase system : blind adjustment of
   a linear equalizer in data communication", IEEE Trans.
   Automatic Contr., Vol. AC-25, June 1980, pp. 385-399.
6. O.Macchi and E. Eweda, "Convergence analysis of self-
   adaptive equalizers", IEEE Trans. Inform. Theory, Vol.
   IT-30, No. 2, March 1984, pp. 161-175.
7. J.E. Mazo "Analysis of decision-directed equalizer conver-
   gence" BSTJ, Dec. 1980, pp.1857-1976.
8. B.Widrow & S.D. Stearns "Adaptive signal processing"
   Printice-Hall, Inc., Englewood chiff, New Jersey, 1985.
9. Shahidu H. Qureshi " Adaptive Equalization" Proceeding
   of the IEEE, Vol. 73, No. 9, Sept. 85 .