IMPROVED DATA DETECTION PROCESSES USING RETRAINING OVER TELEPHONE LINES

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ABSTRACT:
This paper describes two new developed detection processes for a modem over the public switched telephone network. The modem is a synchronous serial system using a 16-point QAM signal with a detector. The detector is preceded by an adaptive filter that is adjusted to make the sampled impulse response of the channel and filter minimum phase. The idea of these two new detectors is to transmit a known retraining data every specified interval and make use of these known data to improve the conventional nonlinear equalizer and near maximum likelihood detector.
Results of computer simulation tests are presented comparing tolerance to additive white Gaussian noise with retraining and with out retraining. The detectors with retraining achieve a better noise performance than conventional detectors. Furthermore the amount of retraining data is varied to find the best compromise between efficiency of data transmission and performance improvement

خلاصة:-
يصف هذا البحث طريقتين لعملية الكشف لمودم يعمل على خط التلفون: المودم يستعمل (16-QAM) مع الكاشف. يسبق الكاشف مرشح متكيف لجعل استجابة القناة المقطعة مع المرشح ذات حد ادنى للطور. فكرة هذان الكاشفان هو ارسال معلومات متفق عليها بشكل دورى والاستفادة من هذه المعلومات لتحسين اداء الكاشف نتائج المحاكاة باستخدام الحاسبة قدمت حيث تمت المقارنة بين اداء الكاشف مع إرسال المعلومات المتفق عليها وبدونها.استخدام المعلومات المتفق عليها ادى الى تحسن اداء الكاشفين بالنسبة للضوضاء. تم ايضا تغيرنسبة المعلومات المتفق عليها لايجاد افضل نسبة التى تجمع بين كفاءة الارسال مع جودة الاداء بالنسبة للضوضاء

KEYWORDS:
Detection,Equalizers,simulation,telephone channel,data transmission

INTRODUCTION:
In the search for a data- transmission system that achieves the fastest possible transmission over a linear channel ,a study must be made of the detection processes that

are the most tolerant to a considerable reduction in the bandwidth of the transmitted signal and hence a considerable intersymbol interference. The intersymbol interference can be removed by means of an equalizer which may be linear or nonlinear. But a better tolerance to noise is achieved through the use of a maximum likelihood sequence estimation implemented with Viterbi algorithm in place of the equalizer (Forney, 1972; Proakis,2001).

Unfortunately with severe intersymbol interference Viterbi algorithm becomes unacceptably complex, so that a simpler but somewhat less effective detector must be used. Considerable advances have been achieved in the development of a potentially cost effective detector, for the application just mentioned (Eyuboglu andQuereshi,1988;Duel-Hallen and Heegard,1989;Gerstacker and Schober,2002;Kamel and Bar-Ness,1994;Olivier et al,2003)(Xiang-guo and Zhi,2005)

A promising technique for overcoming this problem is to modify the Viterbi algorithm detector in such a way as to reduce both the amount of storage required and the number of operations per received data symbol but without reducing greatly the tolerance of the detector to noise, such systems are referred to as near maximum likelihood detector. These detectors, operate basically similar to the Viterbi algorithm but using a different selection process for the stored sequence of possible data symbol values, and only a very few of these sequences (typically 4-16) are stored with the corresponding costs (Clark et al ,1978;Clark and Fairfield,1981;Clark and Clayden,1984;Clark et al 1985;Clark and Abdullah,1987;Alhakim and Abduulah,1990)

* **System model**:

A model of the synchronous serial data transmission is shown in fig.1.

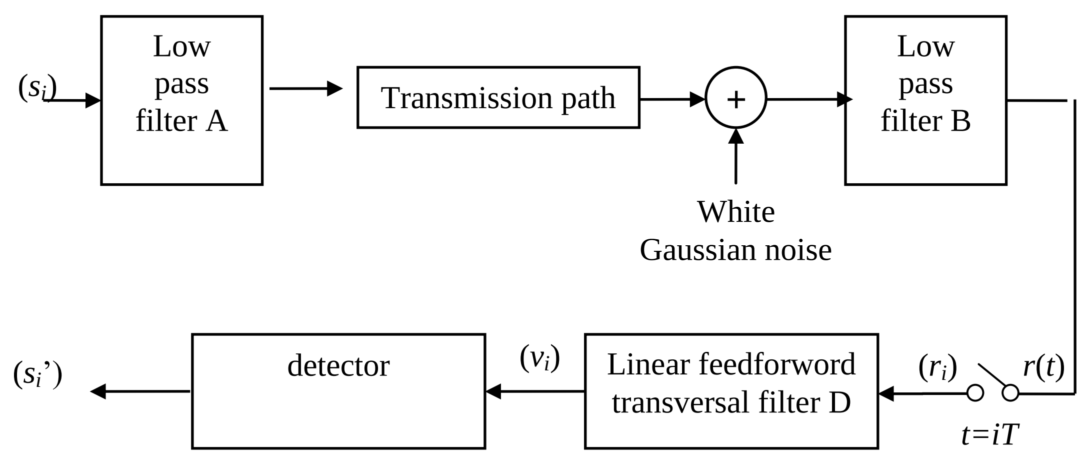

Fig.1 Model of data transmission system

The information to be transmitted is carried by the data symbol $\{s_i\}$ where

$$s_i = a_i + jb_i \qquad (1)$$

And $j = \sqrt{-1}$, a_i = ±1 or ±3 and b_i= ±1 or ±3 .

The $\{s_i\}$ values are statistically independent and equally likely to have any of their 16 possible values (16 point QAM). It is assumed that s_i =0 for i<0, so that s_i is the $(i+1)^{th}$ transmitted data symbol (Fig.2).QAM is used because it is a bandwidth efficient modulation technique

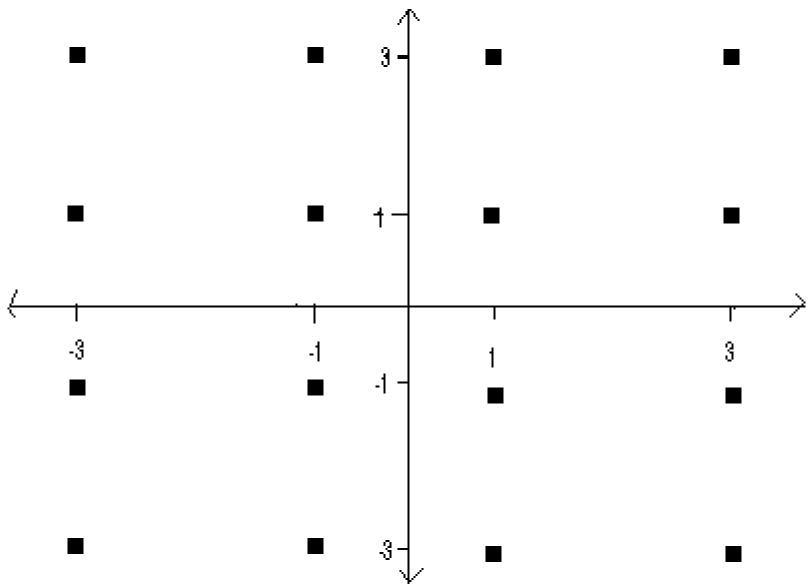

Fig. 2 16-point QAM signal constellation

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The linear baseband channel includes a low-pass filter A (transmitter output filter),a linear baseband transmission path and a low pass filter B(receiver input filter).The transmission path includes a telephone circuit, together with a linear modulator at the transmitter and a linear demodulator at the receiver. The resultant channel has an impulse response $y(t)$,which has ,for practical purposes, a finite duration while Gaussian noise is added to the data signal at the output of the transmission path to give the bandlimited Gaussian noise waveform $w(t)$,with zero mean and fixed variance, at the output of the low pass filter B. The output waveform for the linear baseband channel is

$$r(t) = \sum_i s_i y(t - iT) + w(t) \qquad (2)$$

where $r(t)$, $y(t-iT)$ and $w(t)$ are complex valued in the case of telephone circuits, the shape of $y(t-iT)$ is independent of iT, such that $y(t-iT)$ is $y(t)$ delayed by iT seconds. The received waveform $r(t)$ is sampled at the time instants$\{iT\}$ to give the samples $\{r_i\}$, which are fed to the linear feedforward filter D, whose corresponding output samples are the $\{v_i\}$. The real and imaginary parts of the noise components in the $\{r_i\}$ are statistically independent Gaussian random variable with zero mean and fixed variance. The resultant sampled impulse response of the linear baseband channel and filter is given by the vector (sequence)

$\boldsymbol{E}$=[1 e_1 e_2 …… e_g] (3)

The delay in transmission over the channel and filter D, other than that involved in the time dispersion of the signal, is neglected here, so that the first component of E (with value unity)is taken to have no delay. Also e_i =0 for $i < 0$ and $i > g$.
The linear filter D is an all pass network that adjusts the sampled impulse response of the channel and filter to be minimum phase .This filter may be adjusted adaptively without too much complexity(clark and hau,1987).The ideal adjustment of the filter is assumed throughout this paper. The signal at the output of the filter D, at time $t=iT$, is

$$v_i = \sum_{n=0}^{g} s_{i-h} e_h + u_i \qquad (4)$$

Where the real and imaginary parts of the $\{u_i\}$ are statistically independent Gaussian random variable with zero mean and fixed variance σ^2. The detector has exact prior knowledge of E and the possible values of s_i. It uses this prior knowledge to determine the detected values of $\{s`_i\}$ from its input signals $\{ v_i \}$.

CONVENTIONAL NONLINEAR EQUALIZER:
In the conventional nonlinear equalizer just prior to the detection of si, the equalizer form

$$z_i = \sum_{n=1}^{g} \acute{s}_{i-h} e_h \qquad (5)$$

This is an estimate of the intersymbol interference in z_i and is subtracted from v_i to give the equalized signal

$$f_i = v_i - z_i \qquad (6)$$

Which is fed to a threshold level detector, with correct detection of the data symbols s_{i-1} ,s_{i-2}, ……, s_{i-g} the equalized signal becomes

$$f_i = s`_i\, e_o + w_i \qquad (7)$$

s_i is now detected as its possible value $s`_i$ such that $s`_i\, e_o$ is closest to f_i. The incorrect detection of one or more of the data symbols s_{i-1} ,s_{i-2}, ……, s_{i-g} leads to intersymbol interference in f_i and so greatly increase the probability of error in s_i. Errors in the detected data symbols therefore tend to occur in bursts (Clark and clayden, 1984)

CONVENTIONAL NONLINEAR EQUALIZER WITH RETRAINING:
Here it is similar to the conventional nonlinear equalizer except that at the end of training period it starts with correct detected data symbols $\{s_i\}$(fig. 3)from the known retraining period. Hence there will be a correct cancellation of intersymbol interference at each retraining period (eq. 6) .Fig.3 shows how the sequence of information data and training is transmitted. Furthermore it shows the various amount of training that is transmitted in each case

Retraining is already used in communication systems(such as GSM)(Olivier et al,2003).This detector makes use of it.

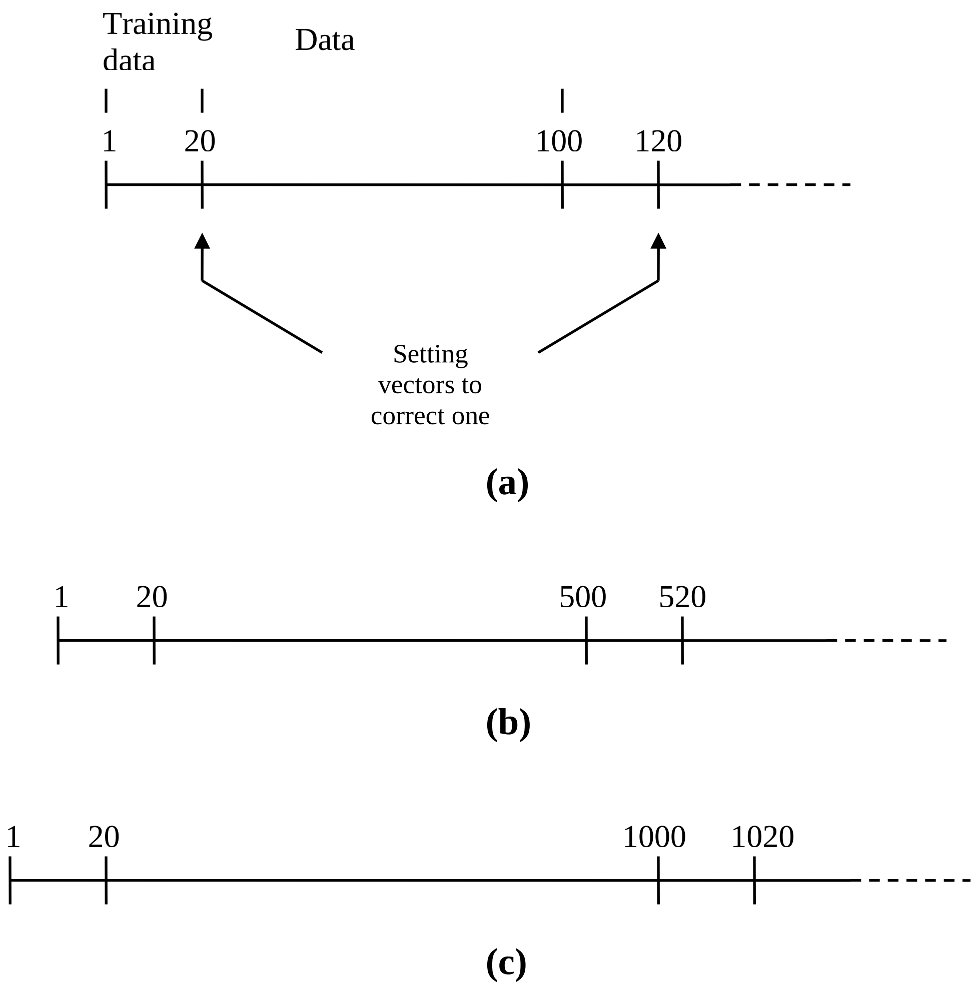

Fig.3 Different retraining cases **(a)** 20-100, **(b)** 20-500, and **(c)** 20-1000

NEAR MAXIMUM LIKELIHOOD DETECTOR (CLARK AND ABDULLAH, 1987):

The data transmission is as shown in fig.1, just prior to the receipt of the signal v_i from the filter D. The detector holds in store 16 n-component vector (sequence)$\{Q_{i-1}\}$where

$$Q_{i-1} = [x_{i-n} \quad x_{i-n+1} \quad \dots \quad x_{i-1}] \qquad (8)$$

And x_{i-h} takes on a possible value of s_{i-h} for h=1,2,… n. Each vector Q_{i-1} represents a possible sequence of values of the data symbols s_{i-n} s_{i-n+1} … s_{i-1}. Each of these vectors is first expanded to give the corresponding vector P_i

$$P_i = [x_{i-n} \;\; x_{i-n+1} \;\; \ldots \; x_{i-1} \; x_i \,] \qquad (9)$$

And hence the corresponding vector Q_i with the smallest cost. The cost is calculated from the equation

$$C_i = C_{i-1} + \left| v_i - \sum x_{i-n} e_n \right|^2 \qquad (10)$$

Where C_{i-1} is the cost of the vector Q_{i-1} from which P_i was derived. Three vectors $\{Q_i\}$ are added to each of the two smallest cost vectors, that differ only in the last component and with smallest cost. One vector Q_i is added to the third and fourth smallest costs of the set of 16,differing only in the last component and with smallest cost. No vectors are added to the four vectors $\{Q_i\}$ having the fifth to eighth smallest cost of the original set of 16 and the remaining eight of this set of vectors are discarded. There are now 16 vectors $\{Q_i\}$ together with their costs. The detected. Value s_{i-n}' of the data symbol s_{i-n} is now taken as the value x_{i-n} in the vector Q_i with smallest cost in the final set of 16 vectors

NEAR MAXIMUM LIKELIHOOD DETECTOR WITH RETRAINING:

Its operation is similar to near maximum likelihood detector above except that at the end of every training period the $\{Q_i\}$ vectors are set to the correct one (correct $s_{i-1}, s_{i-2}, \ldots, s_{i-g}$) (during the training period the data transmitted is known so it starts with correct vector) During normal data transmission if due to noise the minimum cost vector becomes different from the correct one the training period reinitializes it with correct one to start again from a correct vector(fresh start)

COMPUTER SIMULATION TESTS:

Computer simulation tests have been carried out on the various systems described here to determine their relative tolerance to additive white Gaussian noise when operating over models of two different telephone circuits. The systems are as shown in fig.1 and described in sections 3-6The telephone circuits 1 and 2 are close to the typical worst circuits on the public telephone network in the UK. Table 1 shows the sampled impulse response of the linear baseband channel and adaptive filter in fig 1.

Figs 3-6 show the performance of the various systems over the telephone circuits 1 and 2.The signal to noise ratio (SNR)is here taken to be ψ dB where

$$\psi = 10 \log \frac{5}{\sigma^2} \qquad (11)$$

The mean-square value of s_i being 10 and the variance of v_i being $2\sigma^2$

The delay in detection for near maximum likelihood detector has been set equal to n=20

Table 1 : Sampled impulse response of baseband channel and adaptive filter in fig. 1, for each of the two telephone circuits(Clark and clayden,1984)

Telephone circuit 1		Telephone circuit 2	
Real part	Imaginary part	Real part	Imaginary part
1.0000	0.0000	1.0000	0.0000
0.4861	1.0988	0.2544	1.9941
-0.5980	0.0703	-1.7394	-0.2019
0.1702	-0.1938	0.6795	-0.8086
-0.0245	0.1000	0.0408	0.5113
0.0100	-0.0258	-0.1189	-0.1463
-0.0134	0.0110	0.0343	0.0420
0.0056	-0.0042	-0.0185	-0.0364
0.0003	0.0003	0.0139	0.0261
-0.0008	0.0041	-0.0102	-0.0009
0.0000	-0.0061	-0.0019	-0.0034
0.0007	-0.0007	0.0037	-0.0046
0.0037	0.0002	-0.0028	-0.0006
-0.0019	-0.0025	-0.0019	-0.0046
0.0020	0.0008	0.0083	0.0022
0.0005	-0.0002	-0.0056	0.0059
-0.0022	0.0002	-0.0046	-0.0028
0.0007	-0.0005	0.0049	-0.0019
-0.0008	0.0002	-0.0009	0.0037
0.0005	0.0005	-0.0009	0.0003

It can be seen from figs 4 and 5 for nonlinear equalizer with various amount of retraining that for telephone circuit1 at an error rate of 10^{-3} an improvement of around 3 db is gained by using 20% retraining(20-100).Furthermore 1.5 db is gained for 4% retraining(20-500)

For telephone circuit 2 which has more severe intersymbol interference at error rate 10^{-3} there is around 4 db improvement when using 20% retraining and 2 db for 4% retraining

Furthermore in the case of near maximum likelihood detector figs 6 and 7 for telephone circuit 1 at 10^{-3} error rate there is around 1.5 db improvement for 20% retraining over no retraining, and 1 db for 4% retraining

Finally for telephone circuit 2 and near maximum likelihood detector there is 2 db improvement for 20% retraining over the conventional near maximum likelihood detector(no retraining) at an error rate of 10^{-3} and 1 db for 4%retraining

Error rate in the $[s_i]$

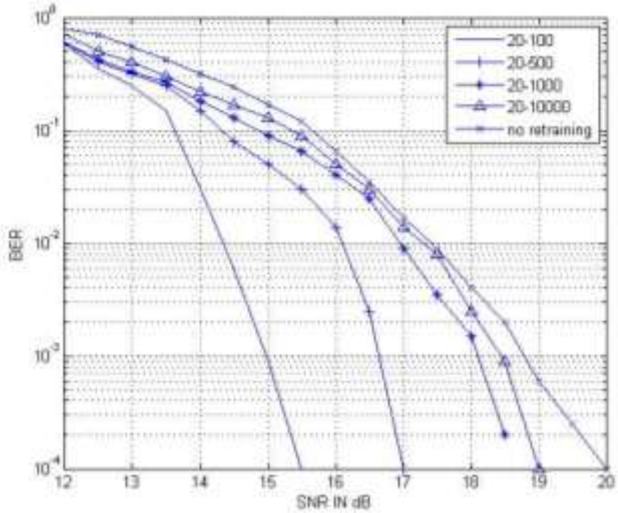

Fig4 Variation of error rate with SNR for nonlinear equalizer over telephone circuit 1

Error rate in the $[s_i]$

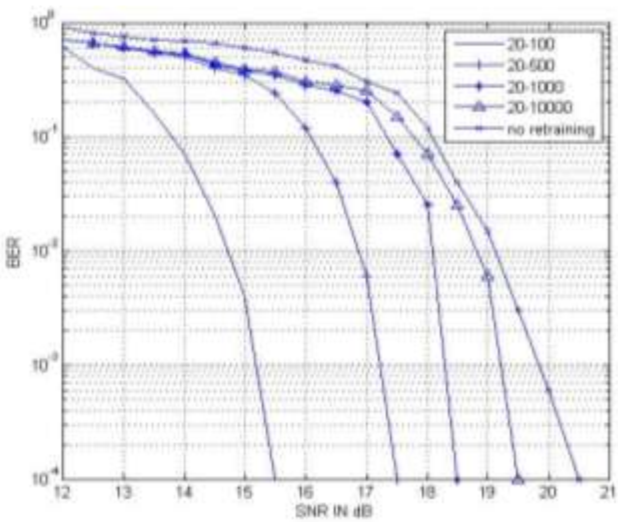

Fig5 Variation of error rate with SNR for nonlinear equalizer over telephone circuit 2

Error rate in the [s_i]

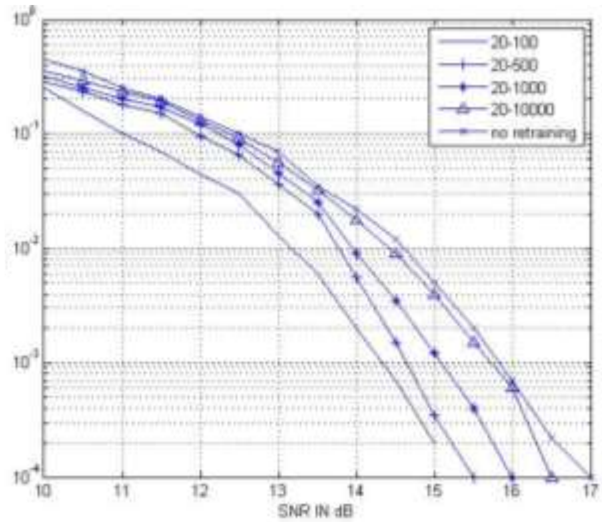

Fig6 Variation of error rate with SNR for near maximum likelihood over telephone circuit 1

Error rate in the [s_i]

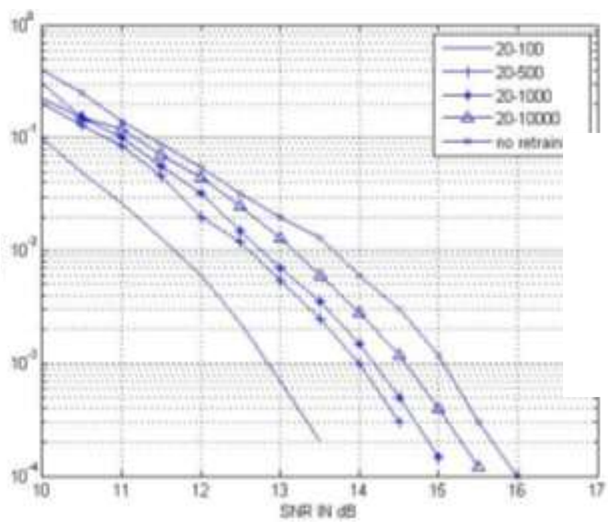

Fig7 Variation of error rate with SNR for near maximum likelihood over telephone circuit 2

CONCLUSIONS:

Retraining of near maximum likelihood detector and conventional nonlinear equalizer improves its noise performance at the expense of reducing efficiency of data transmission. Furthermore it can be concluded that the improvement obtained in the case of nonlinear equalizer is more than that of near maximum likelihood detector. The nonlinear equalizer with retraining seems to be a promising detector that combines both simplicity and performance. The retraining amount could be decided according to the application

Future work may include using retraining with Viterbi algorithm to improve its performance. Also fading channels may be used instead of telephone circuits

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