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AN IMPROVED ADAPTIVE MLSE RECEIVER FOR FADING COMMUNICATIONS CHANNELS

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In a mobile cellular systems the fast varying channel characteristics due to fading and to vehicle speed, require a more efficient design of the adaptive receiver. The paper shows the performance improvement obtained by using adaptation algorithm to the maximum likelihood sequence estimation (MLSE) receiver when the tracking of rapidly time-varying distortion is needed. A variety of the most common adaptive algorithms are considered and a more efficient version of the least mean square algorithm (LMS) to recover the Doppler effect is also proposed.

1.INTRODUCTION

The narrow-band time division multiple access (TDMA) digital cellular systems require adaptive demodulator to combat the intersymbol interference (ISI) resulting from the time-varying multipath propagation of the signal through the channel. Due to the multipath nature of the communication channels the energy associated with the transmission of one symbol is smeared across several symbol periods.
An efficient non linear equalization method to combat ISI is the maximum likelihood sequence estimation (MLSE) which is implemented by means of the Viterbi algorithm (VA).
The MLSE data detection technique requires an estimate of the communications channel response.
There are two aspects for the channel estimation problem:
- channel estimation from a training sequence
- channel tracking

The known training sequence at the beginning or in the middle of each data packet (burst), is used to estimate an initial channel response and can be held for the entire burst. However, depending on the rate at which the channel is time-varying, due to fading and to the vehicle speed, there may be a need to further track the channel variations during the informative data sequence.
The paper first presents classic algorithms, such as the least mean square (LMS) and the recursive least squares (RLS) algorithms, to update the channel coefficients to the MLSE receiver when the tracking of rapidly time-varying distortions is needed and then a modified LMS algorithm is proposed leading to better performance within the burst period.
The performance of the MLSE receiver in the two operating modes, training mode and tracking mode, in term of bit error rate (BER) versus E_b/N_0 is also presented.

2. DIGITAL RECEIVER ARCHITECTURE

A large set of digital modulation schemes of interest for mobile communications, including PAM, QAM and the class of Continuous Phase Modulation (CPM) can be represent with a linear model [1]. The overall communication channel characteristics can be also modelled with a discrete-time equivalent linear model. Therefore, the received signal y(nT) may be written in the form

$$y(nT) = \sum_{k=0}^{N-1} h_n(kT)\, a_{n-k} + w(nT) = \qquad (1)$$

$$= s(nT) + w(nT)$$

where
- n is integer
- $a_n = \pm 1$ are the transmitted data symbols
- T is the sampling period assumed equal to the symbol period
- $h_n(kT)$ is the complex time-varying lowpass equivalent impulse response of the all transmission system including the pulse shape modulation filter, the channel and the receiving filter
- w(nT) are the samples of an additive noise term
- N is the equivalent channel impulse response length
- s(nT) are the samples of the signal component

The best theoretical performance for demodulating operations over linear ISI channels is the MLSE technique. The MLSE criterion leads to a receiver that searches among all possible data sequences to find the sequence which is transformed by the channel impulse response in an estimated signal which is closest to the noisy received signal in the sense of Euclidean distance. In other words it selects the data sequence that minimizes the quantity

$$D = \sum_i | y(iT) - \hat{s}(iT) |^2 \qquad (2)$$

where the summation is computed on the block of the last M received signal samples.

The implementation of the MLSE criterion with the above metric can be efficiently performed by the well-known Viterbi algorithm (VA).

The VA algorithm requires the knowledge of the equivalent channel impulse response $h_n(kT)$ in order to compute the metrics for making decisions on the actual sequence. In the absence of such knowledge the channel must be estimated. The channel estimator is a FIR transversal filter, say of order N.

The MLSE adaptive receiver consists essentially of two blocks: the channel estimator and the Viterbi processor as shown in Fig. 1.

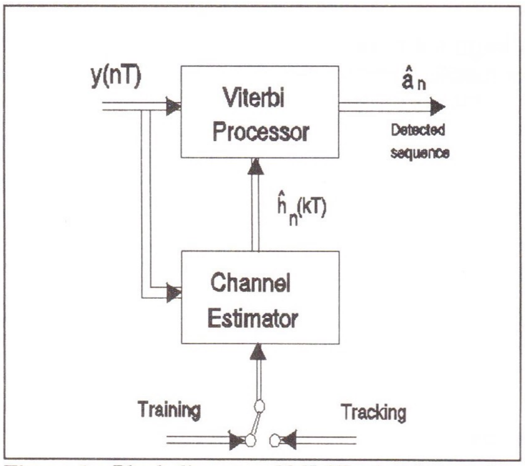

Figure 1 - *Block diagram of MLSE adaptive receiver*

There are two aspects for the channel estimation:

- channel estimation from a known training sequence
- channel tracking to compensate the fast variations of the communications channel characteristics

In the training mode the channel is estimated using the received signal resulting from a known sequence sent by the transmitter. The training sequence often precedes the data blocks and is termed preamble or as in the data packet format adopted by the Group Special Mobiles system (GSM), it is a pseudorandom sequence of 26 bits in the middle of the data packet termed midamble [2].

By using a correlation method, where a portion of the training sequence is correlated with shifted versions of the received signal, an initial estimate of the communication channel during the actual burst can be performed. This estimate can be held fixed during the burst and is updated only the next burst.

Considering a GSM system application, the preamble position in middle of the burst allows to produce a channel impulse response estimate that in many situations is sufficiently accurate for the whole duration of the burst even in presence of small Doppler effects. For this reason the Viterbi algorithm is initialized at the extremes of the midamble and then works on the informative sequence in direct and inverse propagation. As expected the performance of the receiver is worse at the beginning and at the end of the burst, as shown in Fig.2, and when the channel characteristics vary very rapidly as in the case of high speed of the mobile, it is necessary to update the fast channel response variations by a receiver adaptability (tracking mode).

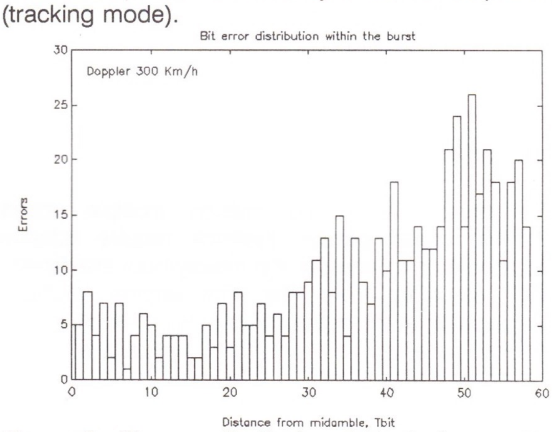

Figure 2 - *Bit errors distribution within the burst ,vehicle speed equal to 300 km/h, Eb/N0=8 dB*

3. MLSE RECEIVER WITH CHANNEL ADAPTIVITY

In the mobile communications system very fast adaptation methods are necessary to compensate for the time and the frequency selective distortions of the channel varying very rapidly with respect to the bit rate. This is particular evident in the case of high vehicle speed of the order of 300 km/h or higher where it is necessary to update the channel coefficients within the burst (tracking mode).

The most commonly used criterion to optimize the channel coefficients is the minimization of the mean square error (MSE) between the received signal and the estimated signal samples. The error is defined as

$$\begin{aligned} e(n) &= y(nT) - \sum_k h_n(kT)\hat{a}(n-k) \\ &= y(nT) - \hat{s}(nT) \end{aligned} \qquad (3)$$

where $h_n(kT)$ is the vector of the estimated channel coefficients at the instant nT, $\hat{a}(nT)$ is the demodulated data symbol.

The minimization of the MSE carries to the Wiener - Hopf solution where the computation of the autocorrelation matrix inverse of the vector of the estimated data is needed. The optimum solution for the channel coefficients is

$$\underline{h}_{OPT}(n) = R^{-1}(n)\, \underline{Z}(n) \qquad (4)$$

R(n) = data autocorrelation matrix at time instant n
$\underline{Z}$(n) = vector of cross correlations between the data symbol and the received signal samples at time n.

This expression is generally too expensive for an implementation. Its usefulness is limited to the evaluation of a bound for the adaptive receiver performance assuming to know the data sequence.
Alternatively, the minimization of the mean square error may be obtained recursively using the LMS algorithm [3]

$$h_{(n+1)}(kT) = h_n(kT) + \Delta e(n)\, \hat{a}(n-k) \qquad k=0,1\ldots N-1, \forall n \qquad (5)$$

where Δ is the step size selected as a compromise between the speed of convergence and the stability. As shown in literature [4], the convergence rate of the LMS algorithm is slow and the LMS algorithm is more sensitive to data demodulation error and noise. For a faster convergence algorithm, the recursive least square (RLS) criterion can be used to update the channel coefficients. Instead of minimizing the MSE of (3) the RLS algorithm minimizes an esponentially weighted squared error

$$E(n) = \sum_{i=0}^{n} w^{n-i} |y(i) - \sum_{k=0}^{N-1} h_i(kT)\, \hat{a}(i-k)|^2 \qquad (6)$$

The exponential weighting factor w is selected to be in the range 0<w<1 and the quantity 1/(1-w) is the fading memory in the estimation of the channel coefficients.
The RLS algorithm has been developed to solve recursively in time the Wiener-Hopf equation for the optimum vector $h_n(kT)$.
The resulting time-update relationship for the channel coefficients is

$$\underline{h}_{n+1} = \underline{h}_n + \underline{K}_{n+1}\, e_n \qquad (7)$$

where $\underline{h}_n$ is the vector of the N channel coefficients. The rapid convergence of the RLS algorithms is controlled by the Kalman gain vector

$$\underline{K}_{n+1} = \frac{P_n \underline{\hat{a}}_n}{w + \underline{\hat{a}}_n^{/} P_n \underline{\hat{a}}_n} \qquad (8)$$

where $\underline{\hat{a}}_n$ is the estimated data vector at time n and P_n is the inverse of the data autocorrelation matrix at time n, computed as

$$P_{n+1} = \frac{1}{w} [P_n + \underline{K}_{n+1} \underline{\hat{a}}_n^{/} P_n] \qquad (9)$$

Eqs. 8 and 9 can be computed recursively. We considered as P_0 a NxN matrix derived from the training sequence and $\underline{\hat{a}}_n$ is the vector of the last N estimated data.
In the RLS algorithm w has to be optimized to fit the speed of system variations and in this paper the optimization of w has been evaluated by computer simulation of different system conditions.
A modified LMS algorithm is proposed and tested in our work which is demostrated very efficient for tracking the channel variations expecially for high speed of the vehicle.
In this case the modified LMS minimizes the sum of squared errors

$$E(n) = \sum_{i=n-L}^{n} |\, y(i) - \sum_{k=0}^{N-1} h_n(kT)\, \hat{a}_{i-k}\, |^2 = \sum_{i=n-L}^{n} |e_n(i)|^2 \qquad (10)$$

where the length of the sum can be selected, (in our case L=15).
The updating of the channel coefficients is effectuated as

$$\underline{h}_{n+1} = \underline{h}_n + \Delta \sum_i e_n(i)\, \underline{\hat{a}}_i \qquad (11)$$

In our approach the adaptation algorithm supplies the channel coefficient estimate to the Viterbi demodulator for use in metric computations.
The channel estimate can be evaluated by using the data sequence of a particular survivor in the branch metric calculation during the Viterbi trellis. The selected data sequence in our case corresponds to the minimum state metric at time nT, and it is used to update the estimate ŝ(nT) of all the survivors. This local estimate is therefore updated continually within the burst during the entire informative sequence. It must be stressed that the selected minimum metric at each time nT is only selected for the updating of channel coefficients. The detection of the transmitted data sequence is performed by the Viterbi algorithm at the end of each burst.

4.RESULTS AND CONCLUSIONS

The performance of the digital MLSE receiver in the case of only training mode and in the one with tracking mode using different adaptive algorithms to update the channel estimate are compared. The performance of the receiver is evaluated using a channel simulator suitable for mobile communications and in particular according to the GSM system recommendations [2].
The simulated channel impairments are:
- flat Gaussian noise
- Rayleigh (or Rice) fading with Doppler frequency shift and multiple echoes according to the ETSI/GSM specifications as representative of rural area channel (RA) [2].

The following assumptions are also made:
- the normalized bandwidth of the premodulation filter in the GMSK transmitter is BT = 0.3 (the bit rate is 1/T= 270.833 kb/s).
- the baseband receiver filter has a 3 - dB bandwidth two-sided equal to 160 kHz
- the equivalent channel impulse response length is N=5
- the receiver structure includes a 16-state Viterbi receiver and a 26 bit midamble is used to set the receiver at the beginning of each burst.

The figures show the performance results of the various tracking adaptive algorithms considering as a lower bound limit the curve traced by the Wiener-Hopf optimum solution that is calculated using the true transmitted sequence. This is the performance of the theoretically optimum adaptive receiver based on the knowledge of the channel impulse response during the whole burst. Of course, in pratice this knowledge is not available and an estimate must be obtained.
We have proved that the use of the tracking mode in the adaptive MLSE receiver is convenient for a narrow TDMA system with the ETSI/GSM characteristics to cope with the rapidly varying channel characteristics within the burst period, due to Doppler effect.
The analyzed adaptation algorithms assure a better performance of the receiver in terms of BER for high vehicle speed.
With respect to the Doppler effect (Fig. 3) the results show that, for tracking purposes, the simple LMS algorithm works about as well as the more complicated RLS techniques on channels with high Doppler effects.
The proposed modified LMS algorithm achieves a performance much better than the others and keeps the BER less than 10^{-2}, which is assumed by ETSI/GSM as a quality threshold for the whole range of specified values of echo delay $0 \div 16 \mu s$ and for a wide range of E_b/N_0.

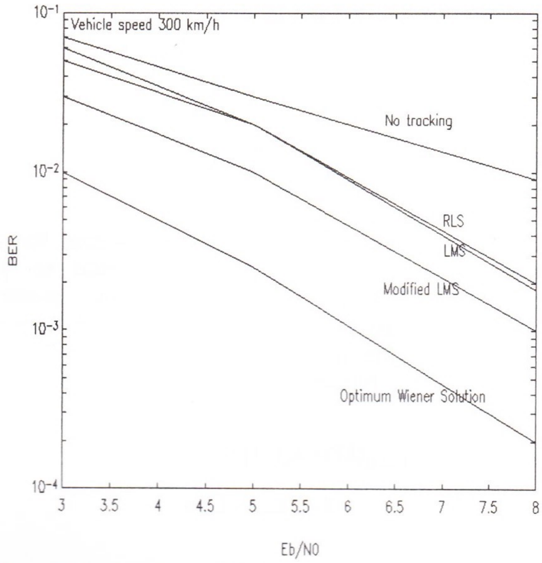

Figure 3 - Vehicle speed of 300 km/h without multipath case

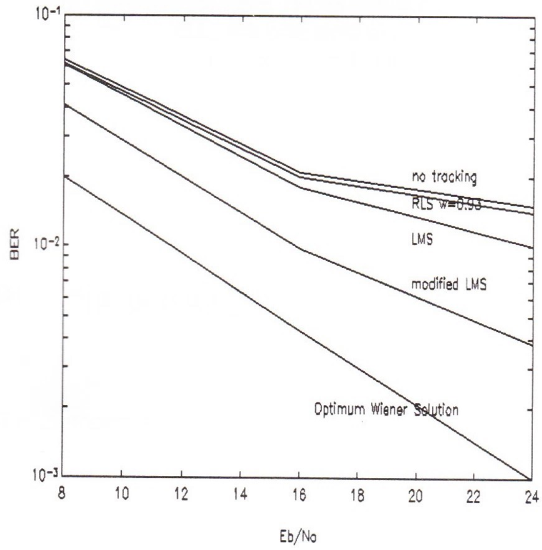

Figure 4 - RA case with vehicle speed of 250 km/h

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