

## A Channel Tracking and Normalization Equalizer

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**Abstract.** An adaptive reduced state maximum a posteriori (RS-MAP) equalizer based on per-survivor processing (PSP) in conjunction with a local decision feedback equalizer (DFE) for the EDGE system with 8PSK modulation is proposed in this paper. A pre-filter prior to the MAP-PSP is exploited to reshape the channel impulse response (CIR) to have approximately a minimum phase property. A trade-off between the performance and complexity can be achieved by using the DFE. The simulations show that the RS-MAP-PSP can improve the performance of the EDGE system, especially for the high coding rate traffic and/or under the high speed environments.

**Keywords:** Equalizer, PSP, EDGE, MAP

### 1. Introduction

Mobile radio communication channels are time-varying channels, which are characterized by the presence of both delay and Doppler spreading, and produce inter-symbol interference (ISI). It is well known that MLSE implemented by the viterbi algorithm (VA) is an efficient detection technique for a digital signal corrupted by ISI and AWGN [1, 2]. Due to the high order modulation and fast time-varying ISI channels specified in EDGE systems [3], the MLSE-PSP is becoming an attractive approach by updating the channel parameters for survivor of each state in the VA and eliminating the tentative decision delay via the PSP [4, 5].

However, with a long CIR length and high order modulation, the number of the states in the VA is extremely large. Sub-optimal methods must be sought to make the MLSE applicable. Mainly, there are two categories for reducing the complexity of MLSE receiver employing the VA. The first category directly shortens the overall effective CIR length handled in the VA by DFE in order to reduce the total number of states in the trellis diagram. The second category selects a subset of total number of trellis states, as in the RSSE [6]. RSSE is an approach that reduces the complexity of VA receiver but still offer at least partial performance gains of MLSE. The RSSE algorithm is suboptimum but less complex alternatives to VA. It can be consider as the combination of VA and DFE-Decided Feedback Estimation. It was suggested by Eyuboglu and Qureshi [8]. Among those various approaches, the delayed decision feedback sequence estimation (DDFSE) [7] in the first category and RSSE with the set partitioning principle in the second one are promising candidates in the sense of making good trade-off between the performance and complexity. The performance of the RSSE-PSP algorithm [9] can't reach the standard of the 3GPP for the high coding rate traffic.

In order to reduce the performance loss and pass the standard mentioned in the 3GPP, a reduced state maximum a posteriori (RS-MAP) algorithm based on PSP is proposed. MAP equalizer is based on the theory of maximum a priority; it involves three variables-forward metrics, branch metrics and backward metrics. The simulations show that the performance of the RS-MAP-PSP proposed in this paper is better than the RSSE-PSP, and the RS-MAP-PSP algorithm can reach the 3GPP performance standard.

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## 2. System Overview

In this paper, a multi-path fading and AWGN are considered. Source consisted of random bits are formatted to the EDGE burst structure and gray mapped to 8PSK symbols. Every burst includes a training sequence of 26 symbols in the middle, 3 tail symbols at either end, and 8.25 guard symbols at one end as shown in Fig.1.



Fig.1: EDGE Burst structure

The gray coding, modulation and pulse shaping filtering of the EDGE signal, which is  $3\pi/8$ -shifted 8PSK, is shown in Fig.2.

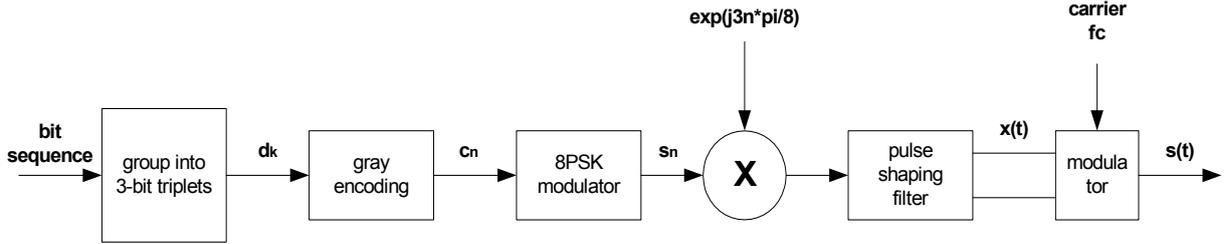


Fig.2: Modulation of EDGE signal

After 8PSK modulation, the signal is corrupted by ISI and AWGN. Thus the received sequence  $y_k$  can be expressed as

$$y_k = \sum_{i=0}^L h_i a_{k-i} + \eta_k \quad (1)$$

where  $h_i$  denotes the discrete overall impulse response with length  $L$ ,  $a_k$  is the transmitted 8PSK symbol and  $\eta_k$  is a AWGN sequence. The received signal  $y_k$  is fed to a pre-filter, in which the settings are calculated based on the estimated CIR by the least squares algorithm with mid-amble. The aim of the pre-filter is to reshape the estimated CIR to have approximately a minimum phase property.

## 3. Prefilter for 8PSK Signals

For high performance of reduced-state equalization such as RS-MAP equalizer, a discrete-time pre-filter is required in front of equalizer to transform the channel impulse response into its minimum phase or maximum phase equivalent. Root-finding, MMSE-DFE and Linear-Prediction (LP) are three common approaches for pre-filter calculation. From the practical implementation consideration, the LP method is much less complicated.

The pre-filter  $A(z)$  can be viewed as a cascade of two filter  $A1(z)$  and  $A2(z)$ , i.e.  $A(z) = A1(z)A2(z)$ . Where  $A1(z)$  is a matched filter, and  $A2(z)$  may be viewed as a noise whitening filter. Basically, the  $A2(z)$  can be solved efficiently through a prediction-error filter:

$$1 - P(z) = 1 - \sum_1^{N_p} p(k)z^{-k} \quad (2)$$

The prediction filter  $P(z)$  is designed for minimization of the output power of the error filter. The optimum coefficients for this criterion are given by the solution of the Yule-Walker equations:  $\Phi_{hh} p = \varphi_{hh}$  where

$$\Phi_{hh} = \begin{bmatrix} \varphi_{hh}(0) & \varphi_{hh}(-1) & \dots & \varphi_{hh}(-N_p + 1) \\ \varphi_{hh}(1) & \varphi_{hh}(0) & \dots & \varphi_{hh}(-N_p + 2) \\ \cdot & \cdot & \cdot & \cdot \\ \cdot & \cdot & \cdot & \cdot \\ \varphi_{hh}(N_p - 1) & \varphi_{hh}(N_p - 2) & \dots & \varphi_{hh}(0) \end{bmatrix}$$

$$p = [p(1) \quad p(2) \quad \dots \quad p(N_p)]^T$$

$$\varphi_{hh} = [\varphi_{hh}(1) \quad \varphi_{hh}(2) \quad \dots \quad \varphi_{hh}(N_p)]^T$$

where  $\varphi_{hh}(k)$  is the autocorrelation of lag  $k$  computed as:

$$\varphi_{hh}(k) = \sum_{m=0}^{L-1-k} h(m) * h^*(m+k) \quad (3)$$

$\Phi_{hh}$  is a Hermitian, positive-definite and Toeplitz matrix. The AR coefficients of Yule-Walker equation can be solved through Levinson-Durbin Algorithm (LDA)

Supposed that the  $n$  taps of the pre-filter based on the estimated CIR are calculated, and labelled as  $P_n$ . Then the CIR and the received signal  $y_k$  are fed to this filter, and their outputs are expressed as

$$pre\_y_k = filter(P_n, 1, y_k) \quad (4)$$

$$pre\_h_i = filter(P_n, 1, h_i) \quad (5)$$

The pre-filtered signal  $pre\_y_k$  is then processed in the RS-MAP-PSP with the initial estimated reshaped CIR  $pre\_h_i$ . In order to track the channel variations, the estimated reshaped CIR  $pre\_h_i$  in the acquisition phase is updated with the LMS or RLS algorithm for every survivor inside the PSP in the tracking phase.

#### 4. Maximum a Posteriori Equalizer

The MAP (Maximum A Posterior) equalization algorithm computes the probability of each transmitted bit given the channel outputs. This provides the optimal soft bit information that can be passed to the channel decoder. The MAP algorithm can be described by forward and backward passes over the trellis with the MAX operation to simplify the practical implementation.

$$\alpha_n(j) = \underset{i}{MAX} \{ \alpha_{n-1}(i) + \gamma_n(i, j) \}$$

$$\beta_n(j) = \underset{i}{MAX} \{ \beta_{n+1}(i) + \gamma_{n+1}(i, j) \}$$

$$\gamma_n(i, j) = | r(n) - \sum_{k=0}^{L-1} f(k)x(n-k) |^2 \quad (6)$$

The MAX operation is

$$MAX(x, y) = \log(e^x + e^y) = \max(x, y) + \log(1 + e^{-|x-y|}) \quad (7)$$

In the reduced-State MAP equalizer, the channel taps are divided into two groups, and then the transition metric will be expressed as

$$\gamma_n(i, j) = | r(n) - \sum_{k=0}^{\mu-1} f(k)x(n-k) - \sum_{k=\mu}^{L-1} f(k)\hat{x}(n-k) |^2 \quad (8)$$

Now we apply the PSP principle [11] to the RS-MAP. When calculating the forward factor alpha, the transition metric from the  $l$ -th state at time index  $k$  to the  $m$ -th state at time index  $k+1$  as

$$\lambda^{l \rightarrow m}(t_k^l \rightarrow t_{k+1}^m) = | y_k - (X(t_k^l \rightarrow t_{k+1}^m), \hat{h}_k^l) |^2 \quad (9)$$

where the expression  $(X(t_k^l \rightarrow t_{k+1}^m))$  is an inner product between the symbol vector at time index  $k$  and the vector  $\hat{h}_k^l$ . The channel updating, by the LMS algorithm, is as follows:

$$\hat{h}_{k+1}^m = \hat{h}_k^l + \beta e^{l \rightarrow m}(t_k^l \rightarrow t_{k+1}^m) X^*(t_k^l \rightarrow t_{k+1}^m) \quad (10)$$

where  $X^*(t_k^l \rightarrow t_{k+1}^m)$  is the symbol vector,  $e$  is the error signal and  $\beta$  is the step size.

The structure of the RS-MAP-PSP equalizer is diagramming in Fig.3

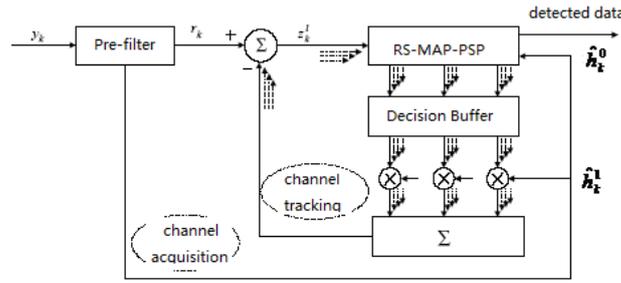


Fig.3: the system structure of the RS-MAP-PSP

## 5. Normalized Soft Information

Since the signal quality of each burst is different, a normalization model is needed to normalize the soft information from the output of the equalizer.

First, the noise energy is estimated using by the hard symbol and the received data;

$$En = \text{sum}(|y - \hat{h} * \text{hard\_inf}|^2) \quad (11)$$

Second, the soft information is divided by the noise energy estimated in the first step.

$$\text{soft\_inf} = \text{soft\_inf} / En \quad (12)$$

## 6. Simulation Result

The simulation results are all based on the TU50 channel environment, and the MCS5 and MCS9 are chosen as the traffic channel. The trellis state of the RS-MAP is set as 8, and the tracking step of the PSP is fixed on the 0.02.

Fig.4 shows the performance comparison between RSSE (64 states) [10] and MAP (8 states). The result is that the performance of MAP equalizer is better about 2dB than RSSE while the FER is 10%. The gain doesn't consider the soft information normalization. If considering the normalization, the gain will be more than 2 dB.

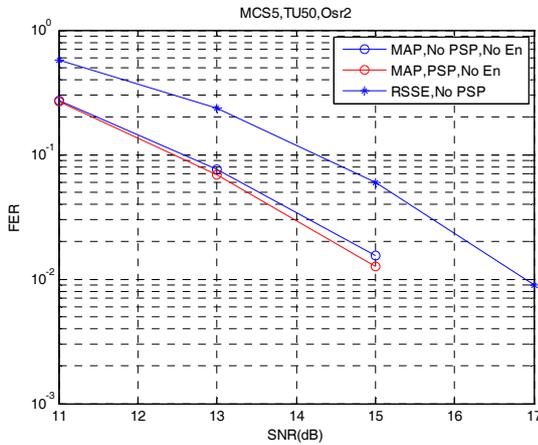


Fig.4: The performance comparison between RSSE and MAP

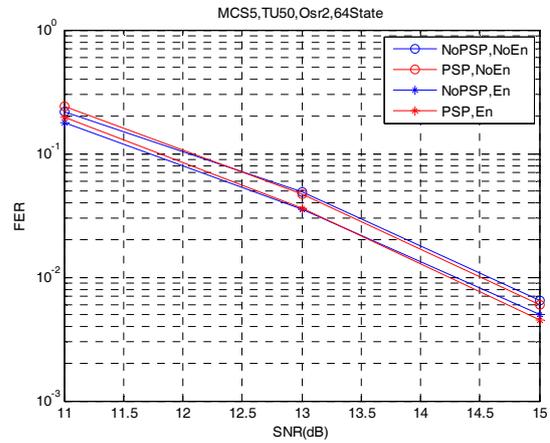


Fig.5: The performance comparison between normalization and not

Fig.5 shows that the normalization based on noise energy can obtain about 0.5dB gain under MCS5 and TU50 conditions, whatever conjoint with PSP or not.

Fig.6 and Fig.7 show the effect of PSP on the performance under low or high coding rate. Taking the MCS5 for example as the low coding rate of traffic, whose results show in the Fig.6. The result shows that the PSP algorithm only obtains about 0.5 dB for low coding rate. The high coding rate, taking MCS9 for example, has more than 4 dB gains due to the PSP algorithm, which shown in the Fig.7.

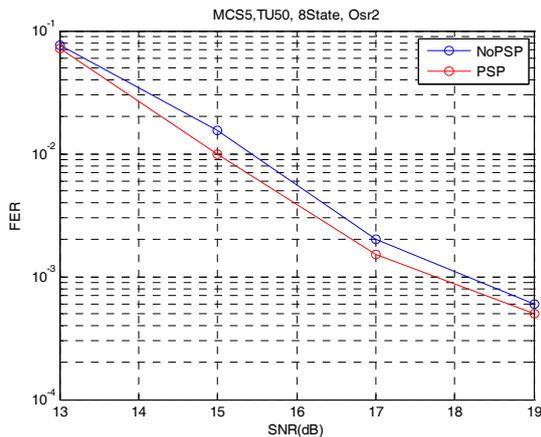


Fig.6: The performance of MCS5 with and without PSP

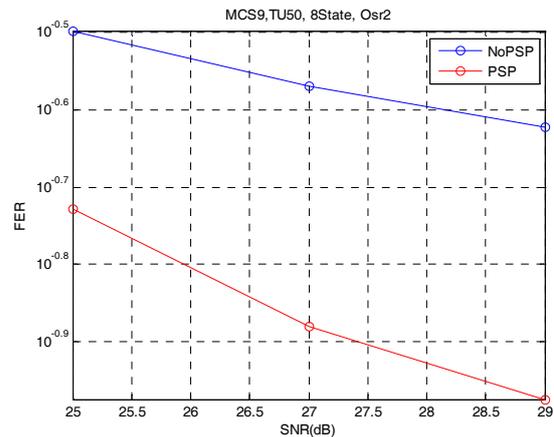


Fig.7: The performance of MCS9 with and without PSP

## 7. Conclusion

This paper proposed a reduced state maximum a posteriori (RS-MAP) algorithm based on PSP to improve the performance of EDGE equalizer. The simulation results show this proposed algorithm can achieve a better FER than the RSSE-PSP equalizer, especially for high coding rate traffic. The noise energy normalization is necessary for the equalizer, which can also contribute performance gain to the whole system. Since the complexity of this algorithm proposed in this paper is higher than RSSE, reducing the complexity is our plan in the next step.

## 8. References

- [1] G. D. Forney, "Maximum-Likelihood Sequence Estimation of Digital Sequence in the Presence of Inter-symbol Interference," *IEEE Transactions on Information Theory*, Vol. 18, No. 5, May 1972, pp. 363-378.
- [2] ETSI, GSM 05.04 V8.1.0 (1999-12), "Digital Cellular Telecommunications System (Phase2+); Radio Transmission reception," GSM 05.04, Version 8.1.0, 1999.
- [3] R. Raheli, A. Polydoros and C-K. Tzou, "Per-survivor Processing: A General Approach to MLSE in Uncertain Environments," *IEEE Transactions on Communications*, Vol. 43, No. 2/3/4, February/March/April 1995, pp. 354-364.
- [4] Z. Li and A. Mämmelä "Adaptive PSP-MLSE Receiver for GSM/EDGE System in Fast Multipath Fading Channels," *IEEE International Conference on Personal and Wireless Communications*, December 2000, Hyderabad, India, pp. 184-188.
- [5] M. V. Eyuboglu and S. U. H. Qureshi, "Reduced-State Sequence Estimation with Set Partitioning and Decision Feedback," *IEEE Transactions on Communications*, Vol. 36, No. 1, January 1988, pp.13-30.
- [6] J. B. Anderson and S. Mohan, "Sequential Coding Algorithm: A Survey and Cost Analysis," *IEEE Transactions on Communications*, Vol. 32, No. 2, February 1984, pp. 169-176.
- [7] S. J. Simmons, "Breadth-First Trellis Decoding with Adaptive Effort," *IEEE Transactions on Communications*, Vol. 38, No. 1, January 1990, pp. 3-12.
- [8] A. Duel-Hallen and C. Heegard, "Delayed Decision-Feedback Sequence Estimation," *IEEE Transactions on Communications*, Vol. 37, No. 3, May 1989, pp.428-436.
- [9] N. Al-Dhahir and J. M. Cioffi, "Fast Computation of Channel-Estimate Based Equalizers in Packet Data Transmission," *IEEE Transactions on Signal Processing*, Vol. 43, No. 11, November 1995, pp. 2462-2473.
- [10] Zhenhong Li, Piirainen O., Mammela, A. "An adaptive RSSE-PSP receiver with a pre-filter for EDGE systems", *IEEE International Conference on Communications*, 2003(5): 3594-3598.