# Mixture Filtering Approaches to Blind Equalization Based on Estimation of Time-Varying and Multi-Path Channels

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***Abstract:*** **In this paper, we propose a number of blind equalization approaches for time-varying and multi-path channels. The approaches employ cost reference particle filter (CRPF) as the symbol estimator, and additionally employ either least mean squares algorithm, recursive least squares algorithm, or H$\infty$ filter (HF) as a channel estimator such that they are jointly employed for the strategy of "Rao-Blackwellization," or equally called "mixture filtering." The novel feature of the proposed approaches is that the blind equalization is performed based on direct channel estimation with unknown noise statistics of the received signals and channel state system while the channel is not directly estimated in the conventional method, and the noise information if known in similar Kalman mixture filtering approach. Simulation results show that the proposed approaches estimate the transmitted symbols and time-varying channel very effectively, and outperform the previously proposed approach which requires the noise information in its application.**

***Index Terms:*** **Blind equalization, cost reference particle filter (CRPF), frequency selective fading, H$\infty$ (H infinity) filter, least mean squares, mixture filtering, Rayleigh fading, recursive least squares.**

## I. INTRODUCTION

GENERALLY, a wireless communication channel is considered as a time-varying and multi-path channel which distorts the transmitted signal, or incurs fading of it. If a wide-band signal is employed in a wireless communication system, the channel is considered as a frequency-selective fading channel when it is safely assumed that the coherence bandwidth of the channel is less than the bandwidth of the transmitted signal: under frequency-selective fading, different frequency components of the signal undergo different gains and phase shifts across the band, and a tapped-delay line model for the received signal describes frequency-selective and slow fading channel which also has significant effect of inter-symbol interference (ISI).

In this paper, we propose approaches that jointly estimate the time-varying channel impulse response (CIR) and transmitted symbols based on the received signal that had gone though frequency-selective and Rayleigh fading, without the aid of a training sequence; besides, the noise statistics of the received signal are not known. The main purpose of the blind equalization is estimating the transmitted symbols regardless of whether the channel is estimated or not. Blind equalization is desirable for some applications, such as multipoint or group communications. The history of the blind equalization may originate from [1] where "a stochastic gradient algorithm" was employed. With the beginning of the work in [1], the blind equalization approaches may be classified into three main categories as follows [2].

Approaches in the first category search for the solution by using the method of "steepest decent," which is a traditional iterative procedure, and has been used to find extremes of nonlinear functions. Due to the difficulty to acquire the correlation of the estimation error and the received signal, usually least mean squares (LMS) algorithm is employed where a one-point sample mean of the estimation error and the received signal replaces the correlation of the error and the received signal. Due to its disadvantage of the slow convergence property, recursive least squares (RLS) algorithm is an alternative which minimizes the "least squares" instead of "mean-square error," and consequently, no statistical information about the received signal and the transmitted input data is required. The reference [3] has been popularly cited in the literature. There are some more papers that proposed the approaches based on "steepest decent" algorithm [1], [4], [5]. A notable feature of the proposed approach in [3] is that any noise statistics are not required to be known nor assumed to be Gaussian, which might be a big advantage in practice. Nonetheless, its well known disadvantage is that the algorithm converges slowly; besides that, it requires a constraint that all elements of the initial equalizing coefficients of the reference taps except for one element should be zero, and the non-zero element must be greater than a threshold that is determined by the true initial channel impulse response for reliable convergence even though the the constraint is a sufficient condition rather than a necessary condition. The approaches in the second category are based on the second-order or higher-order statistics of the received signal. Based on the second-order statistics (autocorrelation) of the received signal, CIR is estimated by using the cyclostationary property of the received signal [6], [7]. The channel also can be estimated explicitly from the forth-order cumulants of the received signal [8], [9]. The tricepstrum equalization algorithm (TEA) based on the forth-order statistics of the received signal was proposed in [10], where the trispectrum is the three dimensional discrete fourier transform of the fourth-order cumulant sequence. Finally, methods in the last category employ the maximum likelihood (ML) ap-

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