# Fast Fading Channel Estimation using Kalman Filter for Turbo Receivers

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***Abstract*** **- An efficient, low complex and near optimal approach to pilot assisted fast fading channel is estimated for single carrier modulation with a turbo receiver. This method is applicable to higher order modulation schemes, where the detector is sensitive to the estimation error. A doubly selective fast fading channel is estimated using a fixed lag Kalman filter. The Kalman filtering is followed by a zero phase low pass filter which acts as a smoother. A method for designing the smoother is introduced. Block processing method is used to reduce the transient effects of the zero phase filter at the edges of the symbol blocks. A first-order autoregressive (AR) model is fitted to the channel variations. For 64 Quadrature Amplitude Modulation (QAM), a complex exponential basis expansion model (CE-BEM) is exploited to capture the varying gains. The long memory of the smoother reduces the error-rate floor associated with low-order channel estimation models. The applicability of the method to 64-QAM scheme is shown through simulation experiment.**

***Key words*****- Basis expansion models (BEMs), channel estimation, fast-fading channels, Kalman filters (KFs), turbo equalization.**



## I. INTRODUCTION

Channel estimation technique is widely used in turbo receivers to estimate the parameters of time and frequency selective fading channels in wireless channels. In turbo receivers the soft information is exchanged iteratively between the equalizer and decorder inorder to improve the performance [1]. Blind and semiblind techniques for channel estimation is employed for slow fading channels but their performance decreases when applied to higher order modulation schemes where the channel estimation needs to be more accurate.

An efficient low complexity channel is estimated with a near optimal error performance. It can be used in high speed and high data rate vehicular communication systems. Other applications include control signaling for remote operated aerial vehicles, high reliability and rapid communications for emergency vehicles, video conferencing on high speed trains, etc., where communication is to be performed over rapidly varying channels.

The channel estimation and equalization is done separately using two cascaded low order kalman filters. A Zero Phase Filter (ZPF ) is used as a smoother inorder to suppress the out of band estimation errors from the channel estimator KF's output. The combination of ZPF and KF reduce the estimation error. The smoothing function is applied to each channel path independently, hence this method efficiently fits on multicore implementations.

The outline of this paper is as follows. In section II, the proposed model and the properties are discussed. In section III the simulation result of the proposed model is shown. In Section IV, some concluding remarks are given.

## II. PROPOSED SYSTEM

In this section, the formulation of the proposed model is stated. The transmitter structure, channel model, receiver, Zero phase filter design, Iterative equalizer and decoder is proposed.

### *A. Transmitter*

A bit-interleaved coded modulation system transmitting over a time varying fading channel is considered as shown in Fig 1. The notation for single carrier signaling and samples are used from [6]. A block of independent data bits $\{b(k), k = 1,2,\dots N_d\}$ is encoded by a convolutional encoder with code rate $R$. The encoded sequence $c(k)$ is given to a bitwise random interleaver $\pi(\cdot)$ of length $N_i$, generating the interleaved coded sequence $\{c(k), k = 1,2\dots N_i\}$. The interleaved data are modulated according to some constellation $X$, mapping every $N_{mod}$ into a constellation point. Following the time multiplexed training scheme in [15], a sequence of pilot symbols $l_p$ is periodically inserted per $l_s$ data symbols to form a transmit sequence $\{s(n), n = 1,2,\dots N\}$. The symbol sequence s(n) is assumed to be zero mean and have unit mean power.

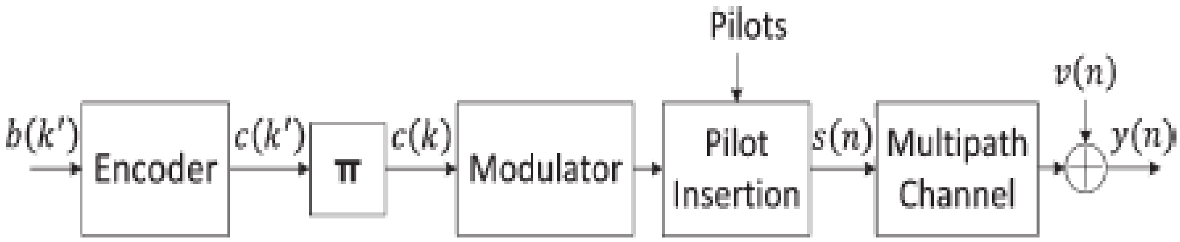


Fig. 1. Transmitter structure

*B. Channel Model*

A doubly selective multipath channel can be modeled as a linear time varying FIR filter with $L+1$ taps. The discrete time signal at the receiver input can be expressed as

$$y(n) = \sum_{l=0}^{L} g(n;l)s(n-l) + v(n)$$

$$= g^T(n)s(n) + v(n) \qquad (1)$$

for $n=1,2,\ldots N,$ where $v(n)$ denotes the Gaussian zero mean complex white noise with variance $\sigma^2_v$, and

$$g(n) := [g(n;0)\ g(n;1) \ldots .\ g(n;L)]^T \qquad (2)$$

Here, a complex exponential basis expansion model (CE-BEM) with Q=1 basis function is used for 4-QAM and 64-QAMThe channel impulse response $g(n;l)$ is expressed in terms of the complex exponential basis expansion model coefficients $hq(n;l)$ as

$$g(n;l)= \sum_{q=1}^{Q} hq(n;l)\ e^{j\omega qn} \qquad (3)$$

Using (1) and (3), the received signal is given as

$$y(n) = \sum_{l=0}^{L}\sum_{q=1}^{Q} hq(n;l)e^{j\omega qn}s(n\text{-}l)+v(n) \quad (4)$$

*C. Receiver*

The turbo receiver consist of channel estimator, equalizer and convolutional decoder modules each exchanging soft information with each other. Channel estimation is done with pilots. The pilots contain zero information for the decoder and are removed from the equalizer output before being forwarded to the decoder.

Data symbols are detected with the equalizer. The output of the equalizer, including the data symbol estimates $s(n)$ and their estimated variance $\sigma^2(n)$, is to generate the extrinsic LLRs for the coded bits $L^M_e\{c(k)\}$. The extrinsic information on a given bit is obtained by subtracting the input LLR from the output LLR to block the positive feedback preventing convergence. For each iteration, the extrinsic information from the detector channel estimation block is fed to the decoder, whereas the information from the decoder generated in the previous iteration is fed to the channel estimation block.

In the decoder, $L^M_e\{c(k)\}$ is deinterleaved to provide the soft input to the SISO convolutional decoder. The SISO decoder produces LLR information on coded bits, which is denoted with $L^D_a\{c(k)\}$. This LLR information is then used to generate updated symbol estimates $s(n)$ and their variance $\gamma(n)$, which are used by the channel estimator and the equalizer. At the same time, the extrinsic information $L^D_e\{c(k)\}$ is extracted to be fed back to the SISO demapper to further improve the next-round decisions on data symbol.

*D.Zero Phase Filter*

The IIR filter of the ZPF is designed to match the characteristics of the fading process. The IIR filter parameters to be determined are the passband edge frequency *fp*, passband ripple *Rp*, stopband attenuation *Rs*, and cutoff edge frequency *fa*. The channel gain variations are band limited to Doppler frequency *fD*. Thus, one sets *fp* = *fD*. The Doppler frequency is considered to have a maximum value. The impact of this over estimation on the performance will be demonstrated through simulation.



Consider a single channel path. The input to the ZPF is the estimated channel gain $\hat{g}(n, l)$ from the KF, which is given by

$$\hat{g}(n; l) = g(n; l) + e(n; l) \qquad (5)$$

where $e(n; l)$ denotes the estimation error of the KF for path $l$. The estimation error $e(n; l)$ is assumed to be uncorrelated with $\hat{g}(n, l)$ and have constant PSD, i.e., $See(f; l) = \sigma 2v$, uniformly distributed over the normalized frequency range of $[-1/2, 1/2]$. At each iteration, the estimated channel gains from the KF are passed through the ZPF with the forward (and backward) magnitude response $A(f)$. The PSD of the output of the ZPF for propagation path $l$, i.e., $S\hat{g}g(f; l)$, can be written as

$$S\hat{g}g(f; l) = A^4(f)\,[Sgg(f; l) + See(f; l)] \qquad (6)$$

where $Sgg(f; l)$ is the PSD of $g(n; l)$, band limited to $[-fD, fD]$.

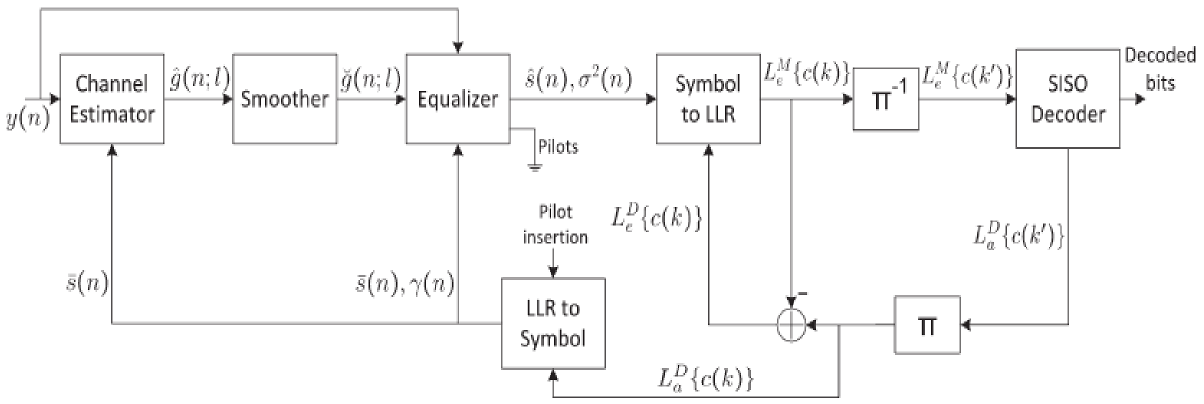


Fig. 2. Receiver structure

## III. SIMULATION RESULTS

The performance of this method is demonstrated by evaluating the BER versus SNR for 64-QAM constellations.

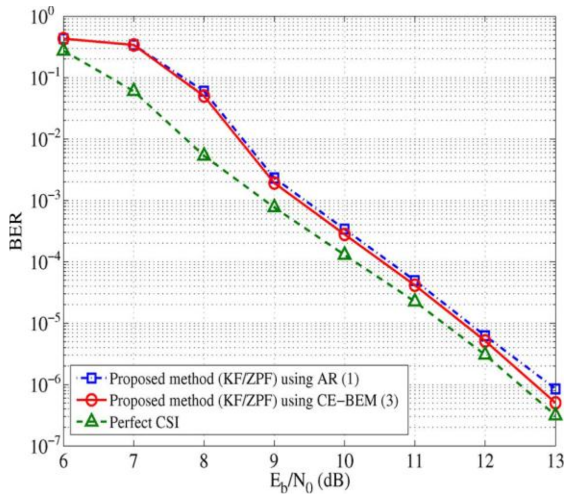


Fig. 3. BER versus $E_b/N_0$ for a 64-QAM receiver

In Fig. 3, the case of a 64-QAM scheme is shown, where Q=1 and Q=3 models performances are compared with the perfect CSI receiver. The EKF method did not reliably converge for the 64-QAM scheme; therefore, the corresponding results are not included in this figure. It can be seen that the Q=1 model performs almost as well as the CE-BEM for BER < $10^{-6}$ but starts to deteriorate afterward. With the proposed method, all the BER curves are within 0.3dB of the perfect-channel receiver. In addition, the system convergence is fast. The average number of iterations it takes for system to converge was approximately three iterations per trial for the case of the 64-QAM receiver.

The 64-QAM setup was also used to contrast the performance of the ZPF with that of FIR and IIR filters in Fig. 4, to justify the use of a ZPF. The IIR filter is designed using the same specifications as the IIR component of the ZPF, except for the passband ripple and stopband attenuation in decibels being doubled (since the magnitude response of a ZPF is equivalent to two cascaded component IIR filters). The FIR filter was designed using the least squares method, where the parameters were selected to be the same as the ZPF designed previously. It can be seen that it requires 2000 taps for an FIR filter to achieve the same performance as a ZPF with only a fifth-order component IIR filter, which is an obvious cost savings. Moreover, the IIR filter introduces a phase distortion that significantly degrades the performance of the receiver.



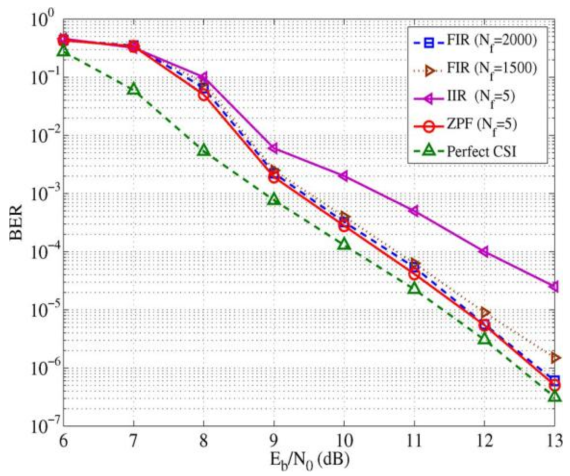


Fig. 4. BER versus $E_b/N_0$ for a 64-QAM scheme, comparing ZPF, ordinary IIR, and FIR filters.

As shown, the proposed method provides excellent performance at the SNR values required for low error reception.

## IV. CONCLUSION

A low cost ZPF is applied to the output of a channel estimator KF to accurately estimate a fast fading channel in a turbo receiver. The long memory of the smoother can reduce the estimation error less than 2dB of the wiener bound without using high order KFs. The simulation results for 64-QAM receiver is shown.

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