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New Channel Estimation Based Decoding Scheme for TURBO-OFDM Systems

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Abstract:

OFDM is an effective technique for combating multipath fading and for high-bit-rate transmission over mobile wireless channels. In using turbo codes for OFDM performance can be sufficiently improved as seen in LTE standard systems. Through this thesis, an efficient channel estimation scheme suitable for LTE standard systems is devised. We estimate the channel by considering it as a problem of estimating vectors from its linear transforms, given the initial probability. Pilot Assisted channel estimation is used with each stage of the estimator iteratively feeding the decoder, so that the algorithm breaks the feedback inside the Turbo decoder and updates the soft inputs from the channel using an updated channel estimate before each iteration. Simulations showed that the proposed joint iterative estimation and decoding technique improves the system's performance over the conventional technique.

Keywords: LTE system, OFDM, turbo codes, channel estimation

1. Introduction

Orthogonal frequency-division multiplexing (OFDM) is a method of encoding digital data on multiple carrier frequencies, which are transmitted in parallel, thereby increasing the symbol duration and reducing the ISI. Therefore, OFDM presents one of the most reliable multiplexing schemes for fading channels. In order to coherently detect a signal being transmitted over a multipath fading channel, accurate channel estimation is essential. Turbo codes have been shown to exhibit near-capacity performance over flat-fading channels with coherent detection and perfect knowledge of the channel response. However, mobile communication systems are characterized by channel responses with time-varying magnitude and phase. Different channel estimation technique had been derived for this purpose.

Channel estimation techniques improve the decoder performance. Pilot-aided techniques are the most commonly used channel estimation methods of which, channel estimation based on iterative techniques are recently in use due to increase in estimation accuracy. But these schemes along with turbo decoding iterations result in a very complex system. Therefore a new channel estimation technique which could be applied to the turbo coded (LTE standard) is to be focused in case of OFDM systems with performance comparable to that of the existing iterative estimation system but will significantly reduce the complexity. Attempts in this direction include an adaptive channel estimation and tracking scheme based on recursive least squares [1]. Preamble aided channel estimation is performed in time-domain (TD). The estimator is then extended to perform decision-directed (DD) channel tracking during data transmission. Also estimation using adaptive filters such as Kalman filter [2] is common. A three-stage estimation scheme is proposed to reduce the complexity and adapt the channel estimates w.r.t the feedback information. In [3], authors incorporate pilot symbols into the transmission. Like pilot tones, pilot symbols are used at the receiver to obtain an estimate of the channel response so that coherent detection can be performed. Following the convention we can use the term pilot symbol assisted modulation (PSAM) to describe systems that incorporate pilot symbols in the transmission. For uncoded systems, the use of pilot symbols is more power efficient than the use of pilot tones.

In this paper, the channel estimation scheme works in synchronism with iterative decoding. The iterative turbo decoder feedback loop is broken and the estimator is inserted and the soft channel input is updated before each stage of decoding, New estimate is formed by the updated decoder decision variable.

2. System Model

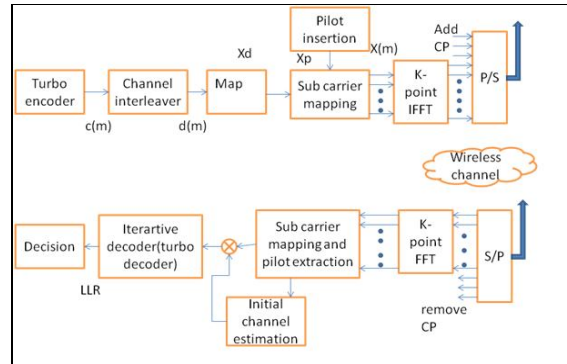


Fig 1:General LTE system model-OFDM

The transmitter and receiver of the LTE system is shown in Figure. The information bits are encoded, interleaved, and mapped into M-ary complex symbols forming the data sequence X_d . This sequence is then pilot sequence is inserted and IFFT is performed to optimize this for OFDM transmission. The information sequence is now in time domain. The data in parallel form is then converted to serial form and transmitted over the wireless communication channel. At the receiver side the received data is first converted back to parallel form. The cyclic prefix is removed and pilot sequence is extracted from the data. This pilot sequence is used to determine the channel characteristics initially using the pilot sequence an initial estimate of the channel transfer function is obtained. This initial estimate is then corrected in each iteration of the turbo decoder with the help of interpolation using an algorithm.

2.1. Turbo Coding-An Overview

LTE, an acronym for Long Term Evolution, commonly marketed as 4G LTE, is a standard for wireless communication of high-speed data for mobile phones and data terminals[5]. Turbo codes usually appropriate for this domain. A turbo code is a concatenation of two convolutional encoders. Each with its own generator polynomial. The input sequence of the information bits is organized in blocks of length N . The first block of data will be encoded by the first encoder which is a rate half recursive systematic encoder. The same block of information bits is interleaved by the interleaver and encoded by second encoder which is also a rate half systematic recursive encoder. The coded bit produced by the encoder is the output of each encoder block. This is a rate 1/3 turbo code, the output of the turbo encoder being the triplet. The code is recursive in that one of the output is the input sequence itself. The parity bits can be punctured to get higher rates.

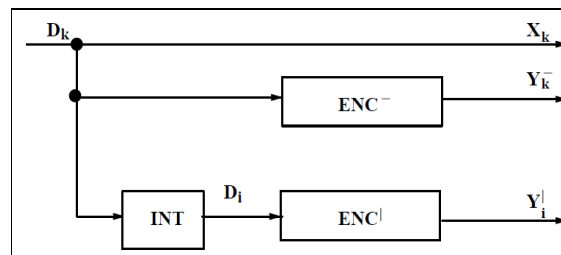


Fig 2:Turbo Coding

The interleaver block, rearranges the order of the information bits for input to the second encoder. The iterative decoding algorithm works best when there are not short cycles in the factor graph that represents the decoder; the interleaver is chosen to avoid short cycles. The main purpose of the interleaver is to increase the minimum distance of the turbo code such that after correction in one dimension the remaining errors should become correctable error patterns in the second dimension. If the number of errors within a code word exceeds the error-correcting code’s capability, it fails to recover the original code word. Interleaving ameliorates this problem by shuffling source symbols across several code words, thereby creating a more uniform distribution of errors. It is a technique commonly used in communication systems to overcome correlated channel noise such as burst error or fading. At the receiver end, the interleaved data is arranged back into the original sequence by the de-interleaver. Coded sequence is then modulated by QPSK.. Data entering into the modulator is separated into two channels called I and Q. Each channel modulates a carrier. The two carrier frequencies are the same, but their phase is offset by 90 degrees (that is, they are “in quadrature”). The two carriers are then combined and transmitted resulting in four states. Pilot sequence is then added, these are the unmodulated data we are transmitting along with the data.

2.2. Channel Estimation

Channel estimation in fading channels with very high mobility usually consists of two steps. Channel estimation at pilot symbols is the first step[7]. The next step, we need to perform interpolation between the pilot symbols to obtain the channel estimate of data symbols (symbols only with unknown data subcarriers), which are transmitted between these pilot symbols. This however assumes that the wireless channel is static during an OFDM symbol period. Pilots are extracted from the FFT output and used to find an initial estimate of the channel transfer function at pilot positions. The estimate is obtained by Least Squares (LS) estimation. N_p pilot signals uniformly inserted in $X(k)$, the channel at pilot subcarriers can be given as:

$$H_p(k) = Y_p(k)/X_p(k)$$

$H_p(k)$ channel at pilot sub-carriers

X_p input at the k^{th} pilot sub-carrier

Y_p output at the k^{th} pilot sub-carrier.

This initial H , the channel transition matrix is used for further computations.

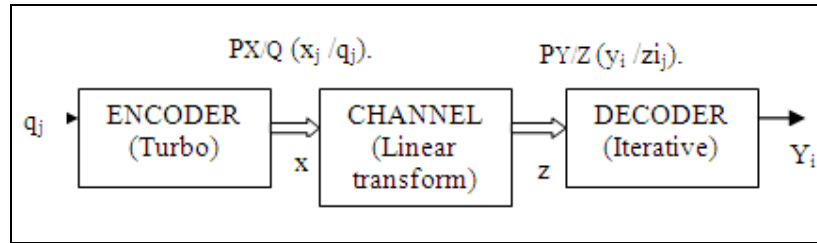


Fig 3: Mathematical representation of the estimation scheme

The approach used here closely resembles that of factor graph approach for loopy belief propagation decoding[11]. An input vector $q \in Q^n$ has components $q_j \in Q$ for some set Q of message sequences and generates an unknown coded sequence $x \in R^n$ through turbo coding described by a conditional distribution $PX/Q(x_j / q_j)$. When x is passed through channel a linear transform $z = Ax$

occurs, as shown in fig.3. Where $A(H)$ is the transform matrix. Finally, each component of z_i of z randomly generates an output component y_i of a vector $y \in Y^m$ through decoding with conditional distribution $PY/Z(y_i / z_i)$. where Y is some output set. The problem is to estimate the transform input x and output z .

2.3. Channel Characteristics

For an additive white Gaussian noise (AWGN) output channel, the output vector y can be written as:

$$y = z + w = Ax + w$$

Where w is a zero mean, Gaussian i.i.d. random vector independent of x being transmitted. The channel transition probability distribution is then given by

$$PY|Z(y|z) = \frac{1}{\sqrt{2\pi\zeta^w}} \exp\left(\frac{-(y-z)^2}{2\zeta^w}\right)$$

Where $\zeta^w > 0$, is the variance of the components of w . For a logistic channel each output y_i is 0 or 1, where the probability that $y_i = 1$ is given by some sigmoidal function such as:

$$PY|Z(y_i = 1|z_i) = \frac{1}{1 + \alpha \exp(-z_i)}$$

Thus, larger values of z_i results in a higher probability of $y_i = 1$.

3. Algorithm For Receiver Section

In this Modified Approximate Message Passing (MAMP) algorithm[11], the scalar operations are defined by two functions, $gout(.)$ and $gin(.)$, called the scalar estimation functions. The algorithm produces a sequence of estimates, $\hat{x}(t)$ and $\hat{z}(t)$ for the unknowns x and z .

3.1. Steps

Given a matrix $A(H)$ obtained from pilot sequence system inputs and outputs q and y and scalar estimation functions $gin(.)$, and $gout(.)$, generate a sequence of estimates through the following recursions;

1. Initialization: Set $t=0, i=j=1$ corresponding to first row and column and $\hat{z}(-1) = 0$ for the first iteration.

2. Computation: For each i , compute

$$\zeta_p^i(t) = \sum_j |a_{ij}|^2 \zeta_x^j(t)$$

$$\hat{p}^i(t) = \sum_j a_{ij} \hat{x}_j(t) - \zeta_p^i(t) \hat{s}^i(t-1)$$

$$\hat{z}_i(t) = a_{ij} \hat{x}_j(t)$$

3. *Updating values:* For each i,

$$\hat{s}_i(t) = g_{out}(t, \hat{p}^i(t), y^i, \zeta_p^i(t))$$

$$\zeta_p^i(t) = -\frac{\partial}{\partial p} g_{out}(t, \hat{p}^i(t), y^i, \zeta_p^i(t))$$

4. *Input side:* For each j,

$$\zeta_r^j(t) = \left[\sum_i |a_{ij}|^2 \zeta_s^i(t) \right]^{-1}$$

$$\hat{r}_j(t) = \hat{x}_j(t) + \zeta_r^j(t) \sum_j a_{ij} \hat{s}_i(t)$$

Where,

$$\hat{x}_j(t+1) = g_{in}(t, \hat{r}^j(t), q_j, \zeta_r^j(t))$$

$$\zeta_r^j(t+1) = \zeta_r^j(t) \frac{\partial}{\partial r} g_{in}(t, \hat{r}^j(t), q_j, \zeta_r^j(t))$$

Then increment t= t+1 and return to step 2 until enough iterations are performed.

4. Scalar Functions Based on MAP Estimation

Through proper selection of the scalar estimation functions $g_{in}(\cdot)$ and $g_{out}(\cdot)$ MAP estimation can be accomplished. The posterior density of x given the system input q and output y is given by the conditional density function

$$P_{\frac{X}{Q}}(x|q, y) = \frac{1}{Z(q, y)} \exp(F(x, Ax, q, y))$$

Where

$$F(x, Ax, q, y) = \sum_{j=1}^m f_{in}(x_j, q_j) + \sum_{i=1}^m f_{out}(z_i, y_i)$$

And

$$f_{out}(z_i, y_i) = \log \frac{P_{Y|Z}(y_i/z_i)}{Z(y/z)}$$

$$f_{in}(x_j, q_j) = \log \frac{P_{X|Q}(x_j/q_j)}{Q(x_j/q_j)}$$

The constant Z(q, y) in (13) is a normalization constant. Given this distribution, the maximum a posteriori (MAP) estimator is the maxima on F(.).

$$\hat{x}^{MAP} = \operatorname{argmax} F(x, z, q, y); z = Ax$$

5. Simulation Results

The proposed system is simulated in MATLAB and performance characteristics analyzed. The coding scheme used is Parallel Concatenation of Convolutional Codes(PCC) of rate 1/3. The block length is same as that of the length of interleaver which is 1024 bits. Pilot sequence is of 5 bits long and is positioned in the middle of the message sequence. The decoding scheme used is Max-log MAP decoding with 2048 iterations.

The performance of this system is compared with that of a normal channel estimation scheme with the help of pn sequence for turbo codes. For ease of implementation, the number of iterations is chosen to be 5. It is found that the performance of the proposed joint channel estimation and decoding scheme is highly improved over the conventional pn sequence based estimation.

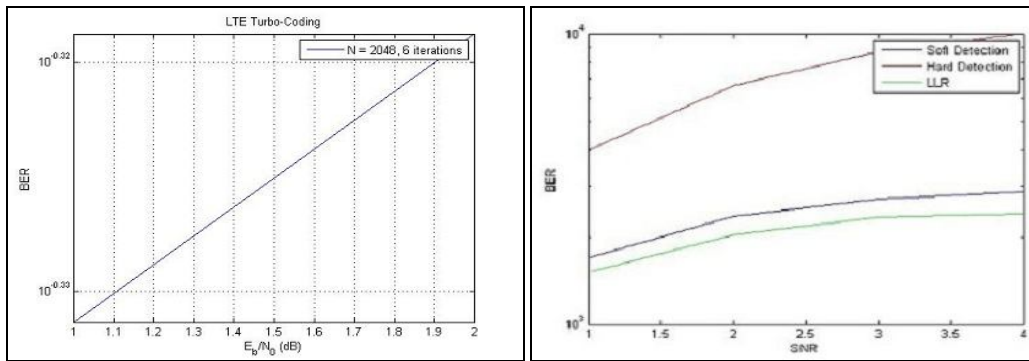


Fig 4:LTE Turbo system BER plot

Fig 5: Comparison of different detection schemes for MAMP LTE system

In addition a comparison of the detection schemes available is also performed i.e the hard decision ,soft decision and LLR decision. It is found that LLR decoding has better performance than the other two for a given SNR.

6. Conclusion

Simulation results shown that the proposed joint estimation scheme for turbo codes is highly effective and have much performance improvement over the existing systems for channel estimation. Moreover since this system does not add to complexity. The estimation accuracy is improved in each satge by iteratively feeding back the channel information to the decoder. This system finds application in widely used present day 4G standard systems such as LTE system. The system output performance does not vary according to the message length.

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