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Combined Channel Estimation and Decoding Scheme For TURBO-OFDM

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Abstract—**OFDM is the present day technology in communication for combating multipath fading and for high-bitrate transmission. Turbo coded OFDM achieves near Shannon capacity limit as seen in LTE standard systems. Through this thesis, an efficient channel estimation scheme suitable for LTE standard systems is devised. The channel is estimated 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 combined iterative estimation and decoding technique improves the system's performance over the conventional technique.**

Keywords—channel estimation; LTE; OFDM;

I. INTRODUCTION

OFDM has developed into a popular scheme for wideband digital communication. It is basically a frequency-division multiplexing (FDM) scheme used as a digital multi-carrier modulation method. The primary advantage of OFDM over single-carrier schemes is its ability to cope with severe channel conditions (for example, attenuation of high frequencies in a long copper wire, narrowband interference and frequency selective fading due to multipath) without complex equalization filters. OFDM is used in combination with any forward error correction (FEC) codes .Turbo codes have been shown to exhibit near-capacity performance over flat-fading channels with coherent detection and perfect knowledge of the channel response. Therefore turbo–coded OFDM systems arewidely in use for 4G standard systems such as LTE. However,mobile communication systems are characterized by channelresponses with time-varying magnitude and phase. Differentchannel estimation technique had been derived for this purpose.

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 reduce the complexity of 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 adaptive channel estimation and trackingscheme based on recursive least squares [1]. Preamble aidedchannel estimation is performed in time-domain (TD). Theestimator is then extended to perform decision-directed (DD)channel tracking during data transmission. Also estimationusing adaptive filters such as Kalman filter [2] is common. Athree-stage estimation scheme is proposed to reduce thecomplexity and adapt the channel estimates w.r.t the feedbackinformation. In [3],authors incorporate pilot symbols into thetransmission. In this paper, the channel estimation schemeworks in synchronism with iterative decoding. The iterativeturbo decoder feedback loop is broken and the estimator isinserted and the soft channel input is updated before each stageof decoding, new estimate is formed by the updated decoderdecision variable.

The paper is organized as follows section II describes the system model for LTE standard, giving an idea of turbo codes and channel estimation. Section III describes our new receiver algorithm and IV indicates the selection of scalar functions for the algorithm, whereas V deals with simulation and results. Section VI concludes the paper follower by VII references.

II. SYSTEM MODEL

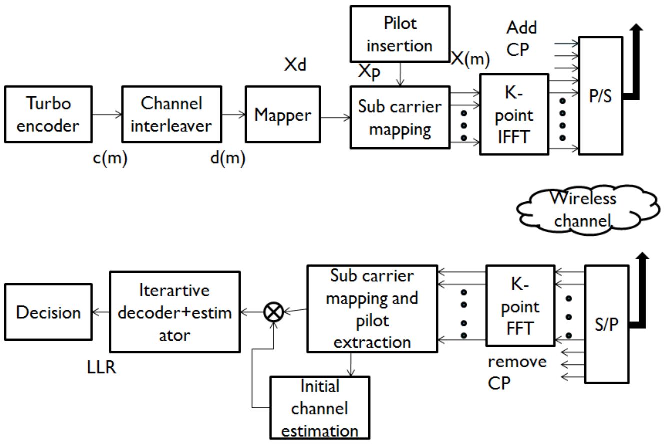

Fig 1:General LTE system model-OFDM

The block level representation of the LTE system is shown inFigure. The information bits are encoded, interleaved, andmapped into M-ary complex symbols forming the datasequence Xd. Then pilot sequence isinserted and IFFT is performed to optimize this for OFDMtransmission. The information sequence is now in time domain.The data in parallel form is then converted to serial form andtransmitted over the wireless communication channel. At thereceiver side the received data is first converted back to parallelform .The cyclic prefix is removed and pilot sequence isextracted from the data. Using this pilotsequence an initial estimate of the channel transfer function isobtained. This initial estimate is then corrected in each iterationof the turbo decoder with the help of interpolation using analgorithm.

A. *Turbo Coding*

The template is used to format your paper and style the text. All A turbo code is a concatenation of two convolutional encoders [4], 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 outputs is the input sequence itself. The parity bits can be punctured to get higher rates. 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. Turbo codes are usually appropriate for this domain.

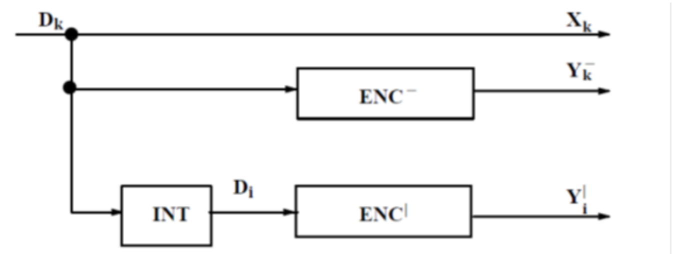

Fig 2:Turbo Coding

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. Pilotsequence is then added, these are the unmodulated data we aretransmitting along with the data.

B. *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. 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:

$$H_p(k) = Y_p(k)/X_p(k) \qquad (1)$$

$\boldsymbol{H_p(k)}$ channel at pilot sub-carriers
$\boldsymbol{X_p}$input at the k_{th}pilot sub-carrier
$\boldsymbol{Y_p}$output at the k_{th}pilot sub-carrier.
This initial *H* ,the channel transition matrix is used for furthercomputations.

Fig 3: Mathematical representation of the estimation scheme

The approach used here closely resembles that of factor graph approach for loopy belief propagation decoding. An input vector q €Q^n has components q_j€ Q for some setQ of message sequences and generates an unknown coded sequence x €R^n through turbo coding described by a conditionaldistribution $P_{X/Q}(x_j/q_j)$.Whenx is passed through channel a linear transform

$$z = Ax \qquad (2)$$

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 € Y m through decoding with conditional distribution $P_{Y/Z}(y_i/z_i)$, where Y is some output set. The problem is to estimate the transform input x and output z.

C. Channel Characteristics

For an additive white Gaussiannoise (AWGN) output channel, the output vector y canbe written as:

$$y = z + w = Ax + w \tag{3}$$

Where w is a zero mean, Gaussian i.i.d. random vector independentof x being transmitted.The channel transition probabilitydistribution is then given by

$$PY|Z(y|z) = \frac{1}{\sqrt{2\Pi\zeta^{\omega}}}\, exp\left(\frac{-(y-z)^2}{2\zeta^{\omega}}\right) \tag{4}$$

Where$\zeta^{\omega} > 0$, 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 sigmoidalfunction such as:

$$PY|Z(yi = 1|zi) = \frac{1}{1+a\exp(-zi)} \tag{5}$$

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

III. ALGORITHM FOR RECEIVER

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

A. 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 $\hat{s}(-1) = 0$ for the first iteration.

2. *Computation:* For each i, compute

$$\zeta_p^i(t) = \sum_j |aij|^2 \zeta_x^j(t) \tag{6}$$

$$\hat{p}^i(t) = \sum_j aij\hat{x}_j(t) - \zeta_p^i(t)\hat{s}^i(t-1) \tag{7}$$

$$\hat{z}_i(t) = aij\hat{x}_j(t) \tag{8}$$

3. *Updating values:* For each i,

$$\hat{s}_i(t) = gout(t, \hat{p}^i(t), yi, \zeta_p^i(t)) \tag{9}$$

4.*Input side:* For each j,

$$\zeta_r^j(t) = [\sum_i |aij|^2 \zeta_s^i(t)]^{-1} \tag{10}$$

$$\hat{r}_j(t) = \hat{x}_j(t) + \zeta_r^j(t)\sum_j aij\hat{s}_i(t) \tag{11}$$

Where,

$$\hat{x}_j(t+1) = gin(t, \hat{r}^j(t), q_j, \zeta_r^j(t)) \tag{12}$$

$$\zeta_r^j(t+1) = \zeta_r^j(t)\frac{\partial}{\partial r} gin(t, \hat{r}^j(t), q_j, \zeta_r^j(t)) \tag{13}$$

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

IV. SCALAR FUNCTIONS USING MAP ALGORITHM

Through proper selectionof the scalar estimation functions gin(.) and gout(.) MAP estimation can be accomplished.The posterior densityof x given the system input q and output y is givenby the conditional density function.

$$P\frac{X}{Q}, y(x\,|q\,,y) = \frac{1}{Z(q,y)}\exp\big(F(x, Ax, q, y)\big) \tag{14}$$

Where

$$F(x, Ax, q, y) = \sum_{j=1}^{n} fin(xj, qj) + \sum_{i=1}^{m} fout(zi, yi) \tag{15}$$

And

$$fout(zi, yi) = \log P_{Y/Z}(y\,/z) \tag{16}$$

$$fin(xj, qj) = \log P_{X/Q}(xj\,/qj) \tag{17}$$

$$\hat{x}^{map} = \arg max\ F(x, z, q, y); z = Ax \tag{18}$$

V. SIMULATION RESULTS

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

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

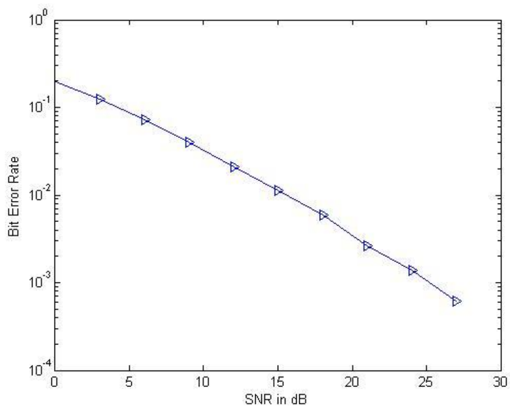

Fig 4:LTE Turbo system BER plot

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

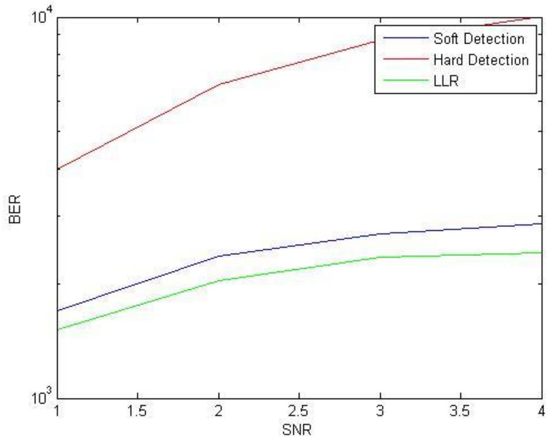

Fig 5: Comparison of different detection schemes for MAMP LTE system

VI. CONCLUSION

Simulation results shows 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 stage 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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