Efficient Channel Estimation Based On LTE In OFDM

Mathe Sowmya^{#1}, CH. Ravi Kumar^{*2} ^{#1}M.Tech, Department of E.C.E & Sir C.R.Reddy College of Engineering ^{#2}Asst.Proffesor, Department of E.C.E & Sir C.R.Reddy College of Engineering

Eluru, India

Abstract— in this paper, we propose an iterative detecting and decoding method in LTE Downlink OFDM system based on reference aided channel estimation. Estimates of the complex channel gain and variance of the additive noise are derived first from known reference symbols and channel estimate filter. After turbo decoding, the decoded bits have been obtained. In order to obtain estimates of channel gain more accuracy, we adopted more reference symbols which modulate the decoded bits at the receiver again to estimate the channel frequency response. Also the mapping position of the OFDM symbols is consistent with first transmission. Through estimation of the channel iteratively, we can use more systems bits as iterative reference symbols. Simulation results shows that through the iterative channel estimation, the initial insert reference symbols have been reduced, the accuracy of channel estimation improved effectively and the bits error rate of the OFDM system can approach the perfect channel estimates.

Key words- channel estimation; turbo; LS; iterative.

I. INTRODUCTION

For the next generation of wireless communication systems OFDM (Orthogonal frequency division multiplexing) is a promising technique. The greatest advantage of OFDM is it can be consider as either multiplexing technique or modulation scheme. The greatest advantage with OFDM is its ability against interfacing in narrowband and frequency selective fading. In simple words in a system which has single carrier (single carrier system) a single fade or interfaces may be a strong reason to fail entire link, but in only a small percentage of the sub-carriers will be affected multi carrier system. The total frequency band is divided into N subchannels which are non-overlapping frequency bands in classical method. With a separate unique symbol each subchannel is modulated and then these modulate N sub channels are frequency-multiplexed. It seems good to avoid spectral overlap of channels to eliminate inter-channel interference. However, it causes to inefficient use of the available frequency spectrum.

In cellular 3G services LTE (Long term evolution) is the next greatest step.LTE technology is a based on a 3GPPstandard that provides for a downlink speed of up to 150 Mbps and an uplink speed of up to 50 Mbps. Fixed wireless and wired standards are already approaching or achieving 100 Mbps or faster, and LTE is a way for cellular communications to operate at that high data rate.

OFDM technology is used in LTE downlink to achieve its high-speed transmission targets. For LTE downlink channel

estimation, recursive wiener filtering is used and it is deduced that the performance can be increased by repetitive iteration and the channel is considered static over duration of one OFDM symbol

The LTE protocol's physical layer is unique because it has asymmetrical modulation and data rates for uplink and downlink. The standard is designed for full-duplex operation, with simultaneous transmission and reception. The radio is optimized for performance on the downlink, because the transmitter at the base station has plenty of power. On the uplink, the radio is optimized more for power consumption than efficiency, because while processing power has increased, mobile device battery power has stayed essentially constant.

In this project, we propose a new method which incorporates an iterative channel estimation and turbo decode to combat fast channel variation. It reduces the consumption of reference symbols in the OFDM time frequency resource units and estimates the channel frequency response accurately at the premise of less loss performance.

I. SYSTEM MODEL

LTE Architecture:

Figure 1 provides a high-level view of LTE architecture. This is a snapshot of the part that most closely interacts with the UE, or mobile device. The entire architecture is much more complex; a complete diagram would show the entire Internet and other aspects of network connectivity supporting handoffs among 3G, 2G, Wi-MAX, and other standards. This particular device shows the e Node B, which is another name for the base station, and the interfaces between the e Node B and UEs. The E-UTRAN is the entire network, which is the "official" standards name for LTE.

- **eNB**: Enhanced Node B, or base station
- UE: User Equipment
- **EPC**: Evolved Packet Core
- **MME**: Mobility Management Entity (Control Plane)
- **SAE**: System Architecture Evolved (User Plane)
- E-UTRAN: Evolved Universal Terrestrial Radio Access

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Fig 1 LTE Architecture Overview

Frames and Packet Timelines: LTE Downlink

Figure 2 shows a time domain view. At the bottom are radio frames. A full frame is 10 ms but we normally think in terms of the 1-ms sub frame, which is the entity that contains the transport block.



Fig 2: Time domain view of the LTE downlink

Within the transport block is the MAC header and any extra space (padding). Within that there is the RLC header, and then within the RLC header there can be a number of PDCPs. There is a somewhat arbitrary relationship between the IP packets coming in, which form the SDUs, and how the RLC

PDUs are formed. Therefore you can make the maximum effective use of radio resources in a fixed period of time.

LTE Downlink Reference Signals Structure:

In order to carry out coherent demodulation in LTE down link, channel estimation is needed at the receiver end. In case of OFDM transmission known reference symbols are added into time-frequency grid for channel estimation. These signals are called LTE Downlink Reference signals. For time domain, reference symbols are slotted-in in the first and the third last elements of resource grid, whereas reference signals are inserted over every six sub-carriers in frequency domain. For an accurate channel estimation over entire gird and reducing noise in channel estimates, a two-dimensional time-frequency interpolation/averaging is required over multiple reference symbols. One reference signal is transmitted from each antenna to estimate the channel quality corresponding to each path when a multiple antenna scheme is applied. In this case, reference signals are mapped on different sub-carries of resource grid for different antennas to refrain from interference. Resource elements used to transmit reference signals from antenna-1 are not reused on antenna-2 for data transmission; these places are filled with zeros. Allocation of these reference symbols is shown in figure 3.



Figure 3: Allocation of Reference Symbols for two antenna transmissions

II. OFDM Structure in LTE Downlink system

In LTE downlink structure, one radio frame consists of 20 sub-frames where each sub frame is divided into two time slots. Each slot has 6 or 7 OFDM symbols. The numbers of subcarriers in each OFDM symbol depends on the numbers of resource blocks used. A physical resource block is defined as N_{SVM}^{DL} consecutive OFDM symbols in the time domain and

 N_{SC}^{RB} consecutive subcarriers of frequency 15 kHz each, in the frequency domain. As per configuration $N_{SYM}^{DL} = 6$ when extend cyclic prefix is used and $N_{SC}^{RB} = 12$. A physical resource block thus consists of $N_{SYM}^{DL} * N_{SC}^{RB}$ resource elements, corresponding to one slot in the time domain and 180 kHz in the frequency domain. Physical resources are numbered from 0 to N_{DL-1}^{DL} in the frequency domain. The reference signal sequence for channel estimation shall be mapped to complex-valued modulation symbols.

Iterative estimation system model

Consider an OFDMLTE system which consists of single-user single-antenna with N subcarriers as shown in Fig. 4. An N \times 1 complex symbol vector, using QPSK or 16-QAM the turbo encoded, interleaved, and rate-matched bits are mapped. The modulated symbols are mapped into the OFDM time-frequency resource grid which is defined in LTE downlink. Also, the reference symbols are mapped to resource grid.

Channel Estimation in LTE

In this session, channel estimation techniques used in LTE down link are described. Channel estimation is a vital part of receivers designs used in mobile communication systems. The effect of the channel on the transmitted information must be estimated in order to recover the transmitted information correctly. The estimation of channel effects is often based on an approximate underlying model of the radio propagation channel. The receiver can precisely recover the transmitted information as long as it can keep track of the varying radio propagation channels. Channel models are described in this session.

Signal Model

In this work, we consider one OFDM symbol to perform channel estimation in LTE down link. Therefore receive symbol at the i^{th} symbols and the k^{th} subcarriers of the resource gird is denoted by Y (i, k), the transmit model can be describe:

$$Y (i, k) = X (i, k) + H (i, k) N (i, k)$$

Therefore we can estimate the frequency response by comparing the received reference symbols and the original reference symbols, then the detected symbols are de-mapped into soft bits, finally, through turbo decoder, we can get decode the transmit system bits.

IV. CHANNEL ESTIMATION TECHNIQUES

Pilot-assisted Channel Estimation

The pilot assisted channel estimation process consists of two steps; first statistical estimation of the channel at OFDM tones consisting of reference symbols is determined using statistical methods including Squares (LS) and Minimum Mean Squares (MMSE) estimates. Different pilots assisted channel estimation schemes can be employed for the estimation of the channel effects on the transmitted signal. The response of the channel at the data subcarriers is subsequently determined by interpolation. The interpolators used for the purpose of estimation are linear, second order, cubic or time domain interpolators derived from both the statistical and deterministic point of view. Various publications can be found that deal with one overall these estimation criteria for pilot assisted channel estimation of OFDM applications from CIR or CFR prospective.

Least Square Estimation

Least Square based parameter estimation approach aims at determining the channel impulse response from the known transmitted reference symbols in the following way:

$$G_{LS} = \left[\frac{Y_{r(1)}}{X_{r(1)}}, \frac{Y_{r(2)}}{X_{r(2)}}, \dots, \frac{Y_{r(N)}}{X_{r(N)}}\right]$$

Where $G_{LS} \in C^{N_r}$ the estimated channel frequency response on the sub carriers which contains reference symbols. This response can be interpolated over full frequency range in order to obtain the channel frequency response for the subcarriers carrying data symbols. The interpolation can be performed in time domain or frequency domain.

The signal received in time domain can be expressed as follows:

$$Y_r = F^H A_r F_L h + \mu$$

The channel can be estimated using Least Squares, in time domain in the following

$$\hat{h} = (S^H S)^{-1} S^H Y_r$$

Solving the above two equations, we get the expression for LS estimates.

$$\hat{h} \approx (F_L^H A_L^H A_r F_L)^{-1} F_L^H Y_r A_R^H F^H Y_r$$

Regularized LS Estimation

In this method, small constant term is added to the diagonal entries for regularizing the Eigen values of the matrix to be inverted. In this case, the channel impulse response becomes of the form,

$$\hat{h}_{reg} = (aI + F_L^H A_L^H A_r F_L)^{-1} F_L^H Y_r A_R^H F^H Y_r$$

The value of α has to be selected such that the inverse matrix is least perturbed.

Minimum Mean Square Estimation

LS channel estimation method has been described which is computationally simple but its performance is not good. Another method to estimate the CIR is minimum mean square estimator (MMSE) which has better performance than LS but it is computationally complex. This method intends at the minimization of the mean square error between the exact and estimated CIRs. In this section we will discuss linear minimum mean square estimator (LMMSE). The CIR can be calculated using LMMSE in the following way

$$\hat{h} = R_{hy_r} R_{y_r y_r}^{-1} Y_r$$

Here $R_{y_r y_r}$ is the auto covariance of vector Y_r and R_{hyr} is the cross covariance of vectors h and Y_r . These covariance matrices for the above equation can be calculated as,

$$\hat{\mathbf{h}} = \mathbf{X}_{r}^{\mathrm{H}} \mathbf{T}_{r}^{\mathrm{H}} (\mathbf{X}_{r} \mathbf{T}_{r} \mathbf{R}_{\mathrm{hh}} \mathbf{X}_{r}^{\mathrm{H}} \mathbf{T}_{r}^{\mathrm{H}} + \sigma_{\mu}^{2} \mathbf{I}_{\mathrm{Nr}}$$

Fundamentals of turbo codes:

In order to explain the proposed parallel Turbo decoder architecture, the fundamentals of Turbo codes are briefly described in this section.

Turbo encoder structure:

As shown in Fig. 4, the Turbo encoding scheme in the LTE standard is a parallel concatenated convolution code with two 8-state constituent encoders and one quadratic permutation polynomial (QPP) inter leaver. The function of the QPP inter leaver is to take a block of N-bit data and produce a permutation of the input data block. From the coding theory perspective, the performance of a Turbo code depends critically on the inter leaver structure. The basic LTE Turbo coding rate is 1/3.It encodes an N-bit information data block into a codeword with 3N+12 data bits, where 12 tail bits are used for trellis termination. The initial value of the shift registers of the 8-state constituent encoders shall be all zeros when starting to encode the input information bits. *Turbo decoder structure:*

The basic structure of a Turbo decoder is functionally illustrated in Fig. 5. A Turbo decoder consists of

two maximum posterior decoders (MAP) separated by an interleave that permutes the input sequence.



Fig.4: Structure of rate 1/3 Turbo encoder in LTE.

Each Turbo iteration is divided into two half iterations. During the first half iteration, MAP decoder 1 is enabled. It receives the soft channel information (soft value L_s for the systematic bit and soft value L_{p1} for the parity bit) and the a priori information L^1 from the other constituent MAP decoder through de-interleaving (π^{-1}) to generate the extrinsic information L^1 at its output.



Fig 5: Basic structure of an iterative Turbo decoder. (a) Iterative decoding based on MAP decoders. (b) Forward/backward recursions on the trellis diagram.

Likewise, during the second half iteration, MAP decoder 2 is enabled, and it receives the soft channel information (soft value L_s for a permuted version of the systematic bit and soft value $L_p 2$ for the parity bit) and the a priori information L^2 from MAP decoder 1 through interleaving to generate the extrinsic information L_e at its output. This iterative process repeats until the decoding has converged

or the maximum n u m b er of iterations has been reached. The MAP algorithm at each constituent MAP decoder computes the loglikelihood ratios (LLRs) of the a posteriori probabilities (APPs) for information bit uk.

MAP algorithms

Corresponding to different contents of encoder memories the encoder can be in one of 2^{M} States for the binary input of alphabet {1,-1}. The trellis structure is exploited to compute the a posteriori *L*-values

$$L(x_{k}) = \log \frac{p(x_{k} = 1 | y)}{p(x_{k} = -1 | y)} = \log \frac{\sum_{(s', s), xk = 1} \lambda_{k}(S', s)}{\sum_{(s', s), xk = -1} \lambda_{k}(S', s)}$$

Where s 'and s represent the states of the encoder element at time k-1 and k, and y are the received sequence of length N, respectively. Dividing the received sequence into three separate terms and applying the Bayes chain rule, we get

$$\lambda_{k}(s',s) = p(y_{k+1,n}|s_{k+1} = s)$$
$$P(s_{k+1} = s, |sk = s').p(s_{k}|y_{1,k-1})$$

Here using forward or backward algorithm the above equation can be evaluated as 3 terms

$$\alpha_{k}(s) = p(s_{k} = s, y_{1,k-1})$$

$$\beta_{k}(s) = p(s_{k} = y_{k+1,n}s_{k} = s)$$

$$\gamma_{k}(s', s) = p(s_{k+1} = s, y_{k})$$

Using the forward and backward recursions $\alpha_k(s)$, $\beta_k(s)$ are computed as

$$\alpha_{k}(s) = \sum_{\substack{s' = \Omega_{k-1}^{(c)}(s) \\ \beta_{k}(s)}} \alpha_{k-1}(s') \gamma_{k-1}(s',s)$$

$$\beta_{k}(s) = \sum_{\substack{s' = \Omega_{k+1}^{(c)}(s) \\ s' = \Omega_{k+1}^{(c)}(s)}} \gamma_{k}(s',s) \beta_{k+1}(s')$$

Here the important is the calculation of the $\gamma_k(s', s)$ is different for both the encoder and decoder, since the existence of treelis codes.

For the equalizer

$$\gamma_k(s',s) = \exp(\sum_{\nu=1}^{2} \frac{1}{2} x_{k,\nu} L(\widehat{x_{k,\nu}}) + \frac{1}{2} u_k L(u_k))$$

For decoder

$$\gamma_k(s',s) = \exp(\sum_{\nu=1}^2 \frac{1}{2} x_{k,\nu} L(\widehat{x_{k,\nu}}) + \frac{1}{2} u_k L(u_k))$$

Over AWGN channel the turbo decoding without the use of Inter State Information

$$L(\widehat{x_{k,y}}) = L_c. y_{k,y} = (4E_b/N_0). y_{k,y}$$

Using the extrinsic Jacobian logarithmic function The log-Map algorithm introduced to evaluates the α_k (s) and β_k (s) and γ_k (s, s').

$$A_{k}(s) = \text{Max'} [D_{k}(s', s) + A_{k-1}(s')]$$

$$B_{k-1}(s') = \text{Max'} [D_{k}(s', s) + B_{k}(s')]$$

Where

$$max'(z_1, z_2) + \log(1 + \exp(-|z_1 - z_2|))$$

 $A_k(s)$, B_k (s), and D_k (s) are the logarithmic values of α_k (s) and β_k (s) and γ_k (s, s'), respectively.

MAP log likelihood ration of the decoder is determined as follows:

L
$$(u_k) = max'_{(s',s),x_{k=1}}[B_k(s) + D_k(s',s) + A_{k-1}(s')] - max'_{(s',s),x_{k=-1}}[B_k(s) + D_k(s',s) + A_{k-1}(s')]$$

In the trellis structure max' operation is performed in first and second part only .the main advantage of the Max-Log-Map is ignores the interference caused by the SNR.

V. SUMULIATION AND PERFORMACE ANALYSIS

In this paper we, Matlab is used to realize the simulation experiment.



Fig3: Comparison of BER for various estimation techniques.

B. Simulation results:

In Fig3, We can see that the performance of the LS estimation is worse in the fading channel. The reason for the low estimation accuracy is that we have little reference symbols to interpolate the channel response. After reconstructing the original transmission OFDM symbols. It means that use more system signals as reference symbol; the performance of the iterative LS estimation could near the MMSE estimation. As a comparison, the BER curve of perfect channel estimation is depicted in the Fig. The simulation result shows that through the proposed iterative channel estimation method, dramatic performance improvements can be achieved with the LS estimation and that the advantage of low implementation complexity makes it possible widely used in the practical OFDM system.

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Mathe Sowmya currently pursuing her M.Tech degree in Sir C.R.Reddy college of Engineering, Affiliated to Andhra University. She was graduated from JNTU Kakinada in the year of 2011.



CH. Ravi Kumar, currently working as an Assistant Professor in Sir C.R.Reddy Engineering College, Eluru. He has done much amount of work towards the new concepts in the era of turbo codes and pursuing Ph.D. degree from JNTU Kakinada. He was post graduated in M.Tech from JNTUK Kakinada and graduated in B.E in SRKR Engineering College, with the both specializations are in Electronic and Communication Engineering. [1]