Study of Various Channel Estimation Techniques in OFDM Mobile Wireless Channel

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Abstract: **In modern world error free transmission is one of the main aims in wireless communications. As the increase in multimedia application, bulky data is being transmitted over wireless communications. Due to the effect of channel fading and the Doppler shifts caused by user mobility, a common problem in the wireless systems, additional technologies are needed to be combat multipath propagation fading and Doppler shifts. The time-variant channel estimation is one of such a crucial technique used to improve the performance of the modern wireless systems with Doppler spread and the multipath spreading. Channel estimation is usually done by estimating the time-varying channel frequency response for the OFDM symbols. . In order to reduce complexity of MIMO system, various detection algorithm such as Zero Forcing (ZF), Minimum Mean Square Error (MMSE), Maximum Likelihood (ML) and LMMSE is studied that reduce Bit Error Rate (BER) by using spatial multiplexing. In time-variant channel estimations using the pilot Sequences technique is a useful channel estimation techniques in mobile wireless communication for accurately estimation of the transmitted information. The main advantage of pilot sequences is to allowing the more accurate representation of the high mobility mobile wireless channels with low complexity. The main goal is to test the most recent proposed method, time-variant estimation of the channel using pilot sequences.**

Keywords: **Doppler Diversity, OFDM, Pilot Sequences, Channel estimation, ZF,ZF-SIC,MMSE,MMSE-SIC,LS,LMMSE.**

I. INTRODUCTION

Channel estimation is an important technique especially in mobile wireless network systems where the wireless channel changes over time, usually caused by transmitter and/or receiver being in motion at vehicular speed. Mobile wireless communication is adversely affected by the multipath interference resulting from reflections from surroundings, such as hills, buildings and other obstacles. In order to provide reliability and high data rates at the receiver, the system needs an accurate estimate of the time-varying channel. Furthermore, mobile wireless systems are one of the main technologies which used to provide services such as data communication, voice, and video with quality of service (QoS) for both mobile users and nomadic. The knowledge of the impulse response of mobile wireless propagation channels in the estimator is an aid in acquiring important information for testing, designing or planning wireless communication systems.

Channel estimation is based on the training sequence of bits and which is unique for a certain transmitter and which is repeated in every transmitted burst [3]. The channel estimator gives the knowledge on the channel impulse response (CIR) to the detector and it estimates separately the CIR for each burst by exploiting transmitted bits and corresponding received bits. Signal detectors must have knowledge concerning the channel impulse response (CIR) of the radio link with known transmitted sequences, which can be done by a separate channel estimator. The channel estimator is shown in fig 1.

The channel estimation method based on OFDM technique. Orthogonal frequency division multiplexing can accommodate high data rate in the mobile wireless systems in order to handle multimedia services. It is important to understand the OFDM technology because the channel estimation is an integral part of OFDM system.

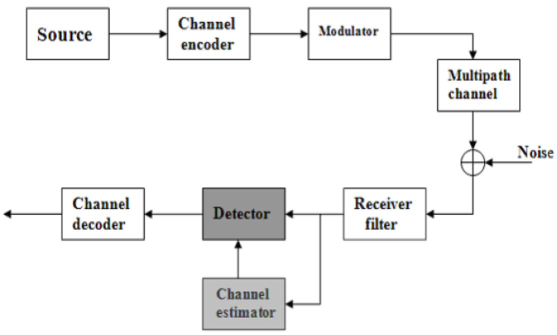

Figure 1. The block diagram of the channel estimator

OFDM technology can be used effectively to avoid the effect of frequency-selective fading and narrowband interference from parallel closely spaced frequencies in mobile networks. One of the desirable features of OFDM is its robustness to the multipath induced intersymbol interference. On the other hand due to the frequency selective fading of the dispersive wireless channel, some sub channels may face deep fades and degrade the overall system performance. In order to compensate the frequency selectivity, techniques such as error correcting code and diversity have to be used [4]-[6].

Doppler shift is caused by the relative motion between the transmitter and receiver. It is considered as Doppler shift in the environment of single-path while Doppler spread in the environment of multipath. For OFDM system, the orthogonality between subcarriers is destroyed by Doppler spread. The greater the relative motion is the stronger the orthogonality is destroyed. Doppler Spread cannot ignore in the scenario [7]. So it is increasingly important to solve the Doppler frequency-spread in high-mobility environment.

Orthogonal Frequency Division Multiplexing (OFDM)

Orthogonal frequency division multiplexing can accommodate high data rate in the mobile wireless systems in order to handle multimedia services. It is important to understand the OFDM technology because the channel estimation is an integral part of OFDM system. OFDM technology can be used effectively to avoid the effect of frequency-selective fading and narrowband interference from parallel closely spaced frequencies in mobile networks. If there is no orthogonality in the channel, inter-channel interference (ICI) can be experienced. With these vital advantages, OFDM technology has been widely used by many wireless standards such as WLAN, WMAN, and DVB [8]. In OFDM scheme, complex filters are not required and time-spreading can be used without any complications in OFDM scheme.

Orthogonal Frequency Division Multiplexing (OFDM) is one of the promising applications, which reduces the multipath fading and makes complex equalizers unnecessary [9]. The concept of using parallel-data transmission and frequency division multiplexing (FDM) was first published in the mid of 1960s. The basic idea was to use parallel data and FDM with overlapping subchannel to avoid the use of high-speed equalization to combat impulsive noise and multipath distortion and fully utilize bandwidth.

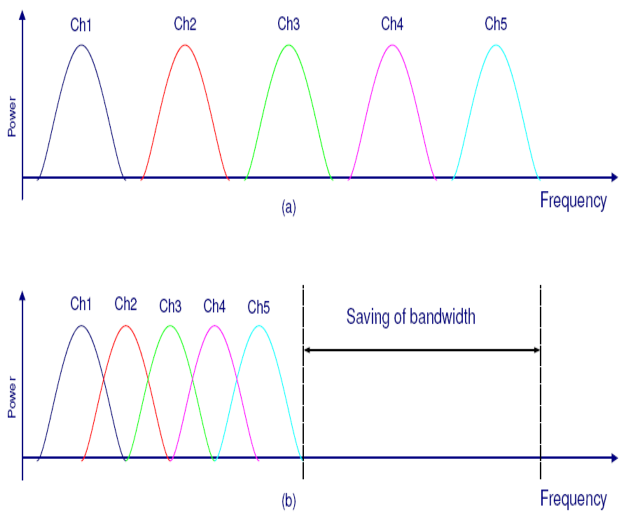

Figure 2 : Concept of OFDM Signal: (a) conventional multicarrier technique (FDM) , and (b) orthogonal frequency division multiplexing technique.

In Figure 2, we can observe the difference between non-overlapping multicarrier modulation technique and overlapping modulation technique. From figure 2(b) it is very clear that by using overlapping modulation technique we can save much more bandwidth than the non-overlapping one [9]. Weinstein and Ebert [10] applied the discrete Fourier transform (DFT) to parallel data transmission system as part of the modulation and demodulation process. In multicarrier transmission, bandwidth divided in many non-overlapping subcarriers but not essential that all subcarriers are orthogonal to each other as shown in figure 3 [9].

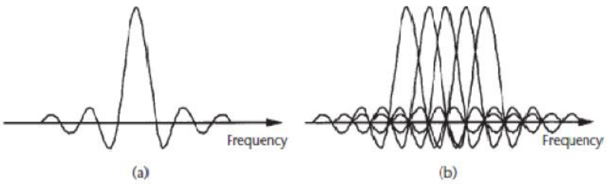

Figure 3: Spectra of (a) an OFDM subchannel, and (b) an OFDM Signal

Orthogonal frequency-division multiplexing (OFDM) is a key technique in the mobile applications of 3G/4G system due to its capability of high rate transmission and robustness to inter-symbol-interference (ISI). It could combat the frequency selective fading effectively because of narrow bandwidth of each subcarrier. Furthermore, its implementation is low-complexity because the signal processing architectures need only IFFT, FFT and simple frequency-domain equalization. However, these advantages no longer exist when the channel response varies rapidly in time-domain because of the mobility of user equipment.

II. METHODOLOGY

In OFDM the concept of parallel data and FDM with overlapping sub channels to avoid the use of high-speed equalization to combat impulsive noise and multipath distortion and fully utilize bandwidth. The working of OFDM is shown in figure 4.

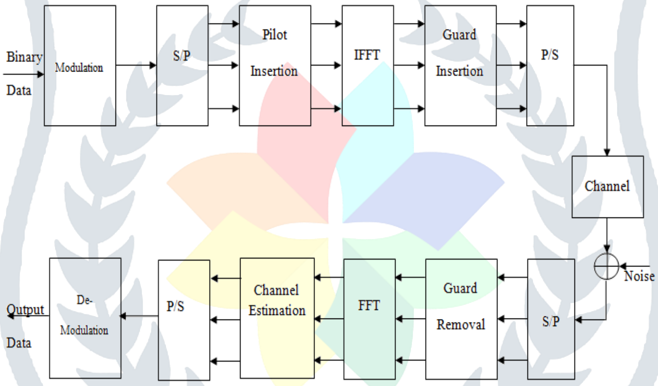

Figure 4: Block diagram of noisy Estimator

Method Used For Training Based Channel Estimation

Usually the Channel Impulse Response (CIR) is estimated based on the known training sequence or pilot sequences, which is transmitted over the channel in every transmission burst. The receiver can utilise the known training bits and the corresponding received samples for estimating CIR typically for each burst separately. There are a few different approaches of channel estimation. In this work the many channel estimation method used for the estimation of the error or distortion occurs in the transmitted signal. A modified and improved Linear Minimum Mean Square Error (LMMSE) and Least Square (LS), Zero Forcing Estimation with ML (Maximal Likelihood) & MRC (Maximal Ratio Combining) methods are used.

Usually the pilot/ training based channel estimation is based on known sequence of bits, which is unique for a certain transmitter and which is repeated in every transmission burst. Thus, the channel estimator is able to estimate CIR for each burst separately by exploiting the known transmitted bits and the corresponding received samples. Detectors comprising of linear detectors (LD) and non-linear detectors (NLD). Linear detectors (LD) consists of zero forcing (ZF) & Minimum mean square error (MMSE) and Non-linear detectors (NLD) consists of Zero forcing-successive Interference cancellation (ZF-SIC) & Minimum mean square error-successive Interference cancellation (MMSE-SIC) [7].

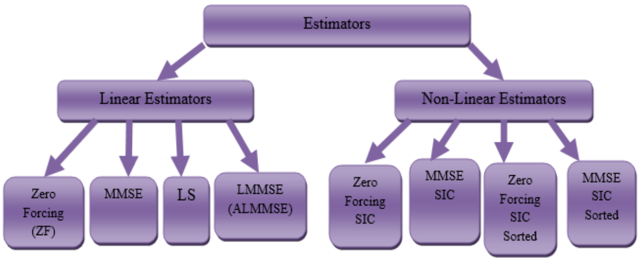

Figure:4 Classification of Various Estimation Techniques

a) **Zero Forcing**

Zero Forcing (ZF) is the form of linear equalization algorithm used in communication systems which applies the inverse of the frequency response to the received signal, to restore signal after the channel. It has many useful applications. In ZF knowing the channel allows recovery of the two or more streams which will be received on top of each other on each antenna. The name Zero Forcing corresponds to bringing down the intersymbol interference (ISI) to zero in the noise free case. This is very much useful when ISI is significant compared to noise. For a channel with frequency response $F(f)$ the zero forcing equalizer $C(f)$ is constructed by $C(f) = 1/F(f)$.

The combination of channel and equalizer gives a flat frequency response and linear phase $F(f)C(f) = 1$.
Actually zero-forcing equalization does not work in most the applications, for the following reasons:

1. Even though the channel impulse response has finite length, the impulse response of the equalizer needs to be infinitely long.

2. At some frequencies the received signal may be weak. To compensate, the magnitude of the zero-forcing filter ("gain") grows very large. As a consequence, any noise added after the channel gets boosted by a large factor and destroys the overall signal-to-noise ratio. Furthermore, the channel may have zeroes in its frequency response that cannot be inverted at all. (Gain * 0 still equals 0).

This second term is many a times more limiting condition. These problems are addressed in the LMMSE equalizer by making a small modification to the denominator of $C(f)$:

$$C(f) = 1/(F(f)+k),$$

Where: k is related to the channel response and the signal SNR.

If the channel response (or channel transfer function) for a particular channel is H(s) then the input signal is multiplied by the reciprocal of it. This is intended to remove the effect of channel from the received signal, in particular the intersymbol interference (ISI).
The ZF removes all ISI, and is perfect when the channel is noiseless. However, when the channel is noisy, the ZF will amplify the noise greatly at frequencies f where channel response H (j2πf) has a small magnitude (near zeroes of the channel) to attempt to invert the channel completely. A more balanced linear equalizer in this case is the minimum mean-square error equalizer, which does not usually eliminate ISI completely but instead minimizes the total power of the noise and ISI components in the output.

b) **Successive Interference Cancellation with optimal ordering***:* In Successive Interference Cancellation, receiver arbitrarily takes one of the estimated symbols, and subtract its effect from the received sym 3.7 nd y_2. Also we can subtract the effect of $\hat{x}_1$ first or $\hat{x}_2$ first. To make that decision, we have to find out the transmit symbols (after multiplication with the channel) which came at higher power at the receiver. The received power at the both the antennas corresponding to the transmitted symbol x_1 is, $P_{x_1} = |h_{1,1}|^2 + |h_{2,1}|^2$.

The received power at the both the antennas corresponding to the transmitted symbol x_2 is,

$$P_{x_2} = |h_{1,2}|^2 + |h_{2,2}|^2.$$

If $P_{x_1} > P_{x_2}$ then the receiver decides to remove the effect of $\hat{x}_1$ from the received vec 3.8 id y_2 and then re-estimate $\hat{x}_2$.
In matrix notation, $r = hx_2 + n$

c) **Minimum Mean Square Error (MMSE)**

The MMSE estimator employs the second-order statistics of the channel conditions to minimize the mean-square error. Denote by Rgg, RHH, Ryy and the auto-covariance matrix of g, H & y respectively, and and by Rgy the cross covariance matrix between g & y. Also denote by σ_N^2 the noise variance $E\,|N|^2$. Assume the channel vector g and the noise are N uncorrelated, it is derived that:

$$R_{HH} = E\overline{HH}^H = E\,(Fg)^H = FR_{gg}F^H$$

$$\hat{H}_{MMSE} = \hat{F}_{g\,MMSE} = F\left[(F^H X^H)^{-1} R_{gg}{}^{-1}\sigma_N^2 + XF\right]^{-1}\bar{Y}$$

3.12

$$\hat{H}_{MMSE} = \left| R_{HH} + \sigma_N^2 \, XX^{H^{-1}} \right| \quad \widehat{H_{LS}}$$

The MMSE estimator yields much better performance than LS estimators, especially under the low SNR scenarios. A major drawback of the MMSE estimator is its high computational X complexity, especially if matrix inversions are needed each time the data in changes.

d) **Least Square (LS)**

B. ***Least square estimation is used to minimize the square distance between the received signal ar*** 3.13 ***ıinal signal. The least square estimates (LS) of the channel at the pilot subcarriers given by the following equation:*** $H_P^{\sim} = (X_P)^{-1} Y_P$

Channel estimation is based on standard LS techniques. We can write the transmitted and the received signals in vector form as:

$$r\ n = \left[r\ n,0\ ,r\ n,1\ ,......,r\ n,k-1\ \right]$$
$$s\ n = \left[s\ n,0\ ,s\ n,1\ ,......,s\ n,k-1\ \right]$$ 3.14

Where r n and s n are the vectors containing samples and respectively for k k0,1,...., 1 , and K is the total number of sub-carriers in an OFDM symbol. By simply r n k s n k, dividing and we get the frequency response of the channel plus some noise. In this way, we can express the estimated channel frequency response by:

$$\widehat{H}\ n,k = \frac{r\ \ n,k}{s\ \ n,k}\ ,\quad for\ k = 0,1,\ldots..K-1$$

Since the transmitted signal is BPSK with unit magnitude is

$$\frac{1}{s\ n,k} = s*\ \ n,k$$ 3.16

We can rewrite the above Equation as:

$$\hat{H}\ n,k = r\ \ n,k\ \ s*\ \ n,k\ ,\quad for\ k = 0,1,\ldots\ldots,K-1$$ 3.17

While implementing this estimation technique, the frequency responses of the channels corresponding nt OFDM blocks and sub-carriers are assumed to be independent of each other. Consequently, none of the correlation properties of the channel is used and the estimation is based on a Gaussian channel model. The block diagram of the LS channel estimator is shown in Figure 3.3.

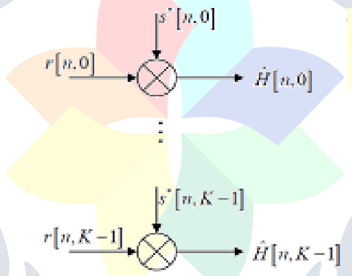

Figure 5: LS Channel Estimator

The advantage of LS algorithm is its simplicity, because no consideration of noise and ICI. So, without using any knowledge of the statistics of the channels, the LS estimators are calculated with very low complexity, but obviously it suffers from a high MSE. LS method, in general, is utilized to get initial channel estimates at the pilot subcarriers, which are then further improved via different methods.

a) **Adaptive Linear Mean Minimum Square Error (ALMMSE)**

C. ***The LMMSE channel estimator is designed to minimize the estimation MSE. The LMMSE estimate of the channel responses given as:***

$$\hat{h}_{lmmse} = R_{hh_{ls}} R^{-1}_{h_{ls}h_{ls}} \hat{h}_{ls} == R_{hh}\left(R_{hh} + \sigma^2{}_n (XX^h)^{-1}\right)^{-1} \hat{h}_{ls}$$

Where:

$$\hat{h}_{ls} = X^{-1} y = \left[\frac{y_0}{x_0}\ \frac{y_1}{x_1} \ldots \frac{y_{N-1}}{x_{N-1}} \right]^T$$

is the least-squares (LS) estimate of h, $\sigma^2 n$ is the variance of the additive channel noise, and the covariance matrices are:

$$R_{hh} = E\{hh^H\}$$

$$R_{hh_{ls}} = E\left\{ h\,\hat{h}_{ls}^{\ H} \right\} \quad R_{h_{ls}h_{ls}} = E\left\{ \hat{h}_{ls}\,\hat{h}_{ls}^{\ H} \right\}$$

The covariance matrices are:

$$R_{HH_P} = E\ \{hh^H\}$$

The LMMSE estimator (12) is of considerable complexity, since a matrix inversion is needed every time the training data in X changes. We reduce the complexity of this estimator by averaging over the transmitted data [10], i.e. we replace the term (XXh)-1 in (Eq. 3.18) with its expectation E (XXh)-1. Assuming the same signal constellation on all tones and equal probability on all constellation points, we get E (XXh)-1 E (1/xk)2I, here I is the identity matrix. Defining the average signal-to-noise ratio as:

$SNR = E|x_k{}^2|/\sigma^2{}_n$

We obtain a simplified estimator as:

$$H_P^{LMMSE} = R_{hh}\left(R_{hh} + \frac{\beta}{SNR} I_P\right)^{-1} H_P^{LS} \qquad 3.25$$

D. ***Where, β = Scaling Factor and given as:*** $\beta = E|x_k{}^2|E|1/x_k|^2$

E. ***Ip = Identity Matrix***

R_{HH_P} represent the cross-correlation matrix between all sub-carriers and the subcarriers with reference signals. $R_{H_P H_P}$ represent the auto-correlation matrix of the sub-carriers with reference signals. The high complexity of LMMSE estimator is due to the inverse matrix. Every time data changes, inversion is needed. The complexity of this estimator can be reduced by averaging the transmitted data. Then, the estimation can be done more accurately.

The performance of LMMSE estimator is much better than LS estimator, especially under the lower Eb / No and ALMMSE estimator could gain 10 -15 dB more of performance than LS [21]. However, because of the required matrix inversions, the computation is very complex when the number of subcarriers of OFDM system increases. Therefore, an important drawback of the ALMMSE estimator can be the high computational complexity.

a) **Maximal-Ratio Combining (MRC)**

MRC is the method of combining diversity in as follows:

1. The signals from each channel are added together,
2. The gain of each channel is made proportional to the RMS signal level and inversely proportional to the mean square noise level in that channel.
3. Different proportionality constants are used for each channel.

It is also known as ratio-squared combining and pre-detection combining. Maximal-ratio combining is the optimum combiner for independent AWGN channels. MRC can restore a signal to its original shape.

In MRC the equalized symbol is,

$$\hat{x}_2 = \frac{h^H r}{h^H h}.$$

Else if $Px_1 \le Px_2$ the receiver decides to subtract effect of $\hat{x}_2$ from the received vector y_1 and y_2, and then re-estimate $\hat{x}_1$.

In matrix notation, $r = hx_1 + n$

Optimal way of combining the information from multiple copies of the received symbols in receive diversity case is to apply Maximal Ratio Combining (MRC). The equalized symbol is, $\hat{x}_1 = \frac{h^H r}{h^H h}$.

Channel and Signal Model

The method of channel estimation in which the Pilot Sequence are added after the signal is modulated and before the signal is transmitted. At the receiver pilot sequences are removed and the signal is extracted from the received signal.

The multipath time-varying channel model is considered in which the impulse response is expressed as:

$$H(n) = \frac{1}{\sqrt{N_p}} \sum_{p=0}^{N_p-1} \alpha_p e^{j\frac{2\pi}{N} f_{D_p}} \delta(n-\tau_p)$$

Where, α_p, f_{Dp} and τ_p are the complex amplitude, normalized Doppler shift and time delay for the *p*th multipath arrival respectively.

For an OFDM signal the transmitted signal is expressed as:

$$X(n) = \frac{1}{N}\sum_{i=0}^{N-1} d_i e^{j\frac{2\pi}{N}in}$$

Where, d_i is BPSK-modulated signal carried by the ith subcarrier, N is the number of subcarriers. After the modulation of OFDM, the pilot sequences are added to the signal and then transmitted.

The received signal can be expressed as:

$$Y(n) = H(n)\,X(n) + W(n)$$

Where, Y(n) is the received signal, X(n) is the transmitted signal, H(n) is the impulse response and W(n) is the additive white Gaussian noise (AWGN).

The received signal is derived from the convolution of the input signal and channel impulse response. After removing the pilot sequences, the received signal can be expressed as:

$$Y(n) = \frac{1}{N\sqrt{N_p}}\sum_{p=0}^{N_p-1} \alpha_p e^{j\frac{2\pi}{N} f_{D_p} n}\sum_{i=0}^{N-1} d_i e^{j\frac{2\pi}{N} i(n-\tau_p)} + W(n)$$

III. EXPECTED RESULTS

The MATLAB/SIMULINK Communication tool box is used for implement the proposed model. In the channel estimation technique, the modified and improved Least Square (LS) method is designed on pilot sequence arrangement. The received signal which is receive at the receiver end which contain the pilot sequences, in which the number of bits are changed is estimated and detect the error in the signal. By which the Doppler diversity and the intersymbol interferences is reduced and the estimation could improve the BER performance.

IV. CONCLUSION

The channel estimation method in which the OFDM transmission technique has emerged as a promising candidate for high data rate and reliable wireless communication systems. Due to channel interference, the accuracy of channel estimation is an essential factor for a good receiver design in order to exploit the full potential capacity of the OFDM systems. The modified and improved LS method has to be proposed for the better performance of the system.

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