

Efficient DFT-based channel estimation for OFDM Systems on multipath channels

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Abstract— An improved discrete Fourier transform (DFT)-based channel estimation for orthogonal frequency division multiplexing systems is proposed. Conventional DFT-based channel estimations improve the performance by suppressing time domain noise. However, they potentially require information on channel impulse responses and may also result in mean-square error (MSE) floor due to incorrect channel information such as channel delay spread. In contrast, our proposed channel estimation can improve the performance by deciding significant channel taps adaptively without requiring any channel statistical information. Significant channel taps are detected on the basis of a predetermined threshold. The optimal threshold to reduce the MSE of the estimation is also derived, and it is confirmed by computer simulation. Simulation results demonstrate that the proposed algorithm can improve the MSE performance 6.5 dB compared with the conventional DFT-based estimation, and the MSE floor is not observed in any channels.

Keywords-Discrete Fourier Transform; DFT; Multiplexing systems; Conventional Channels;

I. INTRODUCTION

Orthogonal frequency division multiplexing (OFDM) has the most promise as a future high data rate wireless communication system due to its advantage in high bit-rate transmission over a time dispersive channels. However, to obtain high spectral efficiency, it is necessary to employ multilevel modulation schemes with non-constant amplitude (e.g. 16-QAM). They require coherent demodulation that needs to estimate and tracks parameters of fading channel. To achieve this, several channel estimation methods are studied and these methods can be divided into two categories. Frequency domain methods, especially the least square (LS) or the linear minimum mean-square error (LMMSE) estimation [1], are included in the first category. The LS estimation is the simplest channel estimation based on parallel Gaussian channel model in frequency domain. Since the LS does not use any information about channels, the performance is not quite acceptable. The LMMSE estimation can achieve better performance by using channel statistics such as channel covariance matrix in frequency domain and average signal-to-noise ratio (SNR). Moreover, since the LMMSE estimation requires very large amount of computation, such as matrix

inversion, there are many attempts to reduce the complexity of the LMMSE. Edfors et al. [1] reduce the computational complexity of a LMMSE channel estimation by means of a singular value decomposition. Noh et al. [2] also propose a low-complexity LMMSE estimation method by partitioning off channel covariance matrix into some small matrices on the basis of coherent bandwidth. However, these modified LMMSE methods have still quite high-computational complexity for practical implementation and require exact channel covariance matrices. The second category of channel estimation focuses on the transform domain by using the discrete Fourier transform (DFT) processing. Zhao and Huang [3] introduce additional DFT processing to obtain the frequency response of LS estimated channel. In contrast to the frequency domain estimation, the transform domain estimation method uses the time domain properties of channels. Since a channel impulse response (CIR) is not longer than the guard interval in OFDM system, the LS and the LMMSE are modified in [4, 5] by limiting the number of channel taps in time domain. Dowler et al. [6] show the performance of various channel estimation methods and yield that the significant performance benefits if the maximum channel delay is known. Minn and Bhargava [7] improve upon this idea by considering only the most significant channel taps. This method, however, can achieve performance improvement only when the channel is sparse, and it is based on statistical channel parameters such as maximum channel delay time or channel sparsity. Further, these methods in [4, 5, 7] result in MSE floor in high SNR due to energy loss caused by missing informative channel taps. Similarly, Athaudage and Jayalath [8] propose a novel delay spread estimation method using cyclic prefix (CP) that can be useful to improve the DFT-based estimation. However, this method requires many OFDM symbols to estimate the accurate delay spread, making it unsuitable for the real-time applications. In this paper, we propose a modified DFT based channel estimation algorithm to cope with the MSE floor and increase the estimation accuracy. In the proposed algorithm, to find all informative channel taps, we detect significant channel taps and determine adaptively the maximum channel delay time with respect to a threshold. The threshold is obtained in order to reduce MSE of estimation and is related to time domain noise power. Hence, the proposed

channel estimation requires two additional blocks One is noise power estimator block and the other is significant channel tap detector (SCTD) block.

II. SYSTEM MODEL AND CHANNEL ESTIMATION SYSTEM DESCRIPTION

We consider an OFDM system that consists of N subcarriers and each subcarrier consists of data symbol X[k], where k represents the subcarrier index. The OFDM transmitter uses an inverse DFT (IDFT) of size N for modulation. Then the transmitted OFDM signal in discrete-time domain can be expressed as

$$y[n] = x[n] \otimes h[n] + w[n], \quad 0 \leq n \leq N-1$$

where n is the time domain sample index of an OFDM signal. To avoid inter-symbol interference and consequent inter-carrier interference, a CP is appended to the OFDM symbol. After passing through a multipath channel and removing CP, one received discrete-time domain OFDM signal y[n] is represented by where \otimes denotes cyclic convolution operation, w[n] is independent and identically distributed additive white Gaussian noise (AWGN) sample in time domain with zero mean and variance and h[n] is the discrete time CIR given by

$$h[n] = \sum_{i=0}^{L-1} \alpha_i [n-1]$$

where α_i represents a different path complex gain that is complex Gaussian, i is the index of the different path delay that is based on sampling time interval, which means there is no channel power loss caused by sampling time mismatch [9] and L is the length of the CIR. For simplicity, time dependence nature of the CIR is suppressed in the notation. At the receiver, we assume that the guard interval is longer than the maximum channel delay and the synchronisation is perfect. Then, the kth subcarrier output in frequency domain can be represented by

$$y[k] = X[k] H[k] + W[k], \quad 0 \leq k \leq N-1$$

A. LS Channel estimation

Coherent detection requires knowledge of the sampled channel frequency response H[k]. In a simple technique to estimate H[k] is to send preamble or pilot signals X[k] at all subcarriers. Then the channel frequency response in all subchannels can be estimated by

$$H_{LS}[k] = \frac{Y[k]}{X[k]} = H[k] + \frac{W[k]}{X[k]}, \quad 0 \leq k \leq N-1$$

This channel estimation algorithm is called the LS estimation. Since LS does not make use of any channel information, no noise power is depressed. As a result, the LS estimation has a large MSE, which does not guarantee a satisfactory performance. The individual MSE of kth subcarrier [5] is

$$MSE_{LS}[K] = \frac{\beta}{SNR}$$

B. DFT-based channel estimation

DFT-based channel estimation exploits a property of OFDM systems having the symbol period much longer than the duration of the CIR. Since the estimated CIR from LS has most of its power concentrated on a few first samples, the DFT-based estimation reduces the noise power that exists in only outside of the CIR part. The basic block diagram of DFT-based estimation is shown in Fig. 1. The nth estimated sample of CIR can be The CIR is typically limited to the length of CIR L which is less than the guard interval and much smaller compared with the number of subcarrier N. Conventionally, the CIR can be described as expressed with the LS estimation, then we have

$$h[n] = \begin{cases} IDFT\{H[k]\}, & 0 \leq n \leq L-1 \\ 0, & 0 \leq n \leq L-1 \end{cases}$$

all information of channels is contained in the first L samples and other samples are only noise. Hence taking the first L samples only and ignoring noise-only samples, we can obtain a better performance. Expressing these processes in equations, we obtain

$$MSE_{DFT}[K] = \frac{L}{N} \frac{\beta}{SNR}$$

the DFT-based channel estimation is denoted as With assumption that channels have sample-spaced impulse, the individual MSE of DFT-based channel estimation is given as where N is constant but L is usually unknown variable depending on channel environments. For example, bad urban has almost two times larger L than typical urban . To express this property, we define two parameters of the channels: La is the length of the CIR at the present channel and Lmax is the maximum value among the La. As shown in (13), the conventional DFT-based channel estimation reduces the MSE to a fraction L/N compared with the LS estimation. However, if we set L as an arbitrary value, which is smaller than La, MSE will increase significantly by losing the energy in the missing channel taps [4]. As a result, we should take L as Lmax to prevent the MSE floor regardless of the lengths of channel impulse responses, which vary as channel environments. However, in this case, the MSE of

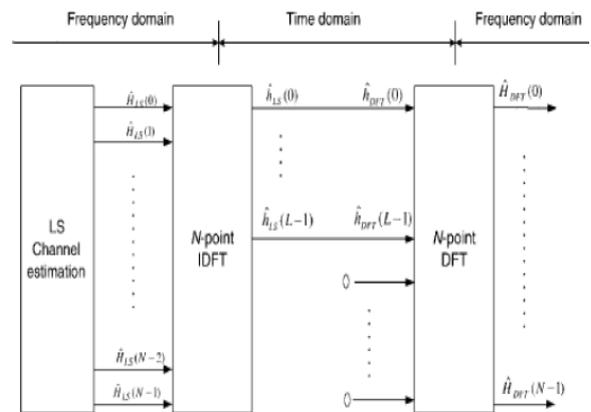


Fig 1: Block diagram of DFT- based channel estimation

conventional DFT-based channel estimation is fixed as (L_{\max}/N) (b/SNR), even though L_a is much smaller than L_{\max} . On the contrary, if we adaptively choose L as the length of the present channel ($L \approx L_a$), we can expect the better performance with small L_a . Hence, we propose an improved estimation algorithm in the next section on the basis of this idea.

III. PROPOSED CHANNEL ESTIMATION

To estimate an unknown CIR that varies according to channel environments, proposed channel estimation has two additional blocks: the noise power estimator block and the SCTD, as shown in . First, in the noise power estimator, we estimate the time domain noise power by averaging the noise-only existing part and decide the threshold (λ), which will be used in SCTD. Expressing the noise estimation in equation, we have With this estimated noise power in the SCTD, we detect significant channel taps. Since L_{\max} is the largest channel

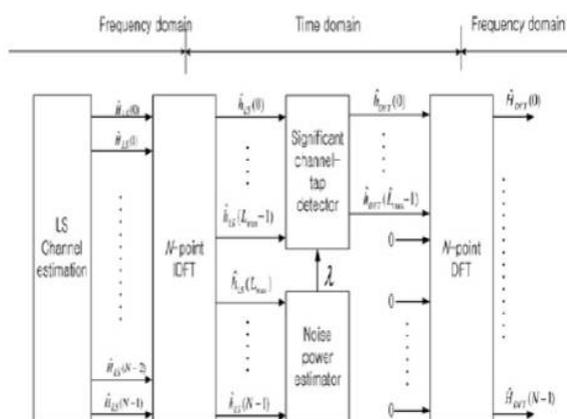


Fig2:Block diagram of the proposed algorithm

where $1/N_b/SNR$ can be expressed with time domain noise power because the estimated channel includes the noise power as well as channel information, we add the noise power to both side of then, we obtain is optimal threshold for the average observations of channel samples. However, in,we should set the threshold λ for an instantaneous observation j not for the mean value . Since the available sample is just one sampled value, the only unbiased point estimator is equal to the observed one sampled data itself . Therefore the optimal threshold λ for instantaneous sample is same as Without losses of generality, training signal $X[k]$ is modulated with average power one, and swt 2 can be replaced as estimated noise power from Then, the threshold is expressed as

$$H_{prop}[k]=DFT\{h_{prop}[n]\}$$

The final frequency response of proposed estimation is Throughout these processing, we can adaptively detect the channel taps against the variable SNR and channel environments.

IV. SIMULATION RESULTS

In this section, we investigate the performance of the proposed channel estimation algorithm on multipath channels.

The MSE and bit-error-rate (BER) performances are examined. An OFDM system with symbols modulated by 16-QAM is used on multipath channels. The system bandwidth is 20 MHz, which is divided into 128 tones with a total symbol period of 8 ms, of which 1.6 ms constitutes the CP, and the carrier frequency is 2.4 GHz. An OFDM symbol thus consists of 160 samples, 32 of which are included in the CP. The channel has L paths determined by rms channel delay, and the amplitude of the each path varies independently with an exponential power delay profile . Unit delay of channel is assumed to be the same as OFDM sample period. Thus, there is no power losses caused by non-sample spaced . We assume channels are static over one OFDM frame, where the preamble is 1 OFDM symbol long and data are composed of 30 OFDM symbols. A new channel is generated at each simulation run.

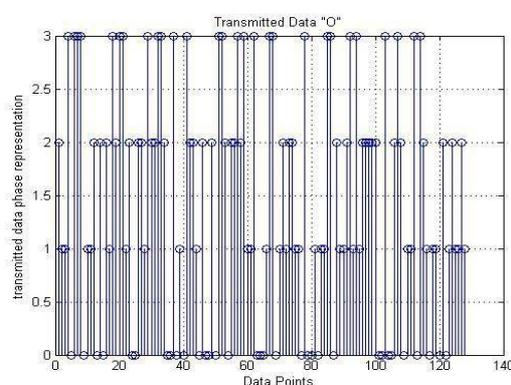


Fig 3.Data transmission points in OFDM

and the system performance is averaged over CIR realisations. We simulate with three different channels shown in . The rms delay of the worst channel has 150 ms that is consisted of 31 sample based on sampling time in this simulation. As a result, the largest channel length (L_{\max}) becomes confirms the MSE performance of conventional DFT-based estimation with and the assumption that L is same as the maximum channel delay tap of present channel (L_a). Usually, the small L_a which is known at the receiver, indicates the small MSE. However, note that if L is smaller than maximum channel delay tap of present channel L_a (e.g. $L \approx 17$ and $L_a \approx 31$), then MSE floor will be caused by missing informative channel taps. As a result, to prevent the MSE floor, it is clear that L should be set as L_{\max} . At the same time, the MSE of channel estimation, however, is fixed as $(31/128)$ (b/SNR), even if an actual channel has L_a smaller than L_{\max} , especially, in case of $L_a \approx 7$ or 17.

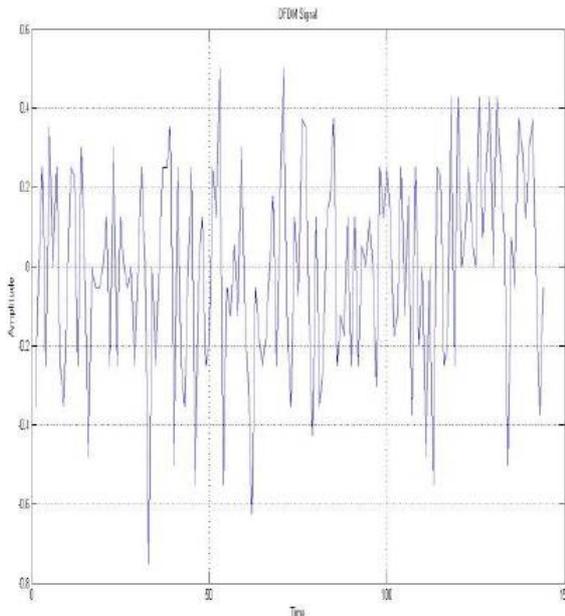


Fig 4:Amplitude level of OFDM signal

shows the performance improvement of proposed DFT-based channel estimation.

Compared with conventional DFT-based channel estimation, proposed channel estimation can reduce the MSE significantly according as the rms delay decreases. Especially, required SNR of proposed estimation is 2.5 dB lower than the conventional DFT-based estimation at rms $\frac{1}{4}$ 80 ms and 6.5 dB lower at rms $\frac{1}{4}$ 30 ms. shows that the average MSE through

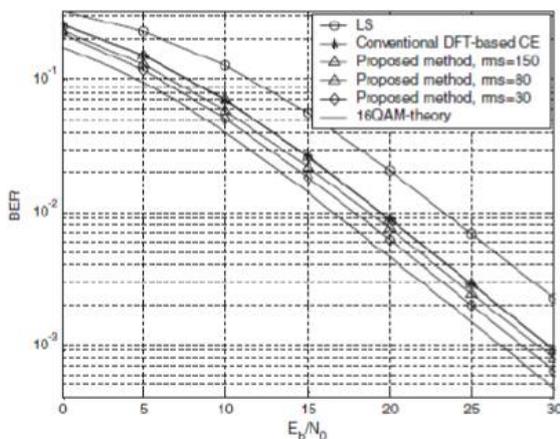


Fig 5:Comparing BER performance with proposed algorithm and conventional DFT-based estimation

several exponential channels for different thresholds at various SNR. From these results, it is obvious that the threshold in (21) is optimal when $1 \frac{1}{4} 2 _ s2$ wt on every SNR. Finally, the BER performance of the proposed method is examined. As shown in , when we apply the proposed

algorithm, the required SNR is $_1.5$ Db lower than the conventional DFT-based estimation with rms $\frac{1}{4}$ 30 ms.

V. CONCLUSION

In this paper, an efficient DFT-based channel estimation for OFDM on multipath environments has been presented. The proposed estimation can improve the performance adaptively against various channel environments and does not require any information about the channel statistics. Moreover, it does not make MSE floor in any channels contrary to conventional DFT-based method. We derive the optimal threshold to determine significant channel taps. The derived threshold is two times of the time domain noise power. The noise power can be estimated by averaging noise-only existing part. The simulation results show that the proposed algorithm can significantly reduce the MSE and improve the BER performance as compared with conventional schemes.

REFERENCES

- [1] Edfors, O., Sandell, M., van de Beek, J.-J., Wilson, S.K., and Borjesson, P.O.: 'OFDM channel estimation by singular value decomposition', IEEE Trans. Commun., 1998, 46, pp. 931–939
- [2] Noh, M., Lee, Y., and Park, H.: 'A low complexity LMMSE channel estimation for OFDM', IEE Proc. Commun., 2006, 153, (5), pp. 645–650
- [3] Zhao, Y., and Huang, A.: 'A novel channel estimation method for OFDM mobile communication systems based on pilot signals and transform-domain processing'. IEEE Vehicular Technology Conf., 1998, vol. 46, pp. 931–939
- [4] van de Beek, J.-J., Edfors, O., Sandell, M., Wilson, S.K., and Borjesson, P.O.: 'On channel estimation in OFDM systems'. IEEE Vehicular Technology Conf., July 1995, vol. 2, pp. 815–819
- [5] Edfors, O., Sandell, M., van de Beek, J.-J., Wilson, S.K., and Borjesson, P.O.: 'Analysis of DFT-based channel estimators for OFDM', Wirel. Pers. Commun., 2000, 12, (1), pp. 55–70
- [6] Dowler, A., Doufexi, A., and Nix, A.: 'Performance evaluation of channel estimation techniques for a mobile fourth generation wide area OFDM system'. IEEE Vehicular Technology Conf., September 2002, vol. 4, pp. 2036–2040
- [7] Minn, H., and Bhargava, V.K.: 'An investigation into time-domain approach for OFDM channel estimation', IEEE Trans. Broadcast., 2000, 64, (4), pp. 240–248
- [8] Athaudage, C.R.N., and Jayalath, A.D.S.: 'Delay-spread estimation using cyclic-prefix in wireless OFDM systems', IEE Proc. Commun., 2004, 151, pp. 559–566
- [9] Yang, B., Cao, Z., and Letaief, K.B.: 'Analysis of low-complexity windowed DFT-based MMSE channel estimation for OFDM systems', IEEE Trans. Commun., 2001, 49, pp. 1977–1987
- [10] Yeh, C.S., and Lin, Y.: 'Channel estimation using pilot tones in OFDM systems', IEEE Trans. Broadcast., 1999, 45, (4), pp. 400–409
- [11] COST 207 TD(86)51-REV 3 (WG1), 'Proposal on channel transferfunctions to be used in GSM tests late 1986', Standard, 1986
- [12] Kay, S.M.: 'Fundamentals of statistical signal processing: estimation theory' (Prentice-Hall, 1993)
- [13] Halford, S., Halford, K., and Webster, M.: 'Evaluating the performance of HRb proposals in the presence of multipath' (Intersil Corporation, 2000), (doc.:IEEE802.11-00/282r2)