

Channel Estimation based on Time-domain Threshold for OFDM Systems

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Abstract— Channel estimation for OFDM systems is usually carried out in the frequency domain by the least-square (LS) method with known pilot symbols. The LS estimator has a merit of low complexity but may suffer from noise because it does not consider any noise effect in obtaining its solution. To strengthen the noise immunity of the LS estimator, we consider estimation noise in time domain. The noise influence at the time-domain channel coefficients could be reduced by reasonable selection of the threshold value. To achieve this, we propose a channel estimation method based on time-domain threshold, where the threshold value is a standard deviation of noise obtained by wavelet decomposition. Computer simulation shows that the estimation performance of the proposed method approaches to that of the known-channel case in terms of bit-error rates after the Viterbi decoder in overall SNRs.

I. INTRODUCTION

Orthogonal frequency division multiplexing (OFDM) has received considerable interests for its advantages in high-bit-rate transmissions over frequency-selective fading channels. In OFDM systems, the entire channel is divided into many narrow subchannels, which are transmitted in parallel. This results in the increase of the time duration corresponding to an OFDM symbol. With the insertion of cyclic prefix for the guard interval (GI), the increased symbol duration reduces the inter-symbol interference (ISI) [1]. This feature has been adopted as a standard for digital audio broadcasting (DAB) [2], Digital Video Broadcasting-Terrestrial (DVB-T), the European digital terrestrial television standard [3] for signal transmission.

Channel estimation for OFDM systems is often carried out in the frequency domain using known symbols, referred to pilots [4], [5]. The Least-Squares (LS) estimator has an advantage of low complexity but its performance is not as good as that of the MMSE estimator [6]. The MMSE estimator shows good performance but requires knowledge of the channel statistics and the signal-to-noise ratio (SNR) and thus can not avoid high computational complexity [7]. For those reasons, the LS estimator has been preferable for

channel estimation. However, the LS estimator is susceptible to noise because it does not consider any noise effect in obtaining its solution.

To compensate for the vulnerability to noise of the LS estimator, we apply a time-domain approach for channel estimation. The first step is to perform an inverse FFT (IFFT) of the frequency-domain estimated channel coefficients. Since the number of meaningful channel coefficients is practically much smaller than the IFFT size, most of channel coefficients will have small values or no energy except for noise. Estimation noise can be removed by forcing the channel coefficients whose gain is less than a threshold value to be zero, where the threshold value becomes a crucial factor of noise-reduction performance and thus should be appropriately chosen [8]. Conclusively, it is required to find the optimal threshold value to get the result of noise-robust channel estimation. To achieve this, we use wavelet decomposition introduced in [9], which transforms and splits frequency-domain channel coefficients into approximation and detail. The standard deviation of the estimation noise can be obtained from the detail and is adopted as a threshold value for noise reduction.

This paper is organized as follows. In section II, a baseband model of the OFDM system with pilot-based signal correction and channel estimation are presented. Section III describes the proposed noise-reduction method based on time-domain approach for robust channel estimation. Simulation results given in Section IV show that the bit error rate (BER) performance obtained after equalization followed by Viterbi decoding of the proposed method is superior to the conventional estimation method and is comparable to the ideal case of a known channel. Section V concludes the paper.

II. SYSTEM DESCRIPTION

A. A Baseband Model of an OFDM System

A baseband model of an OFDM system is shown in Fig. 1. At the transmitter, incoming binary information data are first grouped and mapped according to the modulation scheme. After pilot insertion, the modulated data $X(k)$ are sent to an IFFT (inverse fast Fourier transform) block and are

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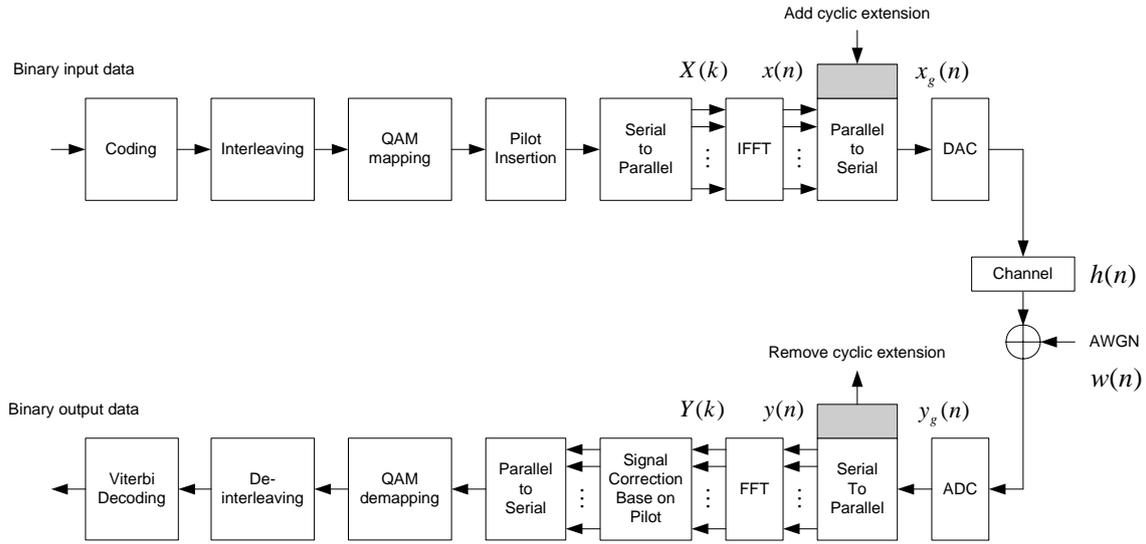


Fig. 1. Baseband model of an OFDM system.

transformed into time-domain signal $x(n)$ as

$$x(n) = IFFT[X(k)] = \sum_{k=0}^{N-1} X(k) \exp(j2\pi kn/N), \quad (1)$$

$$n = 0, 1, \dots, N-1,$$

where N is the number of subcarriers and k is the index of subcarriers. The guard interval is inserted to prevent possible inter-symbol interference (ISI) in OFDM systems using a cyclic prefix which contains a copy of the last part of an OFDM symbol. The transmitted signal passes through the frequency selective fading channel with additive white Gaussian noise (AWGN). At the receiver, the guard interval is removed and the pilot-based signal correction is performed after FFT, followed by decoding.

B. Channel Estimation in an OFDM system

Channel estimation in an OFDM system is often performed in frequency-domain using pilots. Based on the principle of the OFDM transmission scheme, there are two major types of pilot arrangement. The first is block-type and the second is comb-type. The channel estimation with comb-type pilot arrangement consists of two consecutive procedures. The pilot signals are first extracted from the received signal, and the channel coefficients are estimated based on channel estimation schemes. Then, the coefficients of the data subcarriers are interpolated by using the neighboring channel responses at the pilot frequencies. When the duration of the channel impulse response is shorter than the guard interval, there is no ISI between OFDM symbols. Assuming also there is no synchronization error, the FFT of the received signal $Y(k)$ can be formulated as

$$Y(k) = H(k) \cdot X(k) + W(k), \quad (3)$$

where $H(k)$ is the channel transfer function and $W(k)$ is the FFT of the time-domain AWGN, $w(n)$. The received pilot signals are extracted from $Y(k)$. For the pilot subcarriers, the transmitted information $X(k_p)$ is known to the receiver. Therefore, the channel coefficients at pilot frequencies can be estimated simply using

$$\hat{H}(k_p) = \frac{Y(k_p)}{X(k_p)} = H(k_p) + W'(k_p), \quad (4)$$

where $W'(k_p)$ is the noise effect existing at the estimated channel coefficients. The channel estimation scheme used in (4) is based on the LS method [6]. In order to estimate the channel coefficients at data subcarriers, we use the piecewise linear interpolation method due to its simplicity. In the linear interpolation case, two consecutive pilot carriers are used to determine the channel response for data subcarriers. Assuming that the channel coefficient between $H(k_p)$ and $H(k_{p+1})$ is estimated, then the channel response is obtained by

$$\hat{H}(k) = \left(\frac{\hat{H}(k_{p+1}) - \hat{H}(k_p)}{k_{p+1} - k_p} \right) (k - k_p) + \hat{H}(k_p). \quad (5)$$

After interpolation, we can obtain all channel coefficients [10].

III. NOISE REDUCTION BY TIME-DOMAIN THRESHOLD

As we described in section II, the channel coefficients can be estimated based on the LS method with the assumption of no interpolation error as

$$\hat{H}(k) = \frac{Y(k)}{X(k)} = H(k) + W'(k), \quad (6)$$

where $W'(k)$ is added noise existing at the estimated channel coefficients. In OFDM systems, channel estimation is requisite for coherent detection but the LS method is subject to noise because it does not care any noise in obtaining its solution. To overcome the weakness of the LS estimator, we consider a time-domain approach.

As shown in (6), the estimated channel coefficients consist of the FFT of the ideal channel coefficients and the added noise spectrum. Since the time-domain channel coefficients are obtained by IFFT of the estimated channel transfer function, we can regard the time-domain coefficients as

$$\hat{h}(n) = h(n) + w'(n), \quad (7)$$

where $h(n)$ is the impulse response of the channel and $w'(n)$ is the estimation noise. The estimation noise is also AWGN because IFFT is a linear transform and $X(k)$ has an average power of 1 by the normalization of the transmission signal power. Since a large number of time-domain channel coefficients have low energy because the length of the practical communication channel is shorter than that of an OFDM symbol. These ignorable coefficients can be considered to be entirely due to estimation noise, which may be removed by forcing the channel coefficients whose energy is less than a threshold value to be zero [8]. Consequently, it is required to find the optimal threshold value for noise-robust channel estimation.

There have been some useful threshold values for the use of noise reduction. Most of them are highly associated with the energy of the noise added to the signal. One of available threshold values can be determined by estimating the maximum channel tap's energy [8]. Estimated noise power can be used as a threshold value and was obtained by averaging the noise-only existing part in the time-domain channel coefficients in [11]. However, these threshold values do not assure the optimality in the sense of estimation accuracy. To approach the optimality of a threshold value, we introduce a standard deviation of the noise added to the signal, which can be obtained by [9]

$$\hat{\sigma}_k = \frac{\text{median}(|d_i|)}{0.6745}, \quad (8)$$

where d_i are coefficients corresponding to the “detail” obtained by the wavelet decomposition of the estimated channel coefficients obtained in frequency domain. Frequency-domain-estimated channel coefficients are divided into the important coefficients having large absolute values and non-significant coefficients. The unimportant terms referred to the “detail” are used to obtain the standard

TABLE I
MULTI-PATH PROFILE (BRAZIL CHANNEL A)

Delay (μs)	Amplitude (dB)
0.0	0.0
+0.15	-13.8
+2.22	-16.2
+3.05	-14.9
+5.86	-13.6
+5.93	-16.4

TABLE II
MULTI-PATH PROFILE (BRAZIL CHANNEL D)

Delay (μs)	Amplitude (dB)
0.0	-0.1
+0.48	-3.9
+2.07	-2.6
+2.90	-1.3
+5.71	0.0
+5.78	-2.8

deviation of the noise added to the estimated channel coefficients by being taken the median absolute detail divided by 0.6745. This constant proposed in [9] was based on the wavelet coefficients statistics. From the Parseval's theorem, the total average power of the time-domain noise is the sum of the average power in each harmonic component. Therefore, the $\hat{\sigma}_k$ can be directly adopted as a threshold value for noise removal. After adopting the threshold, the modified channel responses are transformed into frequency-domain coefficients in the FFT block. Through these procedures, we get noise-robust channel coefficients.

IV. SIMULATION RESULTS

We performed computer simulations to verify the performance of the proposed method applied to the OFDM system with comb-type pilot arrangement. The pilot spacing is 4. The elementary symbol period was set to be 0.19375 μs referred to the DVB-T standard [11]. “Brazil channel A” and “Brazil channel D” were used in the simulation as a mild channel condition and as a severe one, respectively [12]. These two channels were used for verifying the robustness to channel characteristics of the proposed method. The channel information of “Brazil channel A” and “Brazil channel D” are given in Table I and II, respectively.

We assumed that there were no synchronization error and no frequency offset since the purpose of the simulation is only to compare the performances of the channel estimation with and without the proposed method. The code rate was 1/2 and the generator polynomials of the mother code are $G_1 = 171_{\text{OCT}}$ and $G_2 = 133_{\text{OCT}}$ for output, respectively. The number of subcarriers is 2048 and the OFDM symbol duration is 224 μs . We chose the guard interval to be greater than the maximum delay spread in order to avoid ISI. The guard interval duration is +14 μs . As shown in Table I and II, the maximum channel delays are +5.93 μs and +5.78 μs . We

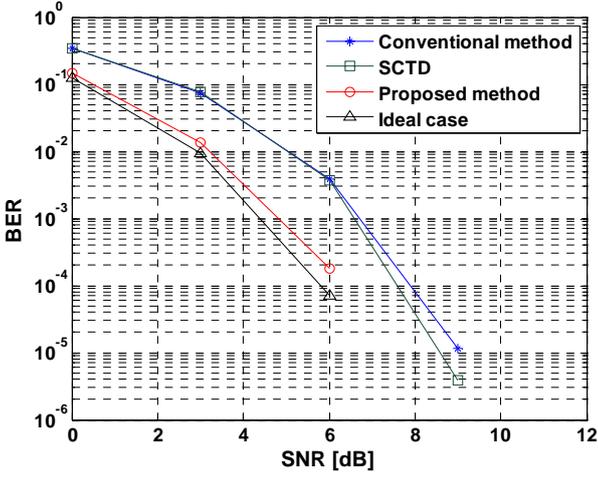


Fig. 2. Comparison of BER performances after one-tap equalization with the estimated channel coefficients obtained by conventional method, the SCTD method, the proposed method, and the original coefficients in case of QPSK at Brazil channel A.

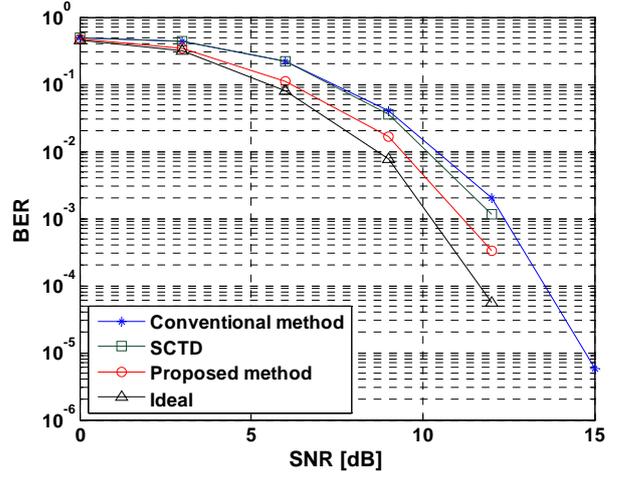


Fig. 3. Comparison of BER performances after one-tap equalization with the estimated channel coefficients obtained by conventional method, the SCTD method, the proposed method, and the original coefficients in case of 16-QAM at Brazil channel A.

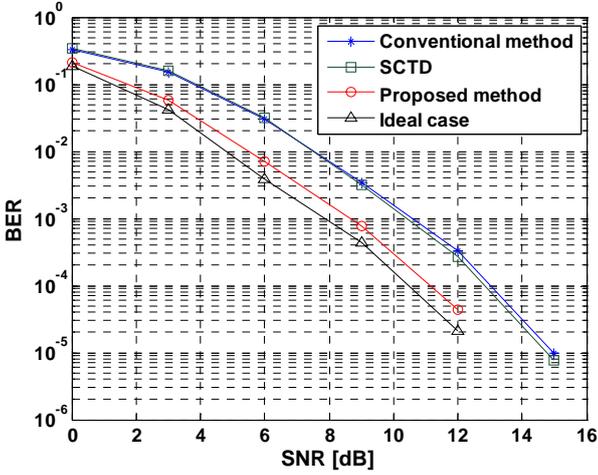


Fig. 4. Comparison of BER performances after one-tap equalization with the estimated channel coefficients obtained by conventional method, the SCTD method, the proposed method, and the original coefficients in case of QPSK at Brazil channel D.

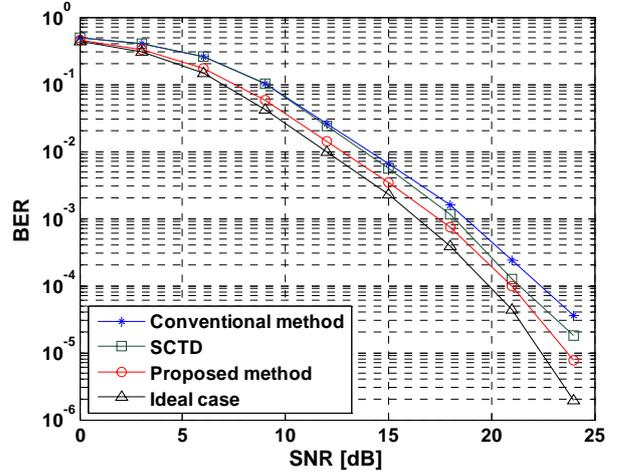


Fig. 5. Comparison of BER performances after one-tap equalization with the estimated channel coefficients obtained by conventional method, the SCTD method, the proposed method, and the original coefficients in case of 16-QAM at Brazil channel D.

used “db4” for a wavelet filter to estimate the threshold values and decomposition level was 1.

We used the performance measure of a BER computed after one-tap equalization with the inverse of estimated channel coefficients followed by Viterbi decoding. We adopted the simplified LLR (Log-Likelihood Ratio) computation for Viterbi decoding [13]. Transmitted data were mapped based on QPSK and 16-QAM. We used a piecewise linear interpolation filter to estimate data subcarriers. The length of the interpolation filter was 7.

Fig. 2 shows the BER curves of the channel estimation after one-tap equalization followed by Viterbi decoding with QPSK modulation under “Brazil channel A.” The legend of “Ideal case” means the channel estimation with the original

channel coefficients and “SCTD” means the performance of the channel estimation method based on the SCTD (significant channel tap detection) studied in [11]. As shown in Fig. 2, the proposed method improves the channel estimation performance in overall SNRs similar to the ideal case. Fig. 3 is the BER curves of the conventional methods and the proposed method in the case of 16-QAM at “Brazil channel A.” The legends are all the same as those in Fig. 2. There is a little degradation of error performance as the bit rate increases. For example, at SNR = 6 dB, the BER performance degradation range is about 0.166. Nevertheless, the proposed method can improve the BER performance of the channel estimator irrespective of the bit rate.

Fig. 4 shows the BER performance of the conventional method, SCTD method and the proposed method in the case

of QPSK at “Brazil channel D.” “Brazil channel D” is severer channel which has an almost 0-dB ghost. We can see that the proposed method can improve the channel-estimation performance in overall SNR conditions almost the same as the ideal case. With the same condition except for the modulation of 16-QAM, the results are shown in Fig. 5. The proposed method improves the BER performance of the conventional method by more than 1.5 dB in overall SNRs.

V. CONCLUSION

We proposed a noise-robust channel estimation method for OFDM systems based on time-domain threshold. The threshold value was determined by obtaining a standard deviation of estimation noise through the wavelet decomposition of the frequency-domain estimated channel coefficients. The estimated channel by the proposed method works as precisely as the known channel in terms of the BER at the end of the Viterbi decoder. Adopting the proposed method in OFDM systems, we may effectively remove estimation noise and thus obtain noise-robust channel coefficients, irrespective of the channel conditions and bit rates.

REFERENCES

- [1] Y. Li, L. J. Cimini, Jr., and N. R. Sollenberger, “Robust channel estimation for OFDM systems with Rapid Dispersive Fading Channels,” *IEEE Trans. Commun.*, vol. 46, pp. 902-915, July 1998.
- [2] B. Le Floch, R. Halbert-Lasalle, and D. Castellain, “Digital audio broadcasting to mobile receivers,” *IEEE Trans. Consum. Electron.*, vol. 35, no. 3, pp. 493-503, Aug. 1989.
- [3] ETSI, “Digital Video Broadcasting (DVB): Framing, channel coding and modulation for digital terrestrial television,” *ETSI EN30 744 V1.3.1*, Aug. 2000.
- [4] M. X. Chang and Y.T. Su, “Model-based channel estimation for OFDM signals in Rayleigh fading,” *IEEE Trans. On Communications*, vol. 50, pp. 540-544, Apr. 2002.
- [5] Y. Li, “Simplified channel estimation for OFDM systems with multiple transmit antennas,” *IEEE Trans. On Wireless Communications*, vol. 1, pp. 67-75, Jan. 2002.
- [6] J.-J. van de Beek, O. Edfors, M. Sandell, S. K. Wilson and P. O. Börjesson, “On channel estimation in OFDM systems,” in *Proc. IEEE 45th Vehicular Technology Conf.*, Chicago, IL, Jul. 1995, pp. 815-819.
- [7] M. Morelli and U. Mengali, “A comparison of pilot-aided channel estimation methods for OFDM systems,” *IEEE Trans. Signal Process.*, vol. 49, no. 12, pp. 3065-3073, Dec. 2001.
- [8] H. Minn and V. K. Bhargava, “An Investigation into Time-Domain Approach for OFDM Channel Estimation,” *IEEE Trans. On Broadcasting*, vol. 46, no. 4, Dec. 2000.
- [9] D. L. Donoho and I. M. Johnstone, “Ideal spatial adaptation by wavelet shrinkage,” *Biometrika*, vol. 81, no. 3, pp. 425-455, 1994.
- [10] M. Hsieh and C. Wei, “Channel estimation for OFDM systems based on comb-type pilot arrangement in frequency selective fading channels,” *IEEE Trans. Consumer Electron.*, vol. 46, no. 7, pp.931-939, Jul. 1998.
- [11] Y. Kang, K. Kim and H. Park, “Efficient DFT-based channel estimation for OFDM systems on multipath channels,” *IET Commun.*, 2007, 1, (2), pp. 197-202.
- [12] Mackenzie, ABERT, and SET, “General description of laboratory tests,” *DTV Field Test Report in Brazil*, July 2000.
- [13] Filippo Tosato and Paola Bisaglia, “Simplified Soft-Output Demapper for Binary Interleaved COFDM with Application to HIPERLAN/2,” *IEEE Trans. On Communications*, vol. 2, pp. 664-668, May 2002.