Noise Reduction for Channel Estimation Based on Pilot-Block Averaging in DVB-T Receivers

You-Seok Lee, Hyoung-Nam Kim, Member, IEEE, Sung Ik Park, Soo In Lee

Abstract — This paper proposes a noise-reduction method to improve the channel-estimation performance in Digital Video Broadcasting-Terrestrial (DVB-T) receivers. In DVB-T, the channel estimation is carried out through two processes of pilot signal estimation and channel coefficient interpolation and thus the performance of the channel estimation is seriously affected by the accuracy of the interpolation. Since the accuracy of the interpolation depends on signal-to-noise ratio, it is important to reduce noise existing at the reference coefficients for the interpolation. To achieve the reduction of noise, we use pilot-insertion type approximation in four consecutive OFDM symbols, which construct one block with the same channel coefficients at the same pilot positions. Considering the fade rate of channels, we average the estimated reference channel coefficients corresponding to the same subcarriers in two or more blocks. Computer simulation shows that the proposed method improves the estimation performance by more than 1 dB in terms of the symbol error rate obtained after equalization.

Index Terms — Channel estimation, orthogonal frequency division multiplexing (OFDM), DVB-T, Interpolation.

I. INTRODUCTION

In Digital Video Broadcasting-Terrestrial (DVB-T), the European digital terrestrial television standard [1], Orthogonal Frequency Division Multiplexing (OFDM) has been adopted for signal transmission. The channel estimation in the DVB-T system is often carried out in frequency domain using known symbols, referred to pilots [2], [3]. The channel-estimation algorithm based on DVB-T pilots is divided into pilot signal estimation and channel coefficient interpolation. The estimation is conventionally carried out using two-dimensional interpolation filter [4]. The first interpolation is executed in time direction and the second in frequency direction and thus the performance of the channel estimation is seriously affected by the accuracy of the interpolation. Since the accuracy of the interpolation depends on signal-to-noise ratio (SNR), it is important to reduce noise existing at the reference coefficients for the more precise channel estimation. To reduce the noise appeared in the reference channel coefficients estimated at pilot subcarriers, we use pilot-insertion type approximation in four consecutive OFDM symbols and bind up the four symbols into one block where the coefficients at the same pilot positions in each symbol are assumed to have the same value. Then, the average of estimated reference coefficients from two blocks or more is applied where the number of blocks is determined according to the fade rate of channels. The average process increases the robustness to noise for the interpolation due to the enhanced accuracy of the reference coefficients and lead us to improve the performance of the channel estimation.

To verify our proposed method, six interpolation schemes were considered and QPSK (Quadrature Phase Shift Keying), 16 QAM (16 Quadrature Amplitude Modulation) and 64 QAM were adopted as constellation schemes.

The paper is organized as follows. In Section II, the baseband model of the DVB-T system with pilot-based signal correction, pilot-insertion type approximation, and block-based channel estimation are presented. Section III describes the proposed noise-reduction method for good channel estimation. Simulation results given in Section IV show that the proposed method improves the SER performance obtained after equalization, compared to the conventional estimation method. Section V concludes the paper.

II. SYSTEM DESCRIPTION

A. Baseband Model of the DVB-T System

A baseband model of the DVB-T system is shown in Fig. 1. Incoming binary source at the OFDM transmitter side is first grouped and mapped according to the modulation scheme. After pilot insertion, the modulated data $X(k)$ is sent to an IFFT (inverse fast Fourier transform) block and is transformed into time-domain signals. 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

![Fig. 1. Baseband model of the DVB-T system.](image-url)
part of the 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 [5].

B. Pilot-Insertion Type Approximation

Based on the principle of the OFDM transmission scheme, there are two major types of pilot arrangement as shown in Fig. 2 [6]. The first kind of pilot arrangement shown in Fig. 2(a) is denoted as block-type pilot arrangement. The pilot signal is assigned to a particular OFDM symbol, which is sent periodically in time domain. The second kind of pilot arrangement shown in Fig. 2(b) is denoted as comb-type pilot arrangement. The pilot signals are uniformly distributed within each OFDM symbol. Since only some subcarriers contain the pilot signal, the channel response of nonpilot subcarriers will be estimated by interpolating neighboring pilot subchannels.

In DVB-T, the pilot cells are inserted every four OFDM symbols at the same position and are uniformly distributed every twelve subcarriers within each OFDM symbol as shown in Fig. 3. Hence the pilot pattern in DVB-T has the propensity of both the block type and the comb type. To estimate the channel coefficients, time-interpolation has to be performed first in order to decrease the pilot spacing to three. Consequently, in DVB-T, two-dimensional interpolation must be carried out to estimate the channel [4]. If a channel is slow fading, however, channel coefficients at the same subcarrier positions are assumed to be unchanged. We use this property for the channel estimation. We construct one block comprising four OFDM symbols as shown in Fig. 4(a). If a channel does not change while the four OFDM symbols are transmitted, the channel coefficients at the same subcarrier position within one block are all the same. It follows that the channel coefficients can be obtained by copying from the neighboring OFDM symbols. This process makes it possible to obtain all channel coefficients only one-dimensional interpolation in frequency direction. As a result, the pilot insertion in DVB-T is considered as a comb-type pilot arrangement distributed every three subcarriers, which is shown in Fig. 4(b). Since the channel coefficients can be completely estimated only by one-dimensional interpolation in comb-type pilot arrangement, we can reduce the computational complexity for the channel estimation in the DVB-T receiver under slow-fading channels.

C. Block-Based Channel Estimation

As we mentioned earlier, the pilot pattern in DVB-T is considered as the comb-type pilot arrangement. The comb-type pilot channel estimation consists of two consecutive algorithms. The first is to estimate the channel at pilot frequencies and the next is to interpolate the estimated channel coefficients because the number of pilot cells is much smaller than unknown data carriers [6]. We represented the block diagram of the channel-estimation algorithm based on comb-type pilots in Fig. 5. As shown in Fig. 5, the pilot signals are first extracted from the received signals after FFT, and the reference channel coefficients for the interpolation are estimated in frequency domain by comparing the pilot signals before and after transmission. Then, the channel responses of the subcarriers carrying data are estimated by interpolating neighboring pilot channel coefficients. When the duration of the channel impulse response is shorter than the guard interval, there is no ISI. Assuming also that there is no synchronization error, the relationship between $Y_p(k)$, $X_p(k)$, and the channel

![Fig. 4. Pilot-insertion type approximation.](image-url)
transfer function at the pilot frequencies $H_p(k)$ can be formulated as:

$$Y_p(k) = H_p(k) \cdot X_p(k) + W_p(k),$$

where $W_p(k)$ is the AWGN for the $k$-th pilot subcarrier after FFT [7] and the subscript $p$ denotes a pilot. Therefore, the frequency response of the channel at pilot frequencies can be estimated simply using:

$$\hat{H}_p(k) = \frac{Y_p(k)}{X_p(k)} = H_p(k) + W_p'(k)$$

where $W_p'(k)$ is the noise effect existing at the estimated channel coefficients. Channel estimation scheme in (2) is based on the Least Square (LS) method [8]. In order to estimate the channel coefficients at data subcarriers, the channel information at pilot frequencies is used for interpolation. After interpolation, we can obtain all channel coefficients $\hat{H}(k)$. In this paper, six interpolation schemes referring to [10] are tested. The formulas of the respective interpolations are given as:

1) Linear interpolation

$$h_{\text{Lin}}(x) = \begin{cases} 1 - |x| & 0 \leq |x| < 1 \\ 0, & \text{elsewhere} \end{cases}$$

2) Blackman-Harris windowed sinc interpolation

$$h_{\text{BH}}(x) = \begin{cases} h(x) \cdot w(x), & 0 \leq |x| < 3 \\ 0, & \text{elsewhere} \end{cases}$$

where $w(x)$ is given as follows:

$$w(x) = w_0 + w_1 \cos \left( \frac{2\pi x}{N} \right) + w_2 \cos \left( \frac{4\pi x}{N} \right)$$

with $N = 6$ and $w_0 = 0.42323$, $w_1 = 0.49755$, $w_2 = 0.07922$ respectively.

3) 4-point Lagrange interpolation

$$h_{\text{Lag}}(x) = \begin{cases} (1/2)|x|^3 - |x|^2 - (1/2)|x| + 1, & 0 \leq |x| < 1 \\ -(1/6)|x|^3 + |x|^2 - (11/6)|x| + 1, & 1 \leq |x| < 2 \\ 0, & \text{elsewhere} \end{cases}$$

4) Truncated sinc interpolation

$$h_{\text{truncated}}(x) = \begin{cases} \sin(\pi x)/\pi x, & 0 \leq |x| < 2 \\ 0, & \text{elsewhere} \end{cases}$$

5) 4-point cubic interpolation

$$h_{\text{cubic}}(x) = \begin{cases} (a + 2)|x|^3 - (a + 3)|x|^2 + 5a|x| - 4a, & 1 \leq |x| < 2 \\ 0, & \text{elsewhere} \end{cases}$$

where $a$ is $-1/2$.

6) Sixth-order Gaussian interpolation

$$h_b^6(x) = G^6(x, 2\gamma) - \gamma G^2(x, \gamma) - \frac{\gamma^2}{4} G^4(x, \gamma)$$

with

$$G^6(x, \beta) = \frac{1}{\sqrt{2\pi \beta}} \exp \left[ -\frac{x^2}{2\beta} \right]$$

and $\gamma = 0.4638115$, $\gamma = 0.8655995$, respectively.

**III. NOISE REDUCTION FOR INTERPOLATION**

As described in the previous sections, the interpolation plays a very important role in channel estimation in DVB-T receivers. Since the accuracy of the interpolation highly
depends on the reference coefficients whose accuracy is associated with the received SNR, it is required to reduce the noise effect at reference channel coefficients for precise interpolation. To achieve this, we propose a noise-reduction method of averaging estimated channel coefficients at the same subcarrier position in two or more blocks as shown in Fig. 6. The block diagram of the proposed method is shown in Fig. 7. Using (2) and letting \( \hat{H}_{p,i}(k) \) be the estimated channel coefficient at the \( k \)-th pilot frequency of the OFDM symbol in the \( i \)-th block produces

\[
\hat{H}_{p,i}(k) = H_{p,i}(k) + W'_{p,i}(k) .
\]

(12)

Averaging \( N \) coefficients in each \( N \) blocks yields a new reference channel coefficient \( \hat{H}_{p,d}(k) \), which is given by:

\[
\hat{H}_{p,d}(k) = \frac{1}{N} \left( H_{p,1}(k) + H_{p,2}(k) + \cdots + H_{p,N}(k) \right) + \frac{1}{N} \left( W'_{p,1}(k) + W'_{p,2}(k) + \cdots + W'_{p,N}(k) \right)
\]

(13)

If a channel is time-invariant, since the channel coefficients are all the same, (4) is reduced as:

\[
\hat{H}_{p,d}(k) = H_{p,org}(k) + \frac{1}{N} \sum_{i=1}^{N} W_{p,i}(k)
\]

(14)

where \( H_{p,org}(k) \) is the original channel coefficient at the pilot frequencies. The noise term, \( \frac{1}{N} \sum_{i=1}^{N} W_{p,i}(k) \) approaches to zero as the number of blocks to be averaged takes a sufficiently large value and the variance \( \sigma^2 / N \) also becomes zero. The reference coefficients for the interpolation are affected by the reduced noise and become almost the same as the original channel coefficients, resulting in the improvement of the channel-estimation performance. The improvement is due to the enhanced accuracy of the reference coefficients required for the interpolation by which we estimate channel coefficients corresponding to the nonpilot positions. The number of blocks used for averaging has to be determined by considering the buffer size for storing data, the fade rate of channels, and the desired accuracy of the estimation.

Note that the proposed method is irrelevant to the choice of interpolation schemes because it intends to improve the interpolation performance with the enhanced accuracy of the reference coefficients, not the interpolation method itself.

IV. SIMULATION RESULTS

We performed computer simulations to verify the performance of the proposed method applied to the DVB-T system. The channel profile was “Brazil channel D” which was the indoor channel used for the Laboratory Test in Brazil [9]. The channel information is given in Table I.

DVB-T system parameters used in the simulation are indicated in Table II. We assumed that there were no synchronization error and no Doppler spread since the purpose of the simulation is only to compare the performances of the channel estimation with and without the proposed method. We did not apply any coding scheme. DVB-T mode was decided to fix onto the 2K and 8 MHz derivative of the standard. Moreover, we chose the guard interval to be greater than the maximum delay spread in order to avoid ISI. As shown in Table I, the maximum channel delay is +5.78 µs and the elementary period is:

\[
T = \frac{7}{64} \, \mu s = 0.19375 \, \mu s
\]

(6)

for the 8 MHz channel [1]. The maximum channel delay corresponds to about 53 elementary symbols.

We used the performance measure of an SER computed after one-tap equalization with the inverse of estimated channel coefficients. Transmitted data were mapped based on QPSK, 16 QAM, and 64 QAM constellations. These three different types of data mapping are used to analyze the performance depending on the bit rates.

<table>
<thead>
<tr>
<th>TABLE I</th>
<th>MULTI-PATH PROFILE (BRAZIL CHANNEL D)</th>
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</thead>
<tbody>
<tr>
<td>Delay (µs)</td>
<td>Amplitude (dB)</td>
</tr>
<tr>
<td>0.0</td>
<td>-0.1</td>
</tr>
<tr>
<td>+0.48</td>
<td>-3.9</td>
</tr>
<tr>
<td>+2.07</td>
<td>-2.6</td>
</tr>
<tr>
<td>+2.90</td>
<td>-1.3</td>
</tr>
<tr>
<td>+5.71</td>
<td>0.0</td>
</tr>
<tr>
<td>+5.78</td>
<td>-2.8</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>TABLE II</th>
<th>SIMULATION PARAMETERS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameters</td>
<td>Specifications</td>
</tr>
<tr>
<td>DVB-T mode</td>
<td>2K</td>
</tr>
<tr>
<td>Number of carriers</td>
<td>1705</td>
</tr>
<tr>
<td>OFDM symbol duration</td>
<td>224 µs</td>
</tr>
<tr>
<td>Guard Interval</td>
<td>1/4 (512)</td>
</tr>
<tr>
<td>Signal Constellation</td>
<td>QPSK, 16QAM, 64QAM</td>
</tr>
<tr>
<td>Channel Model</td>
<td>Brazil channel D</td>
</tr>
</tbody>
</table>
Six interpolation methods were tested: 1) linear interpolation; 2) Blackman-Harris windowed sinc interpolation; 3) 4-point Lagrange interpolation; 4) truncated sinc interpolation; 5) 4-point cubic interpolation; and 6) sixth-order Gaussian interpolation [10]. The length of the interpolation filter was 13.

Fig. 8 shows the SER performance of the interpolation schemes without any noise-reduction methods in the case of the QPSK data mapping. The legends “Linear, Blackman-Harris windowed sinc, 4-point Lagrange, Truncated sinc, 4-point cubic, Sixth-order Gaussian” denote interpolation schemes of comb-type channel estimation based on LS at pilot frequencies. As shown in Fig. 8, the channel estimator with the 4-point cubic interpolation achieves the best performance in high SNRs and the estimator with the truncated sinc interpolation shows the worst performance in overall SNRs. Figs. 9-11 show the performance of the channel estimator incorporating the proposed noise-reduction method with the linear interpolation; the truncated sinc interpolation; and the 4-point cubic interpolation, respectively. The legends “(2blocks) and (4blocks)” represent the number of blocks used for averaging and “Ideal” means the performance of the conventional estimator with the original channel coefficients. From Figs. 9-11, we can see that the SER performance can be improved by more than 1 dB when the proposed method is applied. The proposed method with 4-block averaging shows the better performance than the method with 2-block averaging. Especially, the performance improvement achieved by the proposed method is remarkable for the truncated sinc interpolation scheme.

Fig. 12 shows the SER performance of the channel estimator using the respective interpolations in the case of 16 QAM. The legends are the same as those of Fig. 8. As shown in Fig. 12, there is a little degradation of error performance as the bit rate increases. For example, at SNR = 20 dB, the SER performance degradation range is approximately 0.065. Comparing the results in Fig. 12 with Fig. 8, we can find that the relative SER performances tested in the case of 16 QAM are the same as

![Fig. 8. Estimation performance after equalization using various interpolation schemes with QPSK.](image1)

![Fig. 9. Performance of the estimator with the proposed method using linear interpolation under “Brazil channel D” with QPSK.](image2)

![Fig. 10. Performance of the estimator with the proposed method using truncated sinc interpolation under “Brazil channel D” with QPSK.](image3)

![Fig. 11. Performance of the estimator with the proposed method using 4-point cubic interpolation under “Brazil channel D” with QPSK.](image4)
those in the case of QPSK. Figs. 13-15 show the performance of the channel estimator incorporating the proposed noise-reduction method with the linear interpolation; the truncated sinc interpolation; and the 4-point cubic interpolation, respectively. Even though there is a little degradation of error performance, the proposed method improves the SER performance by more than 1 dB. The proposed method with 4-block averaging also shows the better performance than the method with 2-block averaging.

Fig. 16 shows the SER performance of the channel estimator using the respective interpolation schemes in the case of 64 QAM. The performance degradation is more significant compared with the case of QPSK. At SNR = 20 dB, the degradation range is about 0.285. Figs. 17-19 show the performance of the proposed method with the linear interpolation; the truncated sinc interpolation; and the 4-point cubic interpolation, respectively. Comparing the results in Fig. 16 with Fig. 8, we can find that the relative SER performances tested in the case of 64 QAM are the same as those in the case of QPSK. Figs. 17-19 show the performance of the channel estimator with the proposed method using the linear interpolation; the truncated sinc interpolation; and the 4-point cubic interpolation, respectively. From Figs. 17-19, we can see that the proposed method improves the SER performance of the conventional estimator by more than 1 dB.

In the simulations, we used two and four blocks in averaging coefficients to reduce noise existing in reference coefficients. These blocks, however, result in the delay of 1,792 µs and 3,584 µs, respectively in the receiver. It is natural that as the more blocks are used for average, the amount of noise is reduced but the delay increases and the tracking performance is degraded. It is therefore required to determine the number of blocks used for averaging considering the trade-off between the desired accuracy of the channel estimation and the buffer size for storing blocks of data. As shown in the results, with 2-block averaging we improved by about 1 dB while with 4-block averaging we obtained above 1 dB improvement. The gradient of the performance improvement versus the number of blocks used for averaging is presented in Fig. 16.
of blocks is not so large. We conclude that four blocks, or more or less, are appropriate for the reduction of noise without the burden of hardware complexity and tracking of time-varying channels.

V. CONCLUSION

We proposed a noise-reduction method for interpolation by averaging of the estimated channel coefficients from two or more pilot-blocks based on pilot-insertion type approximation to improve the channel-estimation performance in DVB-T receivers. We applied the proposed method to six interpolation schemes for channel estimation. The estimator using the 4-point cubic interpolation shows the best performance and the estimator with the truncated sinc interpolation shows the worst performance. The proposed method improved the channel estimation performance more than 1 dB in terms of the SER for all SNR conditions, irrespective of interpolation schemes and constellations. The proposed method is expected to contribute to the performance improvement of channel-estimation in any OFDM systems operating under slow-fading channels. In addition, since the proposed noise-reduction method intends to improve the interpolation performance with the enhanced accuracy of the reference coefficients, it may be applied in different fields which use interpolation schemes.

REFERENCES


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