

A Study on Adaptive Channel Estimator for improvement of DVB-T receiving performance

*유창성*손 원

경희대학교 전자공학과

daumi75@passmail.to

A Study on Adaptive Channel Estimator for improvement of DVB-T receiving performance

*Changsung You*Won Sohn

Department of radio engineering, Kyung Hee University

Abstract

In this paper an adaptive channel estimator is proposed and investigated which improves the receiving performance for the DVB-T system. A conventional estimator for the system consists of a two-dimensional Wiener filter which is implemented as a cascade of one-dimensional filters, and the filter is operating with the filter coefficients set which is selected from the four different sets according to the channel environment. Our proposed estimator uses the filter coefficients which is interpolated by the two closest coefficients sets. The proposed scheme shows an improvement of 5 to 10dB in SNR compared to the conventional scheme.

I. INTRODUCTION

The Orthogonal Frequency Division Multiplexing(OFDM) technology has been applied to wireless communication systems due to its high data rate transmission capability with high bandwidth efficiency and robustness to multipath delay. The technology has been used in DVB-T[1].

The transfer function $H(f, t)$ of a wireless

mobile, multipath channel varies not only with frequency f but also with time t . The channel characteristic has to be equalized at the receiver which is using a coherent demodulation. To do so, a channel estimator is required, and many researchers have been working on the channel estimation problem[2,3,6].

Normally the estimation is carried out by the two cascaded orthogonal one -

dimensional filters which have fixed filter coefficients. The channel estimation usually has two steps: the first filtering is executed in time direction and the second in frequency direction in order to estimate channels.

In the 1D case, the Wiener filter for the estimation in time direction is designed for the maximum Doppler shift, and the filter in frequency direction is designed for the worst channel delay spread.

Sanzi and Speidel[2] proposed a channel estimation in which the second filter is adaptive. For the filtering in frequency direction, the filter was designed with the assumption of a maximal delay spread of the channel equal to one of the guard interval durations (7us, 14us, 28us, 56us) according to the channel environments[2].

The maximum delay spread is deviated from any of the guard interval durations in real environments. The deviations will not guarantee the exact channel estimation.

In this paper we propose the channel estimation scheme which consists of two step: the channel delay spread is estimated and the filter coefficients according to the estimated channel delay spread is calculated by an interpolation technique using the two closest ones from the four coefficient sets. The results improve the receiving performance up to 10dB.

The paper is organized as follows: Section II presents the DVB-T system and the channel modelling. Section III presents the conventional channel estimation technique and our proposed one. Section IV presents the performance of the techniques

by computer simulations and analyzes its results.

II. SYSTEM MODELLING

The system model for our studies is based on the DVB-T standard[1] and the system block diagram is shown in Fig. 1.

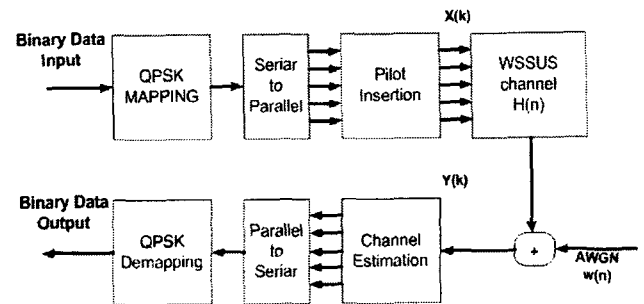


Fig. 1. The system block diagram

We use the wide-sense stationary uncorrelated scattering (WSSUS) channel model[4] for the mobile channel. The frequency response of the channel can be expressed as (1)

$$H(f, t) = \lim_{N \rightarrow \infty} \frac{1}{\sqrt{N}} \sum_{n=1}^N e^{j(\Phi_n + 2\pi f_{D_n} t - 2\pi f \tau_n)} \quad (1)$$

where Φ_n is the phase, f_{D_n} the Doppler frequency and τ_n the delay of the n^{th} path. The Φ_n , f_{D_n} and τ_n are randomly chosen depending on the corresponding joint probability density function $p_{\Phi_n, f_{D_n}, \tau_n}(\Phi_n, f_{D_n}, \tau_n)$ for the considered channel model [2].

We assume that channel characteristic to be approximately unchanged for the duration of the the one OFDM symbol, the guard interval is longer than the delay spread of the channel, and the cyclic prefix avoids inter-carrier interference (ICI) and

ISI.

We can compute the received constellation points as

$$Y_{k,l} = H_{k,l} \cdot X_{k,l} + N_{k,l} \quad (2)$$

where l is the OFDM symbol index, k is the subcarrier index, $X_{k,l}$'s are the transmitted signal constellation points and $N_{k,l}$'s are independent and identically distributed complex Gaussian noise variables. The $H_{k,l}$ are sample values of the channel frequency response,

$$H_{k,l} = H(k\Delta f, lT_s) \quad (3)$$

where $T_s = T_u + T_g$ is the duration of an OFDM symbol plus the guard interval and Δf is the subcarrier distance with $\Delta f = 1/T_u$.

III. CHANNEL ESTIMATION AND ADAPTIVE WIENER FILTER

1. Channel Estimation

The channel frequency response is estimated at pilot positions with

$$\hat{H}_{k_p, l_p} = \frac{Y_{k_p, l_p}}{X_{k_p, l_p}}, \quad (4)$$

whereby $\{k_p, l_p\}$ describes a pilot position and X_{k_p, l_p} is the transmitted pilot signal which is known to the receiver. To allow for coherency response over the whole time/frequency grids, the remaining $H_{k,l}$'s have to be estimated by means of the interpolation based on the known \hat{H}_{k_p, l_p} by,

$$\hat{H}_{k,l} = \sum_{k_p, l_p} u_{k_p, l_p}^{k,l} \cdot \hat{H}_{k_p, l_p}, \quad (5)$$

where $\{k_p, l_p\}$ is a set of locations containing the nearest pilots with respect to the position $\{k, l\}$.

2. Conventional Wiener Filtering

In a conventional Wiener filter, it is assumed that the maximum channel delay spread is equal to a guard interval, and the filter coefficients are set according to the maximum delay spread.

The adaptive Wiener filter proposed by Sanzi and Speidal[2] estimates the nearest guard interval to the actual channel, and the more filter coefficients are chosen among the from different filter sets according to the estimated guard interval.

3. Adaptive Wiener filter Using Linear Interpolation

The previous adaptive Wiener filter employs only four different filter coefficient sets, and the selected filter coefficient set is not optimized according to the near channel environments. When we try to increase the number of the filter coefficient set to solve the problem, the hardware complexity increases. We propose an adaptive Wiener filter where the coefficient set is interpolated based on the four different filter coefficient sets and the estimated delay spread.

In one OFDM symbol duration the channel is estimated every third carrier after filtering in time direction. With the samples of the channel frequency response we calculate the channel impulse response

with an IFFT of order $k(k=1705)$. We obtain the channel impulse response sampled by $T_A = 1/(31705\Delta f)$. We compute the energy E_i for a segment of length $1\mu s$ (22 samples). The number of segments is found by

$$N = \frac{T_g}{1\mu s} \quad (6)$$

The relative energy ΔE_i for each segment is

$$\Delta E_i = \frac{E_i}{E_{\max}} \quad (7)$$

where $i \in \{1, 2, \dots, N\}$. All relative energies which are lower than a certain bound are set to zero. We search for the index of the relative energy, \hat{i} which has a non-zero value and its three following relative energies are all zero. The maximum channel delay spread length is calculated according to

$$\tau_{\max} = \frac{\hat{i}}{N} T_g \quad (8)$$

We calculate the filter coefficient which is interpolated by the two closest coefficients sets. We have the filter coefficient set in case of $\tau_{\max} = 7, 14, 28, 56\mu sec$. We select two filter coefficients sets closest to the estimated delay spread, and calculate the filter coefficient using the linear interpolation as follows :

$$\begin{aligned} C_e(\tau_r) &= (C_{14} - C_7) \frac{\tau_r - 7}{7} + C_7 & 7 \leq \tau_r < 14 \\ &= (C_{28} - C_{14}) \frac{\tau_r - 14}{14} + C_{14} & 14 \leq \tau_r < 28 \\ &= (C_{56} - C_{28}) \frac{\tau_r - 28}{28} + C_{28} & 28 \leq \tau_r < 56 \end{aligned} \quad (9)$$

$C_e(\tau_r)$ is adaptive Wiener filter coefficient

vector according to the estimated delay spread, τ_e is the estimated delay spread.

$C_7, C_{14}, C_{28}, C_{56}$ are filter coefficients for each guard interval.

IV. SIMULATION RESULTS

The OFDM system parameters are specified in Table 1, and we use two channel models (Table. 2) for the simulation. Bad urban and Hilly terrain are modeled with 12 paths channel.

TABLE 1
Simulation Parameter

Parameter	Specification
DVB-T mode	2k mode
FFT size	1705
Pilot arrangement	DVB-T Pilot arrangement
Useful symbol duration	224 μs
Sub-carrier bandwidth	4464Hz

The 97Hz Doppler frequency is used for the mobile speed of 120km/h, and the center frequency of 870MHz .

TABLE 2
CHANNEL MODEL

channel/model	type	τ_{\max}	reference
P1	urban area	0.5 μs	[5]
P2	typical Urban	5 μs	[5]
P3	bad urban area	10 μs	[5]
P4	hilly terrain	20 μs	[5]

We compare the performance of channel estimation on various channel models, and the results are show in Fig. 2. The improvement of the channel P3 is about 10 dB at the BER of 10^{-3} and the improvement

of the channel P4 is 5dB at the BER of 0.02.

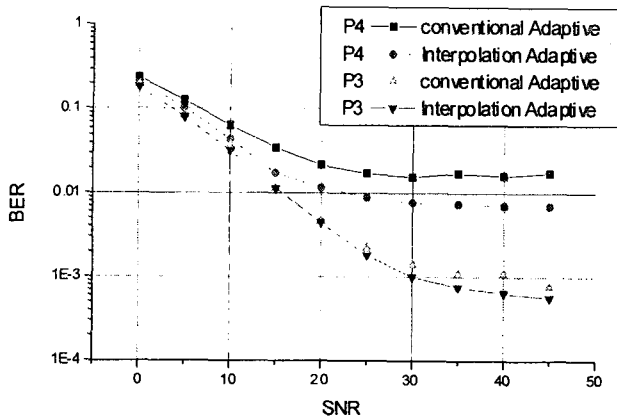


Fig. 2 Comparisons of BERs in channel model

V. CONCLUSION

In this paper we have proposed an adaptive Wiener filter algorithm which outperforms the conventional adaptive Wiener filter algorithm. The proposed algorithm can make upto 10dB improvements for S/N.

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