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# A Study on Channel Estimation Using Two-Dimensional Interpolation Filters for Mobile Digital Terrestrial Television Broadcasting

Yusuke SAKAGUCHI<sup>†a)</sup>, Nonmember, Yuhei NAGAO<sup>†</sup>, Masayuki KUROSAKI<sup>†</sup>, and Hiroshi OCHI<sup>†</sup>, Members

**SUMMARY** This paper presents discussion about channel fluctuation on channel estimation in digital terrestrial television broadcasting. This channel estimation uses a two-dimensional (2D) filter. In our previous work, only a structure of a lattice is considered for generation of nonrectangular 2D filter. We investigate generation of nonrectangular 2D filter with adaptive method, because we should refer to not only a lattice but also channel conditions. From the computer simulations, we show that bit error rate of the proposed filter is improved compared to that of the filter depending on only lattices.

key words: digital terrestrial television broadcasting, channel estimation, scattered pilot, two dimensional filter

## 1. Introduction

Digital terrestrial television broadcasting (DTTB) in Japan began in December 2003. The transmit signals are divided into 13 segments with a bandwidth of 5.57 MHz bandwidth [1]. This enables us to broadcast for various different purposes because we can change the parameters in each segment. Using all of the 13 segments enables high-definition television (HDTV) services to be offered. Moreover, one of the 13 segments is specially designed for one-segment reception, enabling DTTB to mobile terminals by receiving only this segment. However, there have been demands for the reception of all 13 segments by moving terminals because of the development of high-definition car navigation systems in recent years. When DTTB is received by a moving terminal, the performance of the receiver is degraded by multipath fading, the Doppler shift, and so on. Scattered pilots (SPs) are used for channel estimation in DTTB to improve receiver performance. They are inserted nonrectangularly in the time and frequency domains in transmit signals and known a priori by the receiver. Several channel estimation techniques using SPs have been proposed [2], [3]. The general interpolation method in channel estimation is based on a one-dimensional low-pass filter (LPF) [4]-[6]. However, we cannot always interpolate nonrectangular lattices with high performance by these methods designed for rectangular lattices.

To solve this problem, we interpolated nonrectangular

lattices using a two-dimensional (2D) filter that has a nonrectangular spectrum for the SPs in our previous work [7]. However, this 2D filter relates only the positions of SPs. It is not investigated about channel fluctuation. In this paper, we determine cutoff frequency of 2D filter depending on channel conditions. Moreover, we compare the performance of method depending on only lattices and bandlimited filter method depending on channel conditions.

The rest of this paper is organized as follows. Section 2 describes a system model and channel estimation of DTTB. In Sect. 3, a matrix expression of lattice is generalized and a generating process of generalized 2D filters is described. Section 4 describes simulation results of the performance of method depending on only lattices and bandlimited filter method depending on channel conditions. Conclusions are presented in Sect. 5.

#### 2. System Model

## 2.1 Digital Terrestrial Television Broadcasting Systems

The architecture of DTTB is shown in Fig. 1. An information bit sequence is passed to a Reed-Solomon (RS) encoder and a convolutional encoder. At the same time, error correction is applied using multiple interleaving processes: byte, bit, time-domain, and frequency-domain interleaving. After the modulation signal has been interleaved, the SPs are inserted. The process at the receiver is the opposite of that at the transmitter.

SPs are known *a priori* by the receiver. The positions of SPs are shown in Fig. 2. The SPs are inserted once every twelve subcarriers in the frequency domain and once every four OFDM (Orthogonal Frequency Division Multiplexing) symbols in the time domain. Data signals are transmitted at all positions except those of SPs. Consequently, we need high-accuracy interpolation for channel estimation to calculate the channel transfer functions at the data signal positions.

We use the following method for channel estimation in the data part. First, we calculate the received SPs divided by complex standard SPs. Next, these signals are interpolated by some filters. After the signals have been interpolated, we can perform channel equalization with them.

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<sup>&</sup>lt;sup>†</sup>The authors are with the Department of Computer Science and Electronics Engineering, Kyushu Institute of Technology, Iizuka-shi, 820-8502 Japan.

a) E-mail: ysaka@dsp.cse.kyutech.ac.jp

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Fig. 1 Digital terrestrial television broadcasting system model.



## 2.2 Interpolation the Channel Transfer Functions

For estimation of the channel transfer functions in data positions, we calculate the received SPs divided by complex standard SPs. These functions are used in channel equalization block. So, high accuracy interpolation processes improve the performance of receiver.

Conventional interpolation method consists of two 1D filters. First filtering is 1D process in time-domain, and second filtering is 1D process in frequency domain too. However, in the case of nonrectangular lattices, it is generally thought that we cannot perform high-accuracy interpolation with only a 1D process such as rectangular lattices. Therefore, 2D filters that fit the spectrum for nonrectangular lattices are needed to interpolate with high accuracy.

#### 3. Nonrectangular 2D Filters

## 3.1 A Matrix Expression of Lattice

An element on a lattice can be expressed by  $\mathbf{t} = (t_0, t_1)^T$ , where X-Y coordinate of element are  $t_0, t_1$ . In addition, we can express sampling of two-dimentional signals (1) with sampling matrix **V**, where  $\mathbf{n} = (n_0, n_1)^T$  is position of lattice after sampling.

$$\mathbf{t} = \begin{pmatrix} t_0 \\ t_1 \end{pmatrix} = \mathbf{V} \begin{pmatrix} n_0 \\ n_1 \end{pmatrix}$$
(1)





**Fig.4** Interpolation system.

Sampling matrix V can be expressed by (2). This equation is  $2 \times 2$  matrix lined up two vectors, where  $\mathbf{v}_0$  and  $\mathbf{v}_1$  are column vectors of the V.

$$\mathbf{V} = \begin{pmatrix} v_{11} & v_{12} \\ v_{21} & v_{22} \end{pmatrix}, \mathbf{v}_0 = \begin{pmatrix} v_{11} \\ v_{21} \end{pmatrix}, \mathbf{v}_1 = \begin{pmatrix} v_{12} \\ v_{22} \end{pmatrix}$$
(2)

For example, consider sampling matrix with Fig. 3.  $\mathbf{v}_0$  and  $\mathbf{v}_1$  are  $(3, 1)^T$  and  $(0, 4)^T$ , respectively. In this case, sampling matrix is expressed by (3).

$$\mathbf{V} = \begin{pmatrix} v_{11} & v_{12} \\ v_{21} & v_{22} \end{pmatrix} = \begin{pmatrix} 3 & 0 \\ 1 & 4 \end{pmatrix}$$
(3)

Generally, interpolation systems are constructed filtering after the upsampling. The interpolation systems are shown in Figs. 4 and 5.

#### 3.2 Generating Process of 2D Filters

This section shows the process of generating 2D filters that



Fig. 6 Spectrum of nonrectangular 2D filter.

have nonrectangular spectrum from 1D prototype filters [8]. The generated filters have parallelogram spectrum in this section. The process of generating 2D filters consists of three steps.

**Step1** First, we generate a 1D prototype filter p(n). Its spectrum is (4), which does not involve aliasing when we generate a 2D filter. The det(**V**) shows the determinant of the vector **V**.

$$-\frac{\pi}{\det\left(\mathbf{V}\right)},\frac{\pi}{\det\left(\mathbf{V}\right)}$$
(4)

Step2 Next, we generate a 2D filter from the 1D prototype filters, where the 2D filter is  $h^{(s)}(\mathbf{n})$ , which is expressed by

$$h^{(s)}(\mathbf{n}) = p(n_0)p(n_1)^T.$$
 (5)

Step3 Finally, filter  $h^{(s)}(\mathbf{n})$  is down-sampled to make it nonrectangular. This down-sampling is expressed by

$$h(\mathbf{n}) = \det(\hat{\mathbf{V}}) \cdot h^{(s)}(\hat{\mathbf{V}}\mathbf{n}), \tag{6}$$

## where $\hat{V}$ is the cofactor matrix of V.

For example, Fig. 6 shows spectrum of nonrectangular 2D filter using sampling matrix Eq. (3). The filter in Fig. 6 depends on only lattice. We call this 2D filter "original filter."

# 4. Simulation and Discussion about Channel Fluctuation

The bandwidth of 2D filter in Sect. 3 depends on only lattice. However, we can improve the performance by suppressing

Table 1 S	imulation conditions.
Mode	Mode3
Modulation	64QAM
Number of subcarriers	$432 \times 13 + 1 = 5617$
Length of GI	126 µs (1/8)
Channel Model	GSM Typical case for urban area
	(6Taps)
Maximum Doppler frequency fa	a 30 Hz (68.9 km/h), 52 Hz (119.4 km/h)
Shift cycle of fading coefficient	s72 sample (128 shifts in 10FDM)
Error correction (Inner coding)	Convolutional Code (Rate 1/2)
Error correction (Outer coding)	RS Code(204,188)
Channel estimation	Time $\Rightarrow$ Linear / Frequency $\Rightarrow$ LPF
	Two-dimensional filter

Table 2GSM TUx setting (6taps) [9].

Tap number	1	2	3	4	5	6
Relative time ( $\mu$ s)	0.0	0.2	0.5	1.6	2.3	5.0
Average relative power (dB)	-3.0	0.0	-2.0	-6.0	-8.0	-10.0



Fig. 7 Ideal channel estimation value.

noises at unused spectrum. In this section, we evaluate performance of two channel interpolation method. First method depends on only lattice, second method depends on channel conditions and limits bandwidth of 2D filters.

## 4.1 Simulation Conditions

We evaluated the performance of DTTB systems in Fig. 1 by computer simulation. The simulation conditions are summarized in Table 1. The channel is typical urban area model [9] (as shown in Table 2).

For example, channel estimation values are shown in Fig. 7 in the case of ideal channel estimation and maximum doppler frequency Fd = 30 [Hz]. The spectrum of the channel estimation value is shown in Fig. 8. The spectrum is generated from the channel estimation value using the 2D FFT to confirm a cutoff frequency of a 2D interpolation filter.

Figures 9 and 10 show the spectrum in Fig. 8 from a *x*-domain view and *y*-domain view, respectively. The axis in Fig. 9 means delay spreads and the axis in Fig. 8 means subsampled frequency responces. In Fig. 10, we confirm the spectrum like that of Jakes' spectrum. This spread spectrum



Fig. 8 Spectrum of ideal channel estimation value.



is effect of doppler shift. We determine the bandwidth of 2D filter to depend on this spread spectrum.

Normalized cutoff frequency fc is

$$f_{c} = \frac{F_{d} \times s}{F_{s}},\tag{7}$$

where  $F_s$  is sampling frequency, *s* is number of sample in 1 OFDM symbol. Points of FFT are 8192 samples, GI length are 1024 samples (1/8). i.e. s = 8192 + 1024 = 9216.

The bandwidth does not spread in Fig. 9. However, this



Fig. 11 Cutoff frequency (y-domain direction) — BER performance.



Fig. 12 Cutoff frequency (x-domain direction) — BER performance.

component relates the interpolation of diagonal direction. Therefore, doppler shift also affects *x*-domain bandlimited.

## 4.2 Cutoff Frequency-BER Performance

We evaluate BER about some cutoff frequency in the process of generation of 2D filter. In this paper, we construct 2D filter using multistage in Fig. 5. So, we can set up the cutoff frequency of each direction independent. *y* direction is set up by 1st stage filter, *x* direction is set up by second stage. Therefore, we limit cutoff frequency either one of the two direction based on original filter in this subsection. We fix CNR = 25 [dB] in this subsection.

In the case of variable cutoff frequency in each x and y directions, simulation results are shown in Figs. 11 and 12. Both results show that the lowest BER appears on one point and the BER degrades in the rest of the cutoff frequency. In the case of short bandwidth, we cannot de-modulation because of losing signals. When the cutoff frequency corresponds to using bandwidth, BER is the lowest. In the case of more than requires large bandwidth, BER is degraded by the noise of unused spectrum.

We evaluate normalized cutoff frequency  $f_c$  in (7).  $f_c =$ 

Table 3Normalized 2D frequency of using 2D filter.

Maximum Doppler FrEq.	x-domain	<i>y</i> -domain
$F_d = 30 [Hz]$	$5.00 \times 10^{-2}$	$7.50\times10^{-2}$
$F_{d} = 52 [Hz]$	$6.25 \times 10^{-2}$	$8.75 \times 10^{-2}$



**Fig. 13** CNR-BER performance.  $(F_d = 30 [Hz])$ 



0.034 in the case of  $F_d = 30$  [Hz], and  $f_c = 0.059$  in the case of  $F_d = 52$  [Hz]. BER begin to decrease rapidly from up close these  $f_c$  in *y* direction in Fig. 11 and *x* direction in Fig. 12.

## 4.3 CNR-BER Performance

We determine suitable filter bandwidth in each condition, and examine the performance to use generating 2D filter. Determination of bandwidth bases on results in Sect. 4.2. We determine filter bandwidth shown the lowest BER in result described previously, as shown in Table 3. Simulation results are shown in Figs. 13 and 14.

BER performances are improved in both  $F_d = 30 [Hz]$ 

and  $F_d = 52$  [Hz]. When there was no error correction (EC), BER was about  $3.5 \times 10^{-2}$  by original filter and about  $3.0 \times 10^{-2}$  by bandlimited filter with  $F_d = 52$  [Hz], CNR = 40 [dB]. This improvement is very small. On the other hand, BER after Viterbi and RS decoding was about  $1.8 \times 10^{-3}$  by original filter and about  $6.6 \times 10^{-4}$  by bandlimited filter with  $F_d = 52$  [Hz], CNR = 40 [dB]. So, there is a possibility that a small improvement in BER could become a large improvement after error correction in the case of no error correction.

Thus, suitable bandwidth changes by each  $F_d$ . We can generate 2D filter better performance than original filter, which changing filter bandwidth corresponding to speeds of mobile terminals or determine filter bandwidth corresponding to expected maximum speeds of mobile terminals.

## 5. Conclusions

We have proposed a channel estimation method using nonrectangular 2D filters that have low cutoff frequency by channel condition and interpolated SPs for DTTB. The cutoff frequency is determined by maximum doppler frequency. It improves the BER performance because the cutoff frequency is determined by maximum doppler frequency. Our future works are investigating not only doppular shift but also multipath fading, and analyzing highly effective cutoff frequency in theory.

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