Low-Cost Channel Estimation Algorithm for DRM Receiver

Ming Yan[1] and Cheng Yan[2]
[1]GxSOC Research Institute, Communication University of China, Beijing 100024, China
[2]Beijing University of Posts and Telecommunications, Beijing 100876, China
Email: yanming668@gmail.com; yancheng66@aliyun.com

Abstract —In this paper, we consider a digital radio mondiale (DRM) system using orthogonal frequency division multiplexing (OFDM) with coherent detection at the receiver. The DRM pilots are two-dimensional distribution, linear interpolation and Wiener filter interpolation respectively used in the time and frequency domain as the channel estimation algorithm. We propose a low-complexity improved Wiener filter coefficient algorithm to estimate the OFDM channels. Our proposed algorithm differs from traditional algorithms in two ways. Firstly, we divide the DRM pilots into n groups to estimate the channel responses of corresponding OFDM sub-carriers (instead of exploiting the whole pilot at the same time in traditional algorithms). Secondly, we utilize the uniform distribution characteristic of the DRM pilots to reduce the number of Wiener filter coefficients, and this effectively leads to a reduction in computation complexity. In addition, simulation results are also presented to gain further insights.

Index Terms—Channel estimation, digital radio mondiale, wiener filter, OFDM

I. INTRODUCTION

Digital radio mondiale (DRM) is a digital audio broadcasting technology that can deliver sound quality comparable to Frequency Modulation (FM) systems, while working over the traditional Amplitude Modulation (AM) frequency bands, i.e., below 30MHz [1]. DRM utilizes orthogonal frequency division multiplexing (OFDM) transmission technique with coherent demodulation at the receiver. It is well known that channel estimation is critical for coherent detection. Known channel estimation algorithms in the literature can be categorized into two frameworks. The first framework is decision-directed estimation (DDE) [2] and the second framework is pilot-based estimation (PE) [3].

As the DRM transmitted symbol is comprised of pilots, PE is adopted in the DRM system. Due to the two-dimensional (2-D) pilot model, the optimal channel estimator is the 2-D Wiener filter based on the minimum mean squared error (MMSE) criterion [4]. However, the computational complexity is too high for practical implementation. To reduce complexity, the 2-D channel estimation are separated in each dimension, and techniques including linear interpolation, windowed Discrete Fourier Transform (DFT) and Wiener filter interpolation are used for estimating each dimension [5]. Work in [6] gives an OFDM transmission scheme that is suitable for medium frequency and high frequency channels based DRM digital AM broadcasting.

Other previous work includes [7] which provides a pilot design to minimize the coherence of the dictionary matrix used for sparse recovery in OFDM radio system. Reference [8] introduces a comparative performance evaluation of the sampling frequency synchronization method that eliminates the initial sampling frequency offset to reduce the overall synchronization time in DRM receivers, and gives a new DRM synchronization method to satisfy the advanced synchronization performance requirements of DRM receivers.

Although the problem of channel estimation for DRM has been widely explored, there is still possibility for a better estimator which can further reduce the computations and hardware resources. In this paper, we propose a low-complexity improved Wiener filter coefficient algorithm. The proposed algorithm differs from traditional algorithms in two ways. Firstly, we divide the DRM pilots into *n* groups to estimate the channel responses of the corresponding OFDM sub-carriers, whereas the traditional algorithm exploits the whole pilots at the same time. Secondly, since DRM pilots follow a uniform distribution, the number of the Wiener filter coefficients can be greatly reduced. This effectively leads to a reduction in computation complexity.

This paper is organized as follows. The transmission characteristics of DRM is introduced in Section II. In Section III, we describe the interpolation algorithms for channel estimation. In Section IV, we propose the Wiener filter coefficient algorithm for the purpose of channel estimation. Simulation results can be found in Section V and Section VI concludes the paper.

II. TRANSMISSION CHARACTERISTICS OF DRM

A. Transmission Frame Structure of DRM

DRM system includes three different types of data channels, namely, the Main Service Channel (MSC), the Fast Access Channel (FAC) and the Service Description Channel (SDC). The transmitted signal is organized in

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Corresponding author email: yanming668@gmail.com.
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transmission superframes, where each transmission superframe consists of three transmission frames. As shown in Fig. 1, the pilot cells, which are located in the transmission signal, consist of frequency pilots, timing pilots and gain pilots. Channel estimation mainly utilizes the gain pilots.

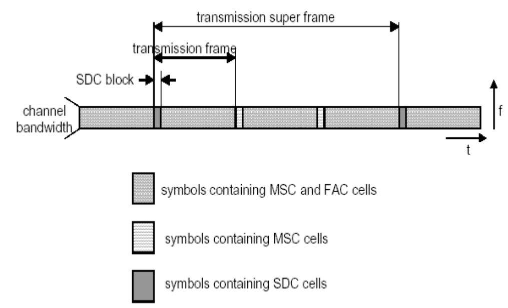

Fig. 1. Time-frequency location of FAC and SDC signals

B. The Channel Models of DRM

We consider a typical multi-path fading channel where the channel can be realized by a wide-sense stationary uncorrelated scattering (WSSUS) model according to the DRM standard as follows,

$$h(t)=\sum_{i=1}^{l}\rho_i c_i(t)\delta(t-\tau_i) \tag{1}$$

where ρ_i is the attenuation of the *i*-th path, τ_i is the delay of the *i*-th path, and $c_i(t)$ can be described by a complex-valued stationary Gaussian random process characterized by its variance and power density spectrum (PDS) as

$$\phi_{h_i}(f)=N_0\frac{1}{\sqrt{2\pi\sigma_d^2(i)}}\exp\{-\frac{(f-f_{sh}(i)^2)}{2\sigma_d^2(i)}\} \tag{2}$$

where $f_{sh}(i)$ is the Doppler shift of the *i*-th path. The Doppler Spread is given by $f_{sp}(i)=2\sigma_d(i)$.

The received symbol can be written as

$$R(k)=T(k)\cdot H(k)+N(k) \tag{3}$$

where *R(k)* is the discrete received symbol in frequency domain, *T(k)* is the transmitted symbol, *H(k)* is the channel response and *N(k)* is the additive white Gaussian noise.

III. Interpolation Algorithms for Channel Estimation

Based on the known pilots, the channel response $\hat{H}(k_p,l_p)$ of the pilots can be calculated by (4) [9]. As channel response of the pilots is known, channel response of the rest sub-carriers can be derived from interpolation algorithm.

$$\hat{H}(k_p,l_p)=\frac{R(k_p,l_p)}{T(k_p,l_p)}=H(k_p,l_p)+\frac{N(k_p,l_p)}{T(k_p,l_p)} \tag{4}$$

where k_p and l_p donate the pilot index in the frequency and time domain, respectively. There are three interpolation algorithms, namely, linear interpolation, wiener filter and windowed DFT. In the next subsections, we will describe the operation of each interpolation algorithm in detail.

A. Linear Interpolation

The simplest implementation is the linear interpolation, which can be implemented in both time and frequency domain. The channel response for *lth* symbol at the $k_p th$ subcarrier can be calculated by (5).

$$\hat{H}(k_p,l)=\hat{H}(k_p,l_p)+\frac{\hat{H}(k_p,l_{p+1})-\hat{H}(k_p,l_p)}{l_{p+1}-l_p}(l-l_p) \tag{5}$$

B. Wiener Filter

Based on mean square error criterion, the 2-D wiener filter interpolation is wildly used for 2-D channel estimation. The Wiener-Hopf equation can be derived by orthogonality principle in [10]-[12].

$$\begin{aligned}&E\{H(k,l)\hat{H}^*(k_p,l_p)\}=\\&\sum_{\{k'_p,l'_p\}\in\Gamma_{k,l}}\omega_{k'_p,l'_p,k,l}E\{H(k'_p,l'_p)\hat{H}^*(k_p,l_p)\},\\&\forall\{k_p,l_p\}\in\Gamma_{k,l}\end{aligned} \tag{6}$$

where k'_p, l'_p donates pilot index in frequency and time domain respectively, $\omega_{k'_p,l'_p,k,l}$ is tap coefficients of wiener filter, $(.)^*$ denotes the complex conjugation, $\Gamma_{k,l}$ is the set of utilized pilots.

As $H(k_p,l_p)$ is zero mean, no correlation with $T(k_p,l_p)$, the cross-correlation function $\theta_{k-k_p,l-l_p}$ can be expressed as

$$\theta_{k-k_p,l-l_p}=E\{H(k,l)\hat{H}^*(k_p,l_p)\} \tag{7}$$

The auto-correlation function $\phi_{k'_p-k_p,l'_p-l_p}$ can be expressed as:

$$\phi_{k'_p-k_p,l'_p-l_p}=E\{H(k'_p,l'_p)\hat{H}^*(k_p,l_p)\} \tag{8}$$

According to (4), the autocorrelation function can be decomposed as

$$\phi_{k'_p-k_p,l'_p-l_p}=\theta_{k'_p-k_p,l'_p-l_p}+\frac{\sigma^2}{E\{|S_{k,l}|\}}\delta_{k'_p-k_p,l'_p-l_p} \tag{9}$$

where k'_p and l'_p donates the pilot index in the frequency and time domain, respectively, $E\{|S_{k,l}|\}$ is the average energy of the transmitted symbols, and σ^2 is the variance of the noise. The tap coefficients of Wiener filter is given by $\omega_{k,l}=\theta_{k,l}\phi^{-1}$, where $\theta_{k,l}$ is the cross-correlation matrix of the sub-carriers and pilots and ϕ^{-1} is auto-correlation inverse matrix of pilots. As large amount of computation is required for the two-dimensional Wiener filter, two cascade one-dimensional Wiener filters can be used to achieve the compromise of performance and complexity. The 1-D Wiener filter interpolation formula at frequency direction is given by

$$\hat{H}_l(k) = \sum_{l\in\Gamma_l} \omega_l(k)\cdot\hat{H}_p \tag{10}$$

$$\omega_l(k) = \theta_l(k)\phi_l^{-1}(k) \tag{11}$$

where $\hat{H}_p$ is the channel response of the pilots at the frequency direction for lth symbol and $w_l(k)$ is the wiener filter coefficients at the kth sub-carrier for lth symbol.

C. Windowed DFT

Windowed DFT is only used for interpolation in the frequency direction. Firstly, the receiver symbols in the frequency domain are transformed to time domain according to IDFT. Then insert 0 to the symbols in the time domain. After transforming back to the frequency domain, the interpolation is completed. In order to bring down signal sidelobes and spectrum spread, it is necessary to be windowed prior to the Inverse Discrete Fourier Transform (IDFT) [13]-[15]. Generally, the Gaussian window is utilized:

$$w(k_e) = e^{-\frac{1}{2}\left(\alpha\frac{k_e - 1/k}{1/k}\right)^2} \quad 0 \le k_e \le K-1 \tag{12}$$

K: the number of gain pilots. Set α to 1.6.

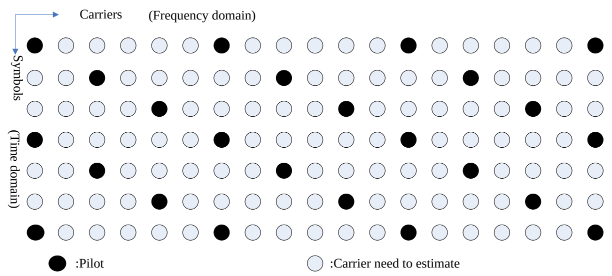

Fig. 2. The transmission frame after the interpolation

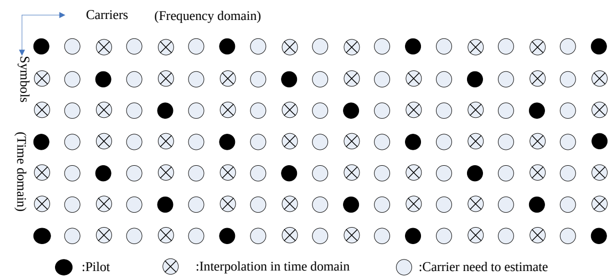

Fig. 3. The transmission frame structure after the interpolation

IV. THE CHANNEL ESTIMATION ALGORITHM FOR DRM

If 2-D channel estimation can be separated for each dimension, the distance of pilots in time and frequency domain are chosen accurately to fulfill the Nyquist sampling theorem [16]:

$$f_{sp}T_s'\cdot D_l < \frac{1}{2} \quad and \quad \tau\Delta f\cdot D_k < \frac{1}{2} \tag{13}$$

where D_l and D_k are the distance of samples in time and frequency direction, respectively. $f_{sp}T_s'$ and $\tau\Delta f$ are the normalized channel bandwidths. As shown in Fig. 2, the pilot distance in time direction equals to 3 samples which allows a maximum Doppler spread of around 6 Hz on the robust B according to the DRM standard. In frequency direction, the spacing is 6 samples, which allows a maximum delay of 1.8ms. In channel 5 the delay spread is defined as 4ms which means that the Nyquist theorem is not fulfilled. With a first interpolation in time domain the distance for interpolation in frequency domain can be reduced to 2 samples and a delay spread of 5.3ms can be tolerated. Fig. 3 shows the transmission frame structure after the interpolation in the time domain.

The channel estimation algorithm for DRM system can be expressed:

Firstly, the channel impulse response of the pilots can be calculated by (4). The linear and wiener filter interpolation are used in the time and frequency domain respectively to complete the whole interpolation process.

Discrete time-frequency cross-correlation function can be divided into the cross-correlation function in the time and frequency domain, respectively.

$$\theta_{k-k_p,l-l_p} = \theta_{k-k_p}\cdot\theta_{l-l_p} \tag{14}$$

The cross-correlation function in time and frequency domain can be divided into two models. One model sets the filter delay τ_{filter} as the channel delay spread τ. The delay power spectrum is:

$$\rho_{filter}(\tau) = \begin{cases} \dfrac{1}{\tau_{filter}} & |\tau| < \dfrac{\tau_{filter}}{2} \\ 0 & \text{others} \end{cases} \tag{15}$$

The cross-correlation function in frequency domain can be expressed as:

$$\theta_{k-k_p} = \frac{\sin(\pi\tau_{filter}(k-k_p)F_s)}{\pi\tau_{filter}(k-k_p)F_s}. \tag{16}$$

where F_s is sampling rate. Set Doppler spectrum in the filter design as the power spectral density:

$$S_{f_{D,filter}}(f_D) = \begin{cases} \dfrac{1}{2f_{D,filter}} & |f_D| < f_{D,filter} \\ 0 & \text{others} \end{cases} \tag{17}$$

So the cross-correlation function in time domain is:

$$\theta_{l-l_p} = \frac{\sin(2\pi f_{D,filter}(l-l_p)T_s')}{2\pi f_{D,filter}(l-l_p)T_s'}. \tag{18}$$

where $T_s' = 1/F_s$.

The other model is that the power spectrum of the delay spread obeys negative exponential distribution and cross-correlation function in frequency domain is:

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$$\theta_{k-k_p} = \frac{1}{1 + j2\pi(k-k_p)\tau_{filter} / T_s'} \tag{19}$$

The classical Doppler spectrum can be expressed as:

$$P(f) = \begin{cases} \frac{1}{\pi f_{\max}} \cdot \frac{1}{\sqrt{1-(f/f_{\max})^2}} & |f| < f_{\max} \\ 0 & \text{others} \end{cases} \tag{20}$$

So the cross-correlation function in time domain is:

$$\theta_{l-l_p} = J_0(2\pi f_{D,filter}(l-l_p)T_s) \tag{21}$$

where T_s is the length of the OFDM symbol.

Before channel estimation, frequency offset and timing delay are compensated, and equation (16) is used as cross-correlation function in frequency domain [17]. The number of the carriers in the frequency domain is 207 when the bandwidth is 10Hz on the Robust B. After the interpolation in the time domain, the number of pilots in frequency domain is 104. To estimate the channel response of the residual carriers, in accordance with (11), the number of Wiener filter coefficient is 207*104. Either the computations or the occupied hardware resources are too large [18].

This paper proposes one improved coefficient algorithm. $i_1, i_2, \cdots, i_{207}$: the sub-carrier indices in frequency domain. $P_1, P_2, \cdots, P_{104}$: the pilot indices in the frequency direction after linear interpolation. Firstly, the pilots are divided into 91 groups:

The 1th group： $P_1, P_2, \cdots, P_{13}$.

The 2th group： $P_2, P_3, \cdots, P_{14}$.

The 3th group： $P_3, P_4, \cdots, P_{15}$.

$\vdots$

The 90th group： $P_{91}, P_{92}, P_{93}, \cdots, P_{103}$.

The 91th group： $P_{92}, P_{93}, P_{94}, \cdots, P_{104}$.

The wiener filter coefficients of $i_1, i_2, i_3, \cdots, i_{14}$ are derived from the pilots of the *1th* group.

The wiener filter coefficients of i_{15}, i_{16} are derived from the pilots of the *2th* group.

The wiener filter coefficients of i_{17}, i_{18} are derived from the pilots of the *3th* group.

$\vdots$

The wiener filter coefficients of i_{190}, i_{191} are derived from the pilots of the *89th* group.

The wiener filter coefficients of $i_{1\,9\,2}, i_{\,1}$ are derived from the pilots of the *90th* group.

The wiener filter coefficients of $i_{194}, i_{195}, \cdots, i_{207}$ are derived from the pilots of the *91th* group.

As the pilots in the frequency domain after the interpolation are obedient to uniform distribution $\phi^1_{k-k_p} = \phi^2_{k-k_p} = \cdots = \phi^{91}_{k-k_p}$, where $\phi^n_{k-k_p}$ is the auto-correlation function matrix of the *nth* group. $\theta^1_{13,14} = \theta^2_{15,16} = \cdots = \theta^{91}_{194,195}$. Where $\theta^n_{x,y}$ is the *xth* and *yth* line of the cross-correlation function matrix of the *nth* group. According to (9), $w^1_{13,14} = w^2_{15,16} = \cdots = w^{91}_{194,195}$. Where $w^n_{x,y}$ is the *xth* and *yth* line of the wiener filter coefficients of the *nth* group. Finally, the number of wiener filter coefficients is 25*13<<207*104. The computations and the hardware resources are greatly reduced.

TABLE I: PILOT NUMBERS FOR EACH MODE IN FREQUENCY DOMAIN AFTER INTERPOLATION

Robust Mode	Spectrum occupancy: 3
A	58
B	104
C	70
D	89

TABLE II: CARRIER NUMBERS FOR EACH MODE

Robust Mode		Spectrum occupancy: 3
A	Kmin	-114
	Kmax	114
B	Kmin	-103
	Kmax	103
C	Kmin	-69
	Kmax	69
D	Kmin	-44
	Kmax	44

TABLE III: THE NUMBERS OF ADDITIONS OF THE CONVENTIONAL AND PROPOSED METHODS

Robust Mode	Conventional method	Proposed method
A	2×58+2×57	2×9+2×8
B	2×104+2×103	2×13+2×12
C	2×70+2×69	2×13+2×12
D	2×89+2×88	2×13+2×12

TABLE IV: THE NUMBERS OF MULTIPLICATIONS OF THE CONVENTIONAL AND PROPOSED METHODS

Robust Mode	Conventional method	Proposed method
A	4×58	4×9
B	4×104	4×13
C	4×70	4×13
D	4×89	4×13

Table I and Table II are pilot numbers after interpolation in the frequency domain and carrier numbers for each mode at spectrum occupancy 3, respectively. From Table III and Table IV, we can see that compared to the conventional method, the additions and multiplications of the proposed method are reduced about 85% in mode A, about 88% in mode B, about 82% in mode C and about 86% in mode D.

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V. Simulation Results

In this paper, the channel bandwidth is 10Hz and the Robust B mode is selected. The useful part of OFDM symbol Tu: 21.33ms, guard interval Tg : 5.33ms, sub-carrier spacing: $46^{7/8}$. The DRM channel model 3 and 5 is chosen. Set a Bit Error Rate (BER) threshold to 10^{3} used to compare the different methods. Linear-wiener linear-linear, linear-DFT, wiener-DFT, and wiener-wiener are on behalf of linear, windowed DFT and wiener filter interpolation in the time and frequency direction, respectively.

As shown in Fig. 4, from the BER curves of linear-wiener and wiener-wiener, we can see that before the BER threshold is reached, there is a minor interval between the two curves. When BER exceeds the threshold, the performance of wiener-wiener is 2 ~ 3dB higher than the linear-wiener.

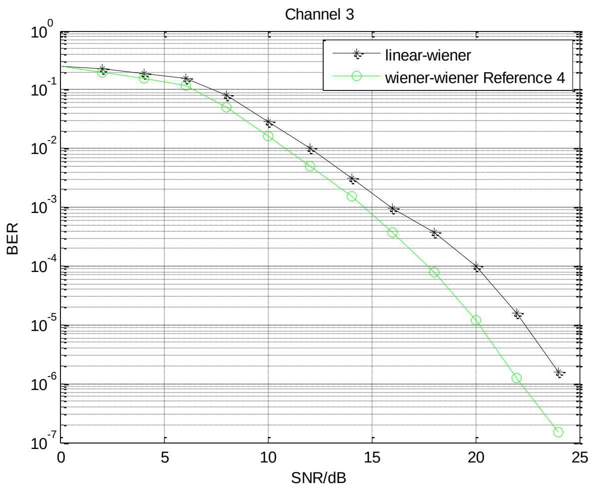

Fig. 4. BER versus SNR for wiener-wiener and linear-wiener method

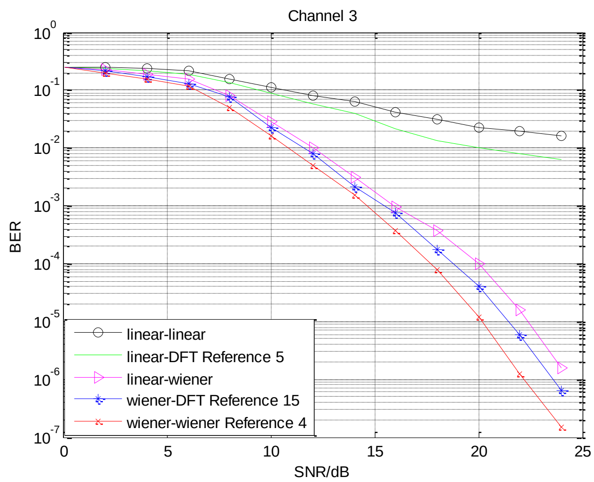

Fig. 5. BER versus SNR for different methods

As shown in Fig. 5, even at a relatively high SNR, the BER of the linear-linear and linear-DFT can't reach the threshold. From the comparison of the BER cures between wiener-wiener and wiener-DFT, we can see that, performance of wiener-wiener is 1 ~ 2dB higher than the wiener-DFT with the same BER. Windowed DFT algorithm can only be used in the frequency domain, and the spectral leakage can't be avoided. Based on the MMSE criterion, the performance of the wiener filter is superior to other algorithms.

As shown in Fig. 6, under condition of the same BER, the performance of the proposed algorithm is about 0.2dB smaller than the traditional algorithm. Simulation results show that the proposed algorithm can fulfill the requirements of the DRM system.

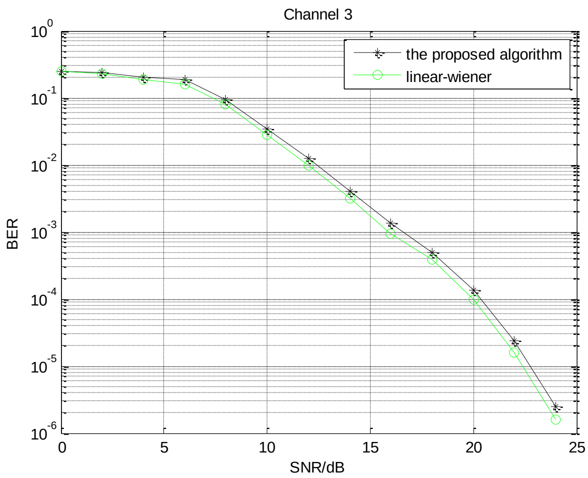

Fig. 6. BER versus SNR for linear-wiener and the proposed method

VI. Conclusions

A larger number of the filter coefficients occupy a lot of memory, and consume a large number of multipliers to implement the filter. The proposed algorithm in this paper is mainly based on two aspects. Firstly, based on the transmission channel characteristics of DRM, the auto-correlation function of pilots and the cross-correlation function between pilots and data positions are only related to the distance between pilots and data. Secondly, the distribution of DRM pilots is uniformly distributed after linear interpolation, so the auto-correlation functions of the pilots in every group are equal.

Theoretically, further reduction of computation complexity can be achieved with the increase of the number of groups. However, the fundamental principles of Wiener Filter are based on MMSE, and the accuracy of channel estimation will reduce if the number of groups increases further. The proposed algorithm is used to reduce the number of the coefficients. The simulation results show that the performance of proposed algorithm can fulfill the requirements of the DRM system.

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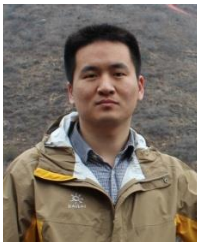
Ming Yan received his B.S. in communication engineering from the Nanjing University of Posts and Telecommunications (NJUPT) in 2002 and his M.S. in signal and information processing from the Communication University of China (CUC) in 2006. He received his Ph.D. in communication and information system from CUC in 2012. In January 2012, Dr. Yan Ming joined the GxSOC Research Institute at CUC as a Research Assistant.



Cheng Yan received his B.S. degree from the School of Electronic Information and Control Engineering at Shandong Polytechnic University Jinan in 2011 and his M.S. in signal and information processing from the Communication University of China (CUC) in 2014. He is currently pursuing a Ph.D. degree at Beijing University of Posts and Telecommunications. His research interests include signal processing, mobile multimedia and wireless communication.