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An Improved Channel Estimation Algorithm for Ultra-Wideband Wireless Communication Systems

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Abstract

Traditional DFT channel estimation algorithm is relatively balanced in terms of complexity and estimation performance, but compared with the four typical channels of UWB, especially in terms of CM1 and CM2 channel with a small number of the diameter, the cyclic prefix length is relatively much larger than the length of their delay, so the back part is still noise. Aimed at this problem, an improved DFT channel estimation algorithm is brought up in this paper, and this algorithm adds the appropriate threshold threshold in the CP to further eliminate noise, and simulation results show that the improved algorithm is better than the original algorithm.

Keywords: OFDM, channel estimation, threshold threshold

1. Introduction

In recent years, wireless personal area networks (WPAN, Wireless Personal Area Network) has become a vital part of communication networks , and also the mainstream technology in 4G wireless communication and control . UWB is one of the main technologies to achieve WPAN, which becomes an ideal choice for its high-speed, low power consumption.

At present, there are two technologies to achieve UWB. One is the early form of narrow pulses based on the baseband, while the other one is now the form based on modulated carrier. Common UWB systems based on carrier modulation have two kinds separately based on the DS - CDMA and MB-OFDM. In this paper, the OFDM-based UWB systems is adopted, because this system uses OFDM technology in the base-band, and proven OFDM technology can also be applied to this system [1]-[3].

Channel estimation is one of the key technologies in UWB communication. So far, there are many channel estimation algorithm. From the point of overall view, there are the auxiliary symbols of non- blind channel estimation, without auxiliary symbols blind channel estimation, and in between semi - blind channel estimation [4].

2. The DFT channel meter algorithm

In OFDM-based UWB systems we have adopted, the 128 - point FFT is used, and the cyclic prefix length is 32. Because the channel multipath delay is generally less than the length of the cyclic prefix, the channel impulse response energy is concentrated in a relatively small number of time domain sampling points. Algorithm based on LS has noise - sensitive shortcomings, so we could transform LS algorithm from frequency domain to the time domain by IFFT, and implement filtering to time-domain channel estimated results in order to reduce the effect of noise and achieve the improvement of the performance of channel estimation. This would be the idea based on DFT transform domain algorithms. The algorithm flow chart is as follows [5][6]:
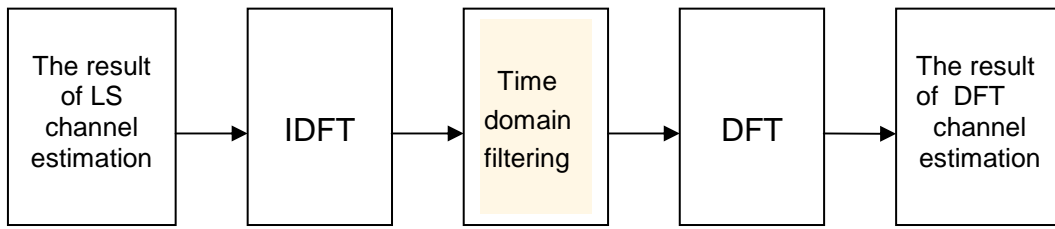
Figure 1. The block diagram of the DFT channel algorithm

Process: First, the estimated channel frequency domain results by the LS algorithm are $\hat{H}_{LS} = [\hat{H}_{LS}(0)\hat{H}_{LS}(1)\cdots\hat{H}_{LS}(N-1)]^T$. By IDFT the time domain results transformed by the frequency domain results are $\hat{h}_{LS} = [\hat{h}_{LS}(0)\hat{h}_{LS}(1)\cdots\hat{h}_{LS}(N-1)]^T$. N is the number of carrier system and N = 128 in the system; filtering algorithm is generally used in the rectangular window, and set L the length of the rectangular window, typically N / 4, and here is 32.

$$h_{timeFilter}(p) = \begin{cases} \hat{h}_{LS}(p) & 0 \le p \le L-1 \\ 0 & L \le p \le N-1 \end{cases}$$

. Then transformed $h_{timeFilter}(p)$ into the frequency domain to get the results of the DFT - based channel estimation that is $\hat{H}_{DFT} = DFT(\hat{H}_{DFT}(0)\hat{H}_{DFT}(1)\cdots\hat{H}_{DFT}(N-1))$.

3. The improved DFT channel estimation algorithm

Through the introduction of the traditional DFT algorithm, we know that the algorithm is to filter the results estimated by the LS algorithm and lower a part of the impact of noise on the estimation results, and thus to get certain improvement in performance . However, it is not very precise to just filter the part exceeded the cyclic prefix length as a noise through a rectangular window. Because in practical UWB systems, some channel length is smaller than the cyclic prefix. In other words, the algorithm only removes the interference noise outside of the noise of the cyclic prefix length in CIR (Channel impulse response and channel impulse response) estimation, and the internal noise takes on inhibition. In view of this situation, under the conditions that the channel meets the integer point sampling channel, this paper presents a improved algorithm based on DFT channel estimation algorithm. [7][8]. The principle block diagram is as follows:
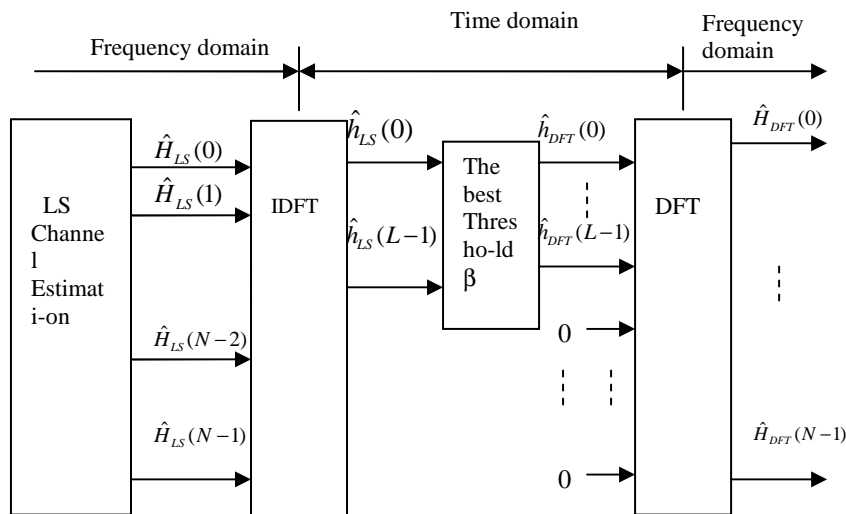

Figure 2. The block diagram of the improved DFT channel estimation algorithm

The algorithm process is as follows:

1.when the cyclic prefix length value is 32, the initial results obtained by LS algorithm transformed into the time domain are: $\hat{h}_{LS} = [\hat{h}_{LS}(0)\hat{h}_{LS}(1)\cdots\hat{h}_{LS}(L-1)]$, where L is not only the cyclic prefix of length 32 , but also the width of the rectangular window .

2.Select the appropriate threshold threshold, and concrete steps are as follows.

1.Define $\partial = \frac{1}{L}\sum_{i=0}^{L-1}\left|\hat{h}_{LS}(i)\right|$, that ∂ is the average value of response amplitude modulus of sampling points in the cyclic prefix. L = 32; make SNR as the signal to noise ratio and β as threshold limit under this SNR. Define $\beta = \lambda\alpha$, according to statistical characteristics of the channel, the range of λ is [0.5, 1.5].

1).Under the SNR, The initial value of λ is set to 0.5, and then he initial value of β the threshold limitation is 0.5∂.

2).Calculate the mean square error, MSE, of the DFT - based improvement algorithm under the SNR, when λ increase every of 0.05,β, the threshold threshold, correspondingly increases.

3).Search the best coefficient λbest according to SNR adaptively in [0.5, 1.5] by 0.05 step in order to find the best threshold threshold βbest. Which means to calculate step by step, every time makes λ increase by 0.5 and calculates the channel estimated results going through the improved algorithm after processing the corresponding threshold β until λ= 1.5 . To identify the value of λ when the MSE difference between the improved algorithm channel estimation and the actual channel results is the minimum, the value obtained is the best coefficient λbest, and the best value of the threshold is βbest=λbest∂.[9][10].

3.Filter the time-domain results processed by the rectangular window filtering of step 1 in LS algorithm using the method of the optimal threshold threshold βbest,.

4.Do the 128 - point DFT transformation to the frequency domain to get the improved estimation results: $\hat{H}_{DFT} = DFT(\hat{h}_{DFT})$

4. Analyses of simulation results

Because our algorithm is mainly aimed at the improvement of CM1 and CM2,concerning of CM3 and CM4, the improvement is finite. So we choose CM2 channel for simulation. The simulation results are shown in Figure 3 and 4.
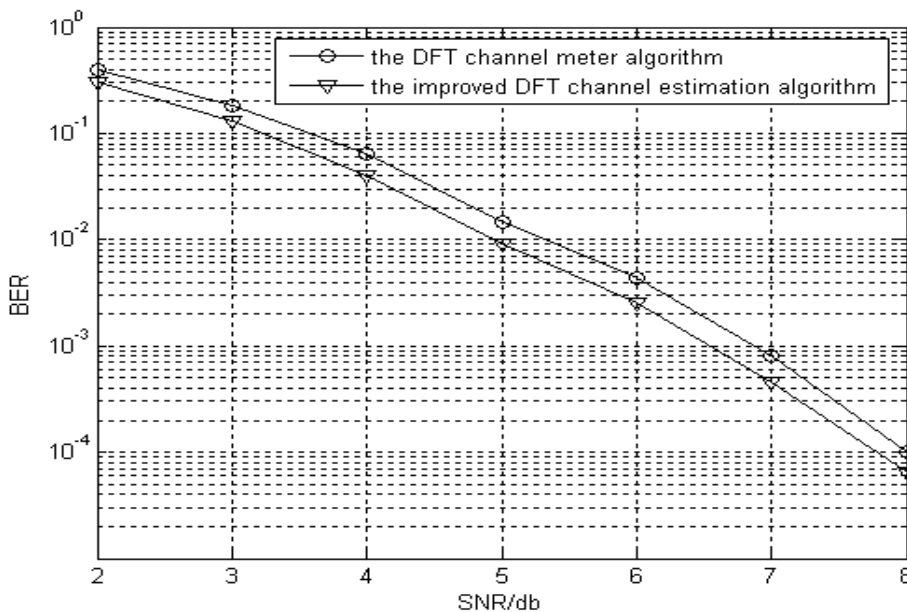

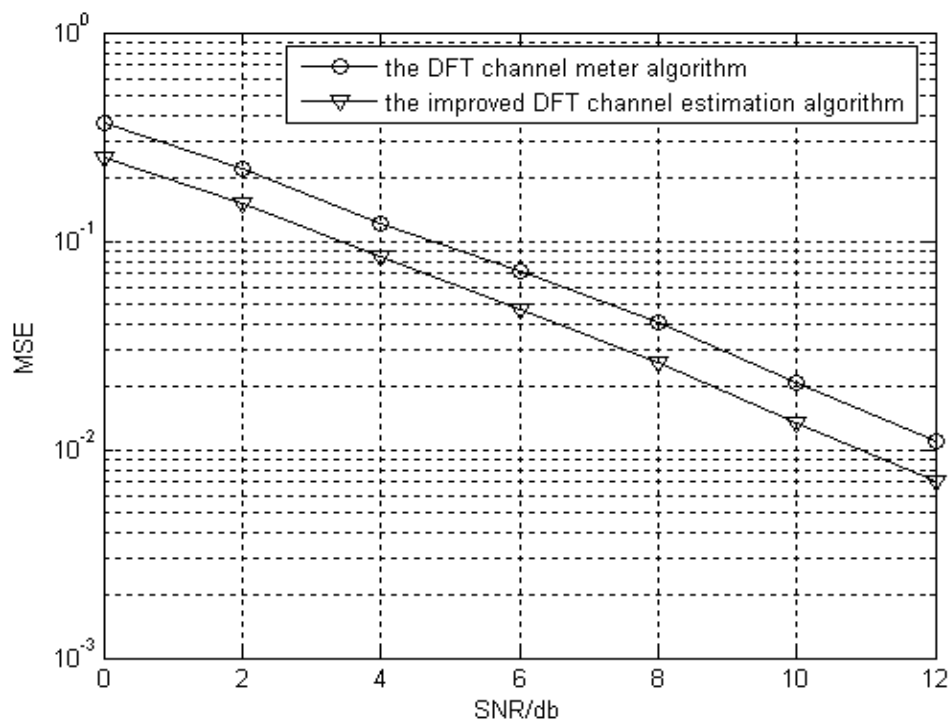Figure 3. The performance of BER under CM2 channel environment

Figure 4. The performance of MSE under CM2 channel environment

From the figure we can see that, compared improved algorithm based on DFT with the conventional DFT-based channel estimation algorithm, performances of the mean square error (MSE) and bit error rate (BER) have improved a lot. It is because improved algorithm makes threshold processing for cyclic prefix length internally, and further removes the noise, thereby reducing the impact of noise on the estimation performance [11].

Simulation results can be seen from the figure displayed, under CM2 channel environment, compared the performance of improved algorithm of MSE with the traditional DFT algorithm, the BER performance has also had some improvement: in the low SNR case, about 0.2dB improvement is made; the MSE performance has achieved the performance gain of about 1.2dB. However, the computational complexity of improved algorithm is larger than the original DFT algorithm in complexity. When calculating the mean value of amplitude mode on sampling points, 31 additions and 1 division are increased and search 20 times for seeking the optimal threshold coefficient adaptively. However, compared to their original DFT algorithm, the amount of increasing computing is not much and there is no excessive increase in the difficulty of hardware implementation and in the mean time achieved a certain degree of performance improvement.

5. Conclusions

The computational complexity of DFT-based channel estimation algorithm is lower than MMSE channel estimation algorithm, and its performance is better than LS channel estimation algorithm, so it has great practical value. The traditional DFT channel estimation algorithm only removes the noise outside the cyclic prefix and doesn't inhibit the internal cyclic prefix noise. Concerning about this part that can be improved of traditional DFT channel estimation algorithm, this paper presents an improved algorithm based on traditional DFT algorithm. By comparison with the conventional DFT algorithm, we can see that the improved algorithm has its own advantages but also has some limitations. The main performances of its advantages are: good performance, and relatively simple. Simulation results given above can explain better performance than the original algorithm and LS algorithm estimates the improved algorithm based on DFT , although slightly worse than the MMSE algorithm performance , but its

complexity is much lower than the MMSE algorithm to implement than MMSE algorithm is simple. The realization of the process only need a reasonable threshold gate limit added on the basis of the conventional DFT algorithm. The inadequacies of the main manifestations are: increases a certain amount of computation, and applies only to satisfy the integer point sampling channel, and its improved space only exists in the diameter of a small number of CM1 and CM2 channel environment. When the channel is non - sampling channel, the energy will not focus on the first L sampling points, but will leak energy, which requires further study and improvement.

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