Study on An Improved Accuracy Algorithm Based on Channel Estimation in 5G Application

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Abstract. 5G has ultra-high spectrum utilization, energy efficiency and resource utilization. Orthogonal frequency division multiplexing modulation is a key technology in order to achieve the characteristics of 5G communication such as ultra-high spectrum utilization. In recent years, a pseudo-random sequence (PN) filled with cyclic prefix time-domain synchronous orthogonal frequency division multiplexing modulation technology has emerged as a modulation technology in high-performance 5G communication systems. And the technology has been maturely applied in the field of terrestrial digital multimedia wireless broadcasting because of its higher channel estimation and synchronization performance. At the same time, 5G has higher requirements in terms of transmission rate and so on, so it has stricter performance requirements for transmission delay control and timing recovery. However, the TDS-OFDM system in 5G application does not allocate pilot subcarriers, which are used for system channel estimation and various parameters. Since the autocorrelation function of the time-domain PN sequence filled in the TDS-OFDM system is not an ideal shock function, when the phase difference with the received sequence is not 0, the autocorrelation value is not 0 as well. When used as a channel estimation, it will cause mutual interference of various multipath components, reduce the accuracy of the estimation, and may also cause some weak multipath Loss of components. This paper proposes an improved channel estimation scheme for TDS-OFDM based systems in 5G application, which can effectively improve the accuracy of channel estimation in the case of high signal-to-noise ratio.

Keywords: Channel estimation, TDS-OFDM, error detection, pseudorandom sequence.

1. Overview of parallel TDS-OFDM systems

In the TDS-OFDM modulation system[1-3], after the analog-to-digital conversion of the received signal, the guard interval is filled by the PN sequence to form the signal, and it is distributed to multiple parallel paths. In addition, the signal and the channel are sampled at different intervals[2,4], and interpolation is performed by the N-point Lagrangian polynomial. After filtering, it outputs the sampling points \( y(k) \) and \( y(k-1) \) of the symbol near the interpolation point and its intermediate sampling point \( y(k-1/2) \), which is used to detect the timing error and is estimated by the MAP criterion Timing error[5]. The structure of the simulation system TDS-OFDM constructed in this paper is shown in Figure 1.
2. System channel estimation and cyclic reconstruction

It can be seen from Figure 2 that the time-domain synchronization header contains three complete PN sequences with overlapping parts, so the multipaths of various lengths can be estimated using these three PN sequences, respectively. Carry out estimation and equalization: (1) Use local known PN sequence and received data time-domain cyclic correlation detection (2) Use iterative interference cancellation method for interference cancellation:

\[ r(n) = \sum_{l=0}^{L-1} x(n - l)h(l) + w(n) \]

Assuming that the receiving end has achieved accurate synchronization, the length of the front and back protection sequences in the filled PN sequence[8] is greater than the front and back delay extension of the channel impulse response. Sliding correlation of the received data with the local PN sequence can be obtained:

\[ R(n) = \frac{1}{M} \sum_{n=0}^{M-1} c^*(i) [r(n + 1)] \]
In the formula, \( c^*(i) \) represents the local PN sequence, which is the conjugate of the PN sequence at the sending end; \( R_{pw}(n) \) represents the cross-correlation value of the PN sequence and noise, that is, the local PN sequence and the received PN sequence are more than one When the radial components are aligned, a correlation peak appears, and the k-th correlation value can be expressed as

\[
R(k) = R_c(0)h(k) + \sum_{l=0}^{l-1} R_c(k-l)h(l) + R_{pw}(k)
\]

In the formula, \( R_c \) represents the autocorrelation function of the PN sequence \( c \). Because the real and imaginary parts of \( c \) take the same value, and the value is a complex sequence with \( c=\pm 1 \).

A tap with a larger channel impulse response amplitude will introduce a larger estimation error to the other taps[9]. A tap with a smaller channel impulse response amplitude will introduce a smaller estimation error to the other taps, so as long as the tap response estimation error with a larger impulse response amplitude is eliminated, the contribution of the channel can improve the accuracy of the channel estimation. That is, first take the channel impulse response taps with a larger amplitude, remove their interference to the channel estimation from the correlation sequence, and then reset the threshold to estimate the amplitude Small tap. The specific method is as follows:

Set a larger threshold value and take the taps of several channel impulse responses with larger amplitudes. Each tap position is represented by \( 0, 1, 2, ..., P-1 \), and their actual response is \( h'={h'(0), h'(1), ..., h'(P-1)} \) means ignoring the influence of noise and other taps, the relationship between them is approximated by the system of equations as

\[
\begin{align*}
H(0) &= h'(0) - \frac{1}{M}[h'(1) + h'(2) + \cdots + h'(P-1)] \\
H(1) &= h'(1) - \frac{1}{M}[h'(0) + h'(2) + \cdots + h'(P-1)] \\
&\vdots \\
H(P-1) &= h'(P-1) - \frac{1}{M}[h'(0) + h'(1) + \cdots + h'(P-2)]
\end{align*}
\]

Solve the system of equations to get a more accurate estimate of the tap at the corresponding position, so that the effect of these taps on other positions can be estimated[10], that is

\[
D = \frac{1}{M} \sum_{i=0}^{P-1} h'(i)
\]

Add D value to all \( R(k) \), and then use lower threshold to get channel response, you can get more accurate channel estimation.

### 3. Interpolation filtering

Suppose the first sampling period is \( T_s \), the channel is sampled at intervals \( T_s \), and the sampling rate satisfies the Nyquist sampling theorem. After the channel samples are converted by analog to digital to generate a weighted analog pulse sequence, the continuous digital interpolation output is
\[ y(n) = \sum x(mT_s)h_I(n - mT_s) \]

\( h_I(n) \) is the interpolation filter impulse response:

\[ h_I(n) = \text{sinc}(\frac{n}{T_s}) \]

Among them, \( \text{sinc} \) is an ideal interpolation filter function, but because its tap coefficient is infinitely long and non-causal, it is necessary to weight and sum an infinite number of samples, which cannot be realized physically. Therefore, in actual interpolation, the filter used is not ideal. In addition, the correct sign can be obtained only by calculating the value of the best sampling time at the receiving end, without the need to completely recover the analog signal from the sampled signal.

Suppose the re-sampling period is \( T_l \), and the output sequence \( y(n) \) is obtained after interpolation. The sampling output at intervals of \( T_l \) is:

\[ y(k) = \sum x(mT_s)h_I(kT_l - mkT_s) \]

In addition, the interval \( T_l \) is not fixed and changes with the output adjustment of the numerically controlled oscillator (NCO). At the same time, \( T_l \) remains synchronized with the local receiver symbol optimal decision interval time \( T \). Among them, \( T \) represents the symbol period.

4. Error detection

The modulation method is QAM[11]: In actual error detection, some points of the timing error result will be wrong, and when the amount of data is large, the average value of the error point may be zero, so there will be no timing error, and the middle point sample jumps The changing situation leads to jitter of the timing clock. In order to eliminate this effect, after determining the best time sampling value, the timing error detection calculation formula is

\[ e(k) = \left[ y_1 \left( k - \frac{1}{2} \right) - a \right] [y_1(k) - y_1(k - 1)] + \left[ y_0 \left( k - \frac{1}{2} \right) - a \right] [y_0(k) - y_0(k - 1)] \]

Among them, \( a = [y(k) - y(k - 1)] \). Compared with the timing recovery waveform, the abscissa is shifted up by a unit, and when it has no timing error, the mean value of the intermediate point calculation sum is zero, which is the same as QPSK modulation.

The timing error \( e(k) \) determines the straight path of the loop filter and the feedback path coefficients \( \alpha \) and \( \beta \) through the feedback loop. The timing loop equation is:

\[ e(k) = A[\tau(k - L) - e(k - L)] * f(k) \]

Among them, \( \tau(k - L) \) and \( e(k) \) are the timing estimate value and its estimation error at time \( k \); \( A \) represents the loop gain coefficient; \( L \) represents the total delay of the loop; \( f(k) \) represents the second-order filter transmission equation. In addition, in order to make the system stable and reliable, it is necessary to ensure that \( BLTu<0.1 \), so when the loop update time is fixed, the maximum loop bandwidth is: \( BL=0.1/Tu \), that is, the loop bandwidth will not change when there is no noise Get bigger[12].
5. Simulation verification and Experiment Results

5.1. Simulation parameters and experimental settings
The performance of the proposed algorithm is simulated and verified. The system parameters are completely in accordance with the PN420 frame structure in the China National Digital TV Terrestrial Broadcasting Standard[14]. The frame body data is 64QAM modulated OFDM symbols[15], and the number of subcarriers is 3780. The performance of the typical channel is simulated and analyzed. The specific simulation system parameters are shown in Table 1:

| The subcarrier number | The number of subcarriers occupied by the data | Time domain synchronization head length | OFDM signal time /ns | Modulation | Transfer efficiency |
|-----------------------|-----------------------------------------------|----------------------------------------|-----------------------|------------|-------------------|
| TDS-OFDM | 3780 | 3780 | 440 | 500 | 64QAM | 90% |
| DVB-T | 2048 | 1512 | 552 | 224 | 64QAM | 67.30% |
| Othercutting-edge system | 8532 | 6048 | 552 | 896 | 64QAM | 84.10% |

5.2. Simulation verification and analysis
It can be seen from Figure 3 that under the typical urban channel, since the longest delay of the multipath is 5us, and the maximum number of multipath delay samples is L=35, it can be seen that the data segment of each symbol is more interfered by the time-domain synchronization header. Big. Therefore, the performance of the data segment without cyclic reconstruction is poor, and there is a bottleneck of error codes. The performance after the cyclic reconstruction algorithm is close to the ideal channel balance. In the figure, it can be seen that the performance between cyclic reconstruction and non-cyclic reconstruction is not much different. This is because under typical urban channels, the longest multipath delay is only 0.5us maximum delay samples, so it is not very good to reflect the superiority of cyclic reconstruction algorithm.

Figure 3. Performance of data cycling reconstruction algorithm in typical urban channel

It can be seen from Fig. 4 that when the bit signal-to-noise ratio is in the range of [0,15], the bit error rate decreases with the increase of the signal-to-noise ratio. By comparison, it is found that the proposed parallel MAP timing estimation algorithm has a lower bit error rate curve and better performance, and the performance difference from the traditional superior MAP algorithm is about 2dB. When the bit error rate is 10-3, the signal-to-noise comparison of this algorithm has a gain of about 4dB and 6dB compared to the Gardner algorithm and the MM algorithm, and when the signal-to-noise ratio is greater than 3dB, the downward trend of the bit error rate is obvious. The reasons are as follows: Before the
MM algorithm determines the timing of the best sampled value, it is necessary to complete carrier recovery and introduce phase noise interference, which affects the timing error detection accuracy; the traditional Gardner algorithm [16] uses interpolation filtering approximation to approximate the function and detect the timing error; in addition, the interpolation order is low, the convergence speed is slow, the error is filtered and evenly distributed to the timing output, resulting in low accuracy; the improved parallel MAP algorithm has the characteristics of the Gardner algorithm, using parallel Lagrangian interpolation, and then Multiple feedback adjustments to accurately interpolate the optimal sampling point position, detect timing error, and finally estimate the timing error by the improved MAP algorithm[17], enhance the correlation between the front and rear frames, resulting in improved timing recovery decision accuracy, which can eliminate timing jitter, Improve accuracy.

Figure 4. Our algorithm and the existing timing synchronization algorithm bit error rate

Figure 5 is the relationship between the modulation error rate of the demodulated output signal and the channel signal-to-noise ratio obtained by simulation of our algorithm model and traditional DVB-T conditions[18]. It can be seen from the simulation results that at low signal-to-noise ratio In this case, the improved algorithm has no significant improvement in MER performance compared to the algorithm without improvement. This is mainly because under low signal-to-noise ratio, the influence of noise on the estimation accuracy is dominant. The improved algorithm The estimation of mutual interference is inaccurate; in addition, in the case of low signal-to-noise ratio, the reliability of the small-amplitude channel impulse response tap itself is poor. In the case of high signal-to-noise ratio, the advantages of the improved algorithm are greater.

Figure 5. MER and SNR curves compared with TVB-T

In the CT8 channel, the system uses PN-related iterative interference cancellation and cyclic reconstruction algorithms to perform better than the traditional DVB-T2K system and DVB-T8K. This is because in our system, the symbol time is longer and the number of subcarriers More, the channel response at the pilot point is more accurate, so the performance is better than DVB-T8K and DVB-T2K,
and our system still has better performance under the highest transmission efficiency and spectrum utilization[19].

Figure 6. A modulation signal analyzer uses different algorithms to receive signal constellation

Figure 6 is a constellation diagram of demodulated signals obtained with and without an improved algorithm in a modulated signal analyzer. The difference in MER can be up to 5dB. When the bit signal-to-noise ratio is in the range [-3,3] As the signal-to-noise ratio increases, the bit error rate decreases. After comparison, it is found that the improved algorithm proposed in this paper has better performance, and the performance difference with the traditional superior MAP algorithm is about 2dB. The reason is that the traditional algorithm needs to complete the carrier recovery before introducing the timing of the best sampling value, and introduce phase noise interference, which affects the timing error detection accuracy. In addition, the interpolation order is low, the convergence speed is slow, and the error is filtered and evenly distributed to the timing output, resulting in low accuracy; the improved parallel MAP algorithm uses parallel Lagrangian interpolation, and then multiple feedback adjustments, accurate within Inserting the best sampling point position can eliminate timing jitter and improve accuracy.

6. Conclusion

This paper presents an improved TDS-OFDM algorithm for multimedia data transmission in 5G communications. This method can achieve the effect of increasing the equivalent signal-to-noise ratio of the output of the demodulator by 5dB and reducing the mean square error of the channel estimation by an order of magnitude under the condition of high signal-to-noise ratio with a slight increase in calculation, which significantly improves the accuracy of the channel estimation Although the improvement of the estimated performance is not obvious under the condition of low and medium signal-to-noise ratio, it is not suitable for general communication receivers. The algorithm does not require the insertion of pilots and cyclic segment prefixes, and can still use simple channels. The balanced mode has the highest transmission efficiency and spectrum utilization rate. Simulation results show that this method has higher estimation accuracy than traditional time-domain system correlation algorithms.

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