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# Analysis of Uplink Channel Estimation Method for LTE System

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**Abstract.** As the evolution of 3GPP Long Term Evolution (LTE) technology as the third generation mobile communication technology, with the increasing demand of mobile Internet users in the world, it has become the focus of mobile communication industry, so academia and industry all of the LTE system key technology research has given high attention. As one of the key technologies of LTE system, channel estimation technology will directly affect the performance of LTE system, so it is very important to study the technology. In this paper, the LTE uplink channel estimation technology in-depth study.

## Introduction

Before the advent of mobile communication, people's information transmission is effected by the time, space, financial, material, meteorological and many other factors. It is because of these factors, which led to the development of society can not be faster. The emergence of mobile communications, so that the first time human beings to achieve cross-regional, cross-regional and even cross-global information transmission, while people's lives and work has undergone a historic change. Although mobile communication has undergone the development from the first generation analog mobile communication system to the second generation cellular digital mobile communication system, and then to the current third generation multimedia high speed digital mobile communication system, but with the development of society, people mobile communication Service requirements are getting higher and higher, the current level of mobile communication technology and service quality is still difficult to meet the growing demands of users, so the world is stepping up efforts to the next generation of mobile communication technology research.

The performance of mobile communication system is mainly affected by the influence and influence of wireless channel. The channel describes all the media that the signal experiences from the sender to the receiver, including the physical medium through which the signal is transmitted from the transmitter to the receiver. In mobile communication, the signal is transmitted using radio waves. As the radio waves in the terrestrial transmission environment transmission, because of different environmental and landform types of energy absorption and penetration, so that radio waves appear direct, reflection, diffraction, refraction and scattering phenomena, but also because of the complex climate And mobile station movement, so that the wireless channel is different from the wired channel, it is not fixed, predictable, but has a strong randomness, which will produce inter-symbol interference, and cause transmission signal amplitude , Frequency and phase distortion, which poses a great challenge to the design of the receiver. In order to restore the signal on the receiver at the receiving end and eliminate the influence of the wireless channel on the signal, people adopt various measures such as source and channel coding, multi-user detection, equalization and diversity. However, the realization of the above technology must know in advance the information of the wireless channel, such as the impact response of the channel, the multi-path delay of the channel and the order of the channel, etc., and to obtain this information must be in the signal reception channel estimation, the research of the proposed algorithm is of great significance to improve the performance of mobile communication system.

## Basic principles of LTE physical layer uplinks

If you want to carry out in-depth research on the key technologies in the LTE uplink, you must understand the transmission mechanism of the LTE system uplink, including the frame structure in the LTE uplink, the resource grid, and the single carrier frequency division (Single-Carrier Frequency Division Multiple Access (SC-FDMA)).

In order to support both paired and unpaired spectrum, the LTE system supports Frequency Division Duplex (FDD) mode and Time Division Duplex (TDD) mode. The corresponding LTE system supports two frame structures.

Frame structure I It is suitable for full-duplex and half-duplex FDD mode. The length of each frame is 307200 10msf sT = 'T = (where sT is the time unit in the LTE physical layer specification, defined as 1 / (15000 2048) ssT ='), by 20 lengths of 15360 0.5msslot sT = T = the time slot. The number of the 20 time slots is 0 ~ 19. A subframe is composed of two adjacent time slots, where the subframes numbered i (i = 0, ×××, 9) contain two time slots numbered 2i and 2i + 1. In FDD mode, up to and down transmission can be used for 10 × 10 frames per 10 ms. The upstream and downstream transmissions are separated in the frequency domain.

Frame structure II, which applies to TDD mode. The length of each frame is the same as frame structure I as 307200 10msf sT = 'T =. Each frame consists of two fields each having a length of 5 ms. Each field consists of five subframes, each of which has a length of 1 ms. There are two types of subframes, one is a regular subframe and the other is a special subframe. The conventional subframe contains two adjacent time slots, where the subframes numbered i (i = 0, ×××, 9) are composed of timeslots numbered 2i and 2i + 1. And the length of each slot is 15360 0.5 msslot sT = 'T =. The special subframe consists of Dw PTS, GP and Up PTS. The length of Dw PTS and Up PTS is configurable, but the total length of Dw PTS, GP and Up PTS is equal to 1ms.

## Research on LTE Uplink Channel Estimation Algorithm

The basic idea of the data-assisted channel estimation method is to rely on the insertion of the pilot sequence known to the sender and the receiver into the data stream to complete the estimation of the channel response. Therefore, how to select the pilot sequence will be channel estimation performance has a very large impact. Different pilot patterns in the same data efficiency, even if the algorithm used will still have the same performance, it is necessary to analyze and compare different pilot patterns to choose a more appropriate pilot pattern while improving effective data utilization while maintaining better channel estimation performance.

The so-called block pilot refers to the pilot sequence in the frequency domain using all the subcarriers to send, and in the time domain is periodically inserted into the data stream to send. Since the retransmission rate of the block pilot is low, it is very suitable for slow fading radio channels, that is, in a symbol block, the channel is considered to remain substantially unchanged. Since the transmission of the pilot sequence in the block pilot utilizes all the subcarriers so that it does not need to be interpolated in the frequency domain at the receiving end, the block pilot is insensitive to frequency selective fading. The so-called comb pilot refers to the pilot sequence in the frequency domain at a certain frequency interval to send periodically, and in the time domain each send data symbols contain pilot symbols. Since comb-like pilots have a high retransmission rate, it is well suited for fast fading wireless channels. In the case of a comb pilot, since the pilot sequence is transmitted using only a partial subcarrier, in order to obtain an estimate of the response to the complete channel, interpolation can only be performed on the basis of the channel response at which the pilot has been obtained, so the comb-like pilot is more sensitive to frequency selective fading.

## Channel Estimation on the Basic Process

Using the pilot to estimate the channel, it is necessary to send the data in the regular insertion of the known pilot signal. In the SC-FDE system, the form of the pilot can be divided into block-like pilot, comb-like pilot and two-dimensional bulk pilot, depending on the direction of the inserted pilot. The block-like pilot is the presence of a pilot, and all subcarriers in a certain SC-FDE symbol are inserted into the pilot, and the subsequent number of SC-FDE symbols is all used to transfer the data. After several symbols, and then transmit the pilot, and then transfer the data, so back and forth, as shown. In this way, the channel state on each subcarrier can be estimated very accurately, but since there is no pilot during the data symbol, the channel information used can only come from the predicted value of the preceding symbol, or by two adjacent the frequency blocks are interpolated. This method is only suitable for the case of burst transmission, in the continuous transmission of data and the channel there is a fast decline in the case, the performance is poor.

From the aspect of improving channel estimation and tracking, the more the pilot is inserted in the frequency domain or time, the better the pilot, the lower the spectrum utilization. According to the two-dimensional Nyquist sampling theorem, the pilot interval inserted in the frequency domain should be less than the coherent bandwidth.

## Adaptive Filtering Channel Estimation Method

In theory, the optimal channel estimation method in the mean square direction is two-dimensional Wiener filtering (2-D Wiener filtering). However, the standard two-dimensional Wiener filtering has considerable complexity, and it is necessary to know the correlation of the channel in the frequency domain with the time domain (or the Doppler power spectrum). In order to reduce the complexity, two-dimensional Wiener filtering can be divided into two one-dimensional Wiener filters, which can be channel-estimated in the time domain and frequency domain, and the performance is not affected by the frequency domain correlation and time domain correlation.

For the time domain channel estimation, since only the power of the larger tap is considered, all simulations in this chapter no longer consider the normal LS frequency domain estimation, but only use the truncated LS estimation of the previous chapter to compare with other time domain estimation methods. For the sake of simplicity, this chapter is referred to as "LS estimation" for "truncated LS estimation". At different signal-to-noise ratios, the MSE performance curves (LS-tdw and LMMSE-tdw) are obtained by filtering the time domain Wiener in the frequency domain LS and LMMSE respectively. The filter length is filterL = 10. It can be seen that MSE performance is improved regardless of whether the frequency domain is estimated using LS or LMMSE, plus time domain Wiener filtering. With the improvement of Doppler, the advantages of Wiener filtering are getting smaller and smaller, because the larger the Doppler frequency shift, the lower the time domain correlation of the channel, the less correlation can be obtained by Wiener filtering, Performance naturally reduced.

When the channel has different Doppler, the MSE performance curve of the time domain Wiener filter has a signal to noise ratio of 10 d B and a filter length of 10. It is clear from this diagram that the greater the performance of the MSE, the worse the performance of the MSE. This is because when the Doppler frequency shift is getting bigger and higher, the correlation between the channel before and after the channel will become low, the use of the time domain correlation can be less, with the time domain filtering method of the benefits naturally reduced The

The random gradient method is derived from the steepest descent method. The Method of Steepest Descent (MSD) is a method of solving the Wiener solution in an iterative way. The steepest descent method first assigns an initial value to the weight vector, calculates the negative gradient direction at that value, and corrects the weight vector in that direction. After obtaining a new weight value, the corresponding weight value is calculated Where the negative gradient direction, so reciprocating, iteration down. Regardless of the initial value, the convergence of the steepest descent method must be a Wiener solution. Since the computational gradient vector still needs to know the autocorrelation matrix of the input signal, it is only theoretical. If the gradient of the gradient vector is used instead of the true gradient, the gradient vector can be avoided and the autocorrelation matrix of the input signal is not needed. It turns out that if the step size of the weight adjustment is within a certain range, this method can eventually be approximated to the Wiener solution. Although the final result with the ideal Wiener solution has some deviation, but the appropriate adjustment step, you can control this deviation within the acceptable range. This is the idea of the famous Least Mean Square (LMS) algorithm. In short, the LMS algorithm is used to replace the gradient with a random gradient, thus avoiding the solution of the exact gradient vector. This feature determines the weight of the adjustment steps can not be too large, or on the one hand may not converge on the other hand the final convergence error is too large. This also determines that the convergence rate of the LMS algorithm is rather slow. LMS is also sensitive to the number of conditions (the ratio of the maximum eigenvalue to the minimum eigenvalue) of the autocorrelation matrix of the input signal. The second method is based on the least squares method, where the cost function is the sum of squares of the weighted errors. The weighting error here is not the error of the weight vector, but the difference between the ideal output of the filter and the actual filter output. Based on the least squares method, the iterative method is recursive Least-Squares (RLS) algorithm. The RLS algorithm is a special case of Kalman filtering, and it is possible to establish a one-to-one correspondence between the variables in the RLS algorithm and the variables in the Kalman filter. The most obvious feature of Kalman filtering is to use the concept of state space to describe the changes in the signal, so it is very flexible to track non-stationary signals, which is LMS algorithm can not be compared.

## Conclusion

Channel estimation is to estimate the frequency response or impulse response of the channel, which is a prerequisite for channel equalization and adaptive resource allocation. The pilot-based channel estimation method is the most commonly used channel estimation method. It is divided into three processes, namely, the selection and insertion of the pilot, the channel of the pilot position, and the channel for the non-pilot location. The pilot form has three kinds of block pilot, comb pilot and bulk pilot, and the density on the time and frequency two-dimensional network should satisfy the two-dimensional Nyquist sampling theorem.

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