Journal of Computers Vol. 28, No. 5, 2017, pp. 303-315
doi:10.3966/199115992017102805026

Adaptive Equalization Algorithm for 60GHz Band in 5G

Ying Ou[1*], Xue-hua Li[1]

[1] Beijing Information Science and Technology University, School of Information & Communication Engineering, Beijing, CHINA
ou_ying1992@163.com, lixuehua@bistu.edu.cn

Received 24 July 2017; Revised 24 August 2017; Accepted 24 September 2017

Abstract. An adaptive equalization algorithm suitable for CM7.1 channel is proposed to address the problem of channel multipath fading in 60 GHz pulse communication system. The steady-state and bit error rate performances of the proposed algorithm are analyzed based on the traditional LMS and RLS algorithms, combined with the CM7.1 IEEE802.15.3c channel model, and compared with the traditional algorithm. Simulation results show that the algorithm reduces the computational complexity and verifies the feasibility of the algorithm in the 60 GHz pulse communication system with the premise of ensuring system reliability. Technical reference is provided for the application of high-capacity fifth-generation, high-frequency hot new empty scene.

Keywords: 60 GHz pulse communication system, adaptive equalization algorithms, LMS algorithm, RLS algorithm

1 Introduction

Mobile internet and the Internet of Things (IoT) have been widely integrated into the lives of people. With its rapid development, the future development of mobile communications proposes higher requirements. At present, research on fifth-generation mobile communication (5G) is popular all over the world. Summarize the future of the Internet and IoT in the main scene and business requirement characteristics the future of the Internet and IoT in the main scene and business requirement characteristics, can extract continuous wide area coverage, hot spot high capacity, low delay and high reliability, low power consumption and large connection are the four major 5G technology scenes [1]. The hot spot high capacity scene mainly for indoor and outdoor local hot region, provide users more than Gigabit experience rate. To this end, 5G will use high frequency spectrum resources to meet the hot zone of high user experience rate and system capacity requirements. The 60 GHz-spectrum wireless communication technology, with rich resources, high data transfer rate, strong anti-interference characteristics, and suitability for small close-range networking, satisfies the higher capacity and speed requirements of 5G.

The physical layer transmission scheme of a 60 GHz communication system is divided into the carrier and pulse communication schemes. The hardware structure of the carrier communication system is complex, the power consumption is high, and it is sensitive to the multipath fading in the channel. It is pointed out that the 60 GHz band can transmit the baseband UWB radio signal, In this way, the 60 GHz pulse signal can be replaced by carrier signal as the carrier of information, which greatly reduces the complexity and cost of the circuit structure of the transceiver, which is the 60 GHz pulse radio. In this paper, the pulse communication technology is applied to the 60 GHz frequency band and the 60 GHz pulse communication system is built, which not only improves the communication capacity and transmission rate of the UWB system, but also reduces the power consumption and cost of the system [2]. A 60 GHz ultra-wide band pulse transmitter is designed and implemented [3]. The pulse period is only 100 ps. Communication is achieved by using the pulse signal in the 60 GHz frequency band. The 60 GHz indoor environment multipath effect is obvious [4]. Signal energy dispersion leads to the significant

* Corresponding Author

reduction of signal-to-noise ratio, which is also caused by the tailing interference of multipath signal at the back of the signal, seriously affecting the performance of the system. The use of an equalizer is one of the effective methods to solve the channel parameter changes caused by distortion [5]. The 60 GHz pulse communication environment uses the Rake receiver and minimum mean square error (MMSE) linear equalization receiver scheme to deal with the multipath channel. However, 60 GHz indoor channel props sometimes degenerates, and an equalizer that can track the channel can be effective against ISI. No literature is presented on the adaptive equalization algorithm in this communication environment. Thus, a 60 GHz pulse communication system with an adaptive equalization receiver scheme is necessary to solve the problem by satisfying the requirements of a 5G communication system with high-speed and highly reliable transmission.

2 Channel Model and System Model

2.1 Signal Representation

For an actual pulse-generating circuit, the most easily generated signal is similar to the Gauss function of the bell signal [6]. In order to get the 60 GHz pulse, the frequency shifting method, UWB pulse signal is multiplied by a fixed frequency sine signal (center frequency for 60 GHz). The realization of the signal generator is very simple, only need to be in the original low frequency signal generator to increase the oscillator can be. The system adopts the method of carrier moving Gauss pulse to the 60 GHz band pulse waveform generation:

$$p(t) = -e^{-\frac{2\pi t^2}{\alpha^2}} \cos(2\pi f_c t), t \in [0, T_p] \quad \textbf{(1)}$$

Where α is the pulse factor, T_p is the pulse duration, f_c is the center frequency, and the 60 GHz communication system generally takes $f_c = 60.5Ghz$. Fig. 1 shows the 60 GHz pulse waveform of the type (1), Fig. 2 is the waveform of the corresponding power spectral density, can be drawn from Fig. 2, m take 0.2ns, 60 GHz Gauss pulse of the main lobe of energy can be very good to meet the FCC spectrum mask requirements, and the spectrum utilization rate is high, so this method is used as the pulse generator of 60 GHz pulse communication system.

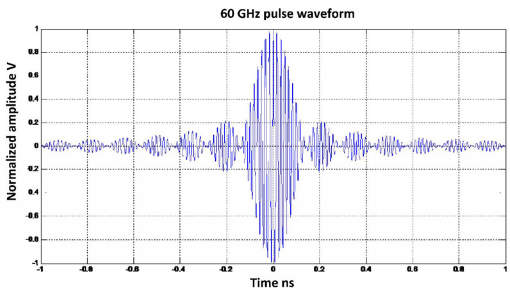

Fig. 1. 60 GHz pulse waveform

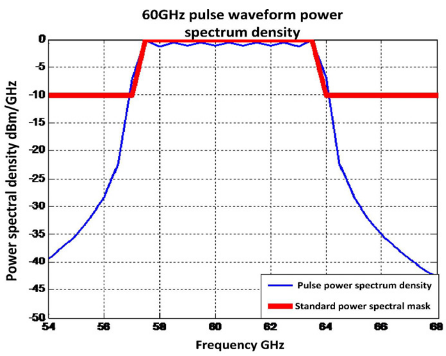

Fig. 2. 60 GHz pulse waveform power spectrum

The PAM signal expression for the transmitter is

$$s(t)=\sum_{j=-\infty}^{+\infty} b_j\, p(t - jT_s) \tag{2}$$

Where T_s is the pulse repetition period and b_j indicates the first j symbol after PAM modulation, DS encoding, and mapping of the binary data sequence.

2.2 Pulse Modulation Mode

60 GHz pulse communication system to achieve reliable data transmission up to tens of gigabits, modulation technology is particularly important. In the choice of modulation scheme, on the one hand, to compare the spectrum utilization of different modulation methods, the system bit error rate, on the other hand, to compare the complexity of different modulation methods. In this paper, based on the 60 GHz pulse signal, the two transmission schemes are analyzed by using pulse position modulation (TH-PPM) and pulse amplitude modulation (DS-PAM).

Time hopping sequence pulse position modulation. TH-PPM divides the time axis into a number of time slices, first, the source sequence is modulated, and then the jump time coding is introduced, a pulse can be generated on a time axis to indicate that a signal is transmitted on a certain segment of a signal, in this way, the position of the pulse can be used to represent the specific information. Fig. 3 shows the schematic diagram of the transmitter:

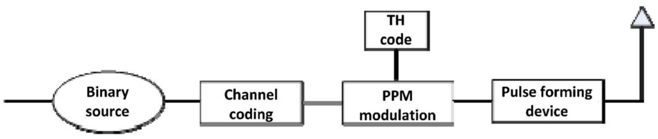

Fig. 3. Schematic of TH-PPM emitter

Function of the chip module generates the binary information sequence, first through the channel coding, pulse position modulation (PPM), at the same time, the use of TH code, the position of the pulse is finally obtained, finally by waveform shaping filter to obtain filter time domain waveform p (t). The transmitting end after a series of modulation, after the formation of the output signal s(t) can be expressed as follows:

$$s(t)=\sum_{j=-\infty}^{+\infty} p\left(t - jT_s - c_jT_c - \delta d_j\right) \tag{3}$$

The upper p(t) is a pulse waveform, T_s is the pulse repetition period, c_jT_c indicates the time offset of the jump, δd_j indicates the time offset caused by PPM, and $jT_s - c_jT_c - \delta d_j$ is the time of the first j pulse. Fig. 4 is obtained after TH-PPM 60 GHz pulse signal waveform, after TH-PPM modulation, the different information bit sequences generated by the source appear in different time slots.

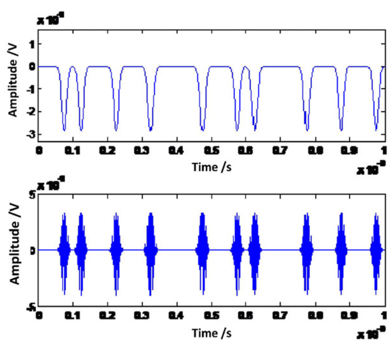

Fig. 4. TH-PPM modulation signal

Direct sequence pulse amplitude modulation. DS-PAM modulation is to modulate the information sequence to the pulse amplitude, and the DS code is introduced to control the polarity of the transmitted pulse, using different amplitude to represent different bits of information, can effectively reduce the interference, achieve multiple access communication, as shown in Fig. 5 for the transmitter schematic diagram:

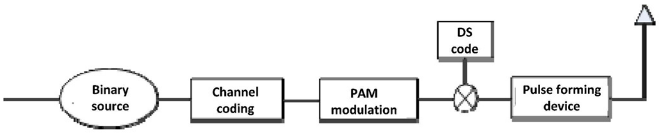

Fig. 5. DS-PAM signal emission schematic

The information sequence generated by the chip firstly carries on the channel coding, then after PAM modulation, and then use the pseudo random code or binary PN code sequence for DS encoding, and finally the modulation signal through impulse response to g(t) pulse forming device. DS-PAM signal can be expressed as:

$$s(t)=\sum_{j=-\infty}^{+\infty} b_j p(t - jT_s) \quad \textbf{(4)}$$

b_j is a modulated binary data sequence, T_s for the pulse repetition cycle. Taking into account the hardware implementation, this paper uses a single polarity pulse PAM modulation, in the transmitter, only need to open and close the diode, than the use of bipolar power consumption is small, relatively easy to achieve hardware. The specific transmission signal as shown in Fig. 6.

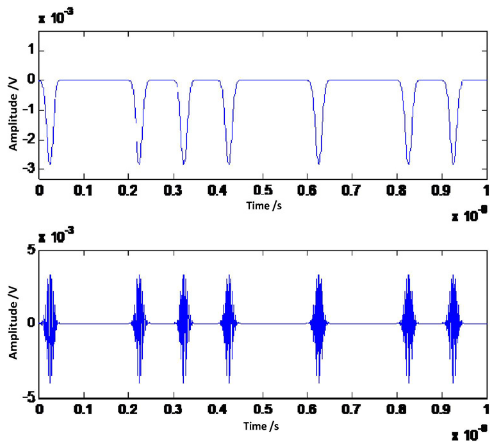

Fig. 6. DS-PAM modulation signal

2.3 Analysis of Multipath Channel Model

60 GHz millimeter wave wireless communication technology is suitable for short-distance communication. According to the working group of the IEEE 802.15.3c establishment of the 60 GHz wireless channel model [7], CM7.1 is employed in the model and in the communication distance of less than 5 m from the indoor communication system. Thus, CM7.1 as the channel model and the channel impulse response in the 60 GHz pulse communication environment and the root mean square delay were analyzed. Path loss is the ratio of received signal power to transmit power, which indicates that the average power is caused by the increase of the transmission distance and the increase of the signal attenuation. The PL of 60 GHz is more serious than the PL of low frequency, because the 60 GHz free space PL and 5 GHz band compared to about an increase of 22dB. In addition, the PL of 60 GHz also added due to the loss caused by oxygen absorption, which poses a challenge to achieve reliable Gigabit wireless transmission, therefore, 60 GHz technology is more suitable for short distance wireless transmission. The PL of UWB and 60 GHz are closely related to the distance and frequency, but there is

no frequency dependent loss model in the IEEE standard, the 60GHz path loss model given in the 3C standard is [7]:

$$PL(d)[dB] = 20\lg\left(\frac{4\pi d_0}{\lambda}\right) + 10nlg\left(\frac{d}{d_0}\right) + X_\sigma[dB] \quad \textbf{(5)}$$

The first two terms represent the average path loss, where λ is the signal wavelength, d_0 is the reference distance, n is the path loss coefficient, depending on the channel environment, n = 1.95 [8]. The third represents shadow fading. Shadow represents the average signal power received by the propagation path change (the new path appears, the old path disappears) within the larger region (the wavelength of a few tens of times). Due to the change of the surrounding environment, the received power at a certain distance will be slightly different from the average received power, which will result in the deviation of the average value of the path loss. The parameter reference value of the PL model of the CM7.1 channel in 60 GHz is shown in Table 1.

Table 1. CM7 model parameters of 60 GHz channel

environment	scene	n	σ_c	σ_r	Antenna configuration
Desktop EnvironmentCM7.1	LOS	1.95	7.27	4.42	Tx-30º, Rx-60º

According to the data provided by the TG3c group [9], the measurement data can be determined. The 60 GHz channel in the time and space domains showed a cluster formation phenomenon, while the direct path component is obvious in the presence of a directional antenna. According to the Saleh-Valenzuela model, the complex base-band expression of the directional channel impulse response is obtained in the angular domain as follows:

$$h(\tau,\phi) = \alpha_{LOS}\delta(\tau,\phi) + \sum_{l=0}^{L}\sum_{k=0}^{K}\alpha_{k,l}\delta(\tau - T_l - \tau_{k,l})\delta(\phi - \theta_l - \omega_{k,l}) \quad \textbf{(6)}$$

where δ is the Dirac impulse function, $\alpha_{LOS}\delta(\tau,\phi)$ is the direct path component ; l is the number of clusters; the number of multipath components that arrived in cluster 1, $\alpha_{k,l}$, $\tau_{k,l}$, and $\omega_{k,l}$ are the multipath components of the complex amplitude, delay, and arrival azimuths, respectively; T_l and θ_l are the units per cluster of delay and average arrival azimuth, respectively. Fig. 7 is a sample of the CM7.1 60 GHz channel impulse response, which has a transmission distance of 3.5 m.

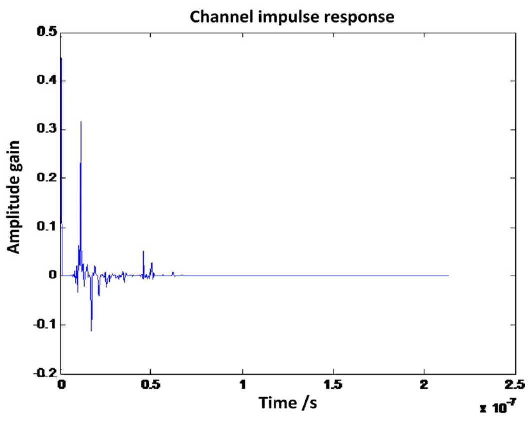

Fig. 7. CM7.1 discrete channel impulse response

Fig. 8 shows the existence of a two-order Gauss pulse output waveform after the channel, In the CM7.1 channel, the average root mean square delay spread is 1.756ns, the maximum value is 7.5ns. And the data rate of the 60 GHz pulse communication system should be more than 1Gbit/s, that is, the maximum value of the symbol period is 1ns, this is much smaller than the mean value of the root mean square delay spread, so the system has a more serious ISI.

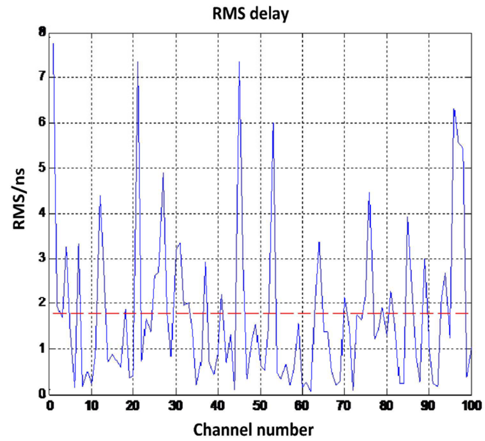

Fig. 8. RMS delay

2.4 System Structures

The equalizer signal received after the multipath channel is transmitted is determined by

$$r(t) = h(t) * s(t) + n(t) \tag{7}$$

The signal through the equalizer, which is the first synthesized signal receiver input, is sampled, r (t) → r (n).

In the equalization algorithm, the weight vector is defined as

$$\boldsymbol{w}(n) = [w_1(n) w_2(n) \cdots w_N(n)] \tag{8}$$

The signal vector is expressed as

$$\text{v}\,\boldsymbol{r}(n) = [r(n) r(n-1) \cdots r(n-N)]^T \tag{9}$$

The signal output is expressed as

$$\boldsymbol{y}(n) = \sum_{i=1}^{N} w_i(n) r(n-i+1) = \boldsymbol{w}^T(n)\boldsymbol{r}(n) \tag{10}$$

The equalizer error is defined as

$$e(n) = \hat{a}_n - y(n) = \hat{a}_n - w^T(n) r(n) \tag{11}$$

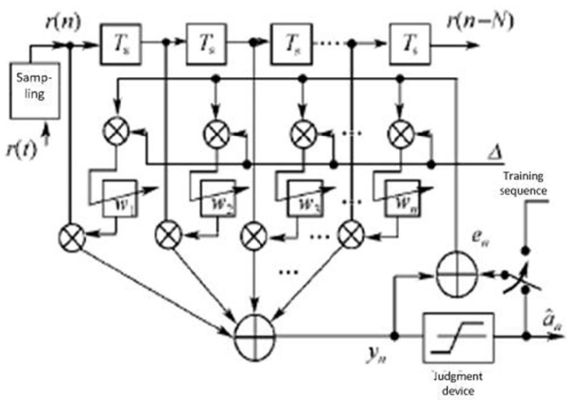

Fig. 9. Basic structure of an adaptive equalizer

3 Adaptive Equalization Algorithm

If the characteristic of the communication transmission channel is known and remains unchanged, so the equalizer only needs to determine a set of the base band signal in the sampling time without the ISI tap coefficient, but in real communication environment, wireless channel is random and time-varying, the equalizer can be updated adaptive equalization coefficient, in order to deal with the unknown channel, and adaptive equalization has the ability to do this [45]. The adaptive equalizer consists of two working modes, namely the training mode and the tracking mode. First, the receiver receives a fixed length and numerical known training sequence (usually a pseudo-random sequence) from the transmitter, the equalizer can be set up according to the training sequence of the appropriate filter parameters to reduce the error rate; Then, send the user data signals to be sent out. In order to compensate the channel distortion caused by multipath, the adaptive equalizer in the receiver uses the recursive algorithm to evaluate the channel characteristics, and adjust the filter parameters to better eliminate the dispersion distortion. Through such a recursive iterative process, After the implementation of the training sequence, the equalizer parameters are close to the optimal value, which can compensate for the distortion. When receiving the user information data, the adaptive equalizer can track the time-varying characteristics of the channel, and ensure good communication quality. Equalizer to track the channel characteristics of the need to change the filter parameters, the common basic algorithm has the least mean square algorithm (LMS), recursive least squares (RLS) and so on.

3.1 LMS Iterative Algorithm

The LMS algorithm is proposed based on the MMSE criterion of the Wiener filter and the steepest descent method. According to the MMSE criterion, the optimal filter tap coefficient vector should minimize the mean square error (MSE) of the performance function. The LMS algorithm has been widely used because it is simple and the variation of the channel statistical characteristics is robust [10].

The MSE performance function is expressed as follows:

$$f(w) = E\{|e(n)|^2\} \tag{12}$$

According to the steepest descent method,

$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) - \mu \nabla_w f(w) \tag{13}$$

$\nabla_w f(w)$ represents the gradient of f(w).

If the $|e(n)|^2$ is taken as the MSE of the estimated value of $E\{|e(n)|^2\}$, then

$$\nabla_w f(w) = \nabla_w E\{|e(n)|^2\} = \nabla_w |e(n)|^2 \tag{14}$$

The following are derived using the formula (equalizer error):

$$\nabla_w |e(n)|^2 = -2e(n)r(n) \tag{15}$$

$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) + 2\mu e(n)x(n) \tag{16}$$

LMS algorithm process is as follows:

(1)Initialization: w (0) =0;

(2)Update:

$n = 1, 2, \ldots;$

$e(n) = \hat{a}_n - y(n)$

$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) + 2\mu e(n)x(n)$

When μ is constant, the above formula is called the basic LMS algorithm; Taking $\mu = \dfrac{\alpha}{\beta + x^H(n)x(n)}$, where $\alpha \in (0,2)$, $\beta \geq 0$, the normalized LMS algorithm is obtained.

The LMS algorithm is the easiest adaptive equalization algorithm, which is not dependent on the model, and therefore demonstrates a robust performance. It is simplified by using the method of instantaneous error approximation instead of the mean of the error. Using this method can reduce computation, especially the calculation of the input signal auto-correlation matrix, which can simplify the iterative formula of the tap coefficients. However, this kind of approximation can cause the system to introduce the noise term of the tap coefficient, which leads to a large amount of steady state offset. Furthermore, convergence LMS algorithm is relatively slow and the adaptability of the non-stationary signal is poor.

3.2 RLS Iterative Algorithm

The essence of the RLS algorithm is a special case of the Kalman filter algorithm. The key to this algorithm is replacing the least mean square criterion with the two-squares minimization of time average according to the time of iterative calculation, and then in accordance with the criteria to determine the filter weight vector w [11], which is based on the criteria for

$$\varepsilon(n) = \sum_{i=1}^{n} \lambda^{n-i} e^2(i) \tag{17}$$

Parameter n is an exponential weighting factor and its value should be selected in the range of $0 < \lambda \leq 1$. The formula is also known as the cumulative square error performance function.

The linear equalizer is expressed as follows:

$$e(n) = \widehat{a}_n - y(n) = \widehat{a}_n - w^T(n) r(n) \tag{18}$$

The $\varepsilon(n)$ of w and its derivative should be equal to zero to find the best weight vector, as shown in the following:

$$\frac{\partial \varepsilon(n)}{\partial w} = 0 \tag{19}$$

Based on the preceding equation, the following three types can be obtained:

$$\sum_{i=1}^{n} \lambda^{n-i} x(i) x^T(i) \boldsymbol{w} = \sum_{i=1}^{n} \lambda^{n-i} d(i) x(i) \tag{20}$$

Order $R_{xx}(n) = \sum_{i=1}^{n} \lambda^{n-i} x(i) x^T(i)$,

$r_{xd}(n) = \sum_{i=1}^{n} \lambda^{n-i} d(i) x(i)$ is available as follows:

$$\boldsymbol{w} = R_{xx}^{-1}(n) r_{xd}(n) \tag{21}$$

which can be obtained by $R_{xx}(n)$ as follows:

$$R_{xx}(n) = \lambda R_{xx}(n-1) + x(n) x^T(n) \tag{22}$$

We can use the matrix reciprocal lemma to obtain the recursive formula of $R_{xx}^{-1}(n)$.

$$R_{xx}^{-1}(n) = C(n) = \frac{1}{\lambda}[R_{xx}^{-1}(n-1) - \frac{R_{xx}^{-1}(n-1) x(n) x^T(n) R_{xx}^{-1}(n-1)}{\lambda + \mu(n)}] \tag{23}$$

where $\mu(n) = x^T(n) C(n-1) x(n)$.

After deduction, the following can be obtained:

$$w(n) = w(n-1) + k(n)e(n|n-1) \tag{24}$$

where k(n) is the gain vector: $k(n) = \dfrac{C(n-1)x(n)}{\lambda + \mu(n)}$.

$$\text{Error: } e(n|n-1) = d(n) - w^T(n-1)x(n) \tag{25}$$

The best value of w(n) at n time can be deduced from the best value of w(n-1) at n-1 time and the correction value.

The correction value is $k(n)[d(n) - w^T(n-1)x(n)]$.

The most significant difference between the RLS and LMS algorithms based on their recurrence formula is the calculation of the gain coefficient. The LMS algorithm uses the input vector multiplied by a constant as the gain coefficient; thus, the calculation is simple, whereas the RLS algorithm uses the complex gain k(n), which causes every moment of the RLS algorithm weights w(n) to adjust to the incoming data with a different step size factor adjustment, Instead of uniform μ pass adjustment, thereby resulting in high computational complexity.

RLS algorithm process is as follows:

(1) Initialization: $\boldsymbol{w}(0) = k(0) = x(0) = 0$, $C(0) = \delta^{-1}I$, in which I represents the unit matrix, and δ takes a small number of normal.

(2) Updating gain factor:

$$\mu(n) = x^T(n)C(n-1)x(n)$$

$$k(n) = \frac{C(n-1)x(n)}{\lambda + \mu(n)}$$

(3) Updating filter parameters:

$$w(n) = w(n-1) + k(n)[d(n) - w^T(n-1)x(n)]$$

(4) Updating the inverse matrix:

$$C(n) = \frac{1}{\lambda}[C(n-1) - k(n)x^T(n)C(n-1)]$$

RLS adaptive algorithm for each iteration of the calculation of the amount of $2.5M^2 + 4.5M$. Compared with the LMS algorithm, the computation of the RLS algorithm is much larger, but the convergence of RLS algorithm is much better than the LMS algorithm, which was widely used.

4 Adaptive Equalization Algorithm for 60 GHz Pulse Communication System

The communication range of the 60 GHz pulse communication system in an indoor communication environment is mainly in tens of meters. Users will not move quickly and the communication environment is stable. However, higher communication capacity on the BER performance is required to achieve high speed. Therefore, the 60 GHz pulse communication system does not require the equalizer to have a super tracking capability, but for the equalizer to be stable after the error value to achieve a lower range.

Therefore, this section is based on the characteristics of the 60 GHz pulse communication multipath channel. This paper designs an adaptive equalization algorithm for the 60 GHz pulse communication environment. The algorithm based on the LMS and RLS adaptive equalization algorithms is analyzed and summarized by analyzing the advantages of both.

In the transmission signal passing through the multipath channel transmission, the received signal for the equalizer is

$$r(t) = h(t) * s(t) + n(t) \tag{26}$$

First, the receiver inputs a synthetic signal sampling, r (t) - r (n). The weight vector and vector signal are defined as

$$\boldsymbol{w}(n)=[w_1(n)w_2(n)\cdots w_N(n)] \tag{27}$$

$$\boldsymbol{r}(n)=[r(n)r(n-1)\cdots r(n-N)]^T \tag{28}$$

The signal output is

$$\boldsymbol{y}(n)=\sum_{i=1}^{N}w_i(n)r(n-i+1)=\boldsymbol{w}^T(n)\boldsymbol{r}(n) \tag{29}$$

The error of the equalizer is

$$e(n)=d(n)-y(n)=d(n)-w^T(n)r(n) \tag{30}$$

First, the weight updating formula is set as follows:

$$\boldsymbol{w}(n+1)=\boldsymbol{w}(n)+\varDelta e(n) \tag{31}$$

The $\varDelta=2\mu x(n)$, order $\mu=C^{\frac{1}{1+an^b}}\mu_0$, C, a, and b, determine the range of attenuation and attenuation speed. Compared with the traditional LMS algorithm, the variable step size LMS algorithm can effectively improve the convergence speed of the equalizer in the premise of ensuring the system stability.

When the error value tends to be stable, the simplified RLS algorithm is adopted to reduce the error. The order is

$$\varDelta=\frac{C(n-1)x(n)}{\lambda+\mu(n)} \tag{32}$$

where $C(n)=\frac{1}{\lambda}[R_{xx}^{-1}(n\text{-}1)\text{-}\frac{R_{xx}^{-1}(n\text{-}1)x(n)x^T(n)R_{xx}^{-1}(n\text{-}1)}{\lambda+\mu(n)}]$. The equalizer passes the variable step LMS algorithm to limit the error in the lower range. Therefore, ignored parameters do not need be extremely low, so that $\lambda=0.95$ can make the system more stable. Using the adaptive algorithm can improve the problem of the steady-state error of the LMS algorithm, and compared to the traditional RLS algorithm, the complexity of the algorithm is much smaller and more suitable for the 60 GHz pulse communication system.

The improved adaptive equalization algorithm process is as follows:

(1) Initialization: $\boldsymbol{w}(0)=k(0)=x(0)=0$, $C(0)=\delta^{-1}I$, where I represents the unit matrix and δ is a small normal number.

(2) Updating of gain coefficient as follows:

$n=1,2,\ldots;$

$e(n)=d(n)-y(n)$,

$\boldsymbol{w}(n+1)=\boldsymbol{w}(n)+2\mu e(n)x(n)$,

$\mu=\frac{\alpha}{\beta+x^H(n)x(n)}$, where $\alpha\in(0,2)$, $\beta\geq 0$.

(3) Judgment convergence state, which is expressed as follows:

$\theta(n)=e(n)-e(n-\lambda)>\varDelta$, where λ is the contrast range and $\varDelta$ is the steady state threshold.

(4) Updating of gain function using

$k(n)=\frac{C(n-1)x(n)}{\lambda+\mu(n)}$.

(5) Updating of inverse matrix using

$C(n)=\frac{1}{\lambda}[C(n\text{-}1)\text{-}k(n)x^T(n)C(n\text{-}1)]$.

5 Results of Simulation Analysis

5.1 Performance Comparison between LMS and RLS Algorithms

Through the preceding description of the algorithm, we know that the LMS algorithm is simple and requires a small amount of computation, but has slow convergence and stability after large errors are encountered. The RLS algorithm converges quickly after a low and steady error is achieved, but the calculation is complex. The simulations using the two kinds of adaptive algorithms show that the RLS algorithm has good tracking performance, but its computation is complex and the noise caused by the recursive calculation for the accumulation of the round-off is highly sensitive.

Fig. 10 shows that the iterative error curve is obtained for the adaptive equalizer in 60 GHz pulse communication system through two kinds of adaptive algorithm simulation. The RLS algorithm only requires an iterative error of about 100 times to reach the steady state error. The LMS algorithm requires 700 to 800 iterations. In addition, the RLS algorithm reaches the steady-state error magnitude after convergence at 11, whereas the LMS algorithm achieves the same in only 10. Therefore, the steady state error of the RLS algorithm is much better than that of the LMS algorithm. However, through the introduction of the two algorithms, we know that the LMS algorithm, which is only $2M+1$ for each iteration of the computation, is much simpler than that of the complex computation of the RLS algorithm, which is $2.5M^2+4.5M$.

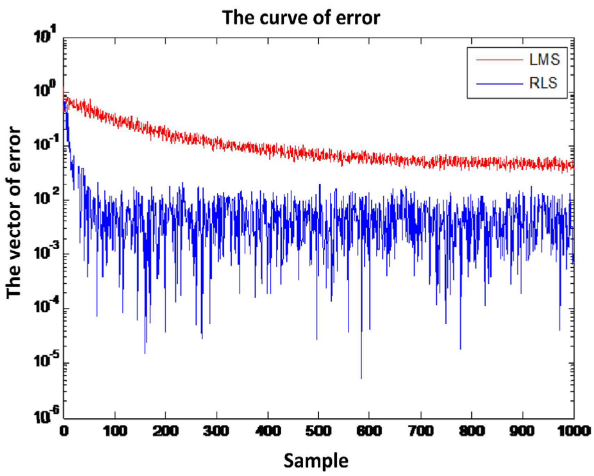

Fig. 10. Performance comparison of LMS and RLS algorithms

5.2 Traditional LMS Adaptive Algorithm and Improved Adaptive Algorithm Performance Comparison

The iterative error curves of the improved adaptive algorithm and the LMS adaptive algorithm are shown in Fig. 11. The improved adaptive algorithm can be seen in the first 400 iterations using the variable step size LMS algorithm, and the convergence of the RLS algorithm continues iteratively until a steady-state error is achieved. Based on this kind of adaptive algorithm, the steady-state error of the system can be 10. The steady state error of the improved adaptive algorithm is lower than that of the LMS algorithm. Meanwhile, the low complexity of the algorithm requires the complex operations to be performed very close to the steady-state error under the condition that is more applicable to the 60 GHz pulse communication system.

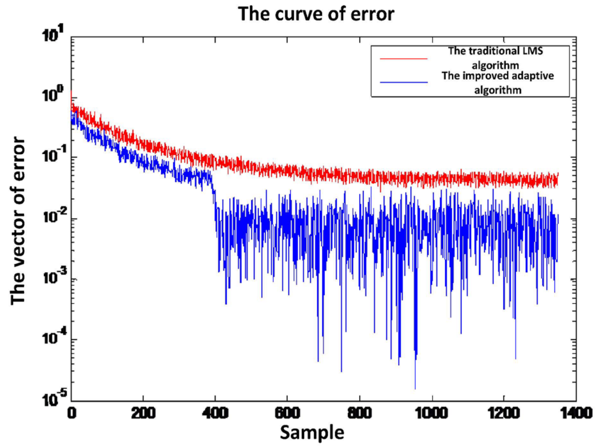

Fig. 11. Pulse communication system adapted to 60 GHz adaptive equalization algorithm and Error traditional LMS algorithm comparison

5.3 Improved Adaptive Equalization Algorithm Performance Simulation

Fig. 12 shows the three kinds of adaptive algorithms in the 60 GHz pulse communication system BER performance after a stable condition is achieved. The improved adaptive algorithm and RLS algorithm can both improve the system performance effectively, thereby eliminating the system ISI. The performance of the LMS algorithm is poor; thus, the system improvement is minimal.

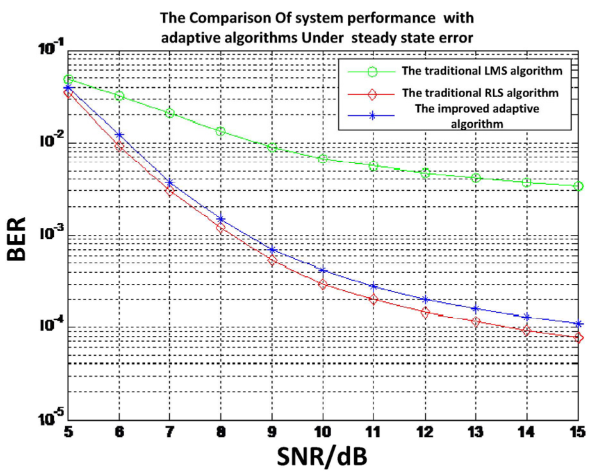

Fig. 12. Three kinds of adaptive algorithm error performance after iterative stabilization

6 Conclusion

An adaptive equalization algorithm for 60 GHz pulse communication system is proposed to achieve the channel characteristics of this system based on research on the receiving structure of the 60 GHz pulse communication system and the adaptive algorithm of the equalizer. First, the 60 GHz pulse communication system channel models and systems are introduced. Then, two traditional adaptive equalization algorithms, LMS and RLS, are introduced. The respective characteristics of these two algorithms are analyzed. Then, an improved adaptive algorithm is designed to enhance the channel characteristics of the 60 GHz pulse communication system. Finally, the performance of the LMS and RLS algorithms are simulated and compared. The convergence speed and steady-state error of the traditional and improved adaptive algorithms are also compared. The two aspects of system performance and implementation complexity, which are applicable to the 60 GHz pulse adaptive receiver scheme of the communication system, are studied. This algorithm needs to switch the updating function, which will increase the difficulty of hardware realization. Therefore, we can explore some methods to improve the algorithm, the design of the optimal performance and easy to achieve the adaptive equalization algorithm.

References

[1] IMT-2020(5G) Promotion Group, White paper on 5G wireless technology architecture, 2015-05.

[2] C.J. Hansen, WiGiG: multi-gigabit wireless communications in the 60 GHz band, IEEE Wireless Communications 18(6)(2011) 6-7.

[3] H. Mopidevi, H.V. Hunerli, E. Cagatay, N. Biyikli, M. Imbert, J. Romeu, L. Jofre, B.A. Cetiner, Three-dimensional micro-fabricated broadband patch antenna for WiGig applications, IEEE Antennas & Wireless Propagation Letters 13(1)(2014) 828-831.

[4] N. Li, Research on the performance of the 60GHz pulse wireless communication system, [thesis] Qingdao: Ocean University of China, 2012.

[5] X. Zhang, X. Tao, P. Zhang, A suitable in ultra-wideband system of low complexity rake receiver, Journal of Beijing University of Posts and telecommunications 28(3)(2005) 93-95.

[6] IEEE Computer Society, Wireless Medium Access Control (MAC) and Physical Layer (PHY) specifications for high-rate wireless Personal Area Networks (WPANs), IEEE. Standard 802.15.3c, 2009.

[7] P. Smulders, Exploiting the 60 GHz band for local wireless multimedia access: prospects and future directions, IEEE Communication Magazine 40(1)(2002) 140-147.

[8] W. Fu, Theoretical study on the characteristics and related key technologies, [dissertation] Chengdu: University of Electronic Science and Technology, 2009.

[9] A. Maltsev, V. Erceg, E. Perahia, Channel models for 60 GHz WLAN systems, IEEE, 2010, 802: 802.11-09.

[10] B. Widrow, S.D. Stearns, Adaptive Signal Processing, Prentice-Hall, Englewood Cliffs, NJ, 1985.

[11] Z.Z. Wang, Research on adaptive equalization algorithm, [dissertation] Liaoning: Dalian Maritime University, 2007.