Turbo Equalization Technique for Data Link Communication Systems

Ao-Shuang Yang, Shi-Ming Li, Yi-Jie Wu, Yue-Qi Wang, and Ju-Hao Tan

Abstract The development of data link has greatly changed the operation mode of modern war. The complexity of wireless communication channel makes the high rate of data transmission become one of the important research topics in data link communication technology. In order to enhance the reliability of the system, channel equalization technology can be used to eliminate ISI (Inter Symbol Interference). Equalization is similar to "whitening filter", which is to "smooth" the frequency response of the channel. Different from the traditional equalization method, the equalization and decoding is combined in the Turbo equalization technology. By recycling the soft information, the bit error rate can be reduced. The Turbo equalization technology can make full use of known information. The soft information can be recycled in each iteration. The simulation results show that the performance of data link communication system can be improved by the Turbo equalization technique. This paper mainly studies the Turbo equalization technology for data link communication systems.

1 Introduction

Due to the influence of multipath transmission and fading, inter symbol crosstalk ISI often occurs in the channel. Channel equalization technology is a method to deal with this characteristic of the channel.

In the mid-1960s, adaptive filtering technology was introduced into equalization technology, and zero forcing algorithm was proposed under the peak distortion criterion [1]. In the early 1980s, Ungerboeck proposed the minimum mean square error algorithm [2], the algorithm has the advantages of small computation and good stability, but the disadvantage is slow convergence. In recent years, turbo code technology has gradually matured. Inspired by the idea of Turbo code, Douillard et al. Proposed the idea of Turbo Iterative Equalization [3, 4]. Bauch et al. proved the good effect of Turbo Equalization Technology Based on map algorithm [5],

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but its huge amount of computation is the obstacle that it cannot be well applied in practice. T. Oberg et al. Proposed using linear equalizer instead of decision feedback equalizer to avoid the error transmission problem of Turbo equalizer [6].

The work is arranged as follows: Sect. 2 gives the principle of Turbo equalization. The algorithm of Turbo equalizer is introduced in Sect. 3. In Sect. 4, the system model is presented, and according to the results of simulation analysis are carried out.

2 Turbo Equalization Theory

2.1 Transmitter Principle

Turbo equalization technology is based on the idea of Turbo code encoding and decoding. Therefore, the transmitter principle of Turbo equalization is similar to the Turbo code encoding part, which is mainly composed of encoder, interleaver and modulator modules (Fig. 1).

Firstly, the data a_k with length K enters the encoder for encoding, and the encoder outputs the data b_k with length N. Then, it enters the interleaver. The main purpose of interleaving is to reduce burst errors. The main principle is to rearrange the information data and change the order. The output result of the interleaver c_k is converted into a sequence x_k for transmission after appropriate modulation.

2.2 Receiver Principle

The receiving part of Turbo equalization is similar to the decoding part of Turbo code and is not done all at once. The receive port is mainly composed of equalizer and decoder. It is connected by interleaver, deinterleaver and modem to transfer soft information to each other.Also the receiving part form a multi-iteration process to complete the final equalization and decode.

As shown in Fig. 2, in the first equalization process, there is no prior information. So the equalizer only calculates the value Y_k output through the channel to obtain the posterior information of the transmitted symbol $L_E^{pos}\left(\hat{x}_k\right)$, and the posterior information gets the soft information $L_D^{pri}\left(\hat{x}_k\right)$ after the deinterleaver module. Then the output information of the decoder is deducted from the input information of the decoder, and the prior information needed in the iterative process $L_E^{pri}\left(\hat{x}_k\right)$ is obtained after the interleaver. Then the prior information and the output value of the channel enter the equalizer together, and the first iteration process begins. After several iterations, no new external information is generated, and the iteration process can be stopped. Finally, the output of the decoder $L_D^{pos}\left(\hat{x}_k\right)$ should be decided, as shown in the formula:

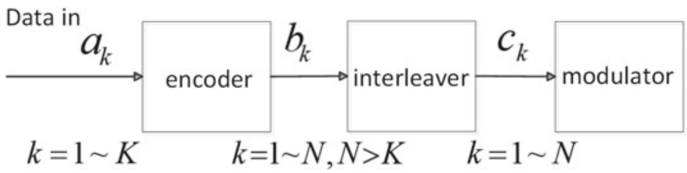

Fig. 1 Transmitter structure

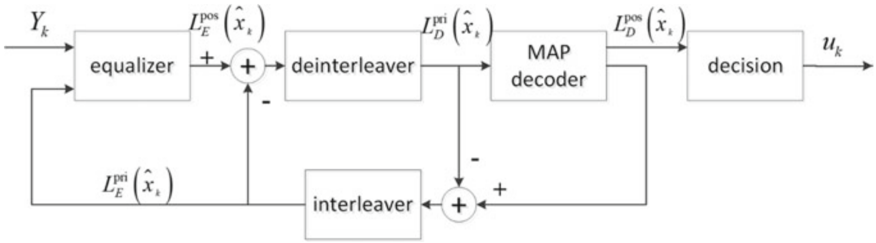

Fig. 2 Receiver structure

$$u_k = \begin{cases} 1, L_D^{\text{pos}}(\hat{u}_k) \geq 0 \\ 0, L_D^{\text{pos}}(\hat{u}_k) < 0 \end{cases} \tag{1}$$

In the whole process of receiving part, each module transmits. soft information to each other, that is, the method of SISO (Soft Input Soft Output) equalization decoding [3]. It is expressed in the form of logarithmic likelihood ratio, which avoids the error transmission due to the new interference caused by direct transmission of information.

3 Turbo Equalization Algorithm

3.1 Maximum a Posterior Probability (MAP) Algorithm

If the impulse response length of the channel is L, according to Watterson model [7], it can be equivalent to a discrete delay linear circuit with a delay of $L - 1$.That is to say, the state of the channel is 2^{L-1}. The classical Prokais B channel is taken as an example for analysis, and its delay circuit structure is shown in Fig. 3. BPSK (Binary Phase Shift Keying) modulation is adopted. Without the influence of noise, the output v_k at time k channel is:

$$v_k = \sum_{l=0}^{L} h_l x_{k-l} \ , \ k = 1, 2, \cdots N \tag{2}$$

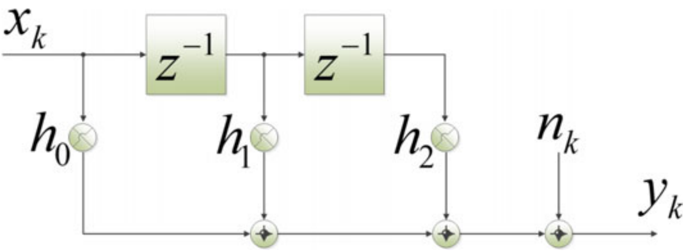

Fig. 3 Equivalent delay circuit structure of Prokais B Channel

where x_k is input signal, h_l is taps of channel.

Therefore, the output of the moment channel is determined by the channel taps, and the state transfer path is presented in a grid diagram (see Fig. 4). The four states of the channel are $s_0 = (1, 1)$ $s_1 = (-1, 1)$ $s_2 = (1, -1)$ $s_3 = (-1, -1)$. These four states can be expressed as sets $S = \{s_0, s_1, s_2, s_3\}$.

Based on the input and output values in the grid diagram and the transition path of the state, the desired posterior probability value $P(x_k|\boldsymbol{y})$ can be calculated.

Then, after processing by equalizer, the soft information of the sequence, namely LLR (Log Likelihood Ratio) value $L(c_k|\boldsymbol{y})$, is obtained:

$$L(c_k|\boldsymbol{y}) = \ln \frac{\sum_{\forall(s_i,s_j)\Rightarrow x_{i,j}=+1} \alpha_k(s_i) \cdot \gamma_k(s_i, s_j)\beta_{k+1}(s_j)}{\sum_{\forall(s_i,s_j)\Rightarrow x_{i,j}=-1} \alpha_k(s_i) \cdot \gamma_k(s_i, s_j)\beta_{k+1}(s_j)} \quad (3)$$

Among them, the posterior probability is calculated according to the joint distribution of the current state and the next time state. To make $\alpha_k(S_k) = p(S_k, y_1, y_2, \cdots y_{k-1})$, $\gamma_k(S_k, S_{k+1}) = p(S_{k+1}, y_k|S_k)$, $\beta_{k+1}(S_{k+1}) = p(y_{k+1}, y_{k+2}, \cdots y_N|S_{k+1})$. And $\alpha_k(S_k)$ $\beta_k(S_k)$ can be obtained by recursive calculation as follows:

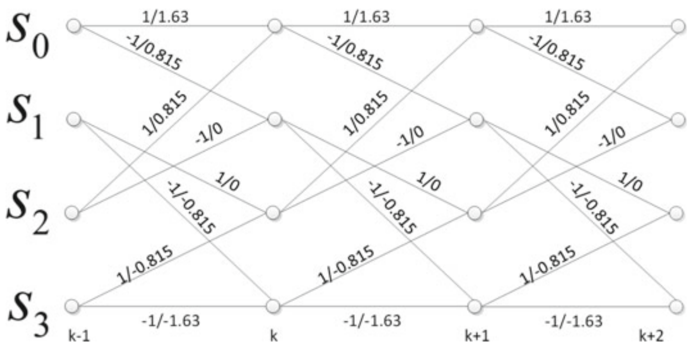

Fig. 4 Arrow diagram of Prokais B Channel

$$\alpha_{k+1}(S_{k+1}) = \sum_{\forall S_k \in S} \alpha_k(S_k)\gamma_k(S_k, S_{k+1}) \tag{4}$$

$$\beta_k(S_k) = \sum_{\forall S_{k+1} \in S} \beta_{k+1}(S_{k+1})\gamma_k(S_k, S_{k+1}) \tag{5}$$

$$\gamma_k(s_i, s_j) = \begin{cases} P(x_k = x_{i,j}) \cdot p(y_k | v_k = v_{i,j}), & (i, j) \in \beta \\ 0, & (i, j) \notin \beta \end{cases} \tag{6}$$

As the calculation method is based on all the observed objects, the performance of MAP algorithm is relatively the best among the equalization algorithms. However, the above analysis shows that the shortcomings of MAP algorithm are also obvious, and it is difficult to realize due to the large amount of calculation in practical engineering practice.

3.2 Minimum Mean Square Error (MMSE) Algorithm

The Turbo equalization based on MMSE algorithm reduces the space of observed samples compared with MAP algorithm, so the calculation of equalization can be reduced to a certain extent. Turbo equalization based on MMSE algorithm is SISO algorithm. Instead of the traditional MMSE equalization method, it adopts an iterative method. Each iteration recalculates the soft information to be transmitted, and then changes the filter coefficient value in real time.

As mentioned above, the MAP algorithm-based Turbo equalization calculates the LLR value of the output posterior probability according to the observation sample space composed of all observed values. The size of the new observation space is determined by the order of linear MMSE filter. After the observation sample space is reduced, the calculation method of LLR value of posterior probability in MMSE algorithm is shown in formula (7):

$$L_E^{pos}(x_k) = \ln \frac{\Pr(x_k = +1 | \hat{x}_k)}{\Pr(x_k = -1 | \hat{x}_k)} \tag{7}$$

where $\hat{x}_k$ is the estimated value of x_k under MMSE algorithm. Since the receiver is not aware of the data sent by the sender x_k, the sent data is estimated. If the mathematical expectation $\overline{x}_k$ and variance v_k at the time k are known, the estimated value $\hat{x}_k$ of the sent data can be calculated:

$$\hat{x}_k = \overline{x}_k + v_k s^H \left(\sigma_w^2 I_N + H V_k H^H\right)^{-1} (z_k - H\overline{x}_k) \tag{8}$$

The tap coefficients of the channel are expressed by matrix $\boldsymbol{H}$, and the mathematical expectation $\overline{\boldsymbol{x}}_{\boldsymbol{k}}$ and variance $\boldsymbol{V}_{\boldsymbol{k}}$ are expressed by vector.

$$\boldsymbol{H} = \begin{bmatrix} h_{M-1}h_{M-2}\cdots h_0 0\cdots 0 \\ h_{M-1}h_{M-2}\cdots h_0 \\ \ddots\ddots\ddots \\ 0\cdots 0 h_{M-1}h_{M-2}\cdots h_0 \end{bmatrix} \tag{9}$$

$$\overline{\boldsymbol{x}}_{\boldsymbol{k}} = \left[\overline{x}_{k-M-N_2+1}, \cdots, \overline{x}_{k+N_1}\right]^T \tag{10}$$

$$\boldsymbol{V}_{\boldsymbol{k}} = Diag\left(v_{k-M-N_2+1}, \cdots, v_{k+N_1}\right) \tag{11}$$

To make $\boldsymbol{s} = \boldsymbol{H}\left[\boldsymbol{0}_{1\times(N_2+M-1)}, \boldsymbol{1}, \boldsymbol{0}_{1\times N_1}\right]^{\boldsymbol{T}}$.

The prior information of the current moment is set to 0, which meets the requirements of Turbo principle. The LLR calculation of external information no longer carries its own information, that is, the estimated value $\hat{x}_k$ and $L_D^{pri}(x_k)$ mutual independence of the transmitted information. $\hat{x}_k$ Is changed into:

$$\hat{x}_k = \boldsymbol{s}^{\boldsymbol{H}}\mathrm{Cov}(\boldsymbol{z}_{\boldsymbol{k}}, \boldsymbol{z}_{\boldsymbol{k}})^{-1}(\boldsymbol{z}_{\boldsymbol{k}} - \boldsymbol{H}\overline{\boldsymbol{x}}_{\boldsymbol{k}} + (\overline{\boldsymbol{x}}_{\boldsymbol{k}} - \boldsymbol{0})\boldsymbol{s}) \tag{12}$$

The LLR of the external output of the equalizer is:

$$L_E^{ext}(x_k) = \frac{2\boldsymbol{c}_k^{\mathbf{H}}(\boldsymbol{z}_k - \boldsymbol{H}\overline{\boldsymbol{x}}_k + \overline{\boldsymbol{x}}_k\boldsymbol{s})}{\left(1 - \boldsymbol{s}^{\boldsymbol{H}}\boldsymbol{c}_k\right)} \tag{13}$$

One of them $\boldsymbol{c}_k = \left[\sigma_w^2\boldsymbol{I}_{\boldsymbol{N}} + \boldsymbol{H}\boldsymbol{V}_k\boldsymbol{H}^{\boldsymbol{H}} + (1 - v_k)\boldsymbol{s}\boldsymbol{s}^{\boldsymbol{H}}\right]^{-1}\boldsymbol{s}$.

The Turbo equalization based on MMSE algorithm reduces the observation sample space and results in less computation than MAP algorithm, but at the same time, it also pays the price of inferior performance to MAP algorithm. In order to ensure the accuracy of decoding, the channel decoding module based on MMSE algorithm Turbo equalization still adopts MAP algorithm.

4 Performance Simulation Analysis

4.1 Simulation Environment and Parameter Setting

The encoder is (7, 5) convolution code encoder, the interleaver is 1024 bit cyclic interleaver, and the constellation mapping adopts QPSK (Quad Phase Shift Keying) modulation mode. In consideration of the effect of multipath, this paper adopts ISI channel smodel with Gaussian white noise, and carries out simulation under different SNR (Signal Noise Ratio). The SISO-MAP decoder is used for channel decoding.

4.2 *Simulation Analysis:*

Simulation performance of different equalization methods

In this paper, the traditional ZF (Zero Force) equalization, LMS (Least Mean Square) equalization [8] and Turbo equalization using iterative theory are simulated respectively. As shown in Fig. 5, the effects of these equalization methods are bad in low SNR. As the SNR increases, the effects of the traditional equalization methods are still poor, and the BER (Bit Error Rate) decreases slowly with a size of about 10^{-1}, which basically has no effect on improving system performance. However when the SNR increases, BER can be effectively reduced by the Turbo equalization method, especially when the iterative curve drops faster.

Simulation performance of Turbo equalization algorithms

Figure 6 shows the BER curves of Turbo equalization using MAP algorithm and LMMSE algorithm respectively. It can be seen from the figure that the Turbo equalization based on MAP algorithm has better performance. However the MAP equalization algorithm computational complexity with the increase of the number of iterations to exponential growth, because the algorithm complexity is too high, on the practical engineering implementation requires great amount of calculation, a waste of time, and Turbo equalization algorithm based on LMMSE (Linear Minimum Mean Squared Error) rules, can greatly reduce the computational complexity, performance is also better than other algorithms, so comprehensive analysis of the algorithm is easier to project implementation.

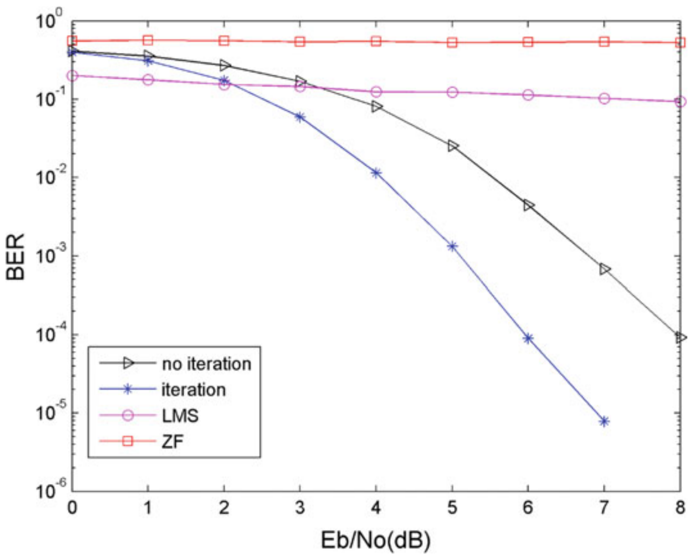

Fig. 5 BER performance comparison in different equalization methods

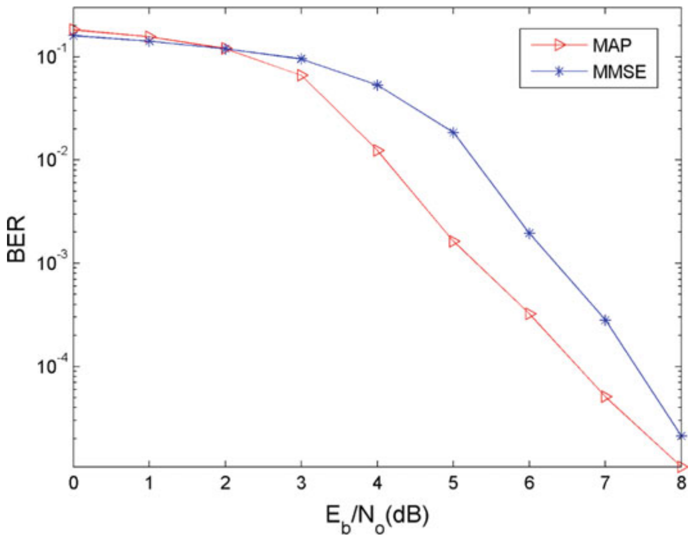

Fig. 6 BER performance comparison of different Turbo equalization algorithms

Turbo equalization simulation performance of different iterations

The number of iterations of Turbo equalizer is set to 4 at the receiving end, and the BER curve is shown in Fig. 7. According to the characteristics of BER curve, it is found that the curve can be divided into three parts: low performance loss, waterfall area and error flat layer area. With the increase of iteration times, the system performance improves gradually. The system makes full use of the external information and gradually reduces the bit error rate of the system by adopting joint equalization and decoding. However, with the increase of iteration times, the improvement of system performance is no longer obvious. Considering that the increase of iteration times also pays the price of greatly increased computation, the number of iterations should be properly controlled in practical engineering applications.

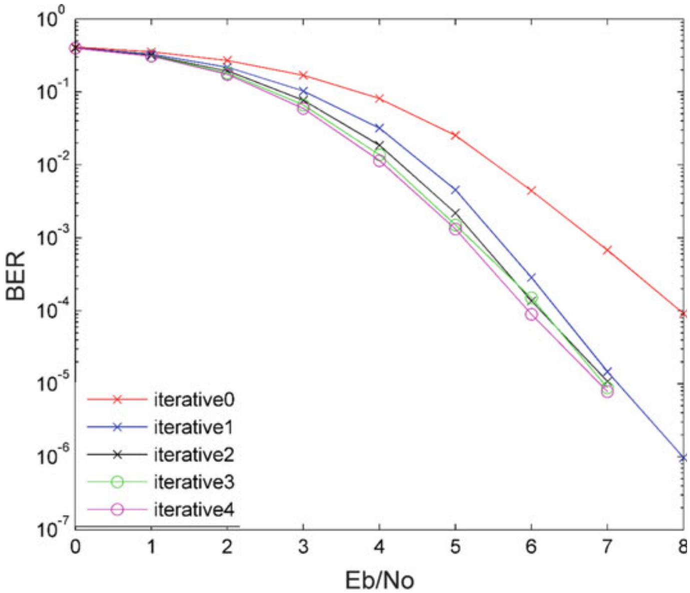

Fig. 7 BER performance comparison of different iterations

5 Conclusion

In this paper, Turbo equalization algorithm of data link communication system and multi-path effect of wireless communication channel are analyzed. Considering the engineering realization, LMMSE equalization algorithm with less computation is combined with soft input and soft output decoder to simulate the system. The simulation results show that the Turbo equalization method improves the system performance greatly when the number of iterations increases within 4 iterations.

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