Research on Symbol Synchronization Technology of LDPC Coding System

Yifeng He\textsuperscript{1,2,*}, Xiuqiang Hao\textsuperscript{1}, Zhemin Duan\textsuperscript{2}, Anzhong Jin\textsuperscript{1}

\textsuperscript{1}Xi’an Institute of Space Radio Technology, Xi’an 710100 China
\textsuperscript{2}Electronic Information College, Northwestern Polytechnical University, Xi’an 710072 China

Abstract. In this paper, the influence of initial estimation of symbol synchronization error and signal-to-noise ratio on the performance of LDPC coding system is analyzed. Combined with the statistical characteristics of iterative decoding output soft information, a symbol synchronization joint signal-to-noise assisted by iterative decoding soft information is proposed. The ratio estimation algorithm combines symbol synchronization, signal-to-noise ratio estimation and iterative decoding through two closely related loops. Through the iterative calculation, signal-to-noise ratio and symbol in the case of extremely low signal-to-noise ratio the estimated value of the synchronization error can be quickly converged, and the decoding performance is close to the coding system with known signal-to-noise ratio and accurate symbol synchronization.

1. Introduction
In 1962, Dr. Gallager proposed an error correction coding scheme based on sparse check matrix, namely low density check code (LDPC), which proved that LDPC code is a good code and gives a simple and effective iteration. Decoding algorithm. However, due to the limited computing power of simulation analysis tools at that time, it was not until 1996 that LDPC codes were found to be a coding scheme that can approximate the channel capacity of additive white Gaussian noise (AWGN). The performance of Turbo codes can be achieved by using LDPC long codes and the Turbo code itself can be regarded as a special case of the LDPC code. Recent studies have shown that the performance of LDPC long codes constructed on irregular graphs is very close to the Shannon limit. At present, the decoding method of LDPC codes mainly uses the BP (Belief Propagation) iterative decoding algorithm proposed by MacKay and Neal, which has very good decoding performance. Although the theoretical performance of LDPC codes is excellent, the performance in practical applications often has a large gap. This is mainly because the synchronization accuracy of the system decreases with the decrease of the signal-to-noise ratio. When the signal-to-noise ratio is low, the traditional synchronization algorithm has already can not work properly, the decoding circuit and the demodulation circuit are out of line, there is a system performance bottleneck without code translatability. How to use the iterative decoding information to assist symbol synchronization is an important research topic of LDPC coding system under low SNR. The change of the symbol synchronization error in the code system of the National Natural Science Foundation of China was changed back to 2007-03-19, and a set of algorithms was proposed. Using 4 times oversampling technology, and obtaining two sets of signals close to the optimal sampling point, respectively decoding the two sets of signals, and weighting the two sets of decoded output soft information, this method effectively overcomes the reception. System performance degradation caused by end symbol synchronization errors.
2. System model and symbol synchronization error on the performance of decoding algorithm

The algorithm and simulation in this paper are based on LDPC coding system (code length l = 1200, code rate 1/4, decoder iteration number upper limit is 20), modulation mode is BPSK, and the code symbol dn with period T is shaped and sent to additive The Gaussian White Noise (AWGN) channel, at the receiving end, oversamples the received signal with a sampling period of TT s = 1/4, and then sends it to the matched filter. The shaping filter and the matched filter use the same square root raised cosine pulse. The roll-off factor is 1. The existence of symbol synchronization error is equivalent to the reduction of the actual signal-to-noise ratio, and the signal-to-noise ratio is proportional to mi in equation (5). It can be seen that the signal-to-noise ratio has a significant impact on the message processing process of the decoder, symbol synchronization. The error not only affects the convergence speed of the iterative process, but also affects the decoding performance of LDPC. Above instruction shows the relationship between signal-to-noise ratio and LDPC decoding performance when the synchronization error is ε = −0.2, ε = 0, and ε = 0.2. It can be seen that at a bit error rate of 3/10, the symbol error of 0.2T loses about 1.3 dB for iterative decoding performance. It can be seen that the existence of symbol error has obvious deterioration of iterative decoding performance, while the literature and this paper use the decoding process of LDPC. The sensitive nature of the symbol synchronization error extracts the synchronization error information from the decoding process.

The estimation of the symbol synchronization error is mostly based on the pulse shape of the symbol itself (such as the cosine roll-off), and the position of the optimal sampling point is estimated by a plurality of sample values within one symbol. However, when the signal-to-noise ratio is very low, the pulse shape of the symbol itself has been destroyed by noise. The traditional symbol error estimation method cannot correctly obtain the optimal sampling position, and the iterative decoding-assisted symbol synchronization method is essentially It is to use the coding gain of Shannon limit code such as LDPC/Turbo code to restore an envelope that can be used for symbol error estimation. In this paper, the sample values of different phases in one symbol are used to form the average value of the output after iterative decoding. Envelope. It shows the mean value of the decoded output amplitude of different symbol phase sampling points under different SNR conditions, forming a set of envelopes for error estimation. It should be noted that when the signal-to-noise ratio is too low, the envelope the shape is not very obvious. In addition, in addition to using the iterative decoding of the mean of the output amplitude, parameters from other iterative processes can also be used to form the symbol error estimation envelope, as Dong U Lee uses the consatiated constraints in the literature to generate Similar to the envelope, good error estimation performance is also obtained.

3. Symbol synchronization joint SNR estimation algorithm

This study shows a cluster of curves, and the estimation of the symbol synchronization error should choose one of them. That is to say, to correctly estimate the symbol synchronization error, firstly, we need to know the signal-to-noise ratio of the received signal, and pass the signal noise. It is used to determine which envelope curve in the curve cluster is used. Summer and Reed give two SNR estimation methods for Turbo coding, but the computational complexity of the two methods is relatively large, and they are based on symbol synchronization. Based on the symbol synchronization algorithm of this paper proposes a symbol synchronization joint SNR estimation algorithm based on LDPC coding system. It can estimate the signal-to-noise ratio of the received signal and the symbol more accurately through several iterations. Synchronization error, combined with the iterative decoding algorithm in this paper, has obtained excellent overall performance. In addition, a practical implementation scheme based on two-dimensional lookup table is presented in this paper, which is convenient for application in engineering practice. This shows the block diagram of the proposed algorithm. The essence is through two closely related loops, namely the SNR estimation loop and the symbol synchronization error estimation loop the symbol synchronization, signal-to-noise ratio estimation and iterative decoding are integrated. The basic working principle is as follows:

By the packet alignment module, all sample values of a code group are extracted from the buffer, a total of 4 × N samples, and N is a code length. The sample points are divided into four groups according
to the symbol synchronization error and sent to the iterative decoder 1 in parallel. The iterative decoder 1 is used to assist the estimation of bit timing and signal to noise ratio. To reduce the computational complexity, the maximum number of iterations is compared. Small, usually set to 3.

The four soft-value outputs of the iterative decoder 1 are sent to the data selector, and the data selector selects the two groups within the synchronization error range $\varepsilon \in -(0.25, 0.25)$ from the four sets of outputs. The signal of the sampling point is sent to the two-dimensional search module. In addition, if there is a case where the packets are not aligned, bit misalignment will occur, and the decoder will converge very slowly, even unable to decode, and the data selection module needs to adjust the packet alignment module.

For the two sets of signal soft values close to the optimal sampling point, respectively obtain the amplitude mean ML, MR of multiple coded packets (usually averaging 5 to 10 packets, and select the mean of 10 coded packets in this paper. Uneven, the estimated value of the symbol synchronization error is large, and the performance will be deteriorated. Under the simulation condition of this paper, the bit error rate is $3 \times 10^{-4}$ and the loss is about 0.3 dB. There are two points PL and PR with interval $\varepsilon = 0.25$. The amplitude mean is closest to ML and MR. According to the signal-to-noise ratio corresponding to the selection curve, the SNR1 of the received signal SNR is sent to the SNRTO 2 $\sigma$ module, according to PL and PR. The position of the symbol synchronization error $\varepsilon$ can be obtained and sent to the interpolator. The data before decoding corresponding to the two sets of signals close to the optimal sampling point is also sent to the interpolator. According to the sampling theorem, two sets of sampling signals can be selected for the raised cosine pulse signal to restore the original received signal waveform. The interpolator recovers the optimal sampling point sample $dn$ according to the two sets of data fed in, and the synchronization error estimate $\varepsilon$, and sends it to the iterative decoder 2, which is used for real information decoding. The maximum number of iterations is set to 20 to 50 (the simulation is set to 20 in this paper).

The SNRTO 2 $\sigma$ module is based on the SNR estimated value SNR1 provided by the two-dimensional search module and the optimal sampling point signal value $dn$ provided by the interpolator, and the estimated noise power 2 $\sigma$ is sent to the two iterative decoding modules, iterating The decoding module adjusts its prior probability at the time of decoding initialization according to this value.

4. Symbol synchronization error correction algorithm

The standard symbol synchronization detector generally uses the maximum likelihood algorithm for synchronization error estimation, and does not utilize the decoded output information of the decoder to assist in the correction of the symbol synchronization error. For low SNR channels with noise power even exceeding the signal power, the existence of synchronization error will make the error of the traditional maximum likelihood (ML-NDA) synchronization error estimation method quite large, even unable to work effectively. The LDPC code is a channel coding technology with strong error correction capability. The decoding output information will help to improve the accuracy of symbol synchronization error estimation in low SNR channels. For a group of LDPC coded data frames with a code length of $n$, if the receiver has obtained frame synchronization, four sets of equally spaced signals with different symbol synchronization errors can be obtained by a 4x oversampling method within one symbol period $TS$. Let $\varepsilon_0$ denote the deviation width of the first set of data $y(\varepsilon_0)$ from the optimal sampling time $y(0)$. For a system with the symbol synchronization error $\varepsilon \in (-0.5, 0.5)$ at the receiving end, $\varepsilon_0 \in (-1, 0)$. For the four sets of interval signals $y_1(\varepsilon_0)$, $y_2(\varepsilon_0+0.25)$, $y_3(\varepsilon_0+0.5)$, $y_4(\varepsilon_0+0.75)$, we can determine the amount of synchronization error $\varepsilon \in (-0.5, 0.5)$ according to the difference in decoding characteristics. Two groups of 0.25, 0.25 are close to the sampling point at the optimal sampling time. When $\varepsilon_0 \in (-1, -0.75)$, bit misalignment occurs in $y_1(\varepsilon_0)$ and $y_2(\varepsilon_0+0.25)$. There is no guarantee that two sets of data in the four sets of data are at the synchronous error $\varepsilon \in (-0.25, 0.25)$. Within the range, it is necessary to shift the data period in which the bit misalignment occurs for one symbol period $TS$. In order to identify the bit misalignment phenomenon, it can be judged by analyzing the decoded output.
result of the group of data by the decoder; this is because the equivalent signal to noise ratio of the data in which the bit misalignment occurs is relatively small.

In order to make the resolution of the decision quantity $\eta(\varepsilon_k)$ large enough, we choose to calculate the number of decoding iterations of the decision quantity $l \in [3, 5]$. If $l$ is too large, the calculation overhead is too large. The following algorithm can be designed to make two sets of data synchronization error $\varepsilon \in (-0.25, 0.25)$ in the four sets of data. The main steps of the algorithm are as follows:

1. After obtaining the judgment amount $\{\eta(\varepsilon_k), 1 \leq k \leq 4\}$ of the four sets of data reception data, the relative sizes between them are compared.

2. If $\eta(\varepsilon_2)$ and $\eta(\varepsilon_3)$ are the two largest decision quantities and are greater than the threshold value $\eta_0$, the corresponding data is considered to be the optimal two sets of data. Otherwise, if $\eta(\varepsilon_1) > \eta(\varepsilon_4)$, the fourth set of data is shifted by one bit; if $\eta(\varepsilon_1) < \eta(\varepsilon_4)$, the first set of data is shifted by one bit. The decoded data is subjected to one decoding iteration, and the obtained decoded output calculates the judgment amount again, and the four sets of data are renumbered in the order of the sampling time, and the step (1) is returned.

3. When $\varepsilon_k \in (-0.25, 0.25)$, the above two steps will fail because the ISI or channel noise power may be too large, which will cause the decoded decision amount to be less than the threshold. In this case, two sets of adjacent data sets having the largest decision amount are selected as the optimum data set from the signal group including the bit shift.

5. Conclusion
This paper firstly analyzes and simulates the influence of signal-to-noise ratio estimation and symbol synchronization error on the performance of LDPC coding system. Based on this, a symbol synchronization combined signal-to-noise ratio estimation algorithm assisted by iterative decoding soft information is proposed. The simulation results show that the algorithm can estimate the signal-to-noise ratio and the symbol synchronization error in the received signal more accurately through several iterations when the signal-to-noise ratio is very low. In addition, this paper also presents a practical two-dimensional table-based implementation, which is convenient for application in engineering practice.

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