Research of LT Code Based on Key Information Feedback in Deep Space Communication

HAI TIAN, DAN-FENG ZHAO, YUN-FAN YANG, AND RUI XUE
College of Information and Communication Engineering, Harbin Engineering University, Harbin 150001, China
Corresponding authors: Hai Tian (tianhai@hrbeu.edu.cn) and Dan-Feng Zhao (zhaodanfeng@hrbeu.edu.cn)

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ABSTRACT CCSDS LDPC code cannot effectively solve the problems of communication interruptions and high communication error rates in deep space communication. An application scheme of digital fountain code for deep space communication is thoroughly studied in this paper. First, an improved joint coding and decoding algorithm for LT code is proposed by adjusting the probability of the source symbol selection to achieve a nonuniform selection coding algorithm and by introducing the idea of real-time decoding on the basis of combining the BP and GE decoding algorithms to disperse the decoding complexity. Second, to ensure that the LT code can be reliably transmitted under different channel conditions, a feedback information extraction method and a retransmission coding method are designed, and an LT code transmission scheme based on key information feedback is proposed. Finally, the concatenation of the LDPC code with the LT code based on key information feedback is proposed. The LDPC code processing can be equivalently run on deep-space channels as erasure channels so that the advantage of the LT code being rateless can still be exploited in deep-space communication. Through simulation experiments, the performance of the cascade codes is analyzed from the two aspects of decoding efficiency and decoding overhead, and the performance of the cascade codes and CCSDS LDPC codes are compared from the perspective of the bit error rate. Simulation results show that the decoding efficiency of the cascade code increases, the decoding overhead decreases, and the communication bit error rate is lower in deep space communication compared with the CCSDS LDPC code. Therefore, the cascade code can improve the reliability of deep space communication.

INDEX TERMS Cascade scheme, CCSDS LDPC code, deep space communication, key information feedback, LT codes.

I. INTRODUCTION
Deep space communication [1] is characterized by long distances, weak signals, long delays, unstable communication and large amounts of data. Therefore, in the process of deep space communication, problems such as long propagation delay, frequent interruption in communication links, high bit error rate and loss of data occur [2]. To accurately recover the transmitted information with low SNR in deep space, channel coding is an effective method due to its characteristics of error correction and error detection [3].

For deep space channels, the convolution code [4], the RS code [5] and the cascade code of both are traditionally widely used. The cascade code is a joint coding and decoding method. Its core is the combination of the convolution code with the RS code to achieve complementary performance, which can effectively obtain better gain in deep space transmission systems. With the development of technology and the improvement in data computing capabilities, a series of advanced coding techniques have been gradually applied in the deep space channel environment. In 1993, Turbo code [6] was proposed as a new concatenated code, and its excellent error correction performance made it widely attractive in the application of deep space communication. In 1998, the superior performance of the LDPC code was rediscovered, showing that LDPC had a bit error performance closer to the Shannon limit [7]. With the continuous development of LDPC codes, these codes have been gradually used in deep space communication. CCSDS has been working on the standardization of LDPC codes and recommended a class of LDPC codes suitable for deep space applications [8]–[10].
These high performance coding techniques demonstrate good performance for a moderate SNR, but their performance is not satisfactory for a low SNR. In addition, LDPC code, Turbo code and other channel coding methods mostly have fixed code rates [11], [12]. For a deep space channel, which is constantly dynamically changing under the influence of electromagnetic interference, node motion and solar wind, communication with a fixed code rate cannot adapt to changes in the channel state, and the code rate is usually set according to the worst link. Therefore, the transmission performance of a fixed-rate coding scheme is still not optimal. At present, the deep space communication system generally uses HARQ technology to ensure the reliable transmission of the communication system on varying channels, but the efficiency is low [13]–[18].

In recent years, the study of fountain code [19], [20] has become a new research direction for deep space channel coding. It has no fixed code length and lower coding and decoding complexity, as well as the advantages of communication without introducing feedback or requiring only a small amount of feedback, which is very suitable for deep space communication [21]–[23]. Therefore, how to combine fountain code with deep space communication to improve the network quality of deep space communication is an urgent concern.

To better apply digital fountain codes to physical channels, the relevant researchers use belief-propagation decoders [24] to successfully extend their application range and achieve certain results [25]–[28]. However, compared with the decoding algorithm on the erasure channel, the operation of this type of decoder is complex, and there are some problems such as the error level [29]. Second, because traditional fountain code must receive enough data packets before it can successfully decoded with high probability, it needs a large amount of storage space and places high requirements on some hardware devices [27]. The probability of successful decoding of fountain code is very low when there are not many data packets at the receiving end, which is not conducive to the research and statistics of very valuable data in deep space communication. Therefore, the application of common fountain code in deep space communication is not ideal.

The other solution is to cascade the digital fountain code with the channel code [30], [31] so that the digital fountain code is still equivalent to that used in the erasure channel through the processing of the channel code in the physical channel to fully exploit its advantage of being rateless [22], [23], [32]. Raptor code is a rateless code that achieves linear complexity coding and decoding using quadratic coding on an LT code platform, which performs well in a binary erasing channel (BEC) [33]–[35]. However, in a noisy channel, the performance is poor. Studies [19], [25], [27] simulated the decoding performance of LT code and Raptor code using the BP decoding algorithm on AWGN. The simulation results show that LT code has an obvious error floor on a noisy channel, while Raptor code does not have an error floor. Studies [19], [36] analyzed the performance of LT codes and Raptor codes in fading channels. Both LT codes and Raptor codes have error floors.

Therefore, to better adapt to the complex and variable deep space channel conditions and further improve deep space transmission capacities, this study uses LT code as the basis to thoroughly study the cascade technology of digital fountain code and channel coding technology and designs a communication system using digital fountain code in the physical layer.

In summary, the main contributions of this paper are as follows.

First, improved coding and decoding algorithms are proposed based on LT code. Through the adjustment of the source symbol selection probability to realize nonuniform selection coding, a memory-based nonuniform selection LT (MNLT) coding algorithm is proposed; based on the joint BP and GE decoding algorithm [37], the idea of real-time decoding is introduced to disperse the decoding complexity, and the BP-GE joint real-time (BP-GE_RT) decoding algorithm is proposed.

Second, aiming at the problem of frequent feedback in the existing schemes [38]–[41], we propose a digital fountain code transmission scheme based on the key information feedback. The key source symbol information extracted from the stop set generation matrix and the channel state information of receiver statistics are fed back to the sender, which provides a basis for the selection of source symbols and the setting of retransmission coding overhead.

Finally, the cascade technology of digital fountain code is studied based on key information feedback and channel coding, and the communication scheme is designed using digital fountain code in the physical layer.

Numerical simulation results show that the improved algorithm can achieve lower decoding failure probability with the same coding overhead and can effectively improve the problem of high complexity in the GE algorithm. The digital fountain code transmission scheme based on key information feedback can effectively reduce the number of coding symbols needed for successful decoding and avoid the problem of the existing scheme requiring multiple feedback and retransmission. The cascade transmission scheme of the LT code based on key information feedback and LDPC code can further reduce the coding overhead. With a small amount of feedback, digital fountain code can be better transmitted by adaptive coding according to channel conditions, thus improving the effectiveness of information transmission under the premise of ensuring the reliability of information transmission. The design of the rateless coding communication scheme in the physical layer is of great engineering significance and will greatly promote the application of rateless code in deep space communication.

The remainder of this paper is organized as follows. In section II, the improved coding and decoding algorithms based on LT code are studied in detail. In section III, the methods of extracting feedback information and coding retransmission are designed, and an LT code transmission scheme

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based on key information feedback is proposed. In section IV, the cascade technology of the LT code based on key information feedback and channel coding is studied, and the communication scheme using digital fountain code in the physical layer is designed. In section V, numerical simulations are performed to simulate the improved coding and decoding algorithms, the LT code transmission scheme based on key information feedback, and the communication scheme using digital fountain code in the physical layer, and the performance of the cascade code is compared with that of the CCSDS LDPC code. We present our conclusions in the final section.

II. RESEARCH ON THE IMPROVED CODING AND DECODING ALGORITHM OF LT CODE

A. MEMORY-BASED NONUNIFORM SELECTION CODING ALGORITHM OF LT CODE

LT code randomly and uniformly selects the source symbols for coding. With the shortening of the code length, the performance declines sharply [42]. The main reason for this decline is that the random selection coding has insufficient coverage with a short code length and is prone to miss the selection and repeat the selection of source symbols, which reduces the coding efficiency and eventually leads to decoding failure [43]. To overcome the above problems, it is necessary to improve the random uniform selection of the source symbols and improve the coding performance by appropriately controlling the selection of the source symbols in the coding process. Different from the existing digital fountain code nonuniform selection coding algorithm, a memory-based nonuniform selection LT (MNLT) coding algorithm for finite code length is proposed in this paper.

The MNLT coding algorithm generates encoded symbols by selecting at least one source symbol each time that is not involved in coding according to the memory (the status of each source symbol in the early stage of coding) to prevent missing source symbols. In addition, through the memorization of the nonuniform selection of source symbols, the connection between low-degree coding symbols is enhanced, and the coding symbols with degree value of 1 and 2 are fully utilized, so more source symbols can be successfully decoded through iteration.

The MNLT coding algorithm can be divided into the following four steps, where $S$ is the set of all source symbols, $S_{12}$ is the set of source symbols encoded with degree values 1 and 2, $S'_{12}$ is the complement set of $S_{12}$, $S_0$ is the set of source symbols not participating in coding, and the number of elements in $S_{12}$ and $S_0$ are represented by $\text{num}_{S_{12}}$ and $\text{num}_{S_0}$, respectively.

1. Initialize each source set:

   $$S = \{s_1, s_2, \ldots, s_k\}, \quad S_0 = S, \quad S_{12} = \phi.$$

2. Select the source symbol according to the degree value.

3. Obtain the corresponding coding symbol by XOR operation on the selected source symbols.

(4) Repeat steps 2 and 3 until enough coding symbols are generated.

Step 2 is the key of the MNLT algorithm, which is divided into the cases where the degree value is 1, the degree value is 2, and the other degree values. When the degree value is 1, if $\text{num}_{S_0} = 0$, a source symbol is randomly selected from $S$ to participate in the coding, and this symbol is copied to $S'_{12}$ without repeating. Otherwise, the source symbol is selected from $S_0$ and moved from $S_0$ to $S_{12}$. When the degree value is 2, if $\text{num}_{S_{12}}$ is less than the total number of source symbols, a source symbol is randomly selected from both $S'_{12}$ and $S_{12}$, and the symbol selected from and from $S'_{12}$ is copied to $S_{12}$. Otherwise, two source symbols are randomly selected from $S$. When the degree value is something else, if $\text{num}_{S_0}>0$, then a source symbol is selected from $S_0$, and the rest are selected from $S$. Otherwise, all the source symbols are selected from $S$, and the symbols selected from $S_0$ are removed each time.

The flow chart of the MNLT code coding algorithm is shown in Figure 1.

A new source symbol can be recovered only when $d - 1$ neighbors of an encoded symbol of degree $d$ are known. Therefore, the probability of recovering a source symbol from an encoded symbol with a degree $d$ is:

$$P_r = \frac{C_i^{d-1} \cdot C_{K-i}^1}{C_K^d} \quad (1)$$

where $i$ is the number of recovered source symbols. It can be seen that the MNLT coding algorithm can ensure that the encoded symbols with degree 1 and 2 can be decoded successfully. Therefore, the $i$ value at $d \geq 3$ is not lower than $(\Omega(1) + \Omega(2)) \cdot K$, that is, $i_{\text{min}} = (\Omega(1) + \Omega(2)) \cdot K$. In RSD, $(\Omega(1) + \Omega(2)) \approx 0.5$, so $i_{\text{min}} \approx 0.5$. However, according to the LT uniform selection-coding principle, may reach $i_{\text{min}}$ only when all the source symbols connected to the coding symbols of the degree values 1 and 2 are not repeated.

$$P_{\text{purerepeat}} = \frac{K!}{K^{\Omega(1)+\Omega(2)} \cdot (K - (\Omega(1)+\Omega(2)))!} \cdot \frac{2}{K(K-2)} \cdot \frac{1}{\Omega(2) \cdot K - 1} / 2 \quad (2)$$

In the LT uniform selection algorithm, the probability $P_{\text{purerepeat}}$ that the source symbols are not repeatedly selected is as shown in (2). That is, the average value of $i$ is much smaller than that of $i_{\text{min}}$. It can be seen from equation (1) that when $d \geq 3$ is much smaller than $K$, $P_r$ is also very small, so when $d \geq 3$, $P_r(\text{LT}) < P_r(\text{MNLT})$.

B. BP AND GE REAL TIME DECODING ALGORITHM OF LT CODE

The belief propagation (BP) algorithm and Gaussian elimination (GE) algorithm are commonly used decoding algorithms of LT codes. The BP decoding algorithm is simple but the performance is poor. The GE decoding algorithm is efficient, but the decoding operation is complicated [44]. Therefore, a more efficient and reasonable decoding method is studied,
which can not only achieve better decoding performance but also control the decoding complexity within a certain range.

The BP decoding algorithm needs a code symbol with the dependency value of 1 to ensure a smooth process of decoding; otherwise, it enters the state of decoding termination. Therefore, the performance of the BP decoding algorithm can be reflected in the decoding failure probability of each step. The current BP decoding state is expressed as \((u, r, c)\), where \(u\) represents the number of undecoded source symbols, \(r\) is the number of encoded symbols with the currently available degree value 1, and \(c\) is the number of encoded symbols with a degree value greater than 1. According to the literature [45], [46], in decoding the encoded symbols with the degree value of \(i\), the probability \(p(i, u)\) of the new source symbol being decoded in the current state is as follows.

\[
\begin{align*}
    p(1, K) &= 1 \\
    p(i, u) &= \binom{K-u}{1} \binom{K-(u+1)}{i-2} / \binom{K}{i}
\end{align*}
\]  
(3)

Further, the state transition probability \(p_u\) of BP decoding can be obtained[37].

\[
    p_u = \frac{\sum_i \Omega_i (\binom{K-u}{1} \binom{K-(u+1)}{i-2} / \binom{K}{i})}{1 - \sum_i \Omega_i (\binom{K-u}{1} \binom{K-(u+1)}{i-2} / \binom{K}{i}) - \sum_i \Omega_i (\binom{K-u}{1} / \binom{K}{i})}
\]  
(4)

According to \(p_u\), the probability of all decoding states transitioning to the decoding stop state \((u > 0, r = 0, c > 0)\) can be deduced, that is, the probability of decoding failure at each step.

If the decoding of the BP algorithm is in the suspended state, the GE algorithm is used to continue decoding, which can solve the problem of decoding failure caused by early termination of decoding. However, this increases the complexity. The key to the complexity of the GE algorithm is its code length. Therefore, we consider a method of real-time decoding to disperse the whole decoding process, which can reduce the decoding complexity of the GE algorithm, and propose a joint decoding method based on BP and GE, which is called the BP-GE_RT (BP-GE real time) decoding algorithm.

First, the decoding process adopts the principle of BP before GE. By using BP decoding, the computation introduced by unnecessary GE operation can be avoided, while GE can continue decoding when BP decoding is suspended. In addition, the idea of real-time decoding is introduced on the basis of the combination of BP and GE, that is, every time an encoded symbol is received, it is decoded once. As soon as the decoding is successful, it immediately stops receiving the encoded symbols and stops the decoding process. Figure 2 is the flow chart of the BP-GE_RT decoding algorithm.

The steps of the BP-GE_RT algorithm are as follows:

1. Receive new encoded symbols and update them based on the known information.
2. Select the corresponding decoding algorithm according to the degree value of the updated encoded symbol. If the
degree value \( d = 0 \), re-execute step 1. If the degree value \( d = 1 \), the BP algorithm is executed until decoding is aborted, and then the GE algorithm is executed. Otherwise, the GE algorithm is directly executed.

(3) Count the number of decoded source symbols. If all are decoded successfully, stop decoding. Otherwise, repeat the previous two steps.

III. RESEARCH ON LT CODE TRANSMISSION SCHEME BASED ON KEY INFORMATION FEEDBACK

A suitable feedback mechanism is introduced in the digital fountain code that can effectively reduce the decoding overhead under different channel conditions but also affects the feedback overhead to different degrees. To further reduce the coding overhead of the digital fountain code and the time of feedback retransmission, a digital fountain transmission scheme is proposed based on key information (KI) feedback based on LT code, which is improved in two parts: feedback information and retransmission coding scheme.

A. THE DESIGN OF FEEDBACK INFORMATION

Undecoded source symbols are the focus of feedback. In fact, there may be a connection between the undecoded source symbols, and as long as some of the key source symbols can be recovered, the remaining source symbols can also be successfully recovered. Therefore, the generator matrix after the decoding is aborted is analyzed, and the key source symbol information is used as feedback information. An example of extracting key source symbols is shown in Figure 3.

Assume that the generation matrix is shown in Figure 3 and the input source symbols are \( s_1, s_2, s_3, s_4, s_5, s_6, s_7, s_8, s_9, s_{10} \). Searching for nonzero elements in rows, the row of element “1” indicates the corresponding source symbol has not been decoded. After decoding once, the undecoded source symbols include \( s_2 \) and \( s_6 \). If \( s_2 \) is assumed to be known, then \( s_6 \) can be decoded, and then \( s_1, s_5, s_9 \) can be decoded. However, the remaining \( s_3, s_8 \) cannot be decoded according to the known conditions. Assuming that \( s_3 \) is known then \( s_8 \) is decoded. Therefore, the key source symbols extracted are \( s_2 \) and \( s_3 \). It is stipulated that the source symbols that meet the conditions are selected according to the principle of serial number from small to large.

After decoding once, the undecoded source symbols are \( s_1, s_2, s_3, s_4, s_5, s_6, s_7, s_8, s_9 \), and after extracting the key information, only \( s_2, s_3 \) need to be fed back. If the two source symbols are successfully received during the retransmission process, other source symbols can be successfully decoded.

The key source symbols are retransmitted without retransmitting all the undecoded source symbols, which can reduce coding redundancy. Retransmitting all the key source symbols can improve the probability of successful decoding after a feedback retransmission, thus reducing the total number of feedback cycles.

The current channel state is also part of the feedback. The reception results of the encoded symbols at the receiver are collected, the current channel erasure probability is calculated, and this information is fed back to the transmitter to provide a reference for determining the coding redundancy during retransmission. We assume that the transmitter sends a total of \( n_k \) encoded symbols, the receiver correctly receives \( n_r \) coded symbols, and the number of key source symbols is \( n_k \);
thus, the channel erasure probability $p_r$ can be estimated as:

$$p_r = 1 - n_r/n_s$$  \hspace{1cm} (5)$$

To ensure that the key source symbols can be successfully decoded after retransmission, the number of encoded symbols received by the receiver cannot be lower than $n_k$. Assuming that the retransmission code redundancy is $R$, then formula (6) must be satisfied.

$$n_k \times (1 - R) \times (1 - p_r) \geq n_k$$  \hspace{1cm} (6)$$

Therefore, the retransmission code overhead should satisfy formula (7):

$$R \geq \frac{n_s - n_r}{n_r}$$  \hspace{1cm} (7)$$

**B. THE DESIGN OF RETRANSMISSION CODING SCHEME**

The rational design of feedback information can make the coding end more targeted for selecting the source symbol for retransmission coding, to improve the effectiveness of coding. However, the retransmitted encoded symbol may still have an indeterminate error across the wireless channel and be erased at the receiving end. In this paper, the system fountain code is used to encode the retransmitted information. The diagram of the system fountain code is shown in Figure 4.

**FIGURE 4.** The diagram of the system fountain code.

The coding results of the system fountain code include the original information and the coded information. The original information is a copy of all the source symbols, which can ensure the full coverage of the source symbols for coding, thereby improving the probability of successful decoding. The coded information is code overhead, and each encoded symbol is randomly connected with different source symbols. When some encoded symbols in the original information are erased in the channel, the related information may be included in the encoded information. As long as the encoded information is correctly received, the source symbol information can still be recovered. The overhead of the encoded information can be adjusted according to the current channel state (satisfying formula (7)).

In summary, the specific steps of the digital fountain code transmission scheme based on key information (KI) feedback are as follows:

1. The encoder continuously encodes and transmits the LT encoded symbols, and the decoder begins to decode for the first time after receiving $n_{tar}$ encoded symbols.

2. If all the source symbols are decoded successfully, the decoder sends an ACK signal to the encoder, and the encoder ends the coding. Otherwise, the decoder sends a feedback message to the encoder containing key source symbol information and the channel erasure probability, then jumps to step 3. If all the retransmitted encoded symbols have been sent and still have not been decoded successfully, the next feedback process is entered.

3. When the encoder receives the feedback information, the encoder adjusts the relevant parameters according to the feedback information and uses the system fountain code to encode and retransmit the extracted key source information.

**IV. DESIGN OF LT CODE TRANSMISSION SCHEME IN DEEP SPACE COMMUNICATION**

Many scholars have studied the cascade of digital fountain code and channel code. Through the processing of channel code, the wireless channel is equivalent to the erase channel, so that the digital fountain code can continue to exert an advantage on the equivalent erase channel. LDPC code has error detection and correction abilities, excellent decoding performance and relatively low coding and decoding complexity, so it is discussed in the scheme of cascading with LT code. The block diagram is shown in Figure 5.

**FIGURE 5.** Block diagram of cascade scheme of LT code and LDPC code.

LDPC code is used to detect and correct the information transmitted through the channel at the bit level. If there is error bit information after decoding, the encoded packet (data frame) is discarded, and then the decoder continues to receive and process the next encoded packet; in contrast, the encoded packet enters the digital fountain code decoding process. By this operation, the actual wireless channel is logically equivalent to an erase channel suitable for the digital fountain code characteristics, so the digital fountain code can process the information in the upper layer. Figure 6 is a schematic diagram of the cascade scheme of the LT code and LDPC code used in this paper.

At the sender, first, the source data information is grouped. Assuming the original information is $B$, the information is divided into $K$ packets (frames), and each packet length is $L$ bits, that is, $B = K \times L$. At this point, a packet corresponds to a source symbol in the LT code. Then, the LT code is encoded.
According to the degree distribution function and the coding algorithm, and the length of each packet after coding is still $L$ bits. Thereafter, LDPC code coding is performed on each packet at a code rate of $r_0 = L/L'$, and the length of the encoded packet is $L'$ bits.

At the receiver, the decoder decodes the encoded packets that are affected by channel interference and noise. First, LDPC decoding is performed. If the decoding fails, the encoded packet is directly discarded; otherwise, fountain code decoding is performed. If the original source information can be completely recovered, the transmission is completed, and the sender stops transmitting the encoded packet.

The key information feedback process in the cascade scheme is also introduced. If the original source information is not recovered, the key information is extracted from the decoding result and fed back to the encoder, and the corresponding encoded packet information is continuously transmitted.

V. NUMERICAL SIMULATION AND ANALYSIS
First, comparative simulation analysis is conducted on the performance of the LT code MNLT coding algorithm and BP-GE_RT decoding algorithm, and joint simulation analysis is conducted. Then, the LT code transmission scheme based on key information feedback is simulated and analyzed. Finally, the simulation and comparative analysis of the cascade scheme performance of digital fountain code and channel code on the wireless channel are carried out.

A. SIMULATION AND ANALYSIS OF MNLT CODING ALGORITHM
Taking coding overhead as the variable, BP decoding is used to simulate the performance of MNLT coding with different code lengths and compared with the doping and nonuniform selecting (DNS) algorithm in the literature [42] and the traditional LT coding algorithm based on uniform selection. The time of simulation is 5000, and the code lengths are 128, 256, 512, and 1024. The simulation results are as shown in Figure 7. It can be seen that when the code length is fixed, as the overhead increases, the BER of the three coding algorithms decreases. In contrast, the MNLT code has the fastest convergence rate, that of DNS is second, and that of LT is the slowest. For code lengths of 128 and 256, when the overhead exceeds 0.2, the waterfall area is entered. For code lengths of 512 and 1024, when the overhead exceeds 0.15, the waterfall area is entered. In addition, the convergence rate increases as the code length increases.

The simulation results show that the coding performance of the MNLT and DNS algorithms is significantly better than that of the LT coding algorithm, and the shorter the code length is, the more obvious the advantage of the MNLT coding algorithm.

B. SIMULATION AND ANALYSIS OF BP-GE_RT DECODING ALGORITHM
In this paper, the three decoding algorithms of BP-GE_RT, BP and GE are compared in terms of complexity and decoding overhead. The time of the simulation is 5000, and the code lengths are 128, 256, 512, 1024, and 2048. The algorithm complexity is represented by the average number of xor calculations when decoding is successful. The more computations there are, the more complex the algorithm. The simulation results are shown in Figure 8. It can be seen that the GE algorithm is obviously more complex than the BP algorithm, and the longer the code length is, the more complex it is. The on the Fly Gaussian Elimination (OFG) algorithm introduces the idea of real-time online decoding based on the GE algorithm, which can reduce decoding complexity. The BP-GE_RT algorithm combines the BP decoding algorithm with the basis of the OFG algorithm and uses the BP algorithm priority principle. The simple xor calculation in the BP algorithm is used to replace the complex calculation of matrix triangulation and elimination in the GE algorithm.

In addition, the BP-GE_RT algorithm introduces the idea of real-time online decoding, which allows timely capture of the moment of decoding success, so only a small number of encoded symbols need to be calculated; this is equivalent to fewer linear equations in the decoding process, and to a certain extent, the decoding complexity can be reduced. In addition, in each independent decoding process,
the BP-GE_RT algorithm decodes the source symbols as soon as possible, while the GE algorithm only begins to recover the source symbols after processing the generated matrix. Therefore, the BP-GE_RT algorithm performs decoding operations where the code length decreases every time, so the decoding complexity can be further reduced.

The decoding overhead of the three decoding algorithms is also simulated, as shown in Figure 9. It can be seen that the GE algorithm only needs to receive encoded symbols with slightly longer code length to decode successfully, compared with the BP algorithm, which requires much more coding overhead. Therefore, the GE algorithm has obvious advantages in decoding overhead. The decoding overhead of the BP-GE_RT algorithm is similar to that of the GE algorithm, because the BP-GE_RT algorithm integrates the idea of solving linear equations of the GE algorithm. Therefore, the BP-GE_RT algorithm not only does not negate the advantage of high efficiency of the GE algorithm but also solves the complex problem of decoding to a great extent.

C. CO-SIMULATION AND ANALYSIS OF CODING AND DECODING ALGORITHMS

In this section, we co-simulate the improved coding and decoding method using the MNLT algorithm and the BP-GE_RT algorithm and compare it with the traditional method using the LT algorithm and the BP decoding algorithm considering that the actual wireless channel conditions cannot be ideal and information is discarded (erased) due to noise interference. Therefore, the performance of the improved coding and decoding algorithm is simulated on the erasure channel.

Figure 10 shows the simulation results for the BER and FER performance of the improved coding and decoding.
method and the traditional method in the erasure channel. The code lengths are 128, 256, 512, and 1024, and the code overhead is 0.5.

It can be seen that for different channel erasure probabilities and different code lengths, the FER and BER of the improved joint decoding algorithm are much lower than those of the traditional LT decoding algorithm. Even when the code lengths are 512 and 1024, the improved algorithm can achieve a BER lower than $10^{-6}$ when the channel erasure probability is lower than 0.2 and as high as $10^{-2}$ or more using the conventional LT algorithm. We can conclude that the improved coding and decoding algorithm can better resist the influence of the channel on code symbol erasure.

**D. SIMULATION AND ANALYSIS OF LT CODE TRANSMISSION SCHEME BASED ON KEY INFORMATION FEEDBACK**

In this paper, the extracted key source symbols are used as the source symbols for retransmission coding, which can reduce the number of retransmission source symbols. In this section, the performance of LT codes based on key information feedback is simulated.

As seen in Figure 11, when the numbers of input source symbols are 128, 256, 512, and 1024, the number of retransmitted source symbols after key information extraction is lower than the number of source symbols that are not recovered when the first decoding fails. Compared with the number of unrecovered source symbols, the number of retransmitted source symbols after key information extraction is reduced by more than half. Therefore, the key source symbol extraction operation can effectively reduce the number of source symbols for retransmission coding.

In the transmission scheme of the LT code based on key information feedback (LT-KI), the receiving end immediately starts decoding after receiving the same number of encoded symbols as the input source symbol. According to the matrix structure generated after the decoding is stopped, the key source information is extracted and fed back, and then the coding is retransmitted according to the system fountain coding method. From the two perspectives of decoding overhead and feedback cycles, the LT-KI transmission scheme is compared with the shift LT (SLT) [38], all random LT (ARLT) [47] and most useful LT (MULT) [48] transmission schemes proposed by scholars. The SLT code was first proposed to combine digital fountain code with feedback. The author took the size of the decoded set as feedback information and designed the SLT degree distribution based on the distribution of RSD. The ARLT coding scheme divides all the source symbols into decoded and undecoded parts at the coder end according to the feedback information provided by the decoder. Only undecoded source symbols are retransmitted to improve the coding effectiveness. On the basis of the ARLT scheme, the MULT scheme is more detailed in the processing of source symbols according to the feedback information and only uses the most useful source symbols in the decoding process as the feedback information.

Figure 12 compares the number of coded symbols required for decoding success for the four transmission schemes under...
different conditions. The channel erasure probabilities are 0.01, 0.05, 0.10, 0.15, 0.20, and 0.30, and the decoding performance varies with channel deletion probability when the code lengths are 128, 256, 512, and 1024.

As seen in Figure 12, the number of encoded symbols required for successful decoding of the four schemes increases with increasing erase probability. The MU-LT scheme has a lower decoding overhead when the erasure probability is lower. As the erasure probability increases, the decoding overhead increases significantly. The decoding overhead of the other three schemes are very close, but in contrast, the LT-KI scheme has the least overhead.

Figure 13 shows the feedback cycles required by different feedback schemes. With increasing input code length, the feedback frequencies of ARLT and MUlt increase rapidly. Since the LT-KI scheme sets the retransmission coding overhead according to the channel conditions, less feedback is needed in different cases to successfully decode, which illustrates the effectiveness of the LT-KI transmission scheme.

E. SIMULATION AND ANALYSIS OF CASCADE SCHEME OF LT-KI CODE AND LDPC CODE

In the cascade scheme, the source information is divided into several packets. The LT code first encodes all the source information in groups, and then the LDPC code encodes each encoded packet at the bit-level. Each packet can be regarded as an input source symbol, so the length of the source symbol is the length of the information packet, and the corresponding number of source symbols is the number of information packets. The length of the input source symbol affects the performance of the channel code, thus affecting the erasure probability of the encoded symbol, while the performance of the LT code is related to the number of source symbols.
In this section, first, the performance of the cascade scheme in wireless communication is simulated for different numbers and lengths of source symbols. In the cascade scheme, the channel code is LDPC code, and the LT code uses the improved coding and decoding algorithm. Each source symbol contains $L$ bits of information. A source symbol can be regarded as a data frame, and the frame error rate (FER) is simulated and analyzed. Then, the decoding overhead and feedback overhead of the cascade transmission scheme based on KI feedback is simulated and analyzed. Finally, the performance of the cascade scheme under different code lengths and code rates on the Gaussian channel and Rayleigh channel is simulated and compared with the LDPC code in the CCSDS standard [8] under the same parameters.

Figure 14 shows the performance simulation curve of the cascade scheme under the AWGN channel when the number of source symbols $K$ is different. Each source symbol length is $L = 512$ bits, and the code rate of the LT code is $2/3$, that is, $1.5K$ LT code coding symbols are generated. The LDPC code with $1/2$ code rate is used as the inner code to encode each symbol after LT coding, and the total rate of the cascade code is $1/3$. Figure 13 also shows the FER performance of the LDPC code when the code length is $L = 512$ bits and the code rate is $1/2$.

In the cascade scheme, the FER of the LDPC code is equivalent to the symbol erasure probability of the LT code. When $E_b/N_0 = 1.2dB$, the number of encoded symbols correctly received by the receiving end exceeds the number of source symbols. Therefore, the FER of the cascade scheme quickly enters the waterfall area after $E_b/N_0$ is greater than $1.2dB$.

It can be seen from Figure 14 that under the same channel conditions, when the length of source symbol is
constant, the performance of the cascade scheme improves with increasing number of source symbols. This occurs because the performance of the LDPC code is also determined after the source symbol length is fixed. The probability that the encoded symbols are erased after passing through the channel is the same, and the performance of the LT code under the same channel conditions is positively related to the number of the source symbols.

The performance of the LDPC coding scheme is also shown in Figure 14. It can be seen that when the signal-to-noise ratio exceeds 1.25dB, the performance of the cascade scheme is significantly better than that of the LDPC coding scheme under the same frame length and code rate conditions; even if the LDPC code uses $L = 1024bits$ information length for coding, its performance compared with that of the $L = 512bits$ cascade scheme is still poor.

The use of cascaded transmission schemes in wireless communication has significant advantages over LDPC coding alone. Through the coding of digital fountain code, the performance requirements for LDPC coding can be reduced regarding both the rate and the FER; additionally, the cascade scheme can reach a lower FER under a lower SNR, thereby obtaining a certain coding gain.

Figure 15 shows the performance as curve of the cascade scheme under the AWGN channel when the source symbol length $L$ differs. The number of source symbols is $K = 1024$, and the lengths of the source symbols are $L = 256bits, 512bits$ and $1024bits$. The fixed LT code rate is $2/3$, the LDPC code rate is $1/2$, and the cascade code rate is $1/3$. Similarly, the FER of the LDPC code is the channel erasure probability of the LT encoded symbol in the cascade scheme.

It can be seen that the longer the source symbol length is, the earlier the performance curve of the cascade scheme enters the waterfall area. Therefore, when the number of source symbols is constant, the larger the value of the source symbol length is, the better the transmission performance of the cascade scheme. This occurs because the longer the source symbol is, the better the performance of the LDPC code, and the lower the probability that the encoded symbol is erased.

The performances of the cascade code over both the AWGN channel and Rayleigh fading channel are studied, as shown in Figure 16 and Figure 17. The code rates are $1/2$ and $1/3$ respectively, and the code lengths are $1024 bits$, $512 bits$ and $256 bits$. It can be seen that the cascade codes have excellent performance over both the AWGN and Rayleigh channels and are superior to LDPC codes by approximately $0.5 − 1.5dB$ for the same FER performance.

The performance of the cascade code and LDPC code that is used as the deep space communication application standard by CCSDS is studied over different channels, as shown in Figure 18. The code length of CCSDS LDPC is $1024bits$, and the code rate is $0.5$. It can be seen that the cascade code has higher coding gain than the CCSDS LDPC code.

Figure 19 shows the encode overhead of the cascade scheme based on KI feedback with different numbers and lengths of source symbols. Figure 19(a) shows that when the source symbol length is fixed, the coding overhead required for successful decoding is almost the same for different numbers of source symbols. Figure 19 (b) shows that when the number of source symbols is fixed, as the source symbol length increases, the coding overhead decreases significantly. Therefore, the transmission performance of the cascade scheme is less affected by the number of source symbols. Therefore, the transmission performance of the cascade scheme is less affected by the number of source symbols and is more affected by the length of the source symbols.

A cascade scheme based on KI feedback can adjust subsequent coding strategies based on feedback information to reduce coding overhead; however, this introduces feedback overhead. Figure 20 shows the number of feedback cycles required for the successful decoding of the cascade scheme based on KI feedback for different numbers and lengths of source symbols. It can be seen that as the signal-to-noise ratio increases, the average number of feedback cycles decreases and is within 2 cycles. Therefore, the cascade transmission scheme can effectively reduce coding overhead at a lower feedback cost and thus better adapt to channel conditions.

In deep space exploration missions, it is important to efficiently and reliably return all kinds of precious image information from distant spacecraft. To ensure high-efficiency and high-reliability transmission of the image data, a cascade code transmission scheme is applied in this paper to a deep space image transmission system.

The error between a reconstructed image and the original image is usually used to measure the quality of the reconstructed image. The evaluation methods include the mean squared error (MSE) and peak signal noise ratio (PSNR). The calculation formula of the MSE is:

$$MSE = \frac{1}{M \times N} \sum_{i=1}^{M} \sum_{j=1}^{N} (f(i,j) - \hat{f}(i,j))^2$$

(8)
where \( f(i, j) \) and \( f'(i, j) \) represent the pixel values of the original image and the reconstructed image at \( (i, j) \), respectively, and the MSE reflects the reconstruction quality of the image. The smaller the MSE value is, the smaller the deviation between the reconstructed image and the original image and vice versa. The calculation formula of the PSNR is:

\[
PSNR = 10\log_{10} \frac{255^2}{MSE} \quad (9)
\]

It can be seen that the PSNR and MSE are essentially the same. The larger the PSNR value is, the smaller the deviation between the reconstructed image and the original image and vice versa. PSNR is usually viewed as a quality metric and MSE as a distortion metric. The set partitioning in hierarchical trees (SPIHT) image compression algorithm has extremely low computational complexity and high-quality recovery characteristics. In addition, this algorithm breaks the limit where the coding efficiency is directly proportional to the complexity, makes reasonable use of multiresolution characteristics after wavelet decomposition, and has good coding performance. “Lena” is chosen as the test image, the image compression algorithm is the SPIHT algorithm, the channel...
The normalized MSE of the reconstructed image is used as a standard for evaluating the quality of the image reconstruction, and the CCSDS LDPC coding scheme and cascade coding scheme are used for image transmission simulation. The CCSDS LDPC code rate is 1/2, and the code length is 1024 bits. In the cascaded code transmission scheme, the LT code coding overhead is 1.5, the LDPC code rate is 3/4, and the code length is 1024 bits. The source compression code rate is 0.1. To ensure the accuracy of the results, the transmission process is repeated 1000 times, and the results are averaged. The simulation results are shown in Figure 22. It can be seen that when the NMSE is greater than 0.95, the quality of the
reconstructed image is good, and an average performance gain of 0.6dB is obtained by using the cascade coding transmission scheme over the CCSDS LDPC coding transmission scheme.

| Algorithm            | Complexity            |
|----------------------|-----------------------|
| LT coding            | $O(\ln(K/\delta))$   |
| LT decoding (BP)     | $O(K\ln(K/\delta))$  |
| LDPC coding          | $O(L^2)$              |
| LDPC decoding (BP)   | $O(L^2)$              |

**F. COMPLEXITY ANALYSIS OF CASCADE SCHEME AND CHANNEL CODING SCHEME**

It can be seen from the simulation that if the LDPC coding transmission scheme needs to achieve the same bit error rate performance as the cascaded transmission scheme, the information bit length of LDPC needs to be more than one order of magnitude longer than the information bit length of the channel coding in the cascaded scheme.

Table 1 shows the algorithm complexity of LT code and LDPC code. The coding complexity of LT code is determined by the average degree $D = O(\ln(K/\delta))$, where $\delta$ is the minimum decoding failure probability and $K$ is the length of LT code. Therefore, the coding complexity of LT code is $O(\ln(K))$. The decoding complexity of LT code is the average degree multiplied by the number of packets required for successful decoding. Theoretically, an LT code with a code length of $K$ can be successfully decoded if it receives $K$ coding packets, so the decoding complexity of LT code is approximately $O(K\ln(K))$. The complexity of LDPC code Gaussian elimination algorithm is $O(L^2)$, where the length of LDPC code is $L$. The complexity of LDPC code BP/ log-BP decoding algorithm is $O(L^2)$.

In summary, for the source information with the length of $B_{bits}$, the cascade transmission scheme can divide the data into $K$ blocks and each block has the length of $L_{bits}$ that is, $B = K \times L_{bits}$. The complexity of the cascade scheme is composed of two parts, one is the complexity of LT code with code length $K$, namely, $C_{LT} = O(K\ln(K)) + O(\ln(K))$. And the other is the complexity of $K$ LDPC codes with information bit length $L_{bits}$, namely, $C_{LDPC} = O(L^2)$.

For LDPC coding transmission scheme, $B_{bits}$ source information is divided into $B = K' \times L_{bits}$. That is, LDPC code information bit length is $L_{bits}$, need to encode $K'$ frames. Therefore, the complexity of LDPC coding scheme is $K'\times L_{bits}$, namely $C'_{LDPC} = O(L^2)$.

Because $L'$ is an order of magnitude larger than $L$, especially when the source data block is large, $L'$ is much larger than $L$, so $C'_{LDPC} \gg C_{LDPC}$. LT code complexity $C_{LT}$ is logarithmic, and the complexity is lower. Therefore, the complexity of LDPC coding transmission scheme is far more complicated than the cascade transmission scheme.

Therefore, as the amount of source information data increases, the complexity of using LDPC codes is higher. The advantage of the cascading scheme is that the larger data is transmitted in blocks, which can improve the transmission reliability and reduce the complexity.

**VI. CONCLUSION**

Aiming at problems such as communication interruption and high communication bit error rate in deep space communication, this study combines the rateless characteristics of LT code with the anti-interference and error detection capabilities of LDPC code to form the LT-LDPC code, which improves the communication success rate of deep space communication.

First, the LT code is optimized, and an improved coding and decoding algorithm is proposed for the LT code. Aiming at the problem of reducing the coding efficiency, such as missing and repeated selection of source symbols in the uniform selection coding algorithm, by adjusting the probability of selecting the source symbols in the coding process, we propose a nonuniform selection coding algorithm based on memory, called the MNLT algorithm; considering the respective advantages of BP and GE algorithms in terms of complexity and decoding efficiency, a joint decoding algorithm using BP and GE is introduced, and the idea of real-time decoding is introduced to further decentralize the decoding complexity. Simulation results show that the improved joint coding and decoding algorithm can achieve lower decoding failure probability and can effectively improve the problem of high complexity in the GE algorithm.

Second, to further reduce the high latency problem in deep space communication, based on the research of existing digital fountain code transmission schemes combined with feedback and in-depth analysis of undecoded source symbols, a digital fountain code transmission scheme based on key information (KI) feedback is proposed. This solution can effectively reduce the number of coding symbols required for successful decoding and can avoid the problem of multiple feedback retransmissions.

Finally, the LT code based on the key information feedback and LDPC code are combined, and the cascade transmission scheme is applied to deep space communication. The LDPC codec and the deep space channel are equivalent to an erasure channel. The performance of the cascade code is analyzed and compared from the aspects of decoding efficiency and decoding overhead, and the communication performance of the deep space communication is analyzed from the perspective of the communication error rate. The simulation results show that the cascade transmission scheme can further reduce coding redundancy. With a small amount of feedback, the digital fountain code can better adaptively transmit according to the channel conditions, thereby improving the effectiveness while ensuring the reliability of information transmission. Compared with CCSDS LDPC codes, the performance of the cascade codes is greatly improved, and in deep space communication, the communication performance is improved.
Because the channel model is not the research focus of the paper, AWGN and Rayleigh channels are used to simulate deep space channels in the simulation. In the next step, we will study the performance of digital fountain code transmission schemes under more channel models.

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**HAI TIAN** received the B.E. degree in communication engineering from Harbin Engineering University, Harbin, China, in 2014, where he is currently pursuing the Ph.D. degree with the College of Information and Communication Engineering. His research interests include channel coding for wireless communications and joint source channel coding (JSCC) schemes for deep space communication.

**DAN-FENG ZHAO** received the B.S. and Ph.D. degrees from the College of Information and Communication Engineering, Harbin Engineering University, Harbin, China, in 1983 and 2002, respectively. He was a Professor and the Assistant Dean of the College of Information and Communication Engineering, Harbin Engineering University, in 2002 and from 2002 to 2012, respectively. From 2009 to 2012, he was the Director of the Teaching Demonstration Center of National Electrical and Electronic Experimental, Harbin Engineering University. In 2010, he was the Adjunct Leader of the Course for Electrical and Electronic Engineering, National Teaching Team Foundation Course. His current research interests include modern communication systems, communications and signal processing, underwater acoustic communication technology, high performance coding and modulation, and video signal processing.

**YUN-FAN YANG** received the B.E. degree in electronic information engineering from Harbin Engineering University, Harbin, China, in 2014, where he is currently pursuing the master’s degree with the College of Information and Communication Engineering. His research interest includes rateless coding for wireless communications.

**RUI XUE** received the M.E. and Ph.D. degrees from Harbin Engineering University, Harbin, China, in 2006 and 2009, respectively. From July 2011 to July 2012, he was a Visiting Scholar with the Nonlinear Signal Processing Laboratory, University of Melbourne, Australia. From August 2010 to September 2013, he was a Post-doctoral Fellow with the College of Automation, Harbin Engineering University, for satellite navigation research. His research interests include radio mobile communication systems, satellite communication, systems and satellite navigation and positioning, and include error-correcting codes, high spectral efficiency modulation, coded modulation, iterative decoding and detection, and so on.

**HAI TIAN** received the B.E. degree in communication engineering from Harbin Engineering University, Harbin, China, in 2014, where he is currently pursuing the Ph.D. degree with the College of Information and Communication Engineering. His research interests include channel coding for wireless communications and joint source channel coding (JSCC) schemes for deep space communication.