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Multiple Access Rateless Network Coding for Machine-to-Machine Communications

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Abstract

In this paper, we propose a novel multiple access rateless network coding scheme for machine-to-machine (M2M) communications. The presented scheme is capable of increasing transmission efficiency by reducing occupied time slots yet with high decoding success rates. Unlike existing state-of-the-art distributed rateless coding schemes, the proposed rateless network coding can dynamically recode by using simple yet effective XOR operations, which is suitable for M2M erasure networks. Simulation results and analysis demonstrate that the proposed scheme outperforms the existing distributed rateless network coding schemes in the scenario of M2M multicast network with heterogeneous erasure features.

Keywords

rateless network coding; multiple accesss; machine-to-machine communications (M2M)

1 Introduction

Machine-to-machine (M2M) communication system is expected to support a massive number of devices communicating with each other in a fully automated fashion with minimum or without human intervention [1]. Equipped with networked and real-time processing capabilities, these devices can implement a wide range of applications, such as intelligent transportation systems (ITS), healthcare monitoring, smart metering, energy management and smart grids.

M2M communication system is generally characterized by a massive number of machine-type communication (MTC) devices that have no/low mobility, low computational and storage capabilities, and low power budget [2]. Moreover, most MTC devices suffer from severe congestion and access delay in an M2M system with a large number of devices [3]–[5]. Therefore, the main motivation behind this paper is to propose a coding strategy that exploits the interference in the channel to increase data rates. Our work focuses on the cooperative joint network and coding strategy for MTC devices in multicast settings. These MTC devices disseminate messages to multiple receivers simultaneously with the help of relay nodes.

The rateless code, originally investigated in [6] for single source broadcasting in a single hop network, is deemed as a milestone for packet erasure codes. It can recover the original k information symbols from any $n=k+O(\sqrt{k}\ln^2(k/\theta))$ received coded symbols with the probability $1-\theta$ and the decoding cost of $O(k\ln(k/\theta))$ of operations, where θ is the allowable failure probability to recover the original message after n coded symbols have been received. In addition, the encoding and decoding process of the rateless code is complex, including logarithmic order for Luby Transform (LT) code and linear order for Raptor code. Furthermore, both LT and Raptor codes are able to provide practical capacity-achieving solutions, if their encoding degree distributions are sophisticated designed [7], [8].

The rateless code has been widely applied in cooperative communications [9]–[13]. In [9], the complexity, delay, and memory of different state-of-the-art rateless coding algorithms are analyzed for a multi-hop network. In [10], a superimposed on-the-fly recoding scheme is performed by each transport node in a multi-hop tree network, but it is difficult to implement due to the high decoding complexity. The first distributed LT (DLT) code is proposed in [11], and a new degree distribution, named deconvolved soliton distribution (DSD) is designed. However, all the source nodes and relays are assumed

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to have exact information regarding the number of sources and encoding distributions to adapt relaying schemes. In [12], the authors utilized density evolution and linear programming frameworks to find an optimal combination at each relay node for any network architecture consisting of four sources. This manner can achieve the asymptotical error floor, but has intractable calculation complexity. A relaying protocol for Y-networks, namely Soliton-Like Rateless Coding (SLRC), is introduced in [13]. By enabling probabilistic forwarding and combining packets to reduce the overhead between relay and destination, the aggregate distribution at the destination can still maintain a near-ideal distribution, even if one source left the network. However, the lack of buffer utilization in SLRC relay limits the total encoding and decoding overheads.

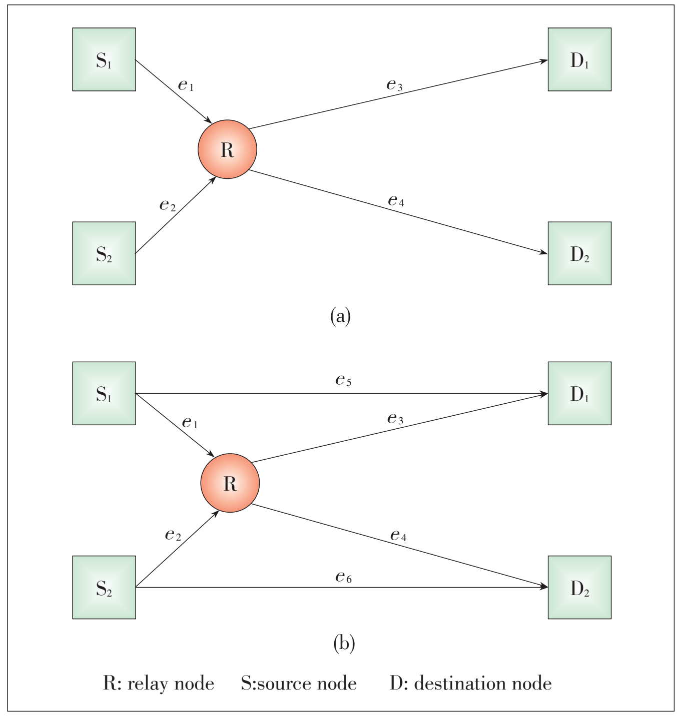
The destination cooperation in interference channels is another feature of M2M communication where one device can act as a relay for another device. A cooperative communication scheme for two mobile users is proposed in [4], which is potentially able to receive and decode each other's messages based on the signal-to-interference-plus-noise ratio (SINR). In [5], The received signal at the destination can be realized as a superposition of coded symbols sent from the relay, which is capacity approaching if an appropriate successive interference cancellation (SIC) is used for decoding [14].

In this paper, we propose an adaptive rateless network coding scheme in an M2M erasure network. First, an simple degree distribution is designed for rateless coding in all the source devices, and the collided devices transmit simultaneously. Then, an optimal relaying strategy is proposed to forward and combine the encoded packets with appropriate proportions, according to different erasure probabilities over the underlying edges. This is particularly suitable for M2M communications with strict power limitations, especially when the data size is small. By doing this, the total time slot of transmission is reduced obviously while the high decoding success rates are maintained. Moreover, we compared the current typical rateless coding relay schemes with our proposed scheme, with the aspects of the complexity and buffer memory. Simulation results show that the proposed rateless network coding scheme outperforms the existing distributed rateless coding schemes under various erasure probability scenarios.

2 Network Model and Rateless Code

2.1 Network Model

In [12] and [13], authors have introduced and optimized the applications of rateless coding in the Y-network model. We attempt to extend the Y-network model to a relay multicast model as shown in **Fig. 1a**. The relay R multicasts the data streams both to destination nodes D_1 and D_2 and guarantees the two source data to be recovered. Due to the special feature of rateless coding, the overheads at two destination nodes are apparently the same as the one in the Y-network and accordingly the DLT and SLRC algorithms are also appropriate for the network model in Fig. 1a.



▲**Figure 1. The proposed network models: (a) relay multicast model; (b) butterfly model.**

We also consider a butterfly network model (**Fig. 1b**), which has two sources, one relay and two destinations. Two direct edges are added to send two separate data streams form S_1 and S_2 respectively. The encoded packets should be processed (recoding) at R, and they can be converged at both D_1 and D_2 in the end. The model uses multicast from source to relay and directly from source to destination as well. This is its remarkable difference from the Y-network model. We define the edges in this model as e_1 to e_6. Each edge has an erasure probability ε_i, which is described as an independent-identical-distribution (i.i.d) Bernoulli variable. We assume that the packet size and the transmission rate of all the edges are equivalent (one time slot one packet). The rateless coding transmission scheme in this network model is described as follows.

1) Step 1: S_1 and S_2 generate the encoded packets with a rateless coding degree distribution [7];
2) Step 2: S_1 multicasts the encoded stream both to R and to D_1, and simultaneously, S_2 multicasts its stream to R and to D_2;
3) Step 3: R generates a new encoded stream by the relaying network coding (NC) scheme with the received encoded packets and multicasts them to D_1 and D_2 simultaneously;
4) Step 4: Once D_1 and D_2 receive enough encoded packets, they start to decode the encoded packets from the source and relay nodes to recover the two sources original packets;

5) Step 5: After successful decoding, D_1 (D_2) transmits a single acknowledgment (ACK) packet indicating the termination of the session.

2.2 Rateless Coding

Rateless coding is a modern forward error correction (FEC) technology. It protects from packet-loss, and can reduce the feedbacks for user acknowledgement due to rarely caring about the erasure probability of the channel. In a single-hop scenario, the original packets are defined as k data symbols, and the encoded packets required by the decoder are defined as n encoded symbols. Furthermore, the overhead is defined as $(n-k)/k$ and the number of encoded symbols sent by the sender is defined as N. Therefore, the expected code rate at the sender is conveyed as $R=\frac{k}{N}=\frac{(1-\varepsilon)k}{n}$, where ε is the erasure probability of the channel. On the other hand, the encoding and decoding complexity of rateless codes is very low, which are expressed as $O(\ln k)$ for LT codes and $O(k)$ for Raptor codes.

As an example of rateless coding, the LT code uses robust soliton distribution (RSD) to achieve the erasure channel capacity under the single hop network. The coding degree distribution is a key element for the successful recovery of data symbols. For a parameter δ and the length k RSD, $\mu(k)$ is defined as:

$$\mu(i)=\frac{\rho(i)+\tau(i)}{\beta},\ for\ i=1,\cdots,k\ , \tag{1}$$

where

$$\rho(i)=\begin{cases}1/i, & for\ i=1\\ 1/i(i-1)\ , & for\ i=2,\cdots,k\end{cases}, \tag{2}$$

$$\tau(i)=\begin{cases}S/ik, & for\ i=1,\cdots,\lfloor k/S\rfloor-1\\ S\ln(S/\delta)/k, & for\ i=\lfloor k/S\rfloor\\ 0, & for\ i=\lfloor k/S\rfloor+1,\cdots,k\end{cases}, \tag{3}$$

$$\beta=\sum_{i=1}^{k}\mu(i)+\tau(i)\ . \tag{4}$$

S is the average number of degree-one symbols, namely ripple size, which is defined by $S=c\ln(k/\delta)\sqrt{k}$, where c>0.

It is worth noting that the LT decoder performs the BP algorithm with the prior knowledge of the degree and associated neighbors. Given a block of encoded symbols, the decoder recursively decodes the data symbols from the bipartite graph connecting the information and encoded symbols. The BP algorithm starts from degree-one symbols, by removing their contributions from the graph in order to produce a smaller graph with another set of degree-one encoded symbols. Then, the new degree-one encoded symbols of this smaller graph are removed again, and iteratively the process continues to recover all data symbols, as described in [7] and [8].

3 Analysis of Relaying Schemes

As the rateless code is used in multi-source relay network, the erasure probability of different paths (multicast and unicast) may influence the relaying strategy and the corresponding performance with NC. Specifically, the relay R may receive no packet from S_1 or S_2 in a time slot due to packet loss in multi-source relay network. Hence, it is an interesting and significant topic to select proper rateless coding algorithms based on NC and relaying strategy for efficient transmissions on lossy network. In this section, we try to consider the conventional methods in Y-networks and butterfly networks, and compare their decoding performance at the destination nodes. Moreover, we propose a new optimized -NC scheme to trade off the decoding performance by selecting proper forwarding and combining probabilities.

3.1 Comparison of Typical Relaying Schemes

We assume that the number of original packets is k and the number of encoded packets generated is N at both the source nodes. In two fixed time slots, the destination nodes D_1/D_2 of butterfly network can receive one encoded packet from S_1/S_2 in the first slot, and then receive one from the relay R in the second slot. It is a limited condition that D_1 and D_2 only receive the maximum $2N$ packets when N encoded packets are sent from the source, if and only if all the edges are lossless. D_1 and D_2 use the BP decoding algorithm to decode the compilations of two encoded streams after $2N$ slots to recover $2k$ original packets, respectively. There are the following four typical relaying schemes in this butterfly network model:

1) Store-and-Forward (SF)

The relay R immediately forwards the packets to the next hop as soon as it receives packets. If two packets arrive simultaneously, R randomly forwards one of them and stores another into the buffer. If the relay R receives no packet, it waits for the next slot. Due to the uncertain storage of packets, this scheme may easily make congestion on R.

2) DLT

With S1 (S2) using DSD, R performs random decision protocol in [11] to combine and forward two received packets. Once the erasure event occurs at one of the edges between sources and relay, R directly forwards another received packet. If no packets arrive, the relay waits for the next slot. These waiting slots at relay lead to low efficiency due to a serious waste of sources. By using considerable low-degree encoded packets, this scheme could scarcely cover all the original packets of two sources, despite of its simple encoding complexity.

3) SLRC

With S_1 (S_2) using RSD, R uses the SLRC relaying scheme to operate the two encoded packets. It forwards most of the low degree packets (degree-one or degree-two) directly and combines other high degree packets, in order to assure the aggregate distribution at destination nodes to be soliton-like. With the SLRC, the transmission time slot is not wasted if the packet on e_1 or e_2 is dropped, by R's choosing two packets from the buffers to combine and forward. Compared with the DLT scheme, this SLRC scheme obtains more gains.

4) eXclusive-OR (XOR)

This scheme is based on the simple eXclusive-OR (XOR) operation of classic NC in the butterfly model. The relay R combines the two received packets into one new packet. If only one packet arrives, the relay R sends the received packet directly, and if no packet arrives, R waits for the next time slot.

Since the relay carries the two received packets by forwarding and combining operations, the recoding complexity comes almost from XOR operation which depends mainly on the erasure probability of the edges. In addition, the relay requires two buffers to store the packets from different sources. Therefore, we compare the four schemes (**Table 1**). We can find that SF has the least recoding complexity, and the complexity of SLRC and DLT depends significantly on their forwarding probability. On the other hand, DLT and XOR need the smallest buffer size of only one packet for each source, while SLRC must store all the packets not forwarded for the next operation.

3.2 Analysis of Decoding Matrix

In this section, we make an intuitive analysis on recovery performance at the destination nodes by the BP decoding matrix, in which the BP decoding algorithm could be described as one unitization process, with the columns denoting the encoded symbols and the rows denoting the original data symbols.

We define A and B as the encoding matrix at S_1 and S_2 respectively, and denote the dimension of the matrix as the subscripts. The decoding matrices at D_1 and D_2 are the aggregation of the forwarding and combining sub-matrices A and B, with $2k$ rows due to the number of data symbols from two sources. Since the lost packets have been removed from the matrix, the number of columns reveals the received encoded symbols by destination nodes exactly.

For the SF scheme, the decoding matrices at D_1 and D_2 are defined as:

$$F^{D_1}_{2k\times\left[(1-\varepsilon_5)N+\left[(1-\varepsilon_1)+(1-\varepsilon_2)\right](1-\varepsilon_3)N/2\right]} = \begin{bmatrix} A^{S_1}_{k\times(1-\varepsilon_5)N} & A^{S_1}_{k\times(1-\varepsilon_1)(1-\varepsilon_3)N/2} & 0 \\ 0 & 0 & B^{S_2}_{k\times(1-\varepsilon_2)(1-\varepsilon_3)N/2} \end{bmatrix}, \quad (5)$$

$$F^{D_2}_{2k\times\left[(1-\varepsilon_6)N+\left[(1-\varepsilon_1)+(1-\varepsilon_2)\right](1-\varepsilon_4)N/2\right]} = \begin{bmatrix} 0 & A^{S_1}_{k\times(1-\varepsilon_1)(1-\varepsilon_4)N/2} & 0 \\ B^{S_2}_{k\times(1-\varepsilon_6)N} & 0 & B^{S_2}_{k\times(1-\varepsilon_2)(1-\varepsilon_4)N/2} \end{bmatrix}. \quad (6)$$

From the matrices F^{D_1} and F^{D_2} in (5) and (6), we know that the SF scheme is only appropriate for unicast like source to destination, since the dimensions of sub-matrices for A and B are extremely unequal. The lack of combination operations makes the destination nodes unable to recover the whole data symbols. Therefore, this scheme is inefficient for multicast in the butterfly network model.

For the SLRC scheme, as an optimized DLT, we only present its decoding matrices that are defined as:

$$H^{D_1}_{2k\times(1-\varepsilon_5+1-\varepsilon_3)N} = \begin{bmatrix} A^{S_1}_{k\times(1-\varepsilon_5)N} & A^{S_1}_{k\times\nu_1(1-\varepsilon_3)N} & 0 & A^{S_1}_{k\times(1-\sum_{i=1}^{2}\nu_i)(1-\varepsilon_3)N} \\ 0 & 0 & B^{S_2}_{k\times\nu_2(1-\varepsilon_3)N} & B^{S_2}_{k\times(1-\sum_{i=1}^{2}\nu_i)(1-\varepsilon_3)N} \end{bmatrix}, \quad (7)$$

$$H^{D_2}_{2k\times(1-\varepsilon_6+1-\varepsilon_4)N} = \begin{bmatrix} 0 & A^{S_1}_{k\times\nu_1(1-\varepsilon_4)N} & 0 & A^{S_1}_{k\times(1-\sum_{i=1}^{2}\nu_i)(1-\varepsilon_4)N} \\ B^{S_2}_{k\times(1-\varepsilon_6)N} & 0 & B^{S_2}_{k\times\nu_2(1-\varepsilon_4)N} & B^{S_2}_{k\times(1-\sum_{i=1}^{2}\nu_i)(1-\varepsilon_4)N} \end{bmatrix}, \quad (8)$$

where ν_i (i=1, 2) is the probability distribution of the relay forwarding packets from S_1 and S_2, while $\tilde{\nu}_i = 1-\nu_i$ is the distribution of the packets into the buffer.

For the XOR Scheme, the decoding matrices are defined as:

$$G^{D_1}_{2k\times\left[(1-\varepsilon_5)+Max\{(1-\varepsilon_1),(1-\varepsilon_2)\}(1-\varepsilon_3)\right]N} = \begin{bmatrix} A^{S_1}_{k\times(1-\varepsilon_5)N} & A^{S_1}_{k\times(1-\varepsilon_1)(1-\varepsilon_3)N} \\ 0 & B^{S_2}_{k\times(1-\varepsilon_2)(1-\varepsilon_3)N} \end{bmatrix}, \quad (9)$$

$$G^{D_2}_{2k\times\left[(1-\varepsilon_6)+Max\{(1-\varepsilon_1),(1-\varepsilon_2)\}(1-\varepsilon_4)\right]N} = \begin{bmatrix} 0 & A^{S_1}_{k\times(1-\varepsilon_1)(1-\varepsilon_4)N} \\ B^{S_2}_{k\times(1-\varepsilon_6)N} & B^{S_2}_{k\times(1-\varepsilon_2)(1-\varepsilon_4)N} \end{bmatrix}. \quad (10)$$

In (9) and (10), G can be segmented into four sub-matrices:

▼**Table 1. The complexity and buffer for the four schemes**

	Relay recoding complexity	Buffer size of relay
SF	0	$N/2$ packets for each source
DLT	$(1-\varepsilon_1)(1-\varepsilon_2)(1-\gamma_i)N$	One packet for each source
SLRC	$(1-\sum_{i=1}^{2}\nu_i)N$	$(1-\nu_i)N$ packets for each source
XOR	$(1-\varepsilon_1)(1-\varepsilon_2)N$	One packet for each source

* γ_i and ν_i are the distributions of the no-forwarding packets for DLT and SLRC
DLT: distributed Luby transform
SF: store-and-forward
SLRC: soliton-like rateless coding
XOR: eXclusive-OR

the two parts in the left are the forwarding matrix A or B, and the two right parts are the combined matrices. Since the combined symbols of this scheme may be lack of degree-one and degree-two packets, it needs enough columns of single sub-matrices A or B on the edge e_5 or e_6 in Fig. 1 to start the BP decoder. In the decoding process, only if all the data symbols of A have been recovered, B can begin to decode.

By comparing these decoding matrices, we can find that the forwarding matrix A or B has occupied such numerous columns in H, as $A^{S_1}_{k\times\nu_1(1-\varepsilon_3)N}$ or $B^{S_2}_{k\times\nu_2(1-\varepsilon_3)N}$. The combination part of encoded symbols as A and B $\begin{bmatrix} A^{S_1}_{k\times(1-\sum_{i=1}^{2}\nu_i)(1-\varepsilon_3)N} \\ B^{S_2}_{k\times(1-\sum_{i=1}^{2}\nu_i)(1-\varepsilon_3)N} \end{bmatrix}$ in the SLRC scheme has much less columns than that of $\begin{bmatrix} A^{S_1}_{k\times(1-\varepsilon_1)(1-\varepsilon_3)N} \\ B^{S_2}_{k\times(1-\varepsilon_2)(1-\varepsilon_3)N} \end{bmatrix}$ in the XOR scheme. SLRC can still recode new packets in the relay to transmit, even if no packets received due to enough large ε_1 and ε_2. However, it has not fully utilized the packets directly from D_1 and D_2 on e_5 and e_6. Note that, only N encoded packets at most could be sent by the sources and relay, which constrains the required time slots to be only $2N$ in the network model, given one packet at each time slot. It renders that the decoding matrix H has many single forwarding columns in A or B which obviously reduces the relevance of encoded symbols from S_1 and S_2. As a result, the large proportion of forwarding packets by the relay cannot give much help to improve the BP decoding performance, especially on the condition of the relatively small erasure probability ε_5 and ε_6.

On the other hand, when ε_5 and ε_6 become larger, the decoding performance is mostly decided by the proportion of forwarding the single packets and XOR combination of two sources' packets in the relay. In the XOR scheme, the number of recovery data symbols would decline very fast due to the lack of low degree encoded symbols for BP decoding. Besides, the XOR operations would be blocked and degraded since two separate packets from two sources could hardly arrive at the relay simultaneously in the large ε_5 and ε_6.

3.3 Proposed Optimized NC Scheme

On the basis of above analysis, we have found that the relaying schemes should forward the low-degree packets for starting BP decoder, by taking into consideration the erasure probabilities of the direct edges e_1 and e_2. On the other side, the relay also needs to remain the enough proportion of combinations of the packets in the buffers, in order to prevent from no received packets from S_1 and S_2 simultaneously. Accordingly, we propose a novel NC scheme with a self-adjusted forwarding probability associated with the variations of ε_5 and ε_6. The basic rule of the proposed scheme is as follows: if ε_1 and ε_2 increase, the relay immediately forwards more low-degree packets; if ε_5 and ε_6 decrease, the relay combines more packets.

The proposed algorithm, named Opt-NC scheme, has the comparable complexity and buffer requirements with the SLRC scheme. We denote the forwarding probabilities λ and θ for encoded packets from S_1 and S_2, which are predetermined to be equivalent to the erasure probabilities ε_5 and ε_6, respectively. Algorithm 1 shows the steps of Opt-NC algorithm.

Algorithm 1: Opt-NC Scheme (at one time slot)

p_1: received encoded packets from S_1;
p_2: received encoded packets from S_2;
d_1: degree of p_1;
d_2: degree of p_2;
λ: forwarding probability of the low degree packets from S_1;
θ: forwarding probability of the low degree packets from S_2;
a=rand();
b=rand();
if $d_1=1\vee 2$ and $d_2=1\vee 2$ and $a<\lambda$ and $b<\theta$
    forward p_1 or p_2 with equal probability;
    put another packet into the buffer of another source;
else if $d_1=1\vee 2$ and $a<\lambda$
    forward p_1 and put p_2 into the buffer of S_2;
else if $d_2=1\vee 2$ and $b<\theta$
    forward p_2 and put p_1 into the buffer of S_1;
else
    put the packets received into the buffers respectively;
    p_{b1}: random choose one packet in the buffer of S_1;
    p_{b2}: random choose one packet in the buffer of S_2;
    $p_{XOR}=p_{b1}$ XOR p_{b2};
    forward p_{XOR};
end if

* $\vee$ means logical operator of OR

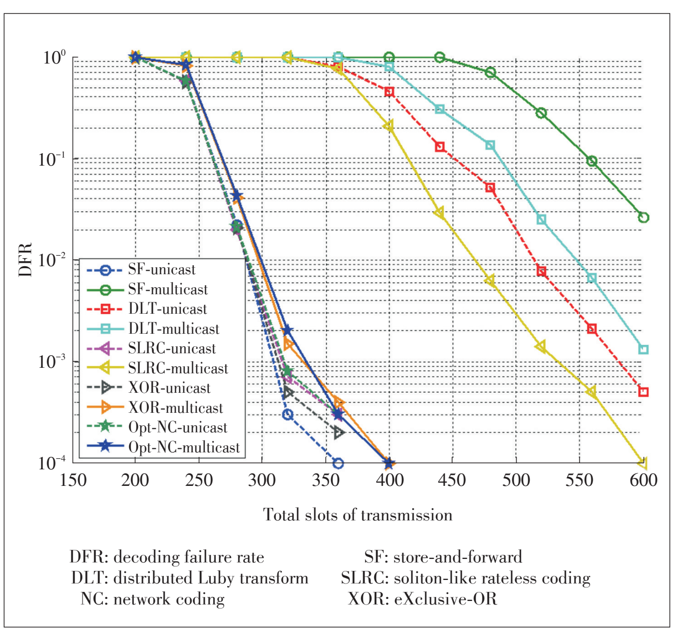
4 Simulation Results and Discussion

We analyze the performance of the above five algorithms in a butterfly network coding system as Fig. 1b. The encoding degree distribution is selected to be RSD with parameters $\delta=0.05$, $c=0.03$. The number of data symbols $k=100$, and the number of encoded symbols from S_1 and S_2 is indicated to be the same as N. We emulate the encoding and decoding procedure using Monte Carlo experiments with 10,000 times. The ratio between the statistics of decoding failure times and total experiment times is defined as decoding failure rate (DFR). In this work, the lowest displayable DFR in our simulation is 10^{-4}. Given the time slots and erasure probabilities of edges, the lower DFR of relaying schemes means outstanding decoding performance. We give the unicast performance and multicast performance respectively to discuss the influences between the single source and double sources.

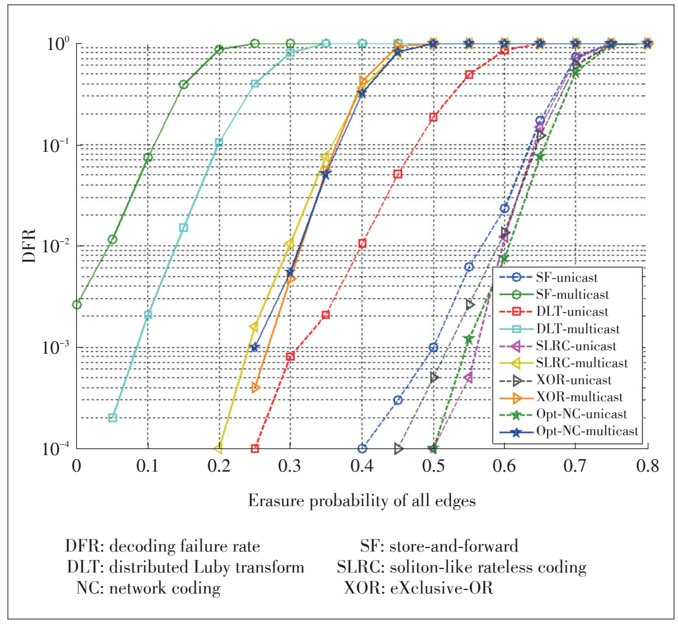
Fig. 2 gives the performance of five schemes in the erasure free network model. We can find that the Opt-NC scheme approaches the unicast and multicast curves of the XOR scheme, which obtain the lowest DFR of 10^{-4} at 400 time slots. The other schemes need at least 600 time slots to recover two sources' data, due to their inefficiency of buffer utilization at the relay. This simulation result proves that the Opt-NC scheme can apply as a typical NC scheme to get the remarkable multicast throughput gains in lossless network data transmission.

With vary erasure probabilities from 0 to 0.8 of all edges among ε_1 to ε_6, and the code length N=360, the decoding performance of five schemes is shown in **Fig. 3**. The unicast results of the schemes reveal the better performance than the multicast results, since the erasure probabilities of ε_1, ε_2, ε_3, ε_4 make the combination operations at the relay inappropriate. It is noted that the SLRC, XOR and Opt-NC schemes have similar multicast decoding performance, with the DFR lower than 10^{-4} and the erasure probability of 0.2. The simulation results indicate that our adaptive Opt-NC scheme integrates the advantages of SLRC and XOR, which also reveals outstanding decoding performance in lossy network.

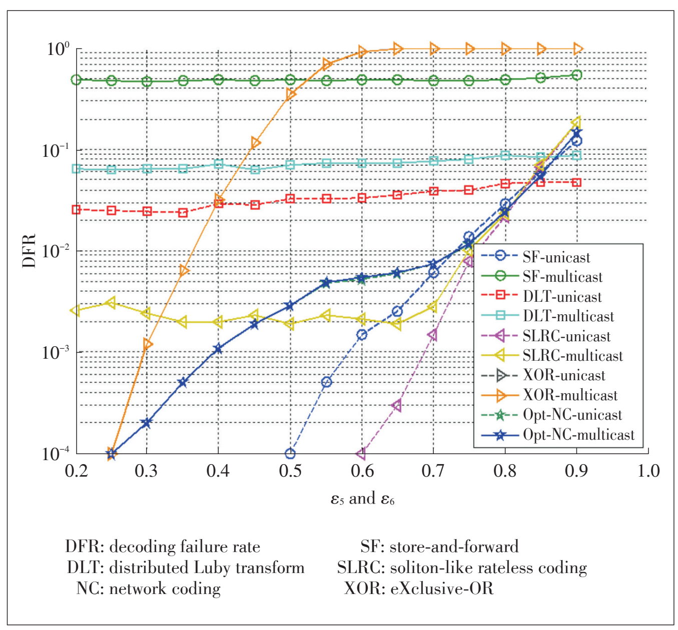
Fig. 4 shows the DFRs of five schemes with the encoded packets of N=250, ε_1 to ε_4 null, and ε_5 and ε_6 from 0.2 to 0.9. Since the multicasts of the SF and DLT are both restricted by the limited number of encoded packets from source, their decoding performance maintains at an inferior level in spite of ε_5 and ε_6 increasing. On the other side, the multicast DFR performance of XOR and that of Opt-NC are almost consistency with their unicasts. If the erasure probabilities of e_5 and e_6 are lower than 0.25, the Opt-NC scheme can get the similar DFR with



▲**Figure 2. All the edges are erasure free, *k*=100.**



▲**Figure 3. All the edges have the same erasure probability, *k*=100, *N*=360.**



▲**Figure 4. Edges e_5 and e_6 are lossy while other edges are lossless, *k*=100, *N*=250.**

that of the XOR scheme. If ε_5 and ε_6 increase from 0.25 to 0.9, the Opt-NC scheme outperforms the XOR scheme by its dynamic property. In addition, compared to the SLRC, the Opt-NC scheme also has a better performance as ε_5 and ε_6 are both lower than 0.45. However, the SLRC gives a lower decoding failure rate as the ε_5 and ε_6 are both in a range of 0.45 to 0.8. Once the erasure probability increases higher than 0.8 (the edges e_5 and e_6 are almost interrupted), the Opt-NC scheme approaches SLRC-multicast with a higher efficiency. In a word, the proposed relaying scheme is an adaptive method to compromise the decoding performance of XOR and SLRC.

5 Conclusions

In this paper, we have studied rateless network coding applied in machine-to-machine communications for multiple access applications. A novel dynamic relaying scheme Opt-NC was proposed that exploits the forwarding and combining operations to obtain an enhanced decoding performance of the decoder at the destination nodes. The Opt-NC scheme has adaptive capability of responding to the vary erasure probability of direct edges. The simulation results show that the proposed relay scheme performs close to the optimal XOR scheme in lossless and lossy network, respectively. Furthermore, the Opt-NC scheme can be used in the physical layer by incorporating the XOR operation and superposition practical modulations.

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