Adaptive Rateless Coding with Feedback for Cooperative Relay Networks

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***Abstract*—Cooperation among nodes is proposed as an effective means of combating fading and for enhancing system's overall capacity and coverage. The robustness of rateless codes makes them particularly attractive for such networks. Yet, the decoding aspects of rateless codes are discarded in most of the traditional distributed networks. In this paper, the intermediate packet decodability of rateless codes is exploited to reduce the number of packets being processed at the relay nodes. This is achieved by harnessing the back channel (from destination to relays) for feedback. The relay transmissions and processing are thereby confined to assist the receiver in decoding the remaining information. It is shown both analytically and through simulation studies that such a scheme achieves significant savings in computation complexity, memory usage and overall energy consumption.**

I. Introduction

In wireless networks due to the scarcity of available bandwidth, effort is mainly focussed on designing schemes with efficient spectrum utilisation under severe degradation and fluctuation of the channel. The potential of spatial diversity in the form of employing multiple antennas to cope with fading as well as to enhance the network capacity is well recognised [1]. However the number of antennas that can be implemented on mobile devices to exploit the spatial diversity is dictated by the operating wavelength. These implementation constraints have fostered the recent interest in cooperative communications, where a virtual antenna array is created by cooperation among the nodes [2], [3]. By relaying signals for each other an improvement in link reliability, range extension and power saving capability can be achieved [2], [3].

In cooperative communication, the overheard information at the intermediate nodes is processed based on the relaying technique employed. Of the forwarding strategies employed at the cooperating nodes, decode and forward based relaying strategies mitigate the effect of noise amplification and thereby offer an improved performance. By integrating forward error correction (FEC) into decode and forward relaying technique, a class of cooperation known as coded cooperation is proposed [4]. In coded cooperation, initially source transmits a block of data together with some parity to both relay and destination. At the relay, the remaining portion of the parity bits (based on the initial code rate employed at source) are generated and forwarded to the destination. After combining the codes from both source and relay nodes, the destination will continue to decode the message. For such scenarios, due to the volatile nature of wireless links, codes that exhibit the property of naturally adapting its rate to time-varying channel conditions offers better bandwidth efficiency. The erasure resilience of rateless codes such as Luby-Transform (LT) codes introduced in [5], ideally suit such relaying technique.

Rateless code property of reconstructing the source data from any subset of encoded packets is particularly attractive for cooperative communication. There are several schemes that utilises rateless coding in cooperative relay networks [6]–[8]. However most of them neglect the inherent advantages of rateless coding or requires considerable complexity and/or resources. A multi-relay cooperative transmission technique is proposed in [6], with mutual information accumulation through rateless codes at the intermediate nodes. A significant performance improvement can be observed when the relay nodes listen to other relays' transmissions. This performance improvement is obtained at the cost of additional spectrum utilisation due to the requirement of orthogonal channels for each node. One of the several distributed fountain coding approaches is discussed in [7], where the data at the sources are encoded in a way that mere XORing at the relay nodes results in a fountain coded packet. However the complexity of such schemes increases with the number of nodes. In [8], an algorithm to modify the encoding/decoding of fountain codes is proposed for online re-coding at the relay nodes. This scheme imposes additional complexity at the relay nodes.

To the best of authors' knowledge in most of the existing works, the intermediate nodes encode all the received source packets. However, by allowing the intermediate node to process all the source packets in cooperative networks, there is always the risk that redundant information may be transmitted across the links. The availability of additional information - the number of input symbols already decoded at the receiver - at the relay, will eliminate such risks. By completely eliminating the redundancies, the transmission across each link supplements the required mutual information at the destination. The relay can be intimated of the decodability at the receiver by harnessing the back channel (from the receiver to the relay) for feedback. An adaptive rateless code assisted with feedback for a relay network is proposed in this paper. In such a scheme, the packets recovered at the receiver are revealed to the relay nodes in an attempt to eliminate redundant transmissions. A feedback channel (from receiver to the transmitter) is exploited in [9] for point-to-multipoint network to design a new class of rateless codes known as shifted LT codes. Further improvement in intermediate performance of rateless codes can be obtained by suitable design of degree distribution, even though such codes are designed to recover the source message with very low overhead [10]. The implication of such an idea of using a feedback message is that significant savings in communication complexity, memory usage and overall energy efficiency can be obtained. The achieved bandwidth efficiency supersedes such a feedback cost which is reflected in the subsequent sections.

The rest of the paper is organized as follows. The network model considered for the analysis and an introduction on the proposed scheme is presented in Section II. This is followed by detailed numerical analysis of the proposed scheme in Section III. In Section IV some simulation results are presented to reveal the advantages of the proposed scheme using rateless codes. Finally, concluding remarks and future extensions of the work are summarized in Section V.

II. Network Model & Proposed Scheme

A three node network comprising of source node (*SN*), relay node (*RN*) and destination node (*DN*) as depicted in [11] is considered for the present analysis. All nodes are constrained to employ half duplex transmission, and are equipped with a single antenna.

A. Received Signal & Channel Model

The received signal at node q over a quasi-static Rayleigh fading channel is given as

$$y_q = h_{pq} x_p + \eta_q \tag{1}$$

where $p \in \{SN, RN\}$ and $q \in \{RN, DN\}$, x_p is the symbol transmitted by node p and η_q is the additive white Gaussian noise (AWGN) at node q with zero mean and variance N_0. h_{pq} is the channel gain between transmitting node p and receiving node q, which is complex Gaussian distributed corresponding to Rayleigh fading amplitudes as in [12]. The channel variance σ^2_{pq} captures the pathloss between nodes and is proportional to $d_{pq}^{-\alpha}$, where d_{pq} is the distance between nodes p and q; and α is the pathloss exponent. Each node is assumed to transmit under equal power constraint, $\varepsilon_s \forall p$ and hence $E[|x_p|^2] = \varepsilon_s$. If the received signal level, $\gamma_q = |h_{pq}|^2 \varepsilon_s / N_0$ is less than the threshold value, γ_t (threshold is dependent on the code employed and on the outage requirement), then the received symbols are declared as an erasure [13]. We no longer assume the availability of channel state information (CSI) at the transmitter, as rateless codes are both robust and efficient without such information. For simplicity of exposition, binary phase shift keying (BPSK) modulation with coherent detector is utilised. The stream of recovered bits is then fed into the decoder.

B. Transmission Process

The source information is grouped into blocks of p_k packets each composed of b information bits. Rateless coding is employed to communicate the p_k packets to the destination node. Rateless encoding and decoding is briefly explained below and a more thorough explanation can be found in [5].

Encoding: In rateless encoding, first an encoded packet degree d is chosen from a degree distribution $\Omega_1, \cdots, \Omega_{p_k}$, where Ω_i is the probability that degree $d = i$ and $\sum_{i=1}^{p_k} \Omega_i = 1$. Subsequently d information packets are randomly selected and XORed to generate the encoded packet.

Decoding: The decoder can be implemented based on the concept of soft decoding on classic belief propagation technique [14], which is typically faster than Gaussian elimination in practise. Specifically the decoding process consists of one step - identifying encoded packets such that the values of all but one of its neighbouring input symbols are known. This step basically utilizes previously recovered source packets to reduce the number of unknown parameters in all encoding packets that utilize these packets. This process is repeated iteratively until no more first degree encoded packets emerge.

Rateless codes are designed to recover “all” source data with minimum number of codewords and feedback channels are required only to signal successful decoding at receiver.

C. Proposed Adaptive Rateless Coding with Feedback

Rateless codes with a low-complexity decoder are designed such that it can recover the source packets with minimum number of encoded packets; which is slightly more than the number of source packets. In general, the number of recovered source packets depends on the number of received encoded packets and on the degree distribution employed for encoding. In traditional cooperative relay networks, rateless coding is considered as a form of continuous incremental relaying where the *RN* monitors the *SN* virtually indefinitely until it has enough mutual information to decode. Initially, the source data is rateless coded and transmitted. Both *RN* and *DN* listen to this broadcast transmission. As soon as either the *RN* or *DN* has accumulated sufficient mutual information to reliably decode the data, an acknowledgment (ACK) is transmitted. If *SN* receives the ACK from *DN* then the next set of data packets are encoded and transmitted. Whereas if the *RN* decodes first, then *SN* ceases transmission and *RN* starts transmitting the encoded packets (*RN* switches from reception to transmission) till *DN* is able to decode correctly.

In most of the existing rateless coded cooperative systems, *RN* re-encodes the entire source packets and transmits to aid the *DN* in decoding. However the broadcast nature of source transmissions facilitates the reception of some encoded packets at the *DN*, whose quality is dependent on the SD channel. Even though the received encoded packets are not sufficient to decode the entire source packets, partial recovery of source packets is still possible. The intermediate decodability of rateless codes at the *DN*, wherein the received encoded packets are not sufficient for successful decoding of entire source packets, is exploited in this paper.

In the proposed adaptive rateless coded system, before assistance by the *RN*, *DN* utilizes the received encoded packets to recover as many source packets as possible. Then by transmitting a feedback message, this additional information is advertised to the *RN* - the source packets already decoded at the receiver. This information is modelled as a number $n \leq p_k$ that is feedback to the *RN* by the *DN*. The n known input packets are removed from the RN buffer and the remaining $(p_k - n)$ packets are rateless coded and transmitted. This basically mitigates the redundant information processing and transmission by the RN. In rateless coding, how well the source data is mixed at the transmitter plays a crucial role and a significant performance improvement can be obtained by encoding only those packets which are not decoded properly at the receiver. However the required feedback for signalling this is small compared to the information bit size as $p_k \ll b$. Throughout the sequel it is assumed that the decoding is deemed to be successful once the received generator matrix is full rank i.e $Rank(G) = p_k$, where G is the generator matrix at the receiving node. The entire scheme can be summarized as as in Algorithm 1.

```
SN encodes the information and transmits coded packets;
if received generator matrix G is full rank then
    if G is full rank at DN then
        SN codes the next sequence and transmits
    else // G is full rank at RN
        DN transmits a feedback of length p_k to signal
        the packet recovery;
        RN encodes a subset of p_k source packets
        (p_k - n) which are not acknowledged by the DN;
        And transmits;
    end
else
    SN transmits encoded packets
end
This process continues until all packets are received.
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Algorithm 1: Algorithm of the proposed adaptive rateless coding with feedback scheme

The drawback of the proposed scheme is that the DN needs to advertise the identity of the received packets and RN needs to buffer the encoded packets to decode the entire source packets. However the achievable bandwidth efficiency of the proposed scheme supersedes such a detriment. This is validated by the analysis provided in the following sections.

III. Numerical Results

In this section, the average number of packets (n) recoverable at the DN before the RN enters into cooperation mode is analysed. Initially the source data is rateless coded and transmitted. Based on the erasure technique presented in Section II, the received symbols are declared as erased and the corresponding probability in a Rayleigh fading channel can be computed as

$$P_e^{pq}(\gamma) = \Pr[\gamma_q < \gamma_t] = 1 - \exp\left(\frac{1-2^R}{\rho_p \sigma_{pq}^2}\right) \tag{2}$$

where $\rho_p = \varepsilon_s / N_0$, ε_s and N_0 corresponds to source transmit power and noise variance respectively; and R is the code-rate employed.

A. Expected number of transmissions required for relay decoding (T_r)

The RN receives the encoded packets from SN based on the channel quality across the SR link. The expected number of transmissions required for proper decoding at RN across the SR link can be computed as

$$T_r = \sum_{t_r = p_k}^{\infty} t_r \Pr(T_r = t_r) \tag{3}$$

where T_r represents the number of transmissions required for the RN to decode successfully, $\Pr(T_r = t_r)$ is the corresponding probability mass function (PMF) and p_k is the number of source information packets. The required minimum number of transmissions is p_k, as minimum p_k transmissions are required to decode all p_k source packets. The upper bound goes till infinity, owing to the rateless property of the code. t_r basically captures the time when RN is ready for cooperation i.e. when the received generator matrix at the RN is full rank. The binary generator matrix is considered to be equiprobable (the same analysis can be performed for other cases), then the probability of having a generator matrix of rank p_k after receiving i unerased encoded packets can be expressed as

$$P_{rank}^{p_k}(i) = \begin{cases} 0 & \text{if } \; i < p_k \\ \prod_{j=0}^{p_k-1}\left(1-2^{j-i}\right) & \text{if } \; i = p_k \\ \frac{2^{-i}(2^{p_k}-1)}{1-2^{p_k-i}} \prod_{j=0}^{p_k-1}\left(1-2^{j-i+1}\right) & \text{if } \; i > p_k \end{cases} \tag{4}$$

As in an erasure channel, packets are declared as erased based on the link quality. Then the probability of receiving the i^{th} encoded packet across SN and RN during t_r^{th} transmission can be computed as

$$P_i^{t_r}(P_e^{SR}) = \left(1-P_e^{SR}\right)\binom{t_r-1}{i-1}\left(P_e^{SR}\right)^{t_r-i}\left(1-P_e^{SR}\right)^{(i-1)} \tag{5}$$

Using (4) and (5) the PMF of receiving a full rank matrix at RN after t_r transmissions can be computed as

$$\Pr(T_r = t_r) = \sum_{i=p_k}^{t_r} P_{rank}^{p_k}(i) P_i^{t_r}(P_e^{SR}) \tag{6}$$

Then, the expected number of transmissions required for relay decoding can be computed by substituting (6) into (3).

B. Expected number of packets recoverable at DN with T_r source transmissions (n)

Based on the SD link quality, the DN also receives encoded packets during the T_r source transmissions. In the proposed scheme, the source packets that are recoverable from the received encoded packets at DN with these T_r transmissions are advertised to the RN. This can be done with the help of a feedback message from DN to RN of size p_k to signal the number of packets (n) recovered at the DN. This is done only once and just before when the RN starts transmitting the encoded packets. The idea is to reduce the number of redundant packets being processed at the RN and thereby the relay transmissions supplement the mutual information at the DN for proper decoding. The number of packets that are recoverable at the DN is a function of the rank of the received generator matrix. The expected number of packets recovered at the DN with T_r source transmissions can be computed as

$$n = \sum_{\tilde{n}=1}^{p_k} \tilde{n} \Pr(n = \tilde{n}) \tag{7}$$

where n represent the number of packets decoded correctly at the *DN* and $\Pr(n = \tilde{n})$ is the corresponding PMF. For analytical simplicity it is assumed that the number of packets that is decoded correctly is equal to the rank of the received generator matrix. However in reality, it is less than or equal to the rank due to the linear combination of packets being transmitted.

Two cases need to be considered to compute the PMF of the number of packets received at the *DN* with T_r source transmissions. For the first case, the *DN* is able to decode correctly all the source packets i.e. the received generator matrix is full rank with T_r or less number of channel uses. Then the corresponding PMF can be computed as

$$\Pr[Rank(G) = p_k] = \sum_{i=p_k}^{T_r} \sum_{j=p_k}^{i} P_{rank}^{p_k}(j) P_j^i(P_e^{SD}) \tag{8}$$

The first summation is due to the total probability theorem where the number of transmission can vary from the minimum limit (p_k) to the number of transmissions (T_r) required for proper decoding at the *RN*. And second summation represents the reception of j packets from those i source transmissions across the SD link and having a full rank matrix.

For the second case, the receiver is able to decode only a subset of p_k packets ($\tilde{n}$) in T_r source transmissions. The corresponding PMF of decoding $\tilde{n}$ packets can be given as

$$\Pr[Rank(G) = \tilde{n}] = \sum_{j=\tilde{n}}^{T_r} P_{lr}(\tilde{n}, j, p_k) \times \binom{T_r}{j} \times \left(1 - P_e^{SD}\right)^j \left(P_e^{SD}\right)^{T_r - j} \tag{9}$$

where $\tilde{n} \in \{1 \text{ to } (p_k - 1)\}$ and $P_{lr}(\tilde{n}, j, p_k)$ is the PMF of receiving a low rank submatrix $\tilde{n}$ from a higher dimension matrix of order $p_k \times j$. The detailed steps to evaluate P_{lr} are shown in Appendix. In (9), the summation signifies the reception of j packets from T_r source transmissions and the received generator matrix is of rank $\tilde{n}$ to decode the $\tilde{n}$ source packets. Therefore the probability of receiving $\tilde{n}$ packets after T_r source transmissions can be evaluated as

$$\Pr(n = \tilde{n}) = \begin{cases} \Pr[Rank(G) = p_k] & \text{if } \tilde{n} = p_k \\ \Pr[Rank(G) = \tilde{n}] & \text{else} \end{cases} \tag{10}$$

By evaluating (7) using equation (10), the expected number of packets recoverable at the *DN* can be computed.

C. Computation Complexity

For the proposed adaptive rateless coded cooperative system assisted by the feedback channel, the computational complexity can be evaluated as the sum of operations for first removing the known n input symbols from the *RN* buffer. And the subsequent operations for decoding an LT code comprised of $(p_k - n)$ input symbols across the RD channel. From [5], the corresponding decoding complexity of an LT code for decoding $(p_k - n)$ input symbols is $O((p_k - n)\log(p_k - n))$. Thereby the proposed scheme allocates more memory at the *RN* for further processing as well as reduces the decoding complexity by exploiting a single feedback from the *DN*.

IV. Simulation Results & Discussion

This section presents the analytical and empirical results to evaluate the performance of the proposed adaptive rateless coded cooperative network. The nodes analysed in this paper are connected with each other through wireless links where the channels between nodes are modelled as Rayleigh as explained in Section II. The typical pathloss exponent (α) for mobile networks is in the range 3-5, and a value of 3.5 is chosen to model an urban environment. For simplicity, the ACK/NACK feedback is assumed to be perfectly received. Without loss of generality, the transmitter and receiver are supposed to use a deterministic random-generator for fountain coding and the binary generator matrix is considered to be equi-probable. Thus the receiver can easily synchronise with the transmitter. The number of source packets (p_k) is taken as 12 and length of each packet (b) is taken as 256 bits. The received packets are considered as erased based on the erasure technique presented in Section II. The *RN* is considered to be placed half way between the source and destination.

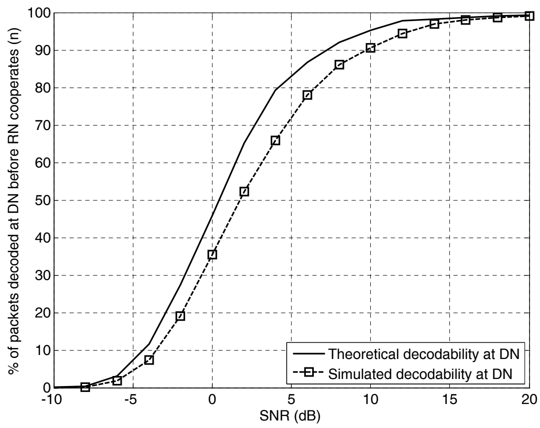

Fig. 1. Variation of percentage of packets decoded at *DN* before *RN* starts cooperating with SNR

Figure 1 illustrates the savings in the number of packets that need to be processed at the *RN*. The numerical results (straight lines) obtained in the former section is compared with the simulated performance (broken lines). As a linear combination of packets are being transmitted, the numerical analysis derived in the former section serves as the upper bound, as the number of packets decodable will be less than or equal to the rank of the received generator matrix. It can be seen that as SD link improves, more source packets are decodable at *DN* owing to availability of more encoded packets. Thereby lesser number of packets needs to be processed at the *RN*. At high SNRs, it is observed that the entire source packets can be recovered from direct transmissions itself and there is no need for cooperation. Thereby the computation complexity and the processing at the *RN* can be reduced significantly by using a single feedback message of size p_k.

The simulation results for the throughput obtained are depicted in Fig. 2. The performance of the proposed scheme is compared with normal rateless coded cooperative systems where the entire source packets are processed at the *RN*.

The results confirm that the proposed scheme provides an improvement in throughput compared with normal relay transmissions at the cost of a single feedback message. The better performance of the proposed scheme can be attributed to the fact that in traditional rateless coded cooperative transmissions, relay encodes those packets which can be recovered at the *DN*. However, the proposed scheme mitigates the redundant packet processing at the *RN* which causes the relay transmissions to supplement the mutual information for recovering the rest of the information at the *DN* and thereby an improvement in throughput.

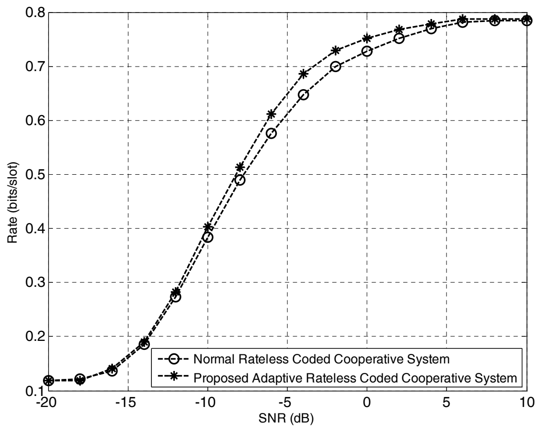

Fig. 2. Throughput performance of the proposed scheme compared to the normal rateless coded cooperative transmissions for varying SNRs

V. Conclusion

In this paper, a novel spectrally efficient adaptive rateless coded cooperative transmission scheme assisted by a feedback is proposed. Adaptation mitigates redundant packet processing at the *RN* by harnessing the reverse feedback channel from *DN* to *RN*. Removal of recovered information from the encoding set ensures that relay transmissions supplements the mutual information to recover the remaining unknown information at the *DN*. Such a scheme thereby reduces the communication complexity and processing at the *RN*. Analytical expressions for computing the expected number of packets recoverable at the receiver has been derived and verified through extensive simulation studies.

Appendix

PMF of receiving a low rank submatrix from a higher dimension matrix

Each new fountain coded packet is associated with an encoding vector over the Galois Field ($GF(2)$) of dimension p_k. Each packet is obtained by the linear combination of p_k source packets as explained in Section II. Consider the received generator matrix to be G, and the p_k source packets can be recovered if p_k linear independent packets are received i.e. the $Rank(G)$ is p_k.

As an all zero column in G does not contain any information, it is assumed that an all zero column is not generated by the encoder. Then the probability of receiving a matrix which is non-zero with one column is $\left(1-\frac{2^0-1}{2^{p_k}}\right)$. With the next encoding set there are 2 columns that are dependent among them, then the probability of having two linear independent columns can be obtained as $1 \times \left(1-\frac{2^1-1}{2^{p_k}}\right)$. By extending this property for R_p received packets, the probability of receiving l independent columns with $l = p_k = R_p$ is given as

$$P_{lr}(l, R_p, p_k) = \prod_{i=0}^{l-1}\left(1-\frac{2^i-1}{2^{p_k}}\right) \tag{11}$$

Then as R_p increases, there are $2^{\left(R_p-l\right)}$ columns that are dependent among them. Then the formula for the probability of receiving l independent columns from R_p columns can be obtained as

$$P_{lr}(l, R_p, p_k) = C \times 2^{\left(R_p-l\right)-p_k} \times \prod_{i=0}^{l-1}\left(1-\left(2^i-1\right)2^{-p_k}\right) \tag{12}$$

where C is a constant. By evaluation, the constant C can be approximated as a geometric series with a common ratio of 2 and initial term to be $2^{\left(R_p-l\right)}$. The number of terms between initial and final terms can be computed as

$$k = \left(R_p-l\right) \times (l-1) + 1 \tag{13}$$

Thus the constant C can be computed as

$$C \approx 2^{\left(R_p-l\right)} \times \left(2^k-1\right) \tag{14}$$

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