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Layered multimedia broadcast using rateless codes with progressive recovery over cooperative MIMO

Zhao Chen^{1,3}, Liuguo Yin^{2,3}, Mai Xu^{1*} and Jianhua Lu¹

Abstract

This article proposes a novel scheme, based on unequal error protected rateless codes, for broadcasting layered multimedia over cooperative multiple-input multiple-output (MIMO). By taking advantage of both cooperation and broadcasting, we first present a two-phase cooperative MIMO broadcast scheme, which exploits distributed diversity in point-to-multipoint communication scenarios. Then, to enhance layered multimedia transmission with progressive recovery, the progressive rateless codes (PRC) is proposed to recover the layered data according to their importance at the expected received ratio of all output packets. Thus, receivers with different packet loss rate can achieve an adaptive recovery of the layered multimedia progressively. Furthermore, based on PRC, a distortion-based layered multimedia broadcast problem is formulated, which optimizes source bitrate of the layered multimedia and coding rates of PRC to improve the quality of experience (QoE) of the multimedia delivery over all receivers. The performance analysis including both analytical results and simulation experiments of Motion-JPEG 2000 broadcast to receivers verifies the superiority of the cooperative broadcast and efficiency of the QoE-driven optimization algorithm.

Keywords: Multimedia broadcast, Cooperative MIMO, Unequal error protection (UEP), Rateless codes

Introduction

Broadcast, as the nature of the wireless medium, offers the promise of overcoming the bandwidth and energy limitation by using one channel to transmit source data to all destinations simultaneously within transmission range [1]. In many communication scenarios, broadcast is often desired and required. Besides, multiple-input multiple-output (MIMO) techniques can be utilized to provide high-rate high-quality communication services in broadcasting.

Diversity, as an efficient way to mitigating the fading arising from multipath propagation, has successfully been extended to relay channels by using distributed relay terminals, referred to as *cooperative diversity* [2]. In contrast with such conventional systems which have only one single destination node and a relatively small number of potential relay nodes, *cooperative broadcast (CB)* [3] has been proposed for multiple destinations to

receive one message in the network, where destinations can be switched between receiving and relay modes. In the receiving mode, each destination node tries to accumulate signal energy from other nodes and decodes the symbol by sufficient signal-to-noise ratio (SNR), and then it switches to the relay mode and retransmit the same symbol. Recently, CB has been shown to provide spatial diversity and achieve better bit error rate (BER) performance [3-5]. In [6], the energy efficiency of cooperative transmissions in a broadcast network is analyzed.

In a broadcast network, multimedia contents play a key role and mainly include the forms of text, audio, image, and video. For example, multimedia broadcast multicast service (MBMS) [7] has been proposed as a standard of 3GPP for providing multimedia service to users via broadcast in 3G UMTS cellular networks. As the state-of-the-art multimedia source compression standard, scalable video coding (SVC) extension of the H.264/AVC Standard [8] and Joint Photographic Experts Group 2000 (JPEG 2000) [9] are both pervasive in broadcast network since they are capable of providing efficient video and image

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information content to the users. One of the most attractions of these standards is that it is able to produce progressive recovery of videos or images by fidelity or resolution, referred to as *layered multimedia*.

In a multimedia broadcast network, layered multimedia transmissions can be achieved once taking advantage of the progressive layers of JPEG 2000 or SVC. However, considering the different importance between layers of the layered multimedia, unequal error protection (UEP) strategies may be applied to protect the layered multimedia from packet losses, in order to maximize the quality of experience (QoE) in the network. UEP property can be realized by various means, one practical technique is to give different redundant information to layers by forward error correction (FEC) codes. Rateless code [10], also known as fountain codes, is one of such FEC codes with capacity-achieving performance. In a rateless code, the original source packets can accurately be recovered from any subset of the encoding packets with the size equal to or only slightly larger than the number of source packets, which means that the redundant packets are maximally utilized. In fact, many studies have been done in the field of applying rateless codes with UEP property for scalable image/video streaming [11-14].

In this article, for layered multimedia broadcast over cooperative MIMO, progressive rateless codes (PRC) is proposed to apply UEP to the broadcast of Motion-JPEG 2000 encoding layered video stream,^a with high efficiency and low complexity. Beyond the proposed PRC, unequal error protected broadcast scheme is established with the purpose that different users can receive broadcasting images/videos with different decoding layers according to their channel qualities. Meanwhile, a distortion-based model of the broadcast scheme using

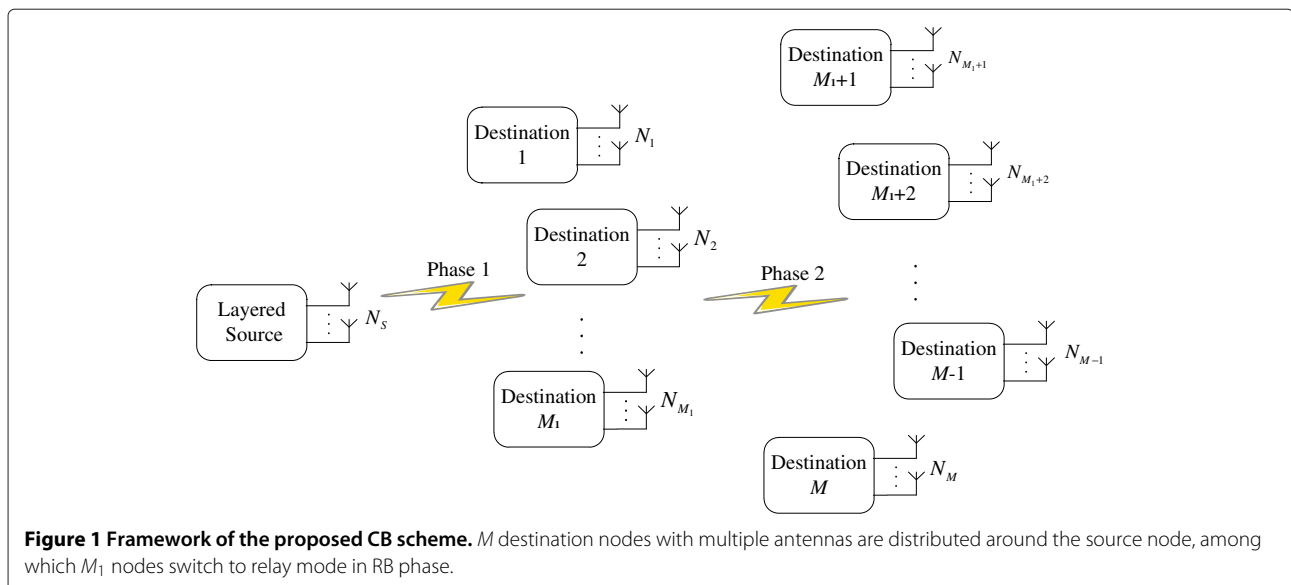
pulse signal-to-noise ratio (PSNR) evaluation is formulated and optimized to improve the QoE of the broadcast system. The framework of such CB scheme can be seen in Figure 1, where a two-phase cooperative broadcasting is illustrated.

The rest of this article is organized as follows. In “Background” section, we introduce some basic concepts of CB and rateless codes. “Rateless codes with progressive recovery” section presents a detailed description and performance analysis of the proposed PRC for layered multimedia transmission. A QoE-driven multimedia broadcast scheme is introduced in “QoE-driven layered multimedia broadcast” section. “Simulation results” section shows the experimental results of the QoE-driven scheme in both CB and traditional broadcast (TB). Finally, we conclude the article in “Conclusions” section.

Background

Cooperative broadcast

In general, due to the medium of wireless communication, the transmitted signals are heard not only by their intended receivers, but also by other neighboring nodes. In conventional point-to-point communications, this may be harmful for other unintended receivers. But it will be beneficial in broadcast scenarios when one message is needed to be transmitted to multiple destinations. In TB schemes, the source node will provide a best-effort transmission service under energy constraint. Each message is broadcasted only once. Some destination nodes with lower receiving SNR will be unable to receive the message, which leads to a higher packet loss probability. If the receiving SNR is below a critical value, the destination node cannot even collect one whole packet. Such node



is called to be out of the coverage of the source node. To broaden the coverage of the source node and improve the SNR of bad receivers, it is better to utilize multi-hop to relay the message to distributed receivers, where cooperative diversity can be exploited [3,4].

We consider a CB scheme over quasi-static flat-fading channels as shown in Figure 1. There are one source node and M destination receiving nodes distributed in the scheme. All nodes are equipped with multiple antennas. We denote N_S as the number of antennas of the source node, N_m as the number of antennas of destination node m , where $1 \leq m \leq M$. Following [4], assume a two-phase decode-and-forward (DF) CB protocol. The two phases are both synchronized in alternate time slots, which are referred to as *source broadcast (SB)* and *relay broadcast (RB)*, respectively. In the first time slot (SB phase), the source node broadcasts one message to all destinations. Suppose that there are M_1 destinations that have successfully decoded the message. Next in the second time slot (RB phase), the source node keeps quiet and the M_1 decoded destination nodes switch to relay mode and start to broadcast the decoded message, while the other $M - M_1$ destination nodes continue receiving, trying to collect the signal energy from all relay nodes. Then it turns to SB phase again in the next time slot.

In RB phase, the relay nodes can form a virtual antenna array. At this point, piloted-assisted channel estimation is feasible at each of the remaining undecoded destinations. Practically, before relaying there should be some knowledge exchange to between the relay nodes. Assuming that all nodes are subject to a half-duplex constraint, so channel state information (CSI) are only available for receiving nodes but not for transmitters. Since CSI is not available at the transmitters, equal power is allocated to every transmit antenna across the virtual antenna array.

For each destination node in the broadcast scheme, the channel quality varies with different pathloss and channel fading. Moreover, the channel quality of some destinations is severely degraded due to deep fading. Those nodes are nearly out of the coverage of the source node. Taking advantage of the virtual multi-antenna array, we can avoid deep fading and notably improve the reliability of the broadcast by exploiting the diversity gain. If we consider to select a group of transmit antennas from all antennas in the decoded nodes to form the virtual array, the maximum diversity gain can be achieved. However, for simplicity we take all the decoded destinations to be relay node without selection as in our model.

For the purpose of preparing for the post-processing in application layer in the next sections, we measure the channel quality of each destination node in the way of packet loss ratio (PLR). The PLR can be derived from SNR with the connection of BER, which is beyond the scope of

this article. With the diversity gain of virtual array, the PLR of destination nodes are all increased, especially for those whose PLRs are critically low. Thus, the coverage of the source node is considerably expanded. Note that though CB can improve the reliability of broadcasting efficiently, the system throughput is halved due to the relay phase. So, a selection strategy is required to obtain better performance between cooperation and non-cooperation, which will be discussed in "Simulation results" section.

Rateless codes

Luby transform (LT) codes [15] or Raptor codes [16], as two state-of-the-art techniques of rateless codes, have been proved to be efficient FEC solutions for erasure channels.^b These codes are universal for different scenarios on packet transmission level regardless of channel packet loss patterns. So, rateless codes are becoming increasingly popular in broadcast network. For example, Raptor codes, with nearly linear encoding/decoding complexity, have been accepted for the application layer FEC scheme in current communication standards, such as 3GPP MBMS [7] and DVB-H [17].

LT codes are the first practical rateless code. Assume that we have k source symbols^c to be transmitted. Let $\Omega(x) = \sum_{i=1}^k \Omega_i x^i$ represent a degree distribution, where Ω_i stands for the probability of degree i and satisfies $\sum_{i=1}^k \Omega_i = 1$. The procedure of generating a encoding symbol is as follows.

1. Select an encoding degree d with distribution $\Omega(x)$.
2. Choose d input symbols randomly and uniformly in k source symbols as neighbors of the encoding symbol.
3. Perform bitwise XOR operation on the d chosen symbols to generate the encoding symbol.

After the above procedure, the encoding symbol is transmitted to the receiver. If $d = 1$, the encoding symbol is just a duplication of the unique input symbol. This procedure will be executed repeatedly and a potentially infinite encoding symbol stream can be generated until enough encoding symbols are collected at the receiver to recover all source symbols.

At the receiver, both belief propagation process (BP) [15] and maximum likelihood decoding (ML) [7] can be applied to the decoding of LT codes. The procedure of BP process is as follows.

1. Initial step: search for receiving symbols with degree one and release them to recover their unique neighbor input symbols to a buffer, called the *ripple*.
2. Process every input symbol in the *ripple* as follows until the *ripple* becomes empty.
 - (a) Remove the input symbol from receiving symbols as a neighbor.

- (b) Release such receiving symbols subsequently with exactly one remaining neighbor and recover their neighbors to the ripple.

BP process fails if at least one source symbol remaining unrecovered in the end. The key point of successful decoding is the perfect design of the degree distribution. Fortunately, it was proved in [15] that such distribution exists and all source symbols can be recovered by any $(1 + \epsilon)k$ encoding symbols. ϵ is the decoding overhead, it has achieved capacity-approaching behavior with very low overhead when $k \rightarrow \infty, \epsilon \rightarrow 0$.

ML decoding, also known as full rank decoding, is executed by solving a set of linear equations in \mathbb{F}_2^k , since each encoding symbol is a linear combination of source symbols. It will be successful if the set of equations is full rank. Compared with BP process in [18], ML decoding has lower decoding overhead but higher decoding complexity.

Raptor codes [16], as an extension of LT codes, have been proposed with linear time encoding and decoding using a pre-coder of low-density parity-check codes. Our UEP approach with progressive recovery will follow Raptor codes with a modified encoding structure, which will be described in the following section.

Rateless codes with progressive recovery

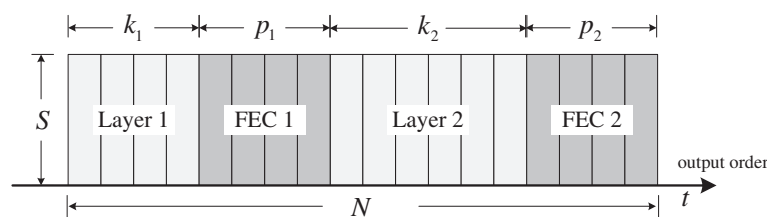
Related work

With applying rateless codes with UEP, various layered delivery techniques have been studied, which can be divided into three groups.

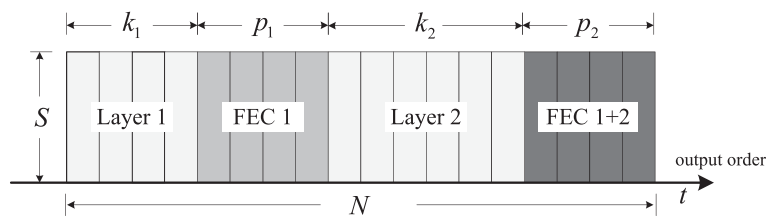
Rahnavard et al. [14], first of all, presented a distribution-based approach. They introduced UEP at the LT encoding stage and designed a non-uniformly degree distribution such that lower layer symbols can be selected with higher probability. With achieving unequal recovery of different layers, the altered distribution weakens the code performance and results in a larger overhead.

Another group of UEP designs are pre-coding-based approaches [19,20]. Without making modifications to the original rateless code structure, firstly layer packets are pre-coded with different code rates proportionally according to their importance, where lower layer packets are assigned to lower pre-coding rate. Then pre-coded packets are passed to a rateless encoder. Since the intermediate performance of rateless code is poor, the recovery of lower layers suffers.

The third and typical one is the redundancy-based strategy [11,12,21]. Stream layers are encoded by different rateless encoders and given redundant symbols proportionally with their importance. There are two types of such strategy. As shown in Figure 2, suppose that there are two stream layers to be delivered as *Layer 2* is dependent on *Layer 1*. In [21], the separate FEC (SP-FEC) protects layered data independently, while the layer-aware FEC (LA-FEC) extends protection following the dependency between stream layers, where *Layer 1* is not only covered by *FEC 1*, but also covered by *FEC 1+2*, together with *Layer 2*. The LA-FEC improves the recovery of *Layer 1* at the expense of *Layer 2*, since the encoding structure of *FEC 1+2* is slightly changed from original rateless codes. Next, we will have a detailed description of



(a) A block of a two-layer SP-FEC with N output symbols.



(b) A block of a two-layer LA-FEC with N output symbols.

Figure 2 Structure of two types of separate unequal protection strategy, whose N output symbols are transmitted from left to right sequentially.

our approach and compare it with the redundancy-based approaches.

Design and implementation

In this section, we propose the PRC to enhance original rateless codes with UEP capability. In our approach, to guarantee the optimized recovery performance, we alter the encoding structure with maintaining the parameters of rateless code, e.g., degree distribution. Meanwhile, with the efficient recovery of rateless codes, the dependency between layers has been satisfied to come to a progressive recovery of the layered multimedia.

We consider a layered multimedia data stream to be transmitted over an erasure channel. Assume that an L -layer video stream is partitioned into several source blocks with the size of K symbols, where the importance of symbols decreasing from *Layer 1* to *Layer L*. Let k_i be the number of source symbols of *Layer i*, so that $K = \sum_{i=1}^L k_i$. Let S be the symbol length in bytes, thus each layer has $k_i \cdot S$ bytes and the total length of the block will be $K \cdot S = \sum_{i=1}^L k_i \cdot S$ bytes. Note that $\frac{k_i}{K}$ is a constant for *Layer i* in a certain layered stream as the block size K changes.

Given total broadcasting bandwidth, the overall coding rate $\gamma = \frac{K}{N}$ is fixed for all possible K , where N is the number of output symbols for each source block, i.e., output block size, to protect the layered data stream from packet losses. Thus, the total length of redundant symbols is $(N - K) \cdot S$ bytes. Based on these conditions, our PRC approach will generate encoding symbols in a parallel way.

Before rateless encoding, all L layers are reshaped with symbol lengths of $\{s_1, s_2, \dots, s_L\}$ bytes, respectively, ensuring $S = \sum_{i=1}^L s_i$. Then the number of *Reshaped Symbols* in *Layer i* becomes $k_i^* = \frac{k_i \cdot S}{s_i}$. Each reshaped layer is

passed through a rateless encoder to generate N reshaped encoding symbols, where an output symbol is formed by packing L reshaped encoding symbols, one from each encoding layer. So, there will be N output symbols with each symbol packing encoding data from all layers. As shown in Figure 3, a two-layer PRC layered delivery is illustrated, where an output symbol is generated by combining two reshaped encoding symbols.

At the decoder, assume that R output symbols are received, of course $R \leq N$ due to packet losses. The received symbols are first unpacked to separate reshaped symbols of each layer, which are then passed to L different rateless decoders, respectively. Lastly the message blocks are recovered layer-by-layer at the decoders. The decoding method adopted in our approach is ML decoding.

Recovery performance analysis

In this section, we will make a combinational analysis of recovery probability of PRC, in comparison with SP-FEC and LA-FEC in Figure 2. To make a fair comparison, for *Layer i*, we have redundant data of the equal length in all approaches, i.e., $p_i \cdot S = (N - k_i^*) \cdot s_i$, where p_i is the number of redundant symbols. And then we have the number of output symbols $n_i = k_i + p_i$, the coding rate $r_i = \frac{k_i}{n_i}$. Thus, we obtain

$$k_i^* = \frac{k_i \cdot S}{s_i} = \frac{k_i \cdot S}{n_i \cdot S/N} = N \cdot \frac{k_i}{n_i} = N \cdot r_i \quad (1)$$

which shows that the coding rate of reshaped *Layer i* is also r_i . In SP-FEC, let $\eta_i = \frac{n_i}{N}$ be the output ratio of *Layer i*, which will be a constant once r_i is determined.

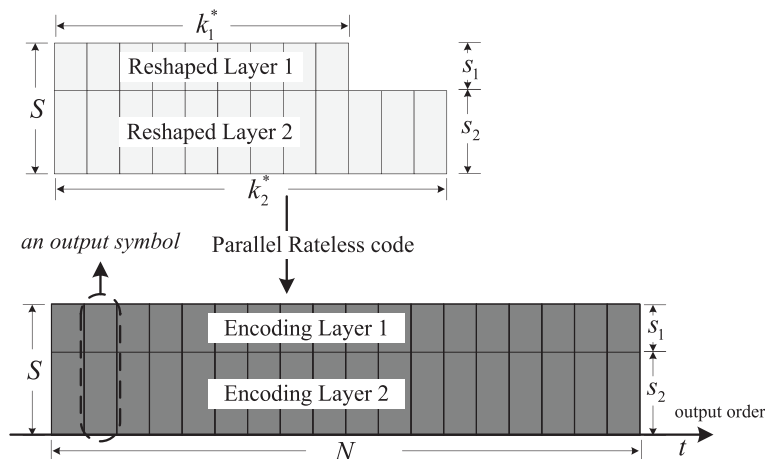


Figure 3 Encoding procedure of a block of a two-layer PRC, whose N output symbols are transmitted from left to right sequentially.

Let $\Pr_i(R)$ be the recovery probability of *Layer i* in PRC. Without loss of generality, we consider the ideal recovery of rateless codes, i.e., k_i^* source symbols can be recovered as soon as $R \geq k_i^*$ encoding symbols are received.^d Then, we have

$$\Pr_i(R) = \begin{cases} 1, & R \geq k_i^* \\ 0, & R < k_i^* \end{cases} \quad (2)$$

which indicates that for *Layer i*, it can be recovered from at most $N - k_i^*$ symbol losses with probability 1. Therefore, to recover layered data stream progressively from *Layer 1* to *Layer L*, we can make $k_1^* \leq k_2^* \leq \dots \leq k_L^*$. From Equation (1), we know that it can be also represented by $r_1 \leq r_2 \leq \dots \leq r_L$. So we can protect *Layer i* by assigning suitable r_i , expecting to recover it after receiving r_i ratio of output symbols.

Following the results in [21], we only investigate SP-FEC without considering dependency for simplicity. The analysis of LA-FEC will be similar to that and when N grows large, there will be no difference between the two approaches.

In SP-FEC, let $\Pr'_i(R)$ be the recovery probability of *Layer i*. With the ideal recovery assumption of rateless codes, it can be recovered by at least k_i out of n_i output symbols from *Layer i*.

- For $R < k_i$, $\Pr'_i(R) = 0$.

- For $k_i \leq R < N - (n_i - k_i)$,

$$\Pr'_i(R) = \sum_{x=k_i}^{\min\{R, n_i\}} \frac{\binom{n_i}{x} \binom{N-n_i}{R-x}}{\binom{N}{R}} = \sum_{x=k_i}^{\min\{R, n_i\}} \frac{\binom{R}{x} \binom{N-R}{n_i-x}}{\binom{N}{n_i}} \quad (3)$$

- For $R \geq N - (n_i - k_i)$, $\Pr'_i(R) = 1$.

It is clear that $\Pr'_i(R)$ is the tail probability of a hypergeometric distribution with parameters of $X \sim \mathcal{H}(R, n_i, N)$, i.e.,

$$\Pr'_i(R) = P(X \geq k_i | R, n_i, N) \quad (4)$$

Let $r = R/N$ be the received ratio of all output symbols. In Figure 4, several curves of recovery probability are shown, normalized by the received ratio. Note that in practice typically multimedia codecs work well under a packet loss rate of no more than 10^{-4} , so the recovery performance can be measured by the received ratio where the recovery probability goes above $1 - 10^{-4}$, which is called *Successful Received Ratio (SRR)*. From Figure 4, we can see that the SRR of PRC is much smaller than SP-FEC for both *Layers 1* and 2.

To find the relationship between the two approaches, we will show some properties of $\Pr'_i(R)$ as the following lemmas.

Lemma 1. $\Pr'_i(R)$ is an non-decreasing function of the number of received symbols R .

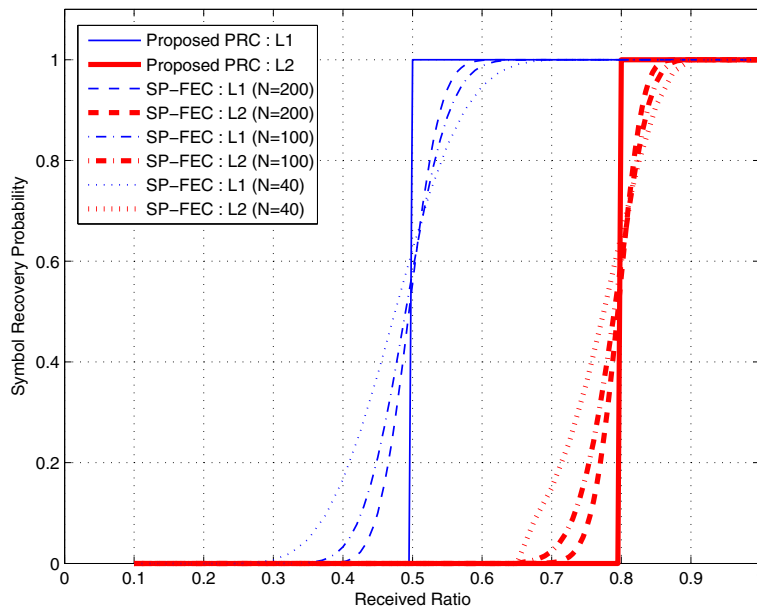


Figure 4 Recovery probability of the two layers with different received ratio of output symbols using SP-FEC and the proposed PRC. Note that L1 and L2 denote *Layer 1* and *Layer 2*, respectively. The coding rate for the two layers are $r_1 = 1/2$ and $r_2 = 4/5$.

Proof. From Equation (3), if $R < k_i$ or $R \geq N - (n_i - k_i)$, $\text{Pr}'_i(R)$ is a constant. Otherwise we can obtain

$$\begin{aligned} \text{Pr}'_i(R+1) &= \frac{\sum_{x=k_i}^{\min\{R, n_i\}} \binom{R}{x} \binom{N-R-1}{n_i-x} + \sum_{x=k_i-1}^{\min\{R, n_i-1\}} \binom{R}{x} \binom{N-R-1}{n_i-x-1}}{\binom{N}{n_i}} \\ &= \text{Pr}'_i(R) + \frac{\binom{R}{k_i-1} \binom{N-R-1}{n_i-k_i}}{\binom{N}{n_i}} \end{aligned}$$

which shows that $\text{Pr}'_i(R)$ is increasing strictly when $k_i \leq R < N - (n_i - k_i)$. \square

Lemma 1 shows the recovery probability of each *Layer i* increases with the number of the received symbols, which is in accordance with our intuition. Furthermore, given a fixed received ratio of output symbols, we have Lemma 2 when the output block size N increases with a fixed overall coding rate γ .

Lemma 2. *If r is any constant and $r > r_i$, $\text{Pr}'_i(N \cdot r)$ is an increasing function of the number of output symbols N .*

Proof. The mean μ and the variance σ^2 of X are given by

$$\mu = R \cdot \frac{n_i}{N}, \sigma^2 = N \cdot r(1-r) \frac{n_i}{N} \left(1 - \frac{n_i}{N}\right) \quad (5)$$

If N is large enough, X can be approximated by a normal distribution, if the following conditions can be satisfied

1. $\frac{R}{N} \rightarrow r$ is a constant.
2. $\frac{n_i}{N} \rightarrow \eta_i$ is a constant.

Then X approaches to $\mathcal{N}(\mu, \sigma^2)$, where

$$P(X = x|R, n_i, N) \approx \frac{1}{\sqrt{2\pi}\sigma} e^{-\frac{(x-\mu)^2}{2\sigma^2}} \quad (6)$$

Since $k_i = n_i r_i = N \eta_i r_i$, we can derive with Equation (4),

$$\text{Pr}'_i(Nr) \approx 1 - \Phi\left(\frac{k_i - \mu}{\sigma}\right) = \Phi\left(\frac{(r - r_i)\sqrt{N}}{\sqrt{r(1-r)(1-\eta_i)}}\right) \quad (7)$$

where $\Phi(x)$ is the cumulative distribution function of standard normal. With the monotonicity property of $\Phi(x)$, it is straightforward to show that $\text{Pr}'_i(Nr)$ increases with N if $r > r_i$. \square

Lemma 2 shows that the recovery probability of each *Layer i* will increase with N when a certain ratio of symbols are received, as long as the ratio is more than r_i . If the PLR is no more than $1 - r_i$, we can improve the performance by assigning a larger block size K .

Lemma 3. *If r is any constant and $r > r_i$, when $N \rightarrow \infty$,*

$$\lim_{N \rightarrow \infty} \text{Pr}'_i(N \cdot r) = \text{Pr}_i(N \cdot r) \quad (8)$$

Proof. First, we will show that $\text{Pr}'_i(N \cdot r) \rightarrow 1$. This is quite easy to be shown from Equation (7), since $\Phi(x) \rightarrow 1$ when x goes to infinity.

Recall Equation (1), $R > Nr_i = k_i^*$ when $r > r_i$. So we have $\text{Pr}_i(N \cdot r) = 1$, which concludes the assertion. \square

Lemma 3 shows that the asymptotic recovering probability of SP-FEC is equal to PRC, which means PRC seems to be optimal for SP-FEC. We also notice that the SRR of SP-FEC approaches to that of PRC when N grows. In other words, for moderate output block size N , PRC will theoretically outperforms SP-FEC with lower overhead.

In Figures 5 and 6, we have performed two cases of numerical simulations on ($N = 1000, K = 650$) and ($N = 500, K = 325$) to evaluate the two-layer progressive recovery of PRC compared with SP-FEC and LA-FEC [21]. In our simulations, we apply the Raptor codes specified in [7] and each packet contains one symbol. It can be seen in the figures that both *Layers 1* and *2* of the PRC are recovered progressively around 50 and 80% as expected with a very low overhead 2% ($N = 1000$) to 4% ($N = 500$).

Figure 5 shows the PLR performance of *Layer 1* of all the three approaches with the received ratio near 50%. It is clear that the PRC outperforms both SP-FEC and LA-FEC with reducing more than 5% ($N = 1000$) to 7% ($N = 500$) received packets below the PLR of 10^{-4} . When the output block size N increases, the PLR at the same received ratio decreases and the gap between PRC and the other two approaches becomes closer, which meet our conclusions in Lemmas 2 and 3. In addition, it also indicates the advantage of PRC in PLR performance for *Layer 2* in Figure 6.

QoE-driven layered multimedia broadcast

In this section, we consider to apply UEP to delivering the layered multimedia stream based on PRC, aiming at recovering the stream adaptively at receivers according to their channel conditions. Moreover, by optimizing source bitrate of the layered stream and coding rates of PRC, we have proposed a broadcast scheme that is QoE-driven to maximize the average quality of received multimedia over all receivers.

System model

Since the worst receivers bottleneck the performance of broadcast, we try to serve the receivers adaptively. In our model, we consider a broadcast system of one source node with layered multimedia to be transmitted and D receiving nodes distributed around the source node. For convenience, we label the receiving nodes' set

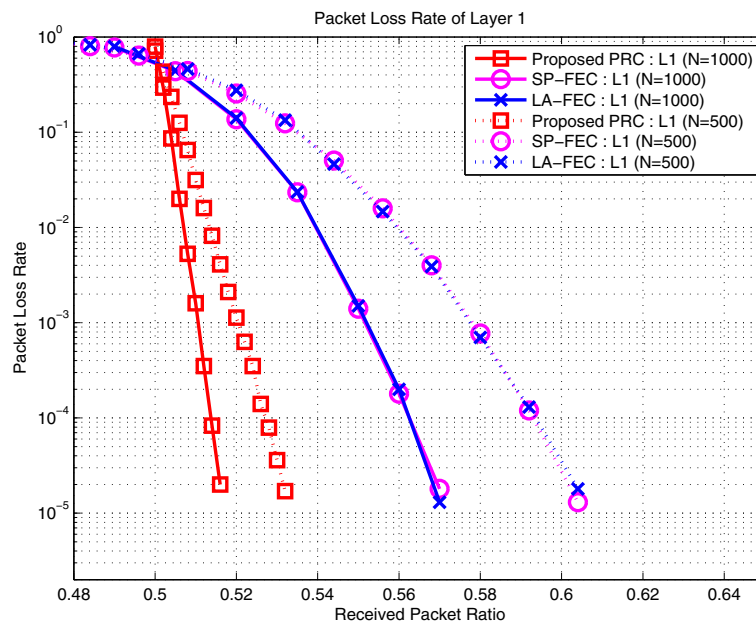


Figure 5 Packet loss rate of *Layer 1* of the proposed PRC versus other UEP approaches after receiving a certain ratio of output packets. Both $N = 1000$ and $N = 500$ are illustrated.

as $\mathcal{D} = \{1, 2, \dots, D\}$. Basically, the transmission bandwidth of source node is limited, resulting in a constrained broadcast bitrate of B . Due to the effect of pathloss and multipath channel fading propagation environment, channel conditions of receiving nodes differ from each other. We assume a slow-fading channel during broadcasting,

leading to a fixed PLR of e_d at each receiving node $d \in \mathcal{D}$. Particularly, the PLRs are sorted in ascending order, i.e., $e_1 \leq e_2 \leq \dots \leq e_D$. It is evident that the channel qualities of receiving nodes are decreasing from 1 to D . A detailed block diagram of the two types of nodes are demonstrated in Figure 7.

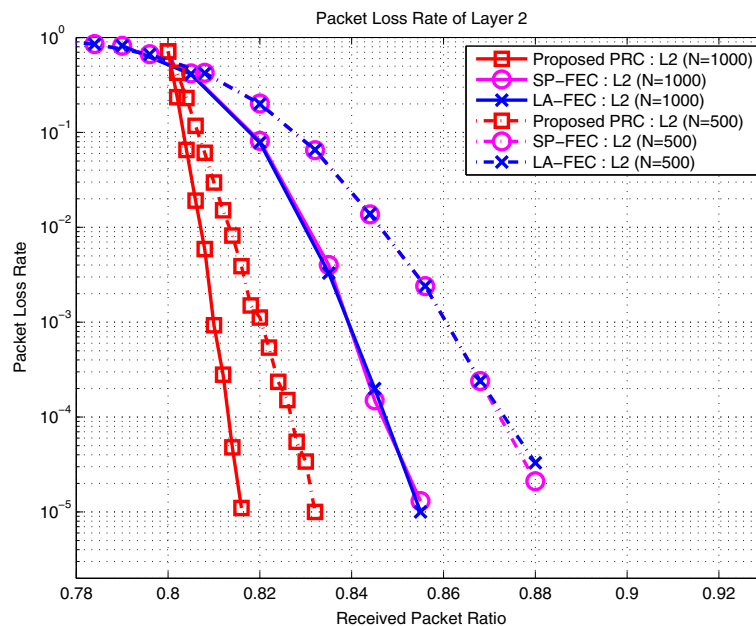
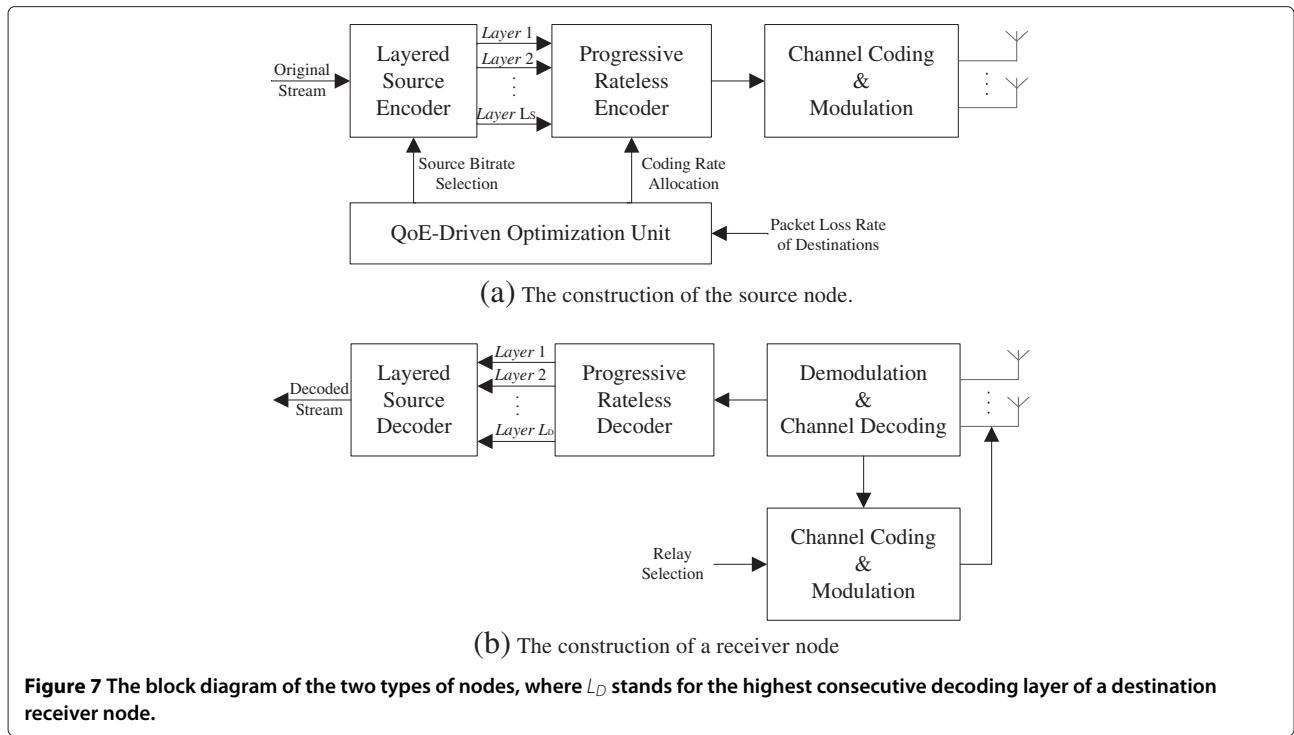


Figure 6 Packet loss rate of *Layer 2* of the proposed PRC versus other UEP approaches after receiving a certain ratio of output packets. Both $N = 1000$ and $N = 500$ are illustrated.



Before the source node starts to transmit, the original multimedia sequence is compressed to L scalable layers, in which the bitrate for the i th layer is given by w_i . Considering the constrained bandwidth B , assume that the layered source is truncated at $Layer L_S$. Then an unequal error protected FEC codes $\mathbf{r} = [r_1, r_2, \dots, r_{L_S}]$ is applied to protect the truncated layered source from packet losses, where each $Layer i$ is encoded by the corresponding coding rate r_i . Hence, we have the total bitrate to be transmitted,

$$B_T(\mathbf{r}) = \sum_{i=1}^{L_S} \frac{w_i}{r_i} \quad (9)$$

where $B_T(\mathbf{r}) \leq B$ need to be satisfied.

Reconstruction of the layered multimedia at the receiver is based on the consecutive correctly recovered layers until the first unrecovered one. For a destination node $d \in \mathcal{D}$, we denote $CP_i(e_d, \mathbf{r})$ as the recovering probability of the first i consecutive layers,

$$CP_i(e_d, \mathbf{r}) = \begin{cases} 1 - p_1(e_d, \mathbf{r}), & i = 0 \\ (1 - p_{i+1}(e_d, \mathbf{r})) \prod_{j=1}^i p_j(e_d, \mathbf{r}), & 1 \leq i \leq L_S - 1 \\ \prod_{j=1}^i p_j(e_d, \mathbf{r}), & i = L_S \end{cases} \quad (10)$$

where $p_i(e_d, \mathbf{r})$ is the independent recovering probability of the i th layer, and it can be obtained from Equation (2) (PRC) or Equation (3) (SP-FEC). In Figure 7, the first L_D consecutive layers are recovered at the receiver node.

In order to assess the QoE of the layered broadcast system, we now consider a distortion-based analysis of the reconstruction quality at each receiving node. In case of the expected PSNR metric for video streams at node d , we have

$$PSNR(d) = \sum_{i=0}^{L_S} PSNR_i \cdot CP_i(e_d, \mathbf{r}) \quad (11)$$

where $PSNR_i$ stands for the PSNR of the first i layers and $PSNR_0 = 0$. It is worth noting that $PSNR_i$ relies on the original video sequence and the scalable encoder, which can be computed before PRC.

The target of the QoE-driven scheme is to minimize the expected average distortion of the reconstructed multimedia over all receiving nodes in the broadcast system, i.e., to maximize the average PSNR. Based on the analysis above, we can formulate the QoE-driven scheme with the truncated layer L_S and the coding rates of UEP \mathbf{r} as follows,

$$\max_{L_S, \mathbf{r}} \overline{PSNR} = \frac{1}{D} \sum_{d=1}^D \sum_{i=0}^{L_S} PSNR_i \cdot CP_i(e_d, \mathbf{r}) \quad (12)$$

subject to

$$\begin{cases} \sum_{i=0}^{L_S} \frac{w_i}{r_i} \leq B \\ L_S \leq L \end{cases} \quad (13)$$

When a user becomes a receiving node, it measures the channel quality and feeds back the PLR to the source node. Once given the layered multimedia to be transmitted, the

conditions including PSNR_i and w_i for i from 1 to L are determined. Then the source node can derive optimized parameters of L_S and \mathbf{r} with applying the proposed PRC to the layered stream.

Optimization of QoE-driven PRC

In this section, we will employ PRC to design the QoE-driven broadcast scheme. Following Equation (2), we can derive

$$p_i(e_d, \mathbf{r}) = \Pr_i(N(1 - e_d)) = u(1 - r_i - e_d) \quad (14)$$

where $u(\cdot)$ denotes the unit step function. Substitute Equation (14) for $p_i(e_d, \mathbf{r})$ in Equation (10), we have

$$\text{CP}_i(e_d, \mathbf{r}) = \begin{cases} 1, & r_i \leq 1 - e_d < r_{i+1} \\ 0, & \text{otherwise} \end{cases} \quad (15)$$

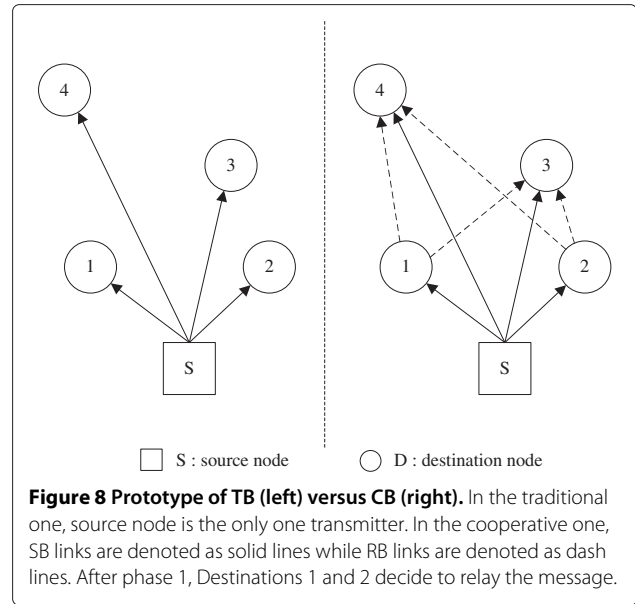
which indicates that the expected consecutive recovering layers are i for the receiving node whose received ratio is in $[r_i, r_{i+1})$. So $\sum_{d=1}^D \text{CP}_i(e_d, \mathbf{r})$ will be the expected number of receiving nodes in \mathcal{D} which have recovered i consecutive layers, i.e., the number of nodes whose received ratio is in $[r_i, r_{i+1})$, we denote it as $G(r_i, r_{i+1})$. Thus, we can rewrite the QoE-driven problem as

$$\max_{L_S, \mathbf{r}} \overline{\text{PSNR}} = \frac{1}{D} \sum_{i=0}^{L_S} \text{PSNR}_i \cdot G(r_i, r_{i+1}) \quad (16)$$

If the PLR of receiving nodes follows an uniformly distribution, i.e., $G(r_i, r_{i+1}) = D \cdot (r_i - r_{i+1})$, the problem will be solved via classical discrete Lagrangian multiplier method. Unfortunately, in fact the receiving nodes are randomly distributed in the broadcast system, so we will concentrate on a general algorithm to solve the problem.

Since the coding rate of rateless codes can flexibly be adjusted in $(0, 1]$, we choose r_i from the set of potential coding rates by the received ratio $\mathcal{R} = \{1 - e_1, 1 - e_2, \dots, 1 - e_D\}$. As the number of layers is limited by layered source encoder and transmission bandwidth, we can try some available value of selected source layers L_S to have the optimal choice. Once given L_S , we have the heuristic algorithm as follows to determine the appropriate \mathbf{r} .

- Step 1: Initialize coding rates \mathbf{r}_0 to recover all L_S layers consecutively at node D with the highest PLR, i.e., for all $1 \leq i \leq L_S$, $r_i = 1 - e_D$. Compute the average PSNR under \mathbf{r}_0 as $\overline{\text{PSNR}}_0$. Set $j = 1$.
- Step 2: Calculate the transmission bandwidth B_T in Equation (9). If $B_T > B$, set $i = L_S$. Otherwise $\mathbf{r} = \mathbf{r}_{j-1}$ and the algorithm is terminated.
- Step 3: Let d be the index of coding rate in \mathcal{R} which equals r_i , such that $r_i = 1 - e_d$. If $i = L_S$ or if $i < L_S$ and $r_i < r_{i+1}$, substitute $r_i = 1 - e_{d-1}$ in \mathbf{r}_{j-1} and



denote it as \mathbf{r}_i^* . Compute the average PSNR under \mathbf{r}_i^* as $\overline{\text{PSNR}}_i^*$ and record the degradation of the average PSNR as $\Delta_i = \overline{\text{PSNR}}_{j-1} - \overline{\text{PSNR}}_i^*$.

- Step 4: If $i = 1$ or if $i < L_S$ and $r_i \geq r_{i+1}$, go to step 5. Otherwise set $i = i - 1$ and go to step 3.
- Step 5: Let $i_j = \arg \min \Delta_i$. Update $\mathbf{r}_j = \mathbf{r}_{i_j}^*$ and $\overline{\text{PSNR}}_j = \overline{\text{PSNR}}_{i_j}^*$. Set $j = j + 1$ and go to step 2.

In step 5 of the algorithm, each time we update the coding rate \mathbf{r} with the minimum degradation of the average PSNR. It will not stop until the transmission bandwidth B is satisfied. The computation complexity is at most $O(L \cdot D^L)$. When the number of receiving nodes D grows to infinity, the computation time for optimization is not acceptable, for the distribution of the receiving nodes is irregular. Nevertheless, we can decrease the complexity by clustering. In each cluster, the PLRs of the receiving nodes within an interval of I_0 are rounded up to the highest one in the cluster. Thus, the number of clusters is $Q = 1/I_0$, and the complexity of solving the problem is at most $O(L \cdot Q^L)$. We can choose suitable I_0 to constrain the processing requirement, noticing that it will lead to a little sacrifice of the average PSNR.

Table 1 The PLRs of destination nodes in TB and CB

	e_1	e_2	e_3	e_4
TB	0.01	0.01	0.4	1
CB	0.007	0.008	0.1	0.3

Simulation results

In this section, we conduct a series of numerical simulations to verify the analysis aforementioned. As shown in Figure 8, two prototypes including TB and Cooperative broadcast are demonstrated. In both prototypes, there are one source node S and a set of four destination nodes $\mathcal{D} = \{1, 2, 3, 4\}$. As described in ‘‘Cooperative broadcast’’ section, each destination node $d \in \mathcal{D}$ is equipped with multiple antennas and can act as a receiver or a relay transmitter. In the right part of this figure, Destinations 1 and 2 decode the message in the first time slot, then they turn to relay nodes to transmit the decoded message simultaneously to Destinations 3 and 4 in the second time slot. Due to the channel fading environment and path loss, the channel qualities, e.g., SNR, of the four receiving nodes differ from each other, which results in different PLRs.

Assume the PLRs of the destination nodes in Table 1. Note that in TB, Destination 4 is out of the coverage of the source node, resulting in PLR of 100%. In addition, the PLR of Destination 3 is very high. With the benefit from CB, Destination 4 can receive from the source under PLR of 30%, while the PLRs of the other three destinations are also decreased.

We use the 720 p high definition video sequence *Mobcal* (25 fps, 1280×720) with five resolution scalable layers which gradually improve the overall video quality. The sequence includes 250 frames in the duration of 10 s, which is encoded by Motion-JPEG 2000. The average PSNR of the sequence of scalable resolution layers is shown in Table 2, corresponding to the encoding bitrate.

Based on the channel conditions of the destination nodes in both prototypes, with the information of layered multimedia from the source encoder, we can perform the proposed QoE-driven optimization algorithm to derive the optimal parameters, i.e., selected source layers L_S and PRC coding rates \mathbf{r} , under the constrained broadcast bitrate. Note that the bitrate of CB includes both the two phases. The expected average PSNR is shown in Figure 9, which can be achieved in case of ideal recovery of rateless codes, which assumes infinite block length. To confirm that ideal results are good approximation of practical applications, we perform simulation experiments.

We select a simulation setting with block length $K = 1000$, symbol size $S = 512$ bytes. Each symbol is transmitted as a packet. According to the optimized parameters

derived in the algorithm, we provide the simulation results in Figure 10. As predicted, the real average PSNR of the simulations decreases a little (no more than 0.3 dB) compared with the expected ones in Figure 9, which indicates the efficiency of the proposed algorithm.

As shown in Figure 10, it can be seen that Cooperative broadcast outperforms TB all the time, since cooperation reduce the PLRs of the worst node effectively. In particular, at some points the traditional one have nearly the same performance as the cooperative one, because all the destinations except for the worst node are served with the best quality video. As the broadcast bitrate increases, the average PSNR of the traditional one will not increase any more.

It is still worth noting that the proposed PRC is always better than the conventional equal error protection strategy we define as *worst case*. Using such strategy, the source node protects all the stream layers with the same coding rate equal to γ and all receiver nodes can recover the equal number of stream layers. Thus, the worst node with highest PLR should be considered, which determines the truncated layer L_S due to the constrained bandwidth B . In the figure, the average PSNR gain of the PRC can be more than 6 dB. Thus, the PRC can improve the average QoE in the broadcast group substantially. Particularly at some intermediate points, i.e., broadcast bitrate exceeds the need of lower layers but not enough for the higher layers, the performance of the two strategies are the same.

In some other distribution of destination nodes, the CB may not be better than the traditional one, since the two-phase protocol is too waste of transmission bandwidth. For example, if the PLRs of the destinations are all too high, there is little that cooperation can do to benefit them. So, the source node needs to decide to cooperate or not before broadcasting. Nevertheless, the proposed PRC will improve the average performance of the broadcast in any case.

Conclusions

This article has presented a layered multimedia broadcast scheme using rateless codes with progressive recovery over cooperative MIMO. In this broadcast scheme, first a two-phase CB protocol is proposed to improve the reliability of the broadcast, which efficiently extends the coverage of the source nodes and decreases the packet loss rate of bad destination nodes. Second, we propose

Table 2 The scalable resolutions and bit rates of Motion-JPEG 2000 encoding for *Mobcal* sequence

	1 Layer	2 Layers	3 Layers	4 Layers	5 Layers
Resolution	80 × 45	160 × 90	320 × 180	640 × 360	1280 × 720
Bit rate (Mbps)	2.015	6.034	16.851	41.733	69.034
PSNR	18.102	19.057	20.971	25.401	35.383

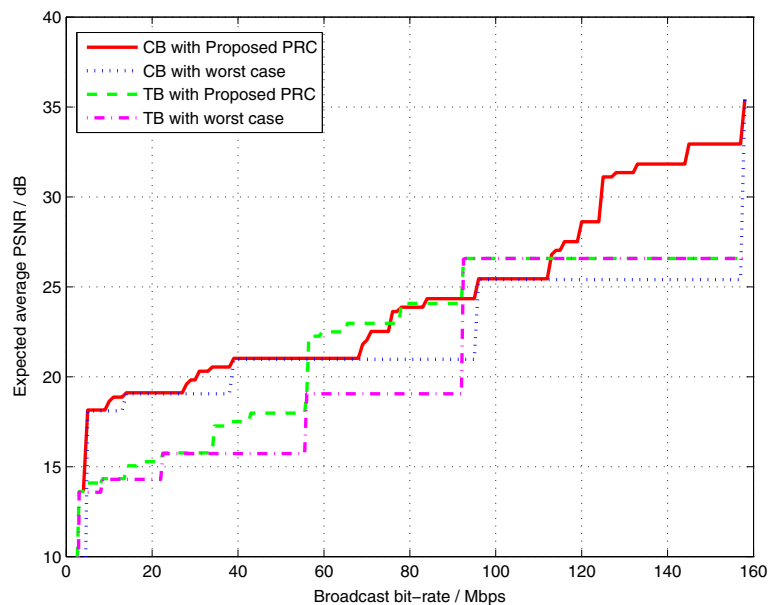


Figure 9 Expected PSNR of the different broadcast schemes with constrained broadcasting bitrate. Note that worst case denotes the conventional EEP strategy satisfying the worst node under layered multimedia.

the PRC to enhance rateless codes with progressive recovery. By assigning unequal redundant packets to each layer with their importance, the PRC can recover layered multimedia at designated received ratio of output symbols, which outperforms the other unequal protected rateless codes such as SP-FEC and LA-FEC in [21] with much lower overhead. Third, based on the PRC and given PLRs of intended destinations in the system, a distortion-

based layered multimedia broadcast optimization problem is formulated to improve the QoE of the broadcast system. By the optimized layered source bitrate and coding rates of PRC, the average transmission quality, i.e., PSNR of videos and images, has been maximized. Using resolution scalable Motion-JPEG 2000 video sequence, the receivers with different PLRs can recover a progressive resolution video stream adaptively. Analytical and

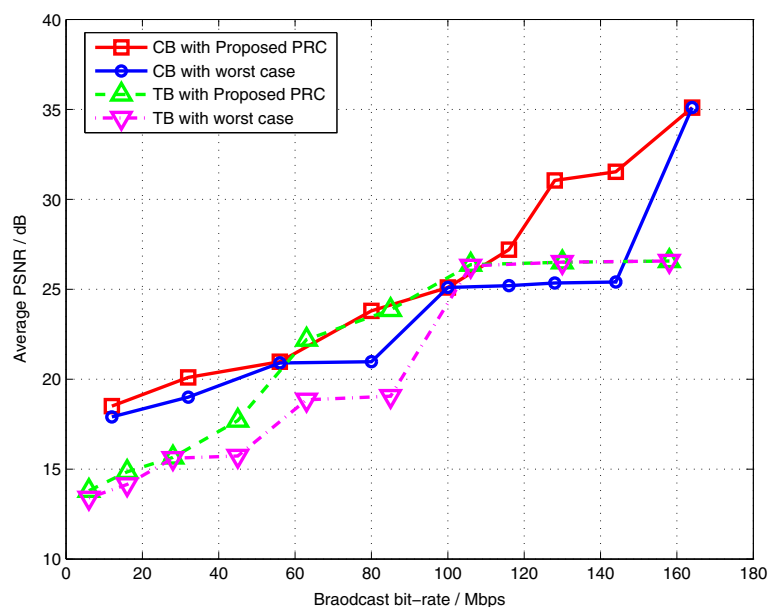


Figure 10 Simulation results for PSNR of the different broadcast schemes with constrained broadcasting bitrate.

experimental results suggest the superiority of the Cooperative broadcast scheme and the efficiency of the QoE-driven optimization algorithm.

Endnotes

^aNote that video can also be transmitted in this broadcast network using Motion-JPEG 2000, each frame of which can be seen as a JPEG 2000 image.

^bFEC gives the receiver an ability to correct errors without data retransmission. Over an erasure channel, the receiver either receives the packet or drop it when error is detected.

^cNote that during the encoding process of rateless codes, a symbol represents the smallest unit of data with the same size. One or more symbols can be contained in a packet.

^dNote that since we implement ML decoding of rateless codes, the overhead can be very low as k grows [18]. For example, when $k > 500$, $\epsilon < 0.01$. So, it is reasonable to make the assumption.

Competing interests

The authors declare that they have no competing interests.

Acknowledgements

This study was supported in part by the National Natural Science Foundation of China (NSFC) under Grants No.61101072, No.61202139 and No.61021001, and in part by the Program for New Century Excellent Talents in University (NCET).

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Received: 2 April 2012 Accepted: 27 September 2012

Published: 30 October 2012

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doi:10.1186/1687-1499-2012-327

Cite this article as: Chen et al.: Layered multimedia broadcast using rateless codes with progressive recovery over cooperative MIMO. *EURASIP Journal on Wireless Communications and Networking* 2012 **2012**:327.

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