

# Joint redundant and random network coding for robust video transmission over lossy networks

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**Abstract.** In this paper a novel unequal packet loss protection scheme R2NC based on low-triangular global coding matrix with ladder-shaped partition is presented, which combines redundant and random network coding for robust H.264/SVC video transmission. Firstly, the error-correcting capabilities of redundant network coding make our scheme resilient to loss. Secondly, the implementation of random network coding at the intermediate nodes with multiple input links can reduce the cost of network bandwidth, thus reducing the end-to-end delay for video transmission. Thirdly, the low-triangular global coding matrix with ladder-shaped partition is maintained throughout the R2NC processes to reduce the impact of global coding matrix's rank deficiency on video transmission and provide unequal erasure protection for H.264/SVC priority layers. The redundant network coding avoids the retransmission of lost packets and improves error-correcting capabilities of lost packets. Based only on the knowledge of the packet loss rates on the output links, the source node and intermediate nodes can make decisions for redundant network coding and random network coding (i.e., how much redundancy to add at this node). However, the redundancy caused by redundant network coding makes the network load increases. In order to improve network throughput, we performed random network coding at the intermediate nodes. Our approach is grounded on the overall distortion of reconstructed video minimization by optimizing the amount of redundancy assigned to each layer. The convex optimization model is constructed under the constraint of network coding and scalable video coding. Experimental results are shown to demonstrate the significant improvement of H.264/SVC video reconstruction quality with R2NC over packet lossy networks.

Keywords: Network coding, scalable video coding, low-triangular global coding matrix with ladder-shaped partition, unequal error protection

## 1. Introduction

Video multicast has received a lot of attention nowadays since it provides a convenient way for users to enjoy video over packet lossy networks. To support quality of service, the link state information such as delay, bandwidth, cost, loss rate, and error rate in the network should be available and manageable [1]. The most important issue is how to effectively use network resources, which is very important to improve transmission efficiency [2]. Besides, packets can be dropped in bottleneck links in the presence of traffic

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congestion [3], so it is important to increase the throughput of the delivery over the network due to the high bit rate of video data.

Network diversity permits to augment the quality of service for video transmission, with increased throughput and improved resilience to failures. Video transmission application can typically benefit from the network diversity [4] by designing multi-path or video transmission strategies. Such schemes, however, generally rely on appropriate source coding, routing mechanisms or coordination of the network peers [5–7] in order to prevent packet duplicates that cause waste of resources. To further improve the quality of the modern communication, new techniques need to be continuously explored and developed [8]. More recently, network coding [9] breaks through the traditional mode, which allows packets to be combined together at the intermediate nodes, and has been proved to have the ability to improve throughput, the video quality and reduce the decoding delay etc. [10,11]. This is a great advantage in balancing network load and improving the network resource utilization. However, unless the global coding coefficient matrix, or simply called the global coding matrix (GCM), is of full rank, the receive node cannot reconstruct all source packets. In addition, it may also increase delay and computational complexity if it is not implemented efficiently. In this paper, robust video transmission scheme with network coding over packet lossy networks is addressed and network coding rank deficiency problem is studied to minimize the overall distortion of reconstructed video.

Reliability and efficiency are both essential to robust video transmission, however, a simple superposition of forward error correction (FEC) codes and network coding may not only reduce the system reliability, but also lower the transmission efficiency. FEC codes (coding in application layer) improve the reliability by adding redundancy, which will reduce the transmission efficiency in terms of bandwidth consumption. In [12] an enhanced multi-point relays selection algorithm is proposed, which leads to less packet loss in the network. Although this algorithm can reduce the redundancy, it is not our concern in this paper. Network coding (coding in network layer) can reduce the number of relayed packets at the cost of higher computational complexity and communication overhead at the intermediate nodes. Therefore the combination of two coding mechanisms implemented in two different layers (network layers and application layers) is essential to improve the reliability and efficiency.

We propose a solution to this problem by introducing redundant network coding and random network coding for H.264/SVC (scalable video coding) video transmission over packet lossy networks. In particular, we apply unequal error protection (UEP) with network coding efficiently for H.264/SVC video to ensure that each scalable layer can be obtained in an incremental order. At the source node, the rearranged packets are encoded by random network coding, and then parity packets are generated by redundant network coding against packets loss. At the intermediate nodes with multiple input links, packets from the same input link are firstly coded with redundant network coding, and then packets from different input links are combined by random network coding. Our approach for combining redundant network coding with random network coding,  $R^2NC$ , has following benefits to video transmission system. First, the robustness of H.264/SVC video transmission system is ensured by redundant network coding, which can correct packet loss. Second, in terms of bandwidth consumption, it is more efficient than FEC codes without network coding, which expands the boundary conditions for the addition of redundancy, because random network coding can reduce the number of relayed packets. Finally,  $R^2NC$  eliminates the need to know the knowledge of neighbors.

In the design of  $R^2NC$  scheme for robust H.264/SVC video transmission, a number of questions still need to be addressed. In particular:

- Q1: How to gracefully combine unequal erasure protection and  $R^2NC$  so that each scalable layer can be obtained in an incremental order? The bitstream rearrangement algorithm is shown to combine unequal erasure protection and  $R^2NC$  for H.264/SVC priority layers.

- Q2: How to generate valid GCM for  $R^2NC$  to reduce the impact of GCM's rank deficiency on video transmission? The global coding matrix with ladder-shaped partition (LTGCM) is maintained throughout  $R^2NC$  process to enable partial decoding of a block and reduce the impact of GCM's rank deficiency.
- Q3: How much redundancy should be added to improve the reliability for H.264/SVC transmission and what percentage of packets from different input links should be combined together to improve the transmission efficiency? We show how to adjust the amount of redundancy after considering the packet loss and link capacity and specify the order in which we perform the two types of coding at the source node and intermediate nodes.

This paper is built on the previous work [13], a practical network coding scheme for H.264/SVC video transmission over the packet lossy networks. The differences are: (i) the effective constructive approaches of LTGCM for  $R^2NC$  scheme are given in detail. (ii) The reasonable redundancy added at the source node and intermediate nodes for  $R^2NC$  is shown, which is ensured by the distributed optimal solutions. (iii) to evaluate the performance of  $R^2NC$  fairly, the packet loss rate, normalized network resource usage and decodability of the scalable unit are addressed. The presented approach is grounded on the overall distortion of reconstructed video minimization formulation. The convex optimization model is constructed under the constraint of network coding and H.264/SVC coding to ensure that each scalable layer is obtained in an incremental order. The effectiveness of the approach is demonstrated by the system implementation and performance evaluation.

This paper is organized as follows. Section 2 outlines the background of network coding applied to the streaming media. Section 3 gives an overview of the system model. Section 4 describes the overall distortion of reconstructed video minimization formulation and solution. Section 5 presents the packet loss protection scheme- $R^2NC$  for video streaming transmission over packet lossy networks in detail. In Section 6, we provide experimental results and performance analysis. Section 7 concludes the paper.

## 2. Related works

### 2.1. Existing works with network coding for video transmission

While most network coding research has been carried out in the field of information theory, its potential benefits for media streaming applications have spurred a lot of interest in the multimedia community. Most existing works based on network coding are designed for robust video transmission system and focus on the application of random linear network coding (RLNC) [14–16]. Nguyen et al. [14] proposed a scheme on multipath transmit with network coding to meet the demand of high bandwidth for video transmission, but the best strategy proposed does not take into account optimization [17] and the variable quality of service. In order to reduce transmission delay caused by network coding for video streaming [18], proposed an effective allocation algorithm for nodes to perform network coding in an overlay network, which minimize the transmission delay for video streaming. However, a clear principle to determine what percentage of nodes to perform network coding is not given. In [11], the effective placement of a limited number of nodes that implement randomized network coding in overlay networks is also addressed in order to reduce the delay for decoding in streaming applications. Taking into account the basic characteristics of streaming data [16], proposed a robust transmission scheme based on UEP with RLNC for scalable video data, but how to assign unequal redundancy of network coding codes to different video layers is not shown. In [19], an effective combination of rateless coding

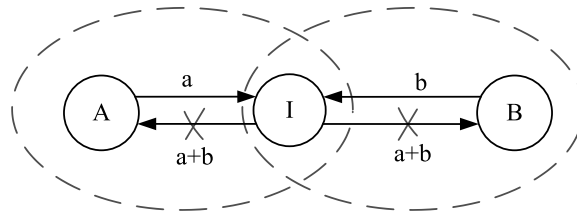


Fig. 1. Example of network coding when there is packet loss over packet lossy networks.

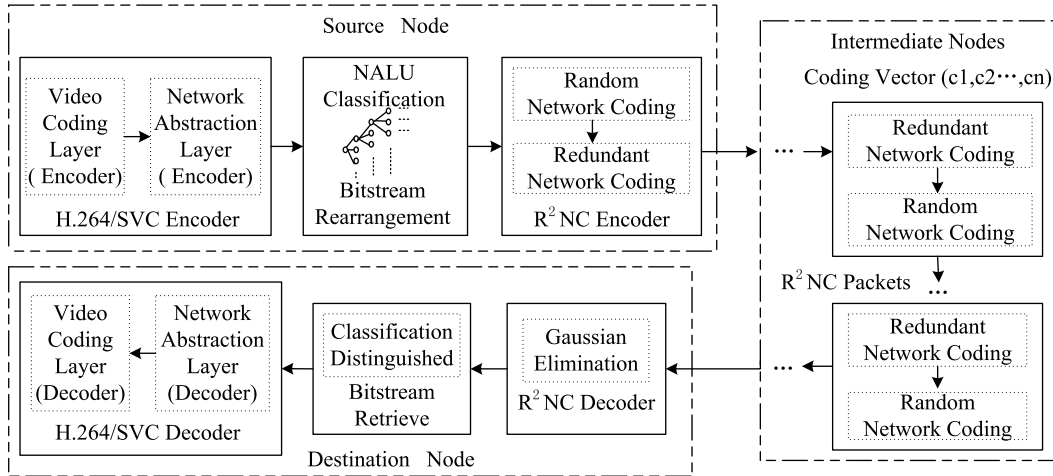
with intra and inter-session network coding to efficiently exploit the path diversity in the streaming overlay. Hulya Seferoglu [20] presented an opportunity network coding mechanism which is exactly the same as in the original COPE for video transmission in the wireless network. The decodability at the receivers is improved, but the intermediate nodes need to learn the contents of the virtual buffers of all their neighbors. After coding packets from different unicast sessions, COPE [21] effectively forwards multiple packets based on the knowledge of what their neighbors have. Without considering packets loss in the network, COPE is certainly the most effective constructive approach. However, in the presence of medium-high loss rate, the coding efficiency of COPE is severely affected. To better illustrate this point, let us discuss the following example.

**Example 1:** Let us consider Fig. 1. For the moment, let us focus only on the neighbors of intermediate node  $I$ , only the packets transmitted via  $I$ , from Node  $A$  to node  $B$  and from node  $B$  to node  $A$ . Obviously, both two nodes are not in the scope of each other's communication. The intermediate node  $I$  in COPE requires the knowledge of its neighbors, and broadcast packet  $a + b$ . In the absence of loss, the probability of successful decoding is 1 and throughput is improved by 0.333, because node  $I$  delivers two packets in three transmissions (with network coding), instead of four (without network coding). Let us revisit this example when there is packet loss. Assume that there is loss only on the link  $I - A$  and  $I - B$ , and the packet loss rate is  $p$ . In this case,  $(1 - p)\%$  of the packets can still be coded together, and the successful decodability is  $1 - p$ . The throughput improved by network coding is decreased due to packets loss. Although the throughput is still improved by NC, the reliability of the system declines because the decodability at the destination node is affected by packets loss.

Our proposed network coding scheme- $R^2NC$  is built on [16,19], a practical network coding scheme for H.264/SVC video transmission over the packet lossy networks. The main differences are: (i) we show how to assign unequal redundancy of  $R^2NC$  codes to different scalable layers based on LTGCM. (ii) we consider the effect of packet loss, in order to generate the right amount of redundancy for each layer at the source node and intermediate nodes. (iii) the intermediate nodes does not need to learn the knowledge of what their neighbors have overheard.

## 2.2. Combination of two coding for video transmission

Both network coding and FEC codes are erasure correction codes, essentially having one thing in common: both are based on a finite field to encode the original packets to a new set of coding packets, and as long as the receiver get enough number of packets, they can be decoded successfully. The main difference is: FEC code is implemented in the end systems, while network coding is carried out at the intermediate nodes. Clearly, both the two codes can be implemented in the streaming system, but mostly traditional methods treat them separately which can not share information. The cooperative work [22] of NEC corporation and the university of California was the first study on network coding and forward error

Fig. 2. System architecture of  $R^2NC$ .

correction coding to improve the performance of video transmission system over the wireless network, and they just planed to optimize the performance of ad hoc network with two coding at present. So far, the results of relevant research has not been publicly published [23] explored the performance of scheme combined network coding with FEC codes in depth. They proposed a scheme joint network coding based on time-domain and FEC in application layer, but the decoding process of two coding were treated separately at the destination nodes, which will result in too much space cost and the end-to-end delay of video transmission system. So far the combination of network coding and FEC coding is just a simple superposition. Firstly, the decoding process of network coding and FEC coding is completed separately at the destination node. Secondly, the combination of network coding and FEC coding can not share information during the coding process. This paper also improves the quality of video streaming by the combination of the two coding-redundant network coding and random network coding, but our scheme is not a simple superposition. The decoding process of the two coding can be completed simultaneously and the boundary conditions of the redundancy added by redundant network coding can be expanded by random network coding at the intermediate nodes.

### 3. System overview

#### 3.1. System description

The architecture of H.264/SVC video transmission system using  $R^2NC$  technique over packet lossy networks is shown in Fig. 2. We study a single-source multicast communication over packet lossy networks, where all nodes are fixed and one source node transmits video to multiple destination nodes. At first, H.264/SVC encoder, bitstream re-arrangement, and  $R^2NC$  encoder are performed at the source node respectively. Then, the redundant network coding and RLNC [9] techniques are conducted at the intermediate nodes. At the destination node, gaussian elimination method is used for  $R^2NC$  decoding, which is followed by bitstream retrieve and video reconstruction.

One concern at the source node is how to setup the classification ranks for the group of pictures (GOP) level and per-frame bitrate. In general, the more classified levels used, the more delicate unequal

protection achieved [24]. The system assembles the classified packets into different blocks of packets (BOP) to implement R<sup>2</sup>NC across packets. More classification stages will result in smaller BOP sizes, which directly affect the capability of the R<sup>2</sup>NC in coping with burst packet losses. Another concern is how to assign the valid coding vectors to combine the packets with random network coding. If invalid coding vectors are assigned, it will lead to the insufficient rank in GCM, and the destination nodes cannot decode packets successfully. The last concern is how much redundancy to be added by redundant network coding at the source node to improve the reliability of robust video transmission system. In this paper, to solve these problems, we propose a generation-based design and specify the order in which we perform the two types of coding.

At the intermediate nodes, how much redundancy should be added to ensure the decodability of each scalable layer at the destination nodes should be carefully decided and how to assign valid coding vectors to generate GCM should also be addressed at the intermediate nodes. In static network environments, where network topology does not change, valid coding vectors can be assigned to all intermediate nodes. In dynamic network environments, where network topology changes dynamically, decentralized coding vector assignment such as RLNC [9] is required. In addition, what percentage of coding packets should be coded together to reduce the number of relayed packets and what parts should remain uncoded must be taken seriously. We show the low-triangular GCM with ladder-shaped partition for R<sup>2</sup>NC and allocate unequal redundancy of R<sup>2</sup>NC codes to different scalable layers after taking into account the packet loss and link capacity on the output links of the intermediate nodes.

### 3.2. Redundant network coding and random network coding

- 1) Redundant network coding (for Error Correction): At the source node, packets with the same group can be used to generate  $P$  parity packets with redundant network coding for error correction. The source node generates the parity packets, depending on the loss rates of the output links, which ensures any  $G$  out of  $G + P$  packets are linearly independent thus can be used to reconstruct the original packets. Each intermediate node first generate the redundant packets for error correction with the redundant network coding along the same input link depending on the packet loss of next hop. The same process is repeated at all intermediate nodes until the decodability of each scalable layer at the destination nodes reaches a specific value.
- 2) Random network coding (for Network Resource Usage): Suppose the intermediate node does not need to decode original packets, it just combines the packets in the same group and updates their global coding vectors. When  $G^i$  packets from link  $i$  are received at an intermediate node, then  $P^i$  redundancy packets would be generated with redundant network coding, which will increase the network load. Random linear network coding can improve the network throughput, which has been proved by [9]. Thus RLNC is used to improve utilization of network bandwidth and reduce the end-to-end transmission delay. This is a great advantage in real-time transfer. In addition, boundary conditions of the redundancy addition is relaxed by random network coding at the intermediate nodes.

### 3.3. Global coding matrix with ladder-shaped partition

One challenge of adapting network coding for video transmission over packet lossy networks is the insufficient rank in global coding matrix. To reduce the impact of GCM's rank deficiency on video transmission, we propose a low-triangular LTGCM for R<sup>2</sup>NC based UEP scheme. Figure 3 shows three types of GCM for NC-based UEP scheme, where submatrix  $M_i$  corresponds to a scalable layer  $i$  for

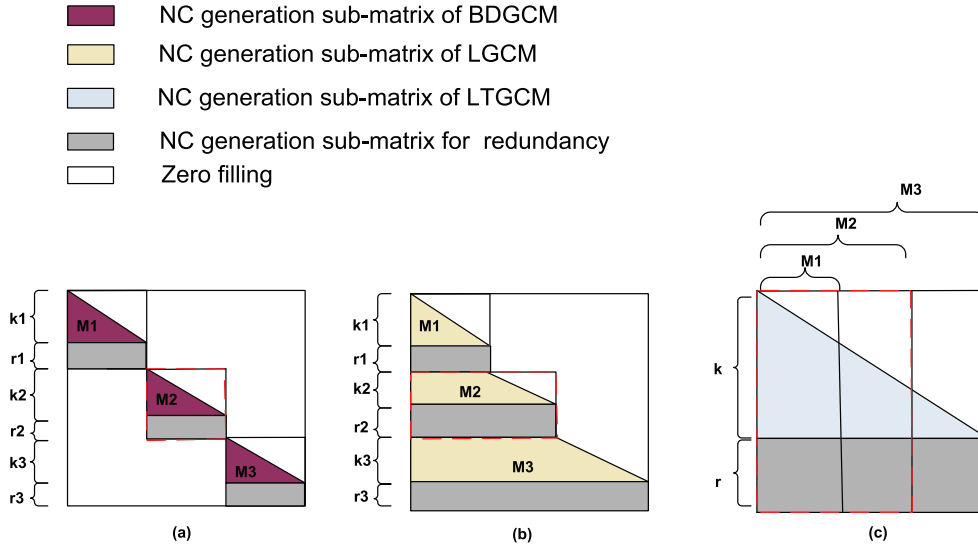


Fig. 3. Three types of GCM for NC-based UEP Scheme: (a) Block diagonal GCM (b) Ladder GCM (c) Low-triangular GCM with ladder-shaped partition.

NC-based UEP scheme. General GCM is a direct consequence of RLNC-based UEP scheme for video transmission, which is not shown here. The coding vectors of general GCM are not linearly independent, which will lead to the insufficient rank in GCM at the destination node and then the scalable layer cannot be decoded with a high decodability. This three types of GCM can all be used to reduce the impact of GCM's rank deficiency on video transmission. Submatrix  $M_i$  consists of two parts. The first part has  $k_i$  rows, which form a non-strict lower triangular matrix, and the second part has  $r_i$  rows. The first part is used to generate network coding packets, and the second part is used to create redundant network coding packets. The white area of three GCM is filled by zeros. We can observe that the coding vectors of low-triangular GCM are all linearly independent which will improve the decodability of the layered video data at the destination nodes, and the general GCM has too much space cost than that of this three GCM. For block diagonal GCM (BDGCM) [25], if redundancy for layer  $i - 1$  is not sufficient, layer  $i - 1$  can not be decoded. For ladder-shaped GCM(LGCM) [25] and LTGCM, if redundancy for layer  $i - 1$  is not sufficient, it can wait for the next layer data till achieving full rank. In another word, the redundancy in layer  $i$  can protect the erasure in layer  $i - 1$ . In addition, LGCM needs more redundant packets than LTGCM to improve the reliability. Therefore, LTGCM is used for our R<sup>2</sup>NC based UEP scheme.

#### 4. Distortion of reconstructed video minimization formulation

##### 4.1. Problem formulation

Although networks pose many security challenges [26,27], the packet loss problem is the most common issue. In order to provide the reliability for H.264/SVC video transmission over packet lossy networks, our goal is to minimize the overall distortion of reconstructed video. In the new H.264/SVC standard, new tools including combined spatial, temporal and quality scalability are added. In this paper, in order to simplify the optimization model, we only consider the temporal and quality scalability. Each temporal

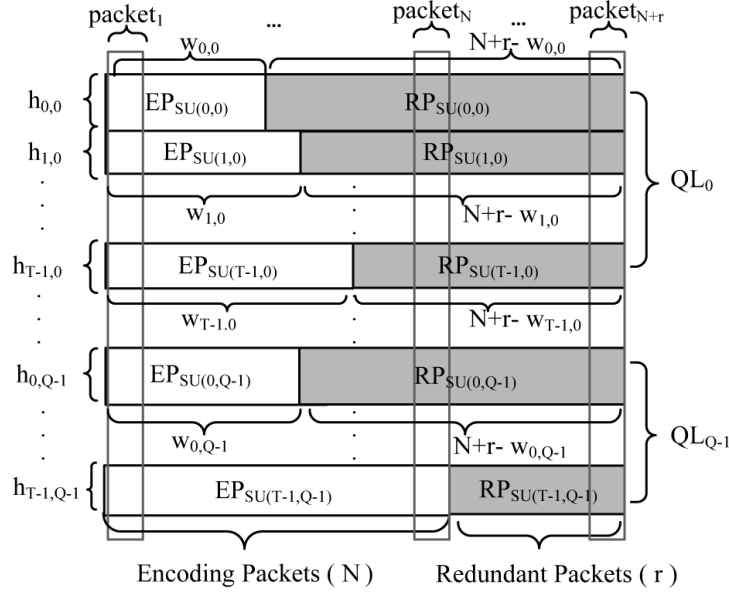


Fig. 4. Bitstream rearrangement with R<sup>2</sup>NC for scalable units (SU).

layer can contain several quality layers. If the temporal layers  $L_0, \dots, L_{i-1}$  can be obtained at the receiver, temporal layer  $L_i$  can be decoded independently. Consequently, the quality layer  $Q_i$  can be decoded as long as the quality layers  $Q_0, \dots, Q_{i-1}$  can be obtained. Each quality layer in each temporal layer is defined as a scalable unit (SU) [28]. Due to the dependency between quality layers, the effect of packet loss is severe when compressed video data is transmitted over packet lossy networks. As a result, we need to reorganize SVC bitstream with prioritization within one GOP to make the transmission more efficient and robust under the same bandwidth condition.

Figure 4 shows unequal protection scheme (UEP) for scalable unit (SU) to rearrange the H.264/SVC bitstream and the package format with R<sup>2</sup>NC. The number of temporal layer is  $T$  and each temporal layer is further divided into  $Q$  quality layers (QL). If we represent  $i$  as the temporal level and  $j$  as the quality level where  $i = 0, 1, \dots, T - 1$  and  $j = 0, 1, \dots, Q - 1$ , the scalable video data for the unit  $(i, j)$  is defined as  $SU(i, j)$  [28]. The width and the height of each  $SU(i, j)$  are  $w_{i,j}$  and  $h_{i,j}$  bytes respectively. The bitstream rearrangement method based on  $SU(i, j)$  is shown in Fig. 4. The data  $EP_{SU(i,j)}$  rearranged generates the encoding packets, and the residual packets are filled with redundancy parity packets  $RP_{SU(i,j)}$  for  $SU(i, j)$  with valid coding vectors instead of zero. In Fig. 4, the white part is the packets which are rearranged by bitstream rearrangement algorithm and then encoded with valid coding vectors of LTGCM. The gray part is the redundant parity packets for each  $SU(i, j)$  generated by redundant network coding based on LTGCM. Here, for convenience, we denote  $w_{T-1,Q-1}$  as  $N$ . We can see the number of the redundant parity packets for each  $SU(i, j)$  is  $N + r - w_{i,j}$ . After the implementation of R<sup>2</sup>NC at the source node,  $N$  coding packets and  $r$  parity packets which are linearly independent are generated, as is shown in Fig. 4. The height of  $SU(i, j)$  is represented as  $h_{i,j}$  and can be calculated as

$$h_{i,j} = \left\lceil \frac{R_{i,j}}{w_{i,j}} \right\rceil \quad (1)$$

here  $R_{(i,j)}$  is the number of source data bytes for  $SU(i, j)$ .



Our object is to find the best R<sup>2</sup>NC code assignment for minimizing the overall distortion of reconstructed video and optimizing the performance and robustness, including the choosing of  $w_{i,j}$ , redundancy  $r$  at the source node, and redundancy  $R$  at the intermediate nodes. In this paper, we adopt the peak signal-to-noise ratio (PSNR) value to measure the amount of distortion. The overall distortion can be calculated as follows

$$D_{overall} = \sum_{i=0}^{T-1} \sum_{j=0}^{Q-1} \delta_{i,j} \cdot p_{i,j}^{su} \quad (2)$$

where  $\delta_{i,j}$  is the PSNR decrement from the erasure of  $SU(i, j)$  and  $p_{i,j}^{su}$  is the loss rate of  $SU(i, j)$  over packet lossy networks with R<sup>2</sup>NC. The  $\delta_{i,j}$  value can be calculated experimentally. If the number of lost packets is greater than the number of the parity packets, the original streaming can not be recovered completely. The  $p_{i,j}^{su}$  can be formulated as

$$p_{i,j}^{su} = \sum_{m=N+r-w_{i,j}+1}^{N+r} C_{N+r}^m (p')^m (1-p')^{N+r-m} \quad (3)$$

Where  $C_{N+r}^m (p')^m (1-p')^{N+r-m}$  is the probability of losing  $m$  packets among  $N+r$  packets over packet lossy networks. The  $p'$  is the average packet loss rate over packet lossy networks with R<sup>2</sup>NC, which is relative to packet loss channels and the R<sup>2</sup>NC mechanism. In order to minimize (2),  $p'$  should be determined. It's a complex problem and may have different results for different topology. In this paper, we use simulation as well as curve fitting to find  $p'$ .

The intermediate nodes are classified into two types simply, nodes with only single input link and nodes with multiple input links.  $N_{SI}$  is defined as the set of nodes with only single input link, and  $N_{MI}$  is the set of nodes with multiple input links. We denote  $I\_link(I)$  as the set of input links of node  $I$ , and  $x_k(i)$  is the number of packets transmitted on the input link  $i$  of node  $k$ . Our goal is to minimize (2), and the problem is formulated as

$$\begin{array}{ll} \text{minimize} & D_{overall} \\ \text{subject} & \text{to} \end{array}$$

$$\sum_{i=0}^{T-1} \sum_{j=0}^{Q-1} h_{i,j} + H \leq M \quad (4)$$

$$w_{i,j} \leq w_{i+1,j}, \quad i = 0, 1, \dots, T-2 \quad (5)$$

$$w_{i,j} \leq w_{i,j+1}, \quad j = 0, 1, \dots, Q-2$$

$$\sum_{i=0}^{T-1} \sum_{j=0}^{Q-1} h_{i,j} \cdot (N+r) \leq B_{tot} \quad (6)$$

$$\forall k \in N_{SI} \quad \text{and} \quad N_{SI} = \{N_1^S, N_2^S, \dots, N_k^S\} \quad (7)$$

$$\frac{x_k \cdot (1 + \alpha(k))}{1 - P_k^{out}} \leq R_k^{out}$$

$$\begin{aligned}
\forall I \in N_{MI} \quad \text{and} \quad N_{MI} &= \{N_1^M, N_2^M, \dots, N_{k^*}^M\} \\
\forall i \in I\_link(I) \quad \text{and} \quad I\_link(I) &= \{l_1, l_2, \dots, l_{i^*}\} \\
\frac{\max\{x_I(i) \cdot (1 + \alpha_I(i))\}}{1 - P_I^{out}} &\leq R_I^{out}
\end{aligned} \tag{8}$$

$$\begin{aligned}
\forall k \in N_{SI}, \quad \forall m \in N_{MI}, \quad \text{and} \quad \forall i' \in I\_link(m) \\
\sum_{k=1}^{k'} \alpha(k) + \sum_{m=1}^{k^*} \sum_{i'=1}^{i^*} \alpha_m(i') = 1
\end{aligned} \tag{9}$$

where  $B_{tot}$  is the total number of bits to be allocated for a GOP and  $M$  represents the length of a packet.  $P_k^{out}$  is the average packet loss rate of the output link of node  $k$  and  $\alpha_k(i)$  is the redundancy rate of  $i$ th input link of node  $k$  with redundant network coding.  $R_k^{out}$  is the capacity of the output link of node  $k$ . Equation (5) means that the smaller width  $w_{i,j}$  is assigned to  $SU(i, j)$  with lower temporal and quality layers than that with higher temporal and quality layers. From Fig. 4, we can also observe that the smaller  $w_{i,j}$ , the more important  $SU(i, j)$ , and the more redundancy allocated for  $SU(i, j)$ , which is ensured by the bitstream rearrangement algorithm and LTGCM on the constraint of Eq. (5). Equation (7) is the capacity constraint for each flow from the single input link of intermediate nodes which perform redundant network coding. The second term of Eq. (7) refers to loss on the output link, which is the amount of redundancy (via redundant network coding) added against loss. Equation (8) is the capacity constraint for the intermediate nodes with multiple input links performing R<sup>2</sup>NC, which determines the redundancy (via redundant network coding) added for each flow from the input link and the amount of packets (via random network coding) combined together. Equation (9) is the boundary constraint of redundancy added (via redundant network coding) at the intermediate nodes.

#### 4.2. Distributed optimal solution

In order to simplify the optimization algorithm, the complex optimization problem need to be decomposed into a serious distributed optimal solutions. In this paper, by relaxing the capacity constraint in Eqs (7) (8) and (9), the optimization problem can be decomposed into two sub-optimization problems with different levels if some variables are fixed. Therefore, the original problem can be decomposed into two sub-problems  $P_a$  and  $P_b$ .

The optimization sub-problem  $P_a$  constitutes a low-level optimization sub-problem, which is a common constrained optimization problem and can be solved by the Lagrange multiplier method. To minimize the overall distortion of reconstructed video, the solution of  $P_a$  includes the choosing of  $w_{i,j}$  and redundant coding packets  $r$  at the source node. The bitstream rearrangement method will be given out by the solution of  $P_a$ .  $P_b$  constitutes a high-level optimization sub-problem, which ensures the effectiveness of redundancy allocation with R<sup>2</sup>NC at the intermediate nodes. The structure of the optimal solution of  $P_b$  has the following interpretation: the number of redundancy packets (via redundant network coding) and the number of forwarding packets at the intermediate nodes are determined by taking into account the packet loss and capacity constraint of output links. The higher the packet loss rate of antidotes on the output links, the smaller  $\alpha(k)$ , and the smaller  $\alpha_m(i')$ . This means, in the extreme case where the packet loss rate is very large, R<sup>2</sup>NC should be turned off.

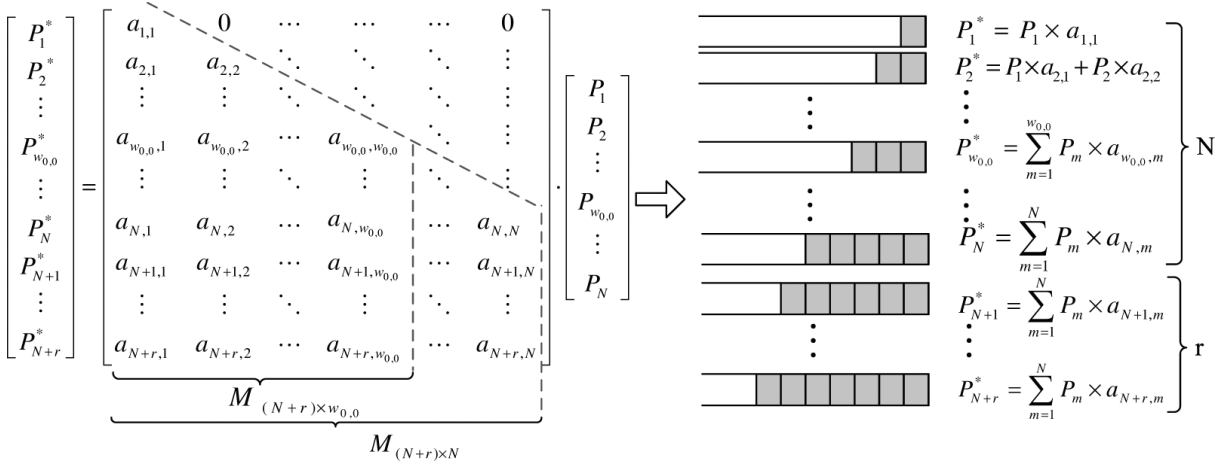


Fig. 5. Illustration of R<sup>2</sup>NC based UEP scheme that preserves LTGCM.

## 5. System implementation

In this section, we propose practical implementations of the R<sup>2</sup>NC scheme at the source node and intermediate nodes for H.264/SVC video transmission.

### 5.1. Operation of source node

Unequal error protection scheme based on R<sup>2</sup>NC for H.264/SVC video transmission: After H.264/SVC encoder as shown in Fig. 2, the importance of  $SU(i, j)$  decreases as the increment of the temporal and quality layer, which is ensured by the bitstream rearrangement algorithm on the constraint of Eq. (5). The unequal protection scheme based on R<sup>2</sup>NC with the proposed LTGCM is performed at the source node. Consider the commonly used generation-based network coding: packets are divided into generations, with size  $G$ . At the source node,  $G$  packets (via bitstream rearrangement algorithm) in the same group are combined together independently with random network coding (assuming large enough field size), and then  $P$  parity packets are generated with redundant network coding depending on the packet loss rates of the output links for error correction. In our work, we generate the parity packets with redundant network coding, which ensures that any  $G$  packets out of  $G + P$  packets are linearly independent (via valid coding vectors), thus can be used to reconstruct  $G$  original packets.

Our proposed R<sup>2</sup>NC based UEP method is performed by LTGCM, as is shown in Fig. 5. The GCM with ladder-shaped partition consists of submatrices  $M_{(N+r) \times w_{0,0}}, \dots, M_{(N+r) \times N}$ , where submatrix  $M_{(N+r) \times w_{i,j}}$  corresponds to  $SU(i, j)$  for R<sup>2</sup>NC. Submatrix  $M_{(N+r) \times w_{i,j}}$  consists of two parts. The first part of  $M_{(N+r) \times w_{i,j}}$  has  $w_{i,j}$  rows, which is used to generate  $w_{i,j}$  encoding packets and the second part has  $N + r - w_{i,j}$  rows, which is used to generate  $N + r - w_{i,j}$  redundant coding packets for  $SU(i, j)$ . The process of R<sup>2</sup>NC based UEP scheme with the low-triangular GCM is shown in Fig. 5.  $P = (P_1^* P_2^* \dots P_{N+r}^*)$  denote a set of NC packets (via R<sup>2</sup>NC) at the source node, and  $a_i$  denote the coding vector associated with  $P_i^*$ .  $(P_1 P_2 \dots P_N)$  is the set of packets generated by bitstream rearrangement algorithm. The proposed LTGCM is generated by  $a_i$ , w.h.p. any matrix  $M_{w_{i,j} \times w_{i,j}}$  of the sub-matrix  $M_{(N+r) \times w_{i,j}}$  of  $SU(i, j)$  can attain full rank. Therefore,  $SU(i, j)$  can be decoded successfully if the destination nodes receive any  $w_{i,j}$  coding packets, which is ensured by the low-triangular global coding matrix. From the GCM with ladder-shaped partition we observed, the larger  $w_{i,j}$ , the more coding packets for  $SU(i, j)$  needed at the destination node, and the smaller decodability of  $SU(i, j)$ .

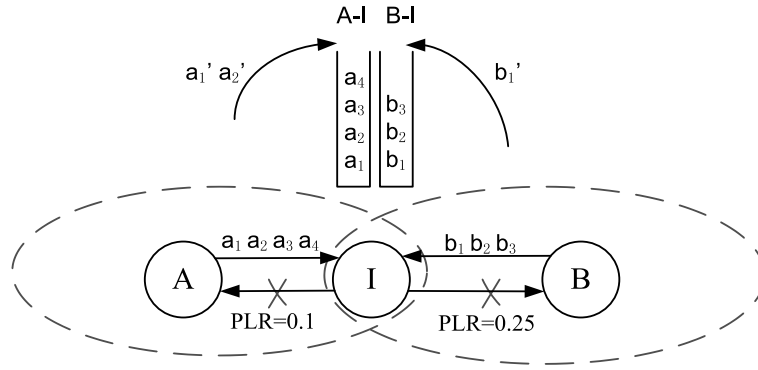


Fig. 6. Example of redundant network coding at intermediate Node I.

## 5.2. Operation of intermediate nodes

### 5.2.1. Receiving a packet and redundant network coding

Supposing the intermediate node does not need to decode, it just combine the packets in the same group and updates their global coding vectors. To reduce the packet loss, the redundant network coding is performed at the intermediate nodes. After  $x_k(i)$  packets in a group are received by intermediate node  $k$  from input link  $i$ ,  $R_k(i)$  redundant coding packets are generated (via redundant network coding) depending on the packet loss rate of the output link. The redundancy  $R_k(i)$  added by the intermediate node  $k$  for input link  $i$  can be calculated as follow

$$R_k(i) = \frac{x_k(i) \cdot P_k^{out}}{1 - P_k^{out}} \quad (10)$$

In Fig. 6, we describe redundant network coding at the intermediate nodes. The packets transmitted by  $A$  and  $B$  are  $a_1, a_2, a_3, a_4$  and  $b_1, b_2, b_3$ , respectively. The packet loss rate over  $I-A$  and  $I-B$  output links are assumed 0.1 and 0.25. Firstly, the redundancy added for input link  $A-I$  is  $R_I(A-I) = 2$ , and the redundant packets ( $a'_1$  and  $a'_2$ ) can be generated by the combination of  $a_1, a_2, a_3$  and  $a_4$  (via linearly independent coding vectors). The redundancy added for  $B-I$  is  $R_I(B-I) = 1$ , and  $b'_1$  is generated by the combination of  $b_1, b_2$  and  $b_3$  (via linearly independent coding vectors).

### 5.2.2. Transmitting a packet and random network coding

After intermediate node  $k$  generates the redundancy  $R_k(i)$  for input link  $i$ , it treats all  $x_k(i) + R_k(i)$  packets as equal parts of the same input link. Considering the actual network load, we expand the boundary conditions of adding redundancy by random network coding at the intermediate nodes and the number of transmitted packets at node  $k$  is  $\max\{x_k(i) + R_k(i)\}$ . For example, in Fig. 6, after random network coding performed at intermediate node  $I$ , the number of relayed packets is 6. Obviously, the bandwidth utilization of output links is improved by random network coding.

## 6. Performance evaluation

### 6.1. Experiment design

Visual studio 2008 is used to build the experimental H.264/SVC transmitting system based on R<sup>2</sup>NC. We use two QCIF video sequences “Foreman” and “Coastguard”. They are encoded using the version

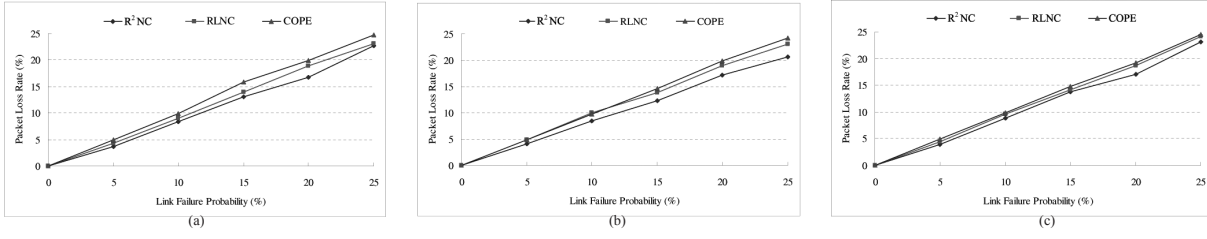


Fig. 7. Packet loss rate on the random network model with different layers: (a) Layer = 6 (b) Layer = 5 (c) Layer = 4.

9 of the joint scalable video model. One spatial layer is encoded with one quality base layer and two quality enhancement layers. For the encoding condition of key pictures with GOP size of 16 frames, the number of maximum temporal layers is 5. The proposed UEP scheme based on R<sup>2</sup>NC is used to test the performance of the video transmission system. We considered various topologies: 4 layers, 5 layers and 6 layers with a source node, multiple intermediate nodes and multiple destination nodes. The packet loss platform of the network can be built using two-state Markov model which can approximate fairly well the packet loss channel behavior. Finally, the video quality received by the destination nodes will be contrasted.

## 6.2. Performance of R<sup>2</sup>NC

To evaluate the performance of R<sup>2</sup>NC fairly, the packet loss rate, normalized network resource usage and decodability of the scalable unit are addressed.

### 6.2.1. Packet loss rate (PLR)

In order to get the average packet loss rate based on R<sup>2</sup>NC for solution of the optimization problem, we carry out simulation on top of the random network model with different layers. We define the average packet loss rate based on R<sup>2</sup>NC as  $p'$ , which is calculated as follow

$$p' = 1 - \frac{1}{N_{dst}} \cdot \sum_{i=1}^{N_{dst}} \frac{r_x^i}{t_x} \quad (11)$$

here  $N_{dst}$  is the number of destination nodes, and  $r_x^i$  and  $t_x$  denote the number of packets received at  $i$ -th destination node ( $i = 1, 2, \dots, N_{dst}$ ) and the number of packets generated at the source node, respectively. We find that the average packet loss rate of R<sup>2</sup>NC is strongly relative to the average link failure probability, as is shown in Fig. 7. We can observe that packet loss rate with R<sup>2</sup>NC is lower than that with other schemes because COPE based method does not take the packet loss into account, and RLNC scheme is performed without valid coding vectors, which will result in more redundant packets compared with R<sup>2</sup>NC scheme.

### 6.2.2. Normalized network resource usage (NRRU)

The normalized network resource usage is defined as  $\eta$ , which is calculated as follow

$$\eta = \frac{1}{N} \cdot \sum_{i=1}^N \frac{K_x^i}{R_x^i} \quad (12)$$

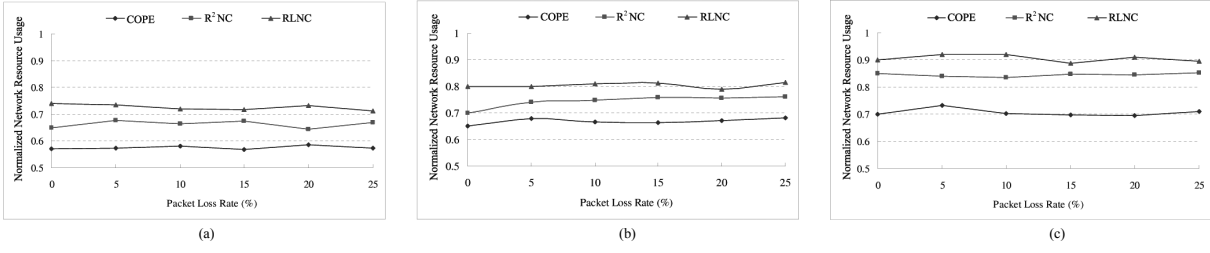


Fig. 8. Normalized network resource usage on the random network model with different layers: (a) Layer = 6 (b) Layer = 5 (c) Layer = 4.

here  $N$  is the number of nodes in the network including the source node and the intermediate nodes.  $K_x^i$  represents the number of packets transmitted by source node or intermediate nodes, and  $R_x^i$  is the number of packets generated by source node or received by the intermediate nodes. Figure 8 gives the normalized network resource usage on top of the random network model with different layers. From Fig. 8 we can observe that the normalized network resource usage with  $R^2NC$  is higher than that with COPE due to the addition of redundancy by  $R^2NC$ . And the NNRU with  $R^2NC$  is lower than that with RLNC because of the valid coding vectors which reduce the number of relayed packets. From Figs 8(a) (b) and (c), we also observe  $\eta$  are all smaller than 1 at different packet loss rate because that: (1) packets from different input links are combined together with random network coding which reduce the number of relayed packets; and (2) the number of links used by all destination nodes is reduced because some links are shared among paths for different destination nodes. In addition, we can observe the more number of intermediate nodes perform random network coding, the more bandwidth usage will be reduced. Therefore random network coding performed at the intermediate nodes can reduce the normalized network resource usage for H.264/SVC video transmission.

### 6.2.3. Scalable unit decodability (SUD)

In order to evaluate the erasure correction capability in the proposed system with  $R^2NC$ , we define the decodability of scalable unit as  $P_{SU(i,j)}$ , which can be calculated as follow

$$P_{SU(i,j)} = \frac{1}{N} \cdot \sum_{m=1}^N D_{SU(i,j)} \quad (13)$$

here  $N$  is the number of GOP in a sequence,  $D_{SU(i,j)}$  denotes whether  $SU(i, j)$  is decoded successfully, and  $D_{SU(i,j)}$  is set to be 1 if  $SU(i, j)$  is decoded. Figure 9 gives the decodability of scalable unit with different network coding schemes at 20% PLR. From Fig. 9(a) (b) and (c) we can observe, the decodability of scalable unit with  $R^2NC$  is higher than that with other schemes and each scalable layer can be obtained in an incremental order on the constraint of Eq. (5). The decodability of base layer  $SU(0, 0)$  with  $R^2NC$  is 1 shown in Fig. 9(a), and the bitstream of important layers can be decoded with a higher decodability.

### 6.3. Performance of video transmission scheme with $R^2NC$

To compare the video quality with different network coding method fairly, all network coding schemes are performed with the same bitstream rearrangement algorithm based on scalable unit. Figure 10 shows the PSNR comparison at 15% PLR for different sequence with different network coding scheme. We

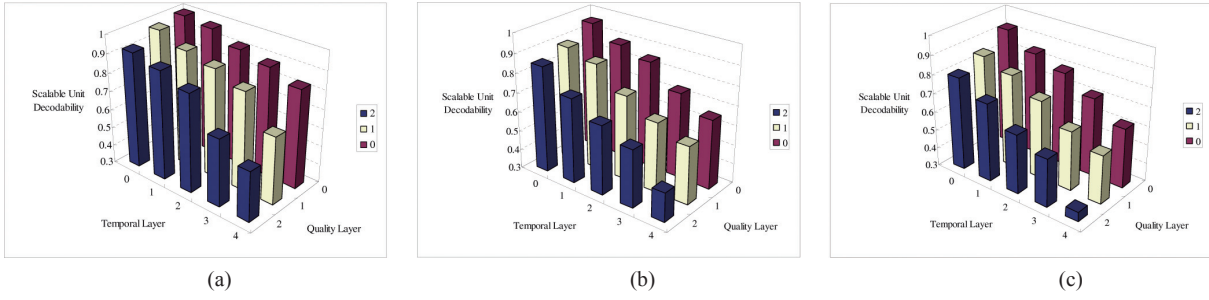


Fig. 9. Scalable unit decodability with different network coding scheme at 20% PLR: (a)  $R^2NC$ + SU-based UEP (b) COPE + SU-based UEP (c) RLNC + SU-based UEP.

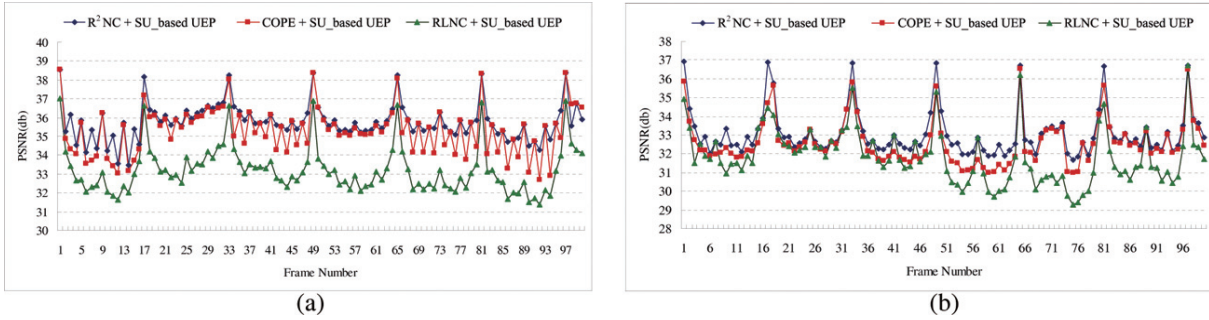


Fig. 10. PSNR comparison at 15% PLR for different sequence: (a) ‘foreman’ (b) ‘coastguard’.

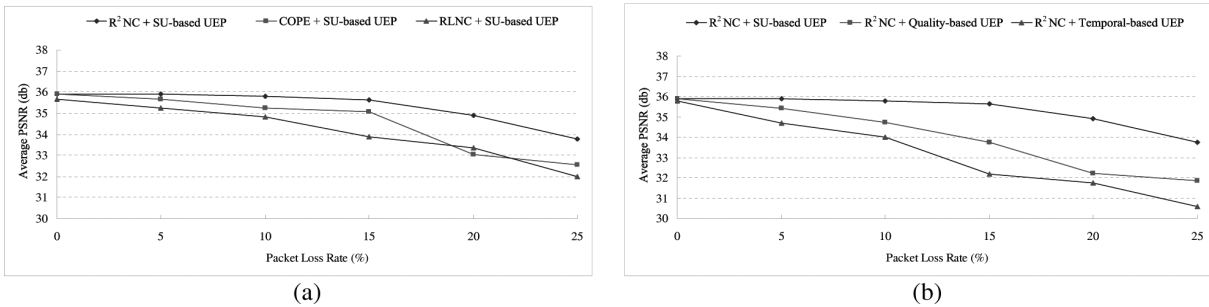


Fig. 11. Average PSNR values comparison at different PLR for sequence ‘foreman’: (a) network coding scheme (b) bitstream rearrangement.

observe the  $R^2NC$  method with proposed LTGCM can provide better PSNR values than other schemes. This is due to COPE based method does not take the packet loss into account and RLNC scheme based on general GCM is performed without using valid coding vectors, which result in a decrease of the decodability of scalable unit.

Figure 11(a) presents the average PSNR values based on different network coding schemes at different PLR with respect to the video quality. We can see that: (1) Average PSNR values are improved for protection  $R^2NC$  scheme against other schemes; (2) When the PLR is very low, all network coding schemes can recover the majority of lost packets; and (3)  $R^2NC$  scheme maintains higher PSNR values compared to other schemes at high packet loss rates from 15% to 25%; Fig. 11(b) shows the average PSNR values with different bitstream rearrangement algorithm. The quality-based UEP scheme is

performed with unequal protection ratio in quality layers without considering the unequal importance of temporal layers. And the temporal-based UEP scheme is performed with unequal protection ratio in temporal layers without considering the unequal importance of quality layers. We can see that: (1) The SU-based UEP scheme can provide better performance than that of other schemes; and (2) The quality-based UEP scheme can provide better performance than that of temporal-based UEP since the temporal-based UEP does not take into account the unequal importance of the quality layers, and the scalable units in lower quality layers obtain rather low protection bits which results in larger distortion. That is, the video transmission system based on R<sup>2</sup>NC+ SU-based UEP method is more reliability and efficient.

## 7. Conclusions

In this paper, a novel unequal packet loss protection scheme-R<sup>2</sup>NC based on LTGCM is presented for robust H.264/SVC video transmission over packet lossy networks. R<sup>2</sup>NC can improve the performance of H.264/SVC video transmission system in two aspects: it is resilient to loss due to the perform of redundant network coding without knowledge of the neighbors and the network bandwidth utilization is improved by random network coding. Therefore, the proposed scheme with R<sup>2</sup>NC can improve the robustness and the playback quality of the streaming system. The experimental results also show that the video transmission system based on R<sup>2</sup>NC can significantly improve the video PSNR values over packet lossy networks.

## Acknowledgments

This work was supported by the project of the National Key Technology R&D Program (Grant No. 2011BAH04B03), the NSFC project (Grant No. 61173039) and the State 863 project of China (Grant No. 2009AA01A135). The authors wish to thank Lin Liu for useful discussions. Part of the work was supported by the Natural Science Foundation of Shaanxi Province, China (Grant No. 2009JM8002-5).

## References

- [1] A. Barolli, E. Spaho, L. Barolli, F. Xhafa and M. Takizawa, Qos routing in ad-hoc networks using ga and multi-objective optimization, *Mobile Information Systems* 7(3) (2011), 169–188.
- [2] P. Nicopolitidis, K. Christidis, G. Papadimitriou, P. Sarigiannidis and A. Pomportsis, Performance evaluation of acoustic underwater data broadcasting exploiting the bandwidth-distance relationship, *Mobile Information Systems* 7 (2011), 285–298.
- [3] N. Qadri, M. Altaf, M. Fleury and M. Ghanbari, Robust video communication over an urban vanet, *Mobile Information Systems* 6(3) (2010), 259–280.
- [4] J. Hailiot and F. Guidec, A protocol for content-based communication in disconnected mobile ad hoc networks, *Mobile Information Systems* 6(2) (2010), 123–154.
- [5] L.M. Contreras and C.J. Bernardos, Rams: A protocol extension to pmipv6 for improving handover performance of multicast traffic, *Journal of Wireless Mobile Networks, Ubiquitous Computing, and Dependable Applications* 2(2) (2011), 67–82.
- [6] A.Z.M. Shahriar and M. Atiquzzaman, Evaluation of the route optimization for nemo in satellite networks, *Journal of Wireless Mobile Networks, Ubiquitous Computing, and Dependable Applications* 2(2) (2011), 46–66.
- [7] A. Durrezi, P. Zhang, M. Durrezi and L. Barolli, Architecture for mobile heterogeneous multi domain networks, *Mobile Information Systems* 6(1) (2010), 49–63.
- [8] F.-Y. Leu, I. You and F. Tang, Emerging wireless and mobile technologies, *Mobile Information Systems* 7(3) (2011), 165–167.



- [9] R. Ahlswede, N. Cai, S.-Y.R. Li and R.W. Yeung, Network information flow, *IEEE Trans Inf Theory* **46**(4) (2000), 1204–1216.
- [10] N. Thomos, J. Chakareski and P. Frossard, Prioritized distributed video delivery with randomized network coding, *IEEE Trans. Multimedia* **13**(4) (2011), 776–787.
- [11] N. Cleju, N. Thomos and P. Frossard, Selection of network coding nodes for minimal playback delay in streaming overlays, *IEEE Trans Multimedia* **13**(5) (2011), 1103–1115.
- [12] T.-H. Lin, H.-C. Chao and I. Woungang, An enhanced mpr-based solution for flooding of broadcast messages in olsr wireless ad hoc networks, *Mobile Information Systems* **6** (2010), 249–257.
- [13] C. Jing, X. Zhang, F. Tang, S. Fowler, H. Cui and X. Dong, R2nc: Redundant and random network coding for robust h.264/svc transmission, in *The 14th International Conference on Network-Based Information Systems*, 2011.
- [14] D. Nguyen, T. Tran, T. Pham and V. Le, Internet media streaming using network coding and path diversity, in *Proc. IEEE Global Telecommunications Conf. IEEE GLOBECOM*, 2008, pp. 1–5.
- [15] M. Montpetit and M. Medard, Video-centric network coding strategies for 4g wireless networks: An overview, in *Proc. 7th IEEE Consumer Communications and Networking Conf. (CCNC)*, 2010, pp. 1–5.
- [16] S. Xiao, J. Lu, Y. Wang and C. Wu, Robust video transmission scheme based on network coding, in *Proc. Sixth Int Intelligent Information Hiding and Multimedia Signal Processing (IHH-MSP) Conf.*, 2010, pp. 591–594.
- [17] N. Ghadiri, A. Baraani-Dastjerdi, N. Ghasem-Aghaee and M.A. Nematbakhsh, Optimizing the performance and robustness of type-2 fuzzy group nearest-neighbor queries, *Mobile Information Systems* **7**(2) (2011), 123–145.
- [18] N. Cleju, N. Thomos and P. Frossard, Network coding node placement for delay minimization in streaming overlays, in *Proc. IEEE Int Communications (ICC) Conf.*, 2010, pp. 1–5.
- [19] J. Saltarin, N. Thomos, E. Bourtsoulatzé and P. Frossard, P2p video streaming with inter-session network coding, in *Proc. IEEE Int Multimedia and Expo (ICME) Conf.*, 2011, pp. 1–6.
- [20] H. Seferoglu and A. Markopoulou, Video-aware opportunistic network coding over wireless networks, *IEEE J Sel Areas Commun* **27**(5) (2009), 713–728.
- [21] S. Katti, H. Rahul, W. Hu, D. Katabi, M. Medard and J. Crowcroft, Xors in the air: Practical wireless network coding, *IEEE/ACM Trans Netw* **16**(3) (2008), 497–510.
- [22] A. Fujimura, S.Y. Oh and M. Gerla, Network coding vs. erasure coding: Reliable multicast in ad hoc networks, in *Proc. IEEE Military Communications Conf. MILCOM*, 2008, pp. 1–7.
- [23] H. Wang and C.-C.J. Kuo, Robust video multicast with joint network coding and al-fec, in *Proc IEEE Int Symp Circuits and Systems ISCAS*, 2008, pp. 2062–2065.
- [24] X. Zhang, X. Peng, D. Wu, T. Porter and R. Haywood, A hierarchical unequal packet loss protection scheme for robust h.264/avc transmission, in *Proc. 6th IEEE Consumer Communications and Networking Conf. CCNC 2009*, 2009, pp. 1–5.
- [25] H. Wang, S. Xiao and C.-C.J. Kuo, Random linear network coding with ladder-shaped global coding matrix for robust video transmission, *J Visual Communication and Image Representation* **22**(3) (2011), 203–212.
- [26] I. You, Y. Hori and K. Sakurai, Enhancing svo logic for mobile ipv6 security protocols, *Journal of Wireless Mobile Networks, Ubiquitous Computing, and Dependable Applications* **2** (2011), 26–52.
- [27] T.A. Zia and A.Y. Zomaya, A lightweight security framework for wireless sensor networks, *Journal of Wireless Mobile Networks, Ubiquitous Computing, and Dependable Applications* **2** (2011), 53–73.
- [28] H. Ha and C. Yim, Layer-weighted unequal error protection for scalable video coding extension of h.264/avc, *IEEE Trans Consum Electron* **54**(2) (2008), 736–744.

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