

QoS Assured Multi-rate H.264 Scalable Video Multi-cast with Network Coding in Lossy Networks

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ABSTRACT

This paper proposes a new multi-rate H.264 scalable video multicast in lossy network using network coding. We first prioritize video layer based on its effect to the end-to-end video quality. Each video layer is routed via the path obtained from the optimization framework under the constraints on quality of service (QoS) guarantees. The data transmissions of lower layers, which are more important for decoding process in video reproduction, are assigned to more reliable paths with QoS guarantee to further improve user's experience. Different destinations may receive different numbers of video layers depending on their network resources. The bottleneck in the network is resolved by using network coding technique to ensure that all destinations can receive the data rates equaling their theoretical data rates, i.e., max flows. The network coding is only applied within the same layer to ensure its QoS guarantee. Simulation and numerical results under randomly generated networks show the advantage of the proposed scheme in terms of objective and subject qualities of the end-to-end video transmission.

Keywords: Scalable Video Coding, Network Coding, Lossy Network, Linear Optimization, Quality of Service

1. INTRODUCTION

The popularity of video applications over wired and wireless networks has been increasing significantly due to the increasing available bandwidth and expansion of communication infrastructure. However, there are several challenging issues in multicasting video contents to customers. First, end users are heterogeneous by nature because their different network connections and display resolutions on their equipment. Second, networks are generally lossy and compressed bit streams are susceptible to the errors. One bit errors may deteriorate video quality greatly. Third, the network may contain bottleneck, which prohibits customers from receiving proper video qualities they deserve. Based on the mentioned chal-

lenges, Scalable Video Coding (SVC) [1] becomes an attractive choice for video transmission in lossy and heterogeneous environment because of its capabilities in rate/quality adjustment and error resilience [2].

There are many previous works proposing various methods for multicasting data to end users using network coding [3], when considering networks are lossy and contains bottlenecks. Zhu *et al.* [4] and Zhao *et al.* [5] proposed a method to improve the throughput of overlay multicast session by using the network coding. However, the packet loss rate was not considered. Thomas *et al.* [6] presented the path and source diversities in transmitting video contents using raptor code and network coding. Sundaram *et al.* [7] proposed multi-rate multicast using layered source coding. Network coding is used to apply the maximum transmission rate to all receivers according to their max-flows. Kim *et al.* [8] proposed a pushback algorithm for multi-resolution multicast giving the maximum total rate achieved by all receivers while guaranteeing decoding probability of the base layer at each receiver. However, both [7] and [8] did not consider lossy network and QoS guarantee. Supitayapornpong *et al.* [9] used an optimization framework for multicasting layered data with QoS guarantee. They included a QoS guarantee constraint in their newly formulated optimization problem. However, these previous works did not cover the multi-rate multicasting layered data with network coding, where QoS guarantee and packet losses are considered.

This paper proposes a new multi-rate H.264 scalable video multicast in lossy network using network coding. We first prioritize video layer based on its effect to the end-to-end video quality. Each video layer is routed via the paths obtained from the optimization framework under the constraints on QoS guarantees. Different destinations may receive different numbers of video layers depending on their max flows. The bottleneck in the network is resolved by using the network coding to ensure that all destinations can receive the rate equaling their max flows. The network coding is only applied within the same video layer.

The rest of this paper is organized as follows. Section 2 presents the video layer prioritization method. Section 3 describes the QoS assured multi-rate video multicast using network coding. Experimental results

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of our proposed scheme compared with no QoS-aware routing are in Section 4. Finally, conclusion of this paper is in Section 5.

2. H.264 SCALABLE VIDEO CODING LAYER PRIORITIZATION

H.264 SVC consists of three types of scalability: Spatial, Temporal, and SNR scalabilities [1]. Spatial, Temporal, and SNR scalabilities describe the subsets of bitstreams representing video contents with a reduced picture size, frame rate, and quality, respectively. SVC bit streams comprise of an array of NALUs packetization, which are grouped into scalable layers, each of which is identified by (d_{id}, q_{id}, t_{id}) identifier [1], where d_{id} (dependency_id) indicates a layer with separable SPS and spatial or CGS layer, q_{id} (quality_id) indicates the quality refinement layer, and t_{id} (temporal_id) indicates temporal resolution. Generally, base layer is an important layer that contains more quality of video contents than enhancement layers. We simply prioritize the scalable layer in the NALUs packets based on only its SNR scalability. The more important layer will be allocated to the more reliable network path compared with that of less important layer. Video layer i is more important than video layer j , when $i < j$. Therefore, the base layer will be given the highest priority. Let Pr_i be the reliability of the path used to convey data layer i . We allocate more important video layers to the path with higher reliabilities, which can represent as

$$\text{Pr}_i \geq \text{Pr}_j, \quad i \leq j. \quad (1)$$

3. QOS ASSURED MULTI-RATE MULTICAST WITH NETWORK CODING

3.1 System Model

Lossy network is modeled by a directed graph $G(N, E)$, where N and E are the sets of nodes and links in the network, respectively. The network has a single source, denoted by S , where D is a set of destination nodes of a multicast transmission. For generality, we define a multicast session as a set of unicast sessions (s, d) , where $d \in D$.

For link $l \in E$, let $t(l)$ and $r(l)$ be the transmitter and receiver nodes of link l , respectively. For each node $n \in N$, let $T_O(n)$ and $T_I(n)$ be a set of outgoing and incoming links of node n , respectively. Define Γ be a set of unicast sessions, where $(s, d) \in \Gamma$.

Each link capacity, which is denoted by c_l , is an integer, where $c_l > 1$. A link with capacity $c_l > 1$ is considered as c_l unit-capacity links in parallel. Link l has a probability of packet loss denoted by p_l , where $0 \leq p_l \leq 1$. Each $(s, d) \in \Gamma$ transmits M layers of data, where L_i is the i^{th} layer with transmission rate r_i . Let the set of layer indices of all (s, d) be I_M , where $I_M = \{0, 1, 2, \dots, M-1\}$.

The probability of a successful packet transmission for L_i , also called the reliability and denoted by $P_i^{(s,d)}$, can be expressed as

$$P_i^{(s,d)} = \prod_{l \in E} (1 - p_l)^{f_{l,i}^{s,d}}, \quad (2)$$

where $f_{l,i}^{s,d}$ indicates whether or not link $l \in E$ is used to transmit a packet of L_i . If it is used, $f_{l,i}^{s,d} = 1$. Otherwise, $f_{l,i}^{s,d} = 0$.

3.2 Optimal Routing Path Selection for Layered Multi-cast

An optimization framework is proposed to select the optimal set of routing paths for a multicast transmission while taking the QoS guarantee into account.

1) *Objective Function*: Let $x_i^{s,d}$ be an indicator variable, where $x_i^{s,d} = 1$ indicates that the packets of the layer L_i are transmitted with QoS guarantee. Otherwise, $x_i^{s,d} = 0$. We define the information value of each layered data as k_i , where $k_i \geq k_j$, when $i > j$. The data of lower layers therefore gain higher priority than that of higher layers. Based on the defined variable, the objective function is to maximize

$$\sum_{i \in I_M} \sum_{(s,d) \in \Gamma} k_i x_i^{s,d}. \quad (3)$$

The objective function is a summation of information values of layers that can be transmitted to each destination. The maximized objective value implies that the number of transmitted layers considering their priorities is maximized.

2) *QoS Guarantee Constraint*: A QoS requirement of L_i of (s, d) is denoted by $q_i^{(s,d)}$, so each destination can have individual QoS requirement, where $0 \leq q_i^{(s,d)} \leq 1$. Based on (2), the constraint on packet transmission of L_i of (s, d) with QoS guarantee can be expressed as

$$P_i^{(s,d)} = \prod_{l \in E} (1 - p_l)^{f_{l,i}^{s,d}} \geq q_i^{(s,d)}. \quad (4)$$

Since (4) leads to a nonlinear optimization which consumes computational complexity greater than that of linear optimization, we simply take the logarithm on both sides of (4), then we obtain

$$\sum_{l \in E} f_{l,i}^{(s,d)} \bar{p}_l \geq \bar{q}_i^{(s,d)}, \quad (5)$$

where $\bar{p}_l = \log(1 - p_l)$ and $\bar{q}_i^{(s,d)} = \log q_i^{(s,d)}$.

3) *Prioritizing Reliability Constraint*: Because different layers of scalable video differently contribute to the end-to-end video reproduction qua-

lity, transmitting packets of these layers should be carefully treated based on their priorities. More important layers will be assigned to the paths with better reliabilities compared with those of less important layers. The constraint of prioritizing reliability constraint can be expressed as

$$\sum_{l \in E} \bar{p}_l f_{l,i}^{(s,d)} \geq \sum_{l \in E} \bar{p}_l f_{l,i+1}^{(s,d)}. \quad (6)$$

4) *Problem Formulation:* To calculate the optimal routing paths of all $(s, d) \in \Gamma$, an integer linear optimization problem is formulated as follow.

Maximize

$$\sum_{i \in I_M} \sum_{(s,d) \in \Gamma} k_i x_i^{(s,d)}, \quad (7a)$$

Subject to

$$\sum_{l \in T_0(n)} f_{l,i}^{(s,d)} - \sum_{l \in T_1(n)} f_{l,i}^{(s,d)} = \begin{cases} r_i x_i^{(s,d)}, n = s \\ -r_i x_i^{(s,d)}, n = d \\ 0, \text{ otherwise} \end{cases} \quad (7b)$$

$$\forall i \in I_M, \forall (s, d) \in \Gamma, \forall n \in N,$$

$$\sum_{l \in E} f_{l,i}^{(s,d)} \bar{p}_l \geq \bar{q}_i^{(s,d)}, \forall i \in I_M, \forall (s, d) \in \Gamma, \quad (7c)$$

$$\sum_{l \in E} f_{l,i}^{(s,d)} \bar{p}_l \geq \sum_{l \in E} f_{l,i+1}^{(s,d)} \bar{p}_l, \forall i \in I_{M-1}, \forall (s, d) \in \Gamma, \quad (7d)$$

$$f_{l,i}^{(s,d)} \leq t_{l,i}, \forall (s, d) \in \Gamma, \forall i \in I, \forall l \in E, \quad (7e)$$

$$\sum_{i \in I} t_{l,i} \leq 1, \forall l \in E, \quad (7f)$$

$$x_i^{(s,d)} \geq x_{i+1}^{(s,d)}, \forall i \in I_{M-1}, \forall (s, d) \in \Gamma, \forall l \in E, \quad (7g)$$

$$f_{l,i}^{(s,d)} \in \{0, 1\}, \forall i \in I_M, \forall (s, d) \in \Gamma, \forall l \in E, \quad (7h)$$

$$t_{l,i} \in \{0, 1\}, \forall i \in I_M, \forall l \in E, \quad (7i)$$

$$x_i^{s,d} \in \{0, 1\}, \forall i \in I_M, \forall (s, d) \in \Gamma, \quad (7j)$$

In the above formulated problem, $t_{l,i}$ is the total capacity on the link l used by the i^{th} layer and $I_{m-1} = \{0, 1, 2, \dots, I_M - 2\}$. Constraint (7b) is the flow conservation constraint. Constraint (7c) is the QoS guarantee constraint. Constraint (7d) is the prioritizing QoS guarantee constraint. Constraints (7e) and (7f) follow the definition of $t_{l,i}$. These two functions establish a link capacity constraint supporting the network coding condition. They allow different (s, d) flows of the same layer L_i to share the link capacity instead of competing for their individual data rates. The intra-layer network coding is therefore performed. Constraint (7g) is the layered data constraint. Constraint (7h), (7i), and (7j) are the possible values of $f_{l,i}^{(s,d)}$, $t_{l,i}$, and $x_i^{(s,d)}$, respectively.

At the end of our algorithm, the optimal set of paths is calculated. These paths will be the input for network code assignment using random network

coding. According to the lower bound of successful decoding probability proposed in [10], which relies on the field size $|F|$ used by randomly selecting the elements of global encoding vectors, we can determine the lower bound of throughput including the effects of successful decoding probability. Nevertheless, we assume that $|F|$ is large enough and the dimension of global encoding vectors is small compared with $|F|$. So, the limit of successful decoding probability converges to 1. We only use network coding within each layer to ensure the QoS guarantee to SVC video transmission.

4. EXPERIMENTAL RESULTS

There are 3 parts of the experiment. First, to illustrate our proposed scheme, we generate an example network as shown in Fig.1(a), where each link has a unit capacity. Each unit capacity is 512 kbps. Table in Fig.1 shows successful transmission probability of each link. Node 1 is a source node and destination nodes are nodes 10, 11, and 12. The video data are encoded 2 layers, i.e., L_0 and L_1 , where their data rates are 2 and 1 unit(s), respectively. The QoS guarantees of L_0 and L_1 in terms of successful transmission probability are 0.90 and 0.86, respectively.

Second, we compare our proposed scheme with the no QoS-aware routing and simulate both two routing schemes on 20 randomly network topologies containing 20 nodes: 1 source and 3 destination nodes. The successful transmission probability of each link is randomly selected. The transmission rates of L_0 , L_1 , and L_2 are 2, 1, and 1 unit(s), respectively. The imposed QoS requirements of L_0 , L_1 , and L_2 for all destinations are 0.90, 0.85, and 0.80, respectively. Python programming language [11] is used with the PuLP [12] package and GLPK [13] solver to calculate the optimal route solution for each scheme.

Finally, JSVM 9.8 reference software [14] is used to encode and decode test sequence. The test sequence is bus.yuv(CIF, 4500 frames) encoded with GOP size equal to 8 and frame rate is 30 fps. The coded stream consists of 1 base layer (L_0) and 2 enhancement layers (L_1 and L_2) with SNR scalability, where the imposed QoS requirements of L_0 , L_1 , and L_2 for all destinations are 0.90, 0.85, and 0.80, respectively.

4.1 Numerical Results with/without Our Proposed Framework in Toy Example

Consider Fig.1., the optimal routing solution is solved by our optimization framework, where each link capacity is allocated for data transmission. Fig.1(b) shows the subgraphs for layers 0 and 1, where the subgraph of layers 0 and 1 are represented by solid lines and dash lines, respectively. At the source node, L_0 is split equally to $L_{0,1}$ and $L_{0,2}$ for transmission. Because of the network topology, nodes 10, 11, and 12 can receive L_0 with reliability 0.9208, 0.9025,

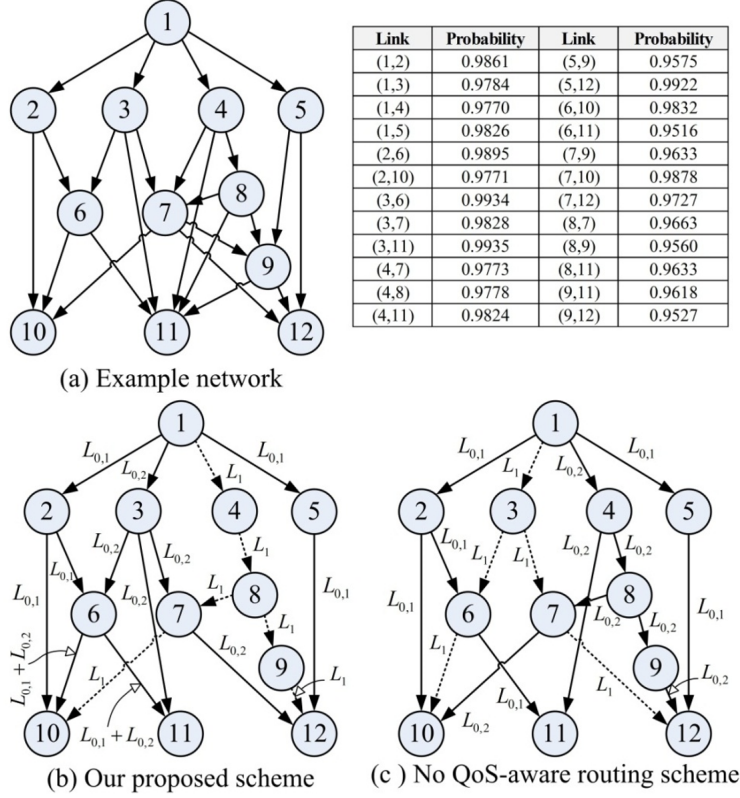


Fig.1: An Example Network: (a) An Original Topology, (b) Subgraphs with QoS Guarantee from our Framework, and (c) Subgraphs without QoS Guarantee.

and 0.9118, respectively. Nodes 10 and 12 can receive L_1 with reliability 0.9117 and 0.8701, respectively. The results demonstrate the multi-rate multicast, where the received rate at each destination is limited by its max flow.

Table 1: Comparison of the reliability obtained by our proposed scheme and No QoS-aware routing.

Destination	Proposed Scheme	No QoS-aware routing
D1(L_0, L_1, L_2)	(0.9085, 0.8793, 0.8463)	(0.7996, 0.8610, 0.9029)
D2(L_0, L_1, L_2)	(0.9096, 0.8822, 0.8426)	(0.7927, 0.8518, 0.8880)
D3(L_0, L_1, L_2)	(0.9067, 0.8750, 0.8429)	(0.7621, 0.8909, 0.8996)

When we simulate the results without our proposed framework, QoS guarantee is not considered. We remove the constraints (7c) and (7d) from the optimization framework. Then the result subgraph of L_0 and L_1 are shown in Fig.1(c). Reliabilities of L_0 at nodes 10, 11, and 12 are 0.8785, 0.8911, and 0.8482, respectively. Reliabilities of L_1 at nodes 10 and 12 are 0.9557 and 0.9353, respectively. As we can see, when we do not consider QoS guarantee, the required reliability of L_0 may not be achieved.

4.2 Simulation of Successful Transmission Probabilities from the Optimization Framework

Twenty topologies are randomly generated for the numerical simulation. Table. I shows the simulation

results of our proposed scheme compared to that of the no QoS-aware routing. These results are average results from 20 random topologies, representing the successful transmission probabilities of L_0 , L_1 , and L_2 of all destinations. It shows that more important layers will be allocated to high reliable paths. Then, lower priority layers will be eligible to use the remained network resources.

4.3 Random Packet Loss Simulation Results

We apply video transmission over the lossy network under 20 random topologies. We consider only SNR scalability. To decode these streams, we use BLSkip error concealment and measure the results using average PSNR of all frames appearing in the sequence.

Fig.2(a) shows the average PSNR of decoded videos at each destination from 20 random topologies. Our proposed scheme provides a higher average PSNR than that of no QoS-aware routing scheme with respect to all successful transmission probabilities simulated in the experiment. The proposed scheme gives an average PSNR of 31.88 dB, 31.84 dB, and 31.86 dB.

Fig.2(b) shows a comparison between average numbers of lost frames in decoded videos at each destination from 20 random topologies using our proposed scheme and those from using the no QoS-aware

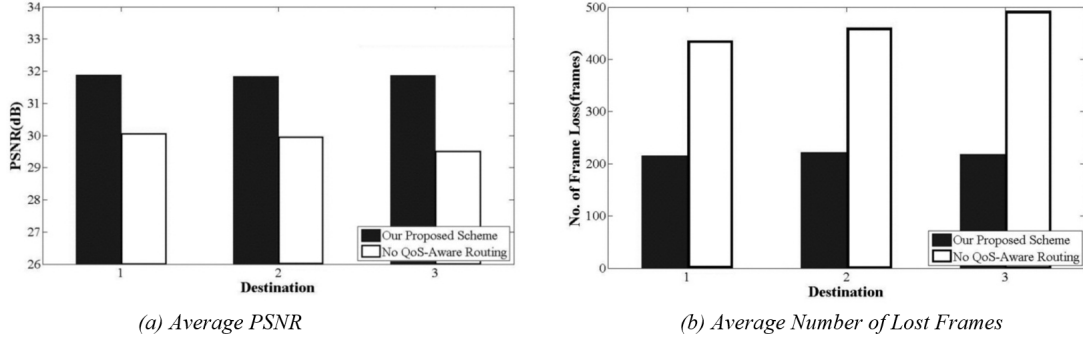


Fig.2: The Objective Qualities of the Reproduced Video Resulted from Different Routing Schemes: (a) Average PSNR and (b) Average Number of Lost Frames.

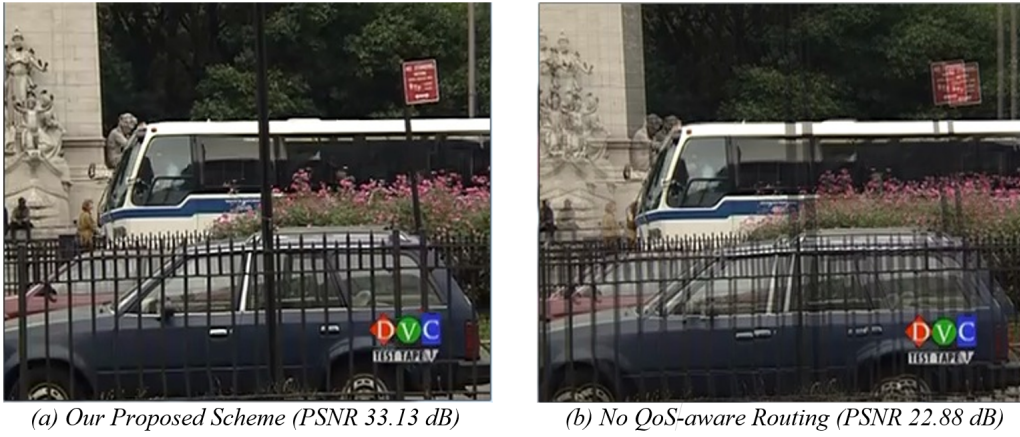


Fig.3: Subjective Quality Comparison of a Sample Frame: (a) Our Proposed Scheme (PSNR 33.13 dB), (b) No QoS-aware Routing (PSNR 22.88 dB).

routing scheme. The number of lost frames from using our proposed scheme is obviously less than that of the no QoS-aware routing scheme. The average numbers of lost frames for each destination are 215.7, 221.75, and 218.3 frames, respectively.

To evaluate the subjective quality of video content, we randomly select a sample video frame from the decoded video at destination 1 according to transmission reliabilities obtained from each routing scheme. The sample frame resulted from using our proposed routing scheme is shown in Fig.3(a) compared with those from the no QoS-aware routing scheme shown in Fig.3(b).

5. CONCLUSION

A QoS assured multi-rate H.264 multicast routing scheme using network coding is proposed. A linear optimization framework is formulated to obtain transmission paths with optimal reliability under several constraints including QoS guarantee constraint and prioritizing reliability constraint. The data transmission of a based layer, which is the most important layer for video reproduction, is assigned to more reliable paths with QoS guarantee to ensure an accept-

able quality of end-to-end video transmission. Our Experimental results, which are performed based on numerical experiments, show the significant improvements in both subjective and objective video qualities compared with those of the scheme without considering QoS guarantee.

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References

- [1] H. Schwarz, D. Marpe, and T. Wiegand, "Overview of the Scalable Video Coding Extension of the H.264/AVC Standard," *IEEE Transactions on Circuits and Systems for Video Technology*, Vol. 17, No.9, pp.1103-1120, 2007.
- [2] Y. Guo, Y. Chen, Y.K. Wang, H. Li, M.M. Hanuysela, and M. Gabbouj, "Error Resilient Coding and Error Concealment in Scalable Video Coding," *IEEE Transactions on Circuits and Systems for Video Technology*, Vol. 19, No.6, pp.781-795, 2007.

- [3] R. Ahlswede, N. Cai, S. Y. R. Li, and R. W. Yeung, "Network Information Flow," *IEEE Transactions on Information Theory*, Vol. 46, No.4, pp. 1204-1216, 2000.
- [4] Y. Zhu, B. Li, and Jiang Guo, "Multicast with Network Coding in Application-layer Overlay Networks," *IEEE Journal on Selected Areas in Communications*, Vol. 22, No.1, pp. 107-120, 2004.
- [5] J. Zhao, F. Yang, Q. Zhang, Z. Zhang, and F. Zhang, "LION: Layered Overlay Multicast with Network Coding," *IEEE Transactions on Multimedia*, Vol.8, No.5, pp. 1021-1032, 2006.
- [6] N. Thomas and P. Frossard, "Network Coding of Rateless Video in Streaming Overlays," *IEEE Transactions on Circuits and Systems for Video Technology*, Vol. 20, No.12, pp.1834-1847, 2010.
- [7] N. Sundaram, P. Ramanathan, and S. Banerjee, "Multirate Media Stream using Network Coding," in *Proc. 43rd Annual Allerton Conference on Communication, Control, and Computing*, 2005.
- [8] M. Kim, D. Lucani, Xiaomeng Shi, F. Zhao, and M. Medard, "Network Coding for Multi-resolution Multicast," *INFOCOM 2010*, pp.1-9, 2010.
- [9] S. Supittayapornpong, P. Saengudomlert, and W. Kumwilaisak, "A Framework for Reliability Aware Layered Multi-cast in Lossy Network with Network Coding," *Computer Communications*, Vol. 33, No.14, pp.1651-1663, September 2010.
- [10] T. Ho, M. Medard, R. Koetter, D. R. Karger, M. Effros, J. Shi, and B. Leong, "A Random Linear Network Coding Approach to Multicast," *IEEE Transactions on Information Theory*, Vol.52, No.10, pp.4413-4430, 2006.
- [11] <http://www.python.org>.
- [12] <http://www.coin-or.org/PuLP/>.
- [13] <http://www.gnu.org/software/glpk/>.
- [14] "JSVM reference software (9.8)," <http://ube.ege.edu.tr/boztok/JSVM/SoftwareManual.pdf>

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