QoS Improvement for Video Streaming over MANET Using Network-Coding

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Abstract—Video streaming (like YouTube) services and related applications become more and more widespread. Therefore, video streaming delivery over a mobile ad-hoc network (MANET) becomes a necessity as an important content delivery infrastructure between the user (content consumer) and the content storage node. Furthermore, user mobility impacts the quality of the delivered video and hence new concepts should be considered. Accordingly, the innovative concept of network coding (NC) emerges as a promising approach for improving the video transmission quality mainly in multicast environment. In this paper, we focus on Quality of Service (QoS) improvement for video streaming over MANET using random network coding. Basically, we consider video coded by H264/SVC codec that generates packets with different priorities and uses the IEEE 802.11e MAC for traffic differentiation. A successful transmission of high priority packets leads to enhance the video transmission quality. Accordingly, we propose a transmission scheme to protect high priority packets from being lost. Our approach, named Multicast Scalable Video Transmission using Classification-Scheduling Algorithms and Network Coding over MANET (and denoted MSVT_CSA_NC), adopts a cross layer solution between the H.264/SVC codec, the network and MAC layers. Moreover, our delivery mechanisms based on random network coding ensure high throughput and low network load over MANET. Simulation results confirm the substantial performance improvement brought by our approach.

Index Terms—H.264/SVC, Cross-layer, MANET, Multicast, Network Coding.

I. INTRODUCTION

Mobile ad-hoc networks (MANET) is a self-configured set of mobile nodes using multi-hop paths to communicate between each other. Mobility represents the most vexing problem in MANET resulting in an unstable network topology. In addition, communication between nodes is based on broadcasting which may produce many redundant packets and create a broadcast storm problem. To overcome this issue, new approaches based on network coding (NC) have been proposed. NC replaces the traditional techniques of store and forward with a new one combining many packets in one coded packet. It improves greatly the wireless networks performance. Mainly, it reduces the number of transmissions in the network resulting in less delay and higher throughput.

The concept of network coding is widely investigated for the network layer (IP) [1] to improve the network throughput in multicast wireless communication. There is also some work about applying NC at the physical layer [2] or at higher layers, such as the transport or application layers [3]. Also, some papers propose modifications at MAC layer of IEEE 802.11 in order to implement NC [4].

Many research works have been done for applying NC in MANET. NC is firstly introduced by [5] as a novel transmission technique for wired multicasting. Authors prove that NC may potentially achieve higher throughput than routing solutions by allowing linear combination of packets at intermediate nodes. Theoretically, linear network coding scheme has been studied by many researchers [6] [7] to achieve the same objective as [5] over multicast transmission. Also, several studies have proposed and investigated random network coding [8] [9]. Because of bandwidth constraint and dynamic topology of MANET, multicast and multi-hop supported routing are very important research issues.

Additionally, to enable an efficient video streaming transmission, scalable video coding (SVC) is an attractive factor applied to original video streaming. H.264/SVC [10] offers a better way of effectively compressing video. It provides three kinds of scalabilities: temporal, spatial and quality. According to H.264/SVC, packets are assigned different priorities. Packets with high priority are the most responsible in guaranteeing high video transmission quality.

Authors in [11] present a random network coding for MANET named CodeCast. This scheme implements NC in all the nodes to achieve high throughput with low overhead and low latency. Added to that, it increases the probability to recover the original packets but it leads to high delay. In fact, when a node receives a block of packets, it decodes, re-encodes and forwards them to its neighbors. Additionally, high priority packets have the same loss probability than low priority packets. Authors in [12] proposed a network coding-based real-time multicast (NCRM) protocol for MANET to reduce the energy consumption. Their method combines the key idea of the protocol for unified multicasting through announcement (PUMA) proposed in [13] with random linear network coding (RLNC). PUMA is a multicast routing protocol that suppresses the unnecessary packets in order to reduce the overhead in the network and the end-to-end-delay. NCRM can reduce the energy consumption and also improve the throughput performance. Moreover, NCRM is shown to be robust in high mobility and high density scenarios. However, this approach sends the whole block of coded packets simultaneously which may lead to large delay, serious network congestion and low packet delivery ratio.
In the light of the sensitivity of low packet delivery ratio and low delay, authors in [14] proposed a new scheme combining partial network coding proposed in [15] and the real-time PUMA (named PNCRM). It is based on RLNC but each vector of packets is transmitted partially. PNCRM increases significantly the reliability and throughput but the performance of PNCRM decreases under the multicast scenario with high traffic load. They conclude that combining PNCRM and NCRM provides optimal performance.

According to this literature survey, there are several approaches for implementing NC in MANET. Although, it is sometimes impossible to update frequently the network topology, mainly when the mobility is highly dynamic. Hence, RLNC is more suitable and it is a performant approach.

Although NC offers higher throughput and better loss protection capability, it does not guarantee to recover high priority packets. It is worth mentioning that priority packet loss is a basic QoS parameter for video coding in MANET to achieve high level of performance; that is, high priority packet loss usually comes at a cost of low QoS. Therefore, minimizing high priority packet loss is the focus of this paper. To tackle the priority packet loss problem for real time traffic in MANET, we propose a two-phase scheme: i) Source-node phase and ii) Intermediate node phase. The source-node phase is based on a classification algorithm to adjust the block size parameter of RLNC. The intermediate-node phase focuses on enhancing the RLNC and making it able to form blocks according to packets priorities.

Additionally, some packets could be lost so that the Global Coefficient Matrix (GCM) [12] associated with RLNC cannot have the full rank for inversion to decode packets in the receiver nodes. Moreover, we consider random NC as a basic approach to ensure high throughput and low network load over MANET.

This paper considers video H.264/SVC encoding that generates packets with different priorities. H.264/SVC is detailed in [10].

The remainder of this paper is organized as follows. In section II, the proposed transmission scheme is outlined. Section III, illustrates the evaluation of our contribution through simulations. Finally, section IV draws the concluding remarks.

II. PROPOSED TRANSMISSION SCHEME (MSVT_CSA_NC)

In the following, the new proposed transmission scheme is detailed. First, we present the proposed packet classification at the source, then the proposed modification on random NC and finally we detail the block scheduling algorithm at MAC layer of intermediate nodes.

A. Source node: SVC packet classification

In this work, it is assumed that the application layer generates equal size SVC packets. In traditional RLNC, the intermediate node is responsible of receiving native packets from the source node and groups them into different b blocks, each of p packets. The RLNC generates equal size SVC packets. In traditional RLNC, the intermediate node is responsible of receiving native packets from the source node and groups them into different b blocks, each of p packets. Packets belonging to a block may have different priorities. Also, in traditional RLNC a high number of out-of-blocks packets may exist. In addition, the priority of out-of-block packets may be high. In this case, some high priority packets are lost, which decreases the transmission quality.

In this work, our main issue before encoding packets is to choose the appropriate size of blocks minimizing the number of out-of-block packets having high priority. First, we propose to classify the packets according to the value of (DID, TID, QID) triplet. Hence, the priority of each packet can be easily identified. For example, with the temporal scalability, the TID field is coded on three bits, leading to 2^3 queues having different priority levels \((i.e.(0,0,0),(0,1,0)\ldots(0,7,0))\). Packets belonging to the same queue have the same priority. Next, we propose to calculate the appropriate block size (denoted \(b_{sz}\)) that minimizes the loss of highest priority packets (base layer).

Let \(N\) denote the number of SVC generated packets and \(L\) the number of priority levels \(\{L_1,L_2,\ldots,L_L\}\). We assume that a level \(L_i\) has \(n_i\) packets. Let \(r_i\) be the number of reminded packets in level \(L_i\) and we assume that \(b_{sz} \in \{3,\ldots,12\}\). The \(b_{sz}\) giving the minimum out-of-block packets, \(r_i\), is defined by:

\[
n_i \mod b_{sz} = r_i
\]  

As a result, level \(L_1\) represents the highest priority level. This level has \(b\) blocks. The last formed block of each level \(L_i\) is noted \(block_{ib}\). The size of the \(block_{ib}\) may be less than the defined block size \(b_{sz}\). In each level \(L_i\), these packets are supposed to be lost since they can’t form a block. In this situation, the probability \(P_{loss(L_i)}\) of packet loss for the level \(L_i\) is given by:

\[
r_i \left\lfloor \frac{(b-1)b_{sz} + r_i}{b} \right\rfloor
\]

Since increasing the number of high priority coded packets guarantees better QoS, we propose a scheme that automati
cally forms an additional block for the different priority queues starting from the highest priority level which is \(L_1\). The proposed scheme assigns the number \(add_i\) packets from the level \(L_i+1\) to the queue \(L_i\), where the priority of \(L_i\) is immediately higher than \(L_i+1\):

\[
add_i = b_{sz} - r_i
\]

As consequence, the number out-of-block packets \(r_i\) in queue \(i\) becomes zero \((r_i = 0)\). Therefore, the probability of packet loss for the priority queue 1 becomes zero and all \(n_i\) packets are coded. Fig. 1 illustrates our proposed priority-based packet loss reduction mechanism with \(b_{sz} = 4\). For example, for 17 high priority packets, our algorithm chooses blocks of size 4 resulting into only 1 out-of-block packet \((i.e.17 \mod 4 = 1)\). So, \(3\) packets \((add_i = 3)\), from the level \(i + 1\) are added to the queue \(i\) having the highest priority.

By this way, it is guaranteed to have the minimum number of non-encoded highest priority packets \((P_{loss(L_1)} = 0)\).

We propose to add a new field in the header structure of each original packet called block size.
The coded packet format consists of the header and data fields. In our proposed coded packet format, the header is constructed from two major fields, called, Code_Priority and Code_Type. The Code_Priority, a 2-bits field, designs the priority level of each coded packet. Its priority level values are in the range of 0 – 2. The coded packets can also be identified by its type named: “PT_Coded”

Case 2: If the packet_type field interpreted by the intermediate node is PT_Coded, then, as shown in equation 7, the receiver node can easily decode the input block of packets.

\[
\begin{pmatrix}
P_1 \\
\vdots \\
P_{b_{sz}}
\end{pmatrix}
= 
\begin{pmatrix}
\alpha_{1,1} & \cdots & \alpha_{1,b_{sz}} \\
\alpha_{2,1} & \cdots & \alpha_{2,b_{sz}} \\
\vdots & \ddots & \vdots \\
\alpha_{k,1} & \cdots & \alpha_{k,b_{sz}}
\end{pmatrix}
\times 
\begin{pmatrix}
c_{1,1} \\
\vdots \\
c_{k,b}
\end{pmatrix}
\]  

(7)

If this node is not a final destination, NC offers its great ability of recoding. Thus, all decoded packets are stored in a local memory and a new combination of packets is generated. Then, this node transmits recoded packets to another node. Algorithm 1 resumes the intermediate node processing.

**Algorithm 1: INTERMEDIATE NODE PROCESSING**

**Input:** (packet \* p)  
1. if p.packet_type == "PT_Original" and N! = 0 then  
   a. if p.block_size == 2 then  
      1. p.block_size = encoded_packets(p)  
      2. Dequencing_packets()  
      3. N ← N − p.block_size  
      4. S = \(\sum_{i=1}^{\text{p.block_size}} \sum_{j=1}^{\text{p.packet.size}} p(i,j)\)  
      5. if S ≥ N then  
         a. Coded_packet = encoded_packets(p)  
         b. high_priority_of_rear_packets.  
      6. if p.packet_type == "PT_Coded" then  
         a. p1, p2, ..., p_{b_{sz}} = decoded_packet(Coded_packet)

2) Block scheduling (MAC Layer): The IEEE 802.11e standard is designed to enhance the MAC mechanism by providing a QoS method called Enhanced Distributed Channel Access (EDCA). EDCA has four transmission queues: AC_BK (for background traffic), AC_BE (for best effort traffic), AC_VI (for video traffic) and AC_VO (for voice traffic). EDCA doesn’t take in consideration the packet priority generated by the SVC coder. In this paper, we propose to adapt IEEE 802.11e EDCA with SVC blocks (produced by NC). In fact, we propose to assign the AC_VO, AC_VI, and AC_BE to SVC blocks with priority \(P_r = 0\), \(P_r = 1\), \(P_r = 2\) respectively.

### III. SIMULATION ENVIRONMENT AND RESULTS

This section is dedicated to evaluate the performance of our propositions by simulations. First, the performance of Codecast [11] is evaluated using SVC video transmission which will serve as a benchmark for our work. Next, the performance of the cross layer MSVT_CSA_NC is evaluated in the context of mobile nodes.
A. Simulation environment and input parameters

To evaluate the performance of our proposed scheme, we consider the network environment shown in Fig. 2.

The evaluation of MSVT_CSA_NC has been carried out with NS-2 simulator. Video streams are coded with version JSVM 9.18 of the SVC reference software. Simulations consider 100 nodes following the Random-Waypoint mobility model, placed in an area of $(1500 \times 1500)$ m$^2$. It is assumed that all the nodes offer MAC level QoS according to the IEEE 802.11e standard. At the network layer, we restrict our attention to employ PUMA as the multicast routing protocol. The considered topology has one source node and 10 receivers. Video streaming is divided into different layers that provide different degrees of quality. The source node transmits three video is a “Foreman” CIF format sequence composed of 300 frames, coded at $30 \text{ frame/s}$ with a GOP (Group of pictures) of 8. The bit rate of the video sequences is assumed constant and it is set to $2 \text{ Mbps}$. All the measurements are made over a period of 120s. In order to generate statistically quality measures, each metric is averaged over 10 simulation runs with various random seeds.

![Fig. 2. Video transmission with the proposed algorithms over MANET](image)

B. Simulation results

In this subsection, we show the effectiveness and the performance of our proposed scheme using a simulated MANET in multicast environment. Fig. 3 compares the E2ED for MSVT_CSA_NC with both Codecast and the conventional multicast transmission without NC represented by PUMA, in MANET. It shows that our scheme provides acceptable AE2ED. Average E2ED for Codecast is higher than MSVT_CSA_NC. In fact, when high priority packets drop increases, more retransmissions are required. So the E2ED increases. In addition, each mobile user in IEEE 802.11e EDCA mode has 4 queues with different priority levels. The waiting time of a packet in a 4-queue node is less than the waiting time in a 1-queue node. Therefore, the associated queuing delay decreases.

![Fig. 3. Average delay: varying node speed.](image)

Where $r_{\text{max}}$ is the number of simulation iterations (in our simulations, $r_{\text{max}}= 10$) and ADSR is the average delay for each simulation run. ADSR is given by:

$$ADSR = \frac{\sum_{n=1}^{n_{\text{max}}} p(n) \cdot \text{Delay of each packet sent}}{\text{Total number of received packets}}$$

where $n_{\text{max}}$ is the number of destination nodes (Our simulations assume $n_{\text{max}}=10$) and $p(n)$ is the number of received packets by a given receiver $n$.

Equations (8) and (9) show the definition of the PSNR between the luminance component of source frame and destination frame.

$$PSNR = 10 \times \log(\frac{255^2}{\text{MSE}}), \quad (8)$$

where MSE is the mean square error defined by:

$$\text{MSE} = \frac{1}{w \times h} \sum_{i=1}^{w} \sum_{j=1}^{h} (A_{ij} - B_{ij})^2, \quad (9)$$

where $w$ and $h$ are the width and height of an image, respectively. Hence, $w \times h$ presents the number of pixels.

The AE2ED is defined by:

$$AE2ED = \frac{\sum_{r=1}^{r_{\text{max}}} (ADSR)}{r_{\text{max}}}$$

Equation (8) and (9) illustrate the video quality expressed in terms of PSNR versus frames number. In this simulation, 300 frames of Foreman sequence have been processed under three different works. Average PSNR value starts from $22.12\text{dB}$ for the conventional multicast transmission and increases up to $35.40\text{dB}$ and $37.07\text{dB}$ for Codecast and MSVT_CSA_NC respectively.

Fig. 5 shows the Average PSNR with different node mobility. It reveals that the average PSNR is high when the speed is very low. However, when the speed increases, average PSNR decreases slightly. Furthermore, the average PSNR of the decoded frames for the video sequence used in our performance study is always greater than the average PSNR of other approaches (Codecast and conventional multicast transmission without NC). This performance is due to the fact that MSVT_CSA_NC decreases the packet loss.
Mean Opinion Score (MOS) is a subjective metric used to measure the digital video quality at the application level. Thus, PSNR for video coding can be converted to MOS. The mapping between PSNR and MOS is given in [16]. Therefore, the PSNR value given by our scheme is considered Excellent. Whereas, Codecast provides a Good MOS compared to the conventional multicast which provides a Poor MOS.

Fig. 6 shows that the effect of node mobility is almost negligible. It also shows that MSVT_CSA_NC provides better and more consistent results than other approaches. In fact, when nodes are moving at a constant speed (20m/s), MSVT_CSA_NC allows a PDR close to 100%. Whereas, the PDR of the conventional multicast represented by PUMA degrades to 81%.

IV. CONCLUSION

In this paper, we have proposed a new cross layer approach named MSVT_CSA_NC based on network coding with dynamic choice of packet size at the application layer and two scheduling algorithms at the network and MAC layer. We have evaluated the performance of multicast transmission of video streaming in MANET. MSVT_CSA_NC can significantly improve the end-to-end-delay and the QoS of received video over multicast transmission. Additionally, the proposed scheme provides significant improvement on network performance under different node mobility.

As an extension of this work, we will investigate the utilization of MSVT_CSA_NC in multicast wireless network with multi source nodes using a variety of sequence videos. Moreover, H.264/SVC is open source. Hence, we will implement our framework in some mobile devices and test the quality of the results.

REFERENCES