The Use of Erasure Coding for Video Streaming Unicast Over Vehicular Ad Hoc Networks*

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***Abstract*—We tackle the issue of providing video streaming support over Vehicular Ad Hoc Networks (VANETs). Video streaming imposes stringent requirements in terms of delivery ratio and delay in order to provide a satisfying level of service at the user's end. In our work, we investigate the use of Erasure Coding to overcome packet loss and to fulfill video streaming requirements of delivery ratio. We have investigated two distinct coding techniques and evaluated thoroughly their peculiarities. This study has shown how Erasure Coding could improve delivery ratio without affecting end-to-end delay.**

I. Introduction

Video streaming from a source to one destination over Vehicular Ad Hoc Networks (VANETs) is useful for a variety of applications. A camera placed along roads or streets could capture a video from an accident and stream it to an incoming ambulance so paramedics can prepare their treatment even before reaching patients. The content sampled by surveillance cameras could be streamed to law enforcement agents for crime prevention or suspect pursuits.

Erasure Coding is a mechanism useful to increasing the delivery ratio in networks highly prone to packet loss such as VANETs. It is used by requesting the source node to transmit a larger amount of data than the minimum necessary for the assemble of all the video content at the receiver. In this manner, if part of the transmitted data is lost, the receiver is still able to obtain a high percentage of the original video.

We have evaluated two distinct coding techniques: Random Linear Coding (RLC) and XOR-based Coding. We could observe that for RLC the size of the block through which the original packets are divided defines the relevance of individual packets and has a direct impact on what the optimum values are depending on the level of packet loss in the network layer. In XOR-based Coding, packets that were encoded based on fewer portions of the original data were more useful than otherwise. We have also concluded that the use of XOR-based Coding outperforms that of RLC.

The use of coding techniques for the purpose of taking advantage of redundancy towards higher delivery ratios has been suggested in different ways before. In [1], the authors proposed a comprehensive protocol named Distributed Rate-Distortion method (DRD) where Raptor code [2] is used as a forwarding error correction (FEC) mechanism operating on the top of Scalable Video Coding streams. This solution builds an overlay network functioning on top of MANETs aimed at organizing the cooperation between peers and clients. They have shown promising results; however, their network model is composed of slow moving nodes (up to 3m/s) when compared to VANETs and video data rates are limited to 200 kbit/s. Yi et al. [3] also use a FEC mechanism to improve delivery ratios but they suggest that a multipath approach is able to decrease the frequency of collisions.

In this work, we perform a thorough investigation on the use of Erasure Coding for video streaming. We investigate the peculiarities of two different coding techniques and we analyze if their use is sufficient to overcome packet loss and reach the necessary delivery ratios for unicast video streaming.

II. Erasure Coding

We have looked into the use of additional redundancy as a suitable solution to overcome packet loss. This is performed by the source node which sends more information than the minimum necessary for receivers to assemble the whole content produced. The use of redundancy does not require any interaction between receiver and source nodes. This synchronistic decoupling is pertinent to video-streaming over VANETs as it can lead to higher delivery ratios with a low impact on end-to-end delay.

There are coding techniques that make an efficient use of the additional space that is used to carry redundant data. These coding techniques take the concept of entropy and by a perspective with the same basic idea of transmitting a larger amount of data than what is necessary to describe the whole content. By this manner, upon the partial reception of the sent data, receivers are able to assemble either the whole content or a large percentage of it.

A. Random Linear Coding (RLC)

Linear Coding [4] has been created to optimize the distribution of information in a multicast scenario but it can be adapted to add redundancy in a network in order to handle packet loss. In this work, for the use of linear coding, video content is coded with network coding by first dividing the original data into blocks. These blocks are then subdivided into η segments of same sizes. Segments (s) are then linearly combined using κ (with $\kappa \geq \eta$) sets of η coefficients (c) to create κ unique

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frames (to be included in the transmitted packets). Each one of these κ unique sets have to be linearly independent of all the other sets. Therefore, nodes that receive η of κ sent packets can decode the original data broadcast. In Random Linear Coding (RLC) [5], the coefficients used to form the vector that encode each segment are randomly chosen from a finite field.

An advantage of using RLC is that all sent packets have the same role in the decoding process. The reception of any η packets from the κ broadcast by the source allows receivers to decode all η original segments.

B. XOR-based Coding

There are many Erasure Coding solutions where coding of the data is based on XOR operations[6], [7] and the principle is built recursively on the property that with $C = A \oplus B$, A and B could be recovered by $C \oplus B$ and $C \oplus A$, respectively. As with RLC, this XOR-based Coding approach divides the original data to be sent into blocks composed of η segments. The data carried by packets ranges from non-encoded information to content assembled based on all η segments. The combination of the selected segments is done through XOR operations; and, when this encoded packet is received, it is possible to decoded it into original content if all but one of the combined segments are obtained by the receiver.

The number of segments used to form a packet is called its degree (d) and this influences greatly the role of the packet. The importance of the degree of a packet is related to how likely it can be used to generate original content at the receiver. A small degree means that the probability of it being decoded is greater; however, it has higher chances of being decoded only into content that has already been received. Therefore, the selection of the degree d has to balance the trade-off between the probability a packet has of being decoded and of the usefulness of decoded information to the receiver.

In our XOR-based Coding, the first η packets sending for each block are of degree 1 ($d = 1$), each contains an unique segment. In this way, if the additional redundancy τ is equal to zero, this solution behaves exactly the same as the underlying unicast solution. Our goal is to use the additional redundancy to handle eventual packet losses from the original data.

The degree of the remaining packets depends on the amount of additional redundancy τ. All extra packets should cover uniformly all η segments of each block. The number of additional packets is $\kappa - \eta$ and the additional redundancy τ is calculated as $\tau = \frac{\kappa}{\eta} - 1$. In this work, we have considered additional redundancy up to 100% and both the degree distribution and segment index selection depends on the indexes (i) of the extra packets. The idea is that up to 50% of redundancy (i.e. extra packet index smaller than $\kappa/2$), the packets combined are chosen based on a modular function of their indexes. When $\tau > 50\%$, the extra packets of index greater than $\kappa/2$ select packets sequentially. The degree distribution is given by:

$$d_i = \begin{cases} max(\lceil \eta/(\kappa - \eta) \rceil, 2) & \text{if } i < \lfloor 0.50\eta \rfloor \\ max(\lceil \eta/(\kappa - \eta - \lfloor \eta/2 \rfloor) \rceil, 2) & \text{otherwise} \end{cases} \tag{1}$$

And the choice of segments used for combination in each packet is either based on an interleaved or sequential pick. If the extra packet index i is less than half of the number of segments in each block, segments are selected in an interleaved manner depending on the modulo of their indexes j. Otherwise, the d_i consecutive segments are chosen. The test to determine if a segment s_j belongs to the set which will be combined to form the additional packet p_i can be described by the following equation:

$$s_j \text{ used at } p_i = \begin{cases} j = i \bmod d_i & \text{if } i < \lfloor 0.50\eta \rfloor \\ \lfloor j/d_i \rfloor = (i - \lfloor 0.50\eta \rfloor) & \text{otherwise} \end{cases} \tag{2}$$

III. Performance Evaluation

In this section, we evaluate the use of Erasure Coding by both Random Linear Coding (RLC) and XOR-based Coding techniques for video streaming over VANETs. This extensive evaluation has permitted us to analyze in detail the advantages and issues of using Erasure Coding.

For the evaluation of Erasure Coding, we need to make use of an underlying unicast protocol and we have chosen to use VIRTUS [8]. VIRTUS is a receiver-based solution where the selection of relay nodes is done for the duration of a time window rather than for every transmitted packet. This is highly adequate for video streaming as several consecutive packets are sent in a short period of time.

A. Simulation Environment

The deployment of these protocols over a real VANET is simply unfeasible at this point. Because of this, it was decided that its performance should be evaluated over a simulation environment. We have chosen to use the most widely accepted network simulator, namely the Network Simulator 2 (NS2) [9]. We have used a communication range of 300 meters and a bandwidth of 10Mbps. The video sent in the simulations is the well-known benchmark Akiyo_cif video encoded with H.264/MPEG-4 AVC 10 codec.

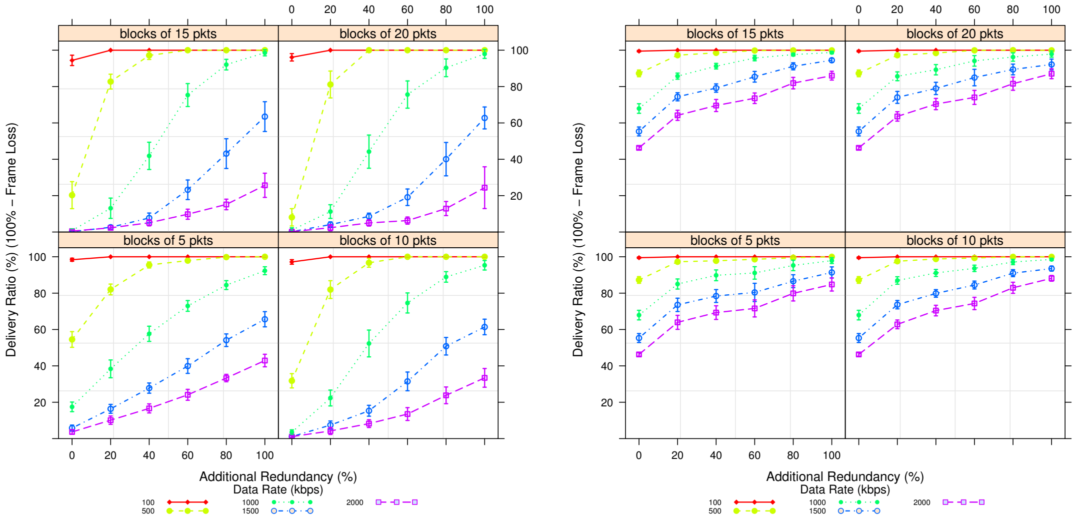
We have used the Freeway$^+$ [10] mobility model to emulate vehicle movement over a highway. The topology used was of a 12 km stretch of highway with 600 vehicles on it. The standard location of the source of the video is at 1 km and the destination is at 11 kms; therefore, the location and destination are at least 1 km far from the road extremes.

B. Results

We have evaluated the use of Erasure Coding under different network data rates. In this way, we are able to observe its use under different levels of packet loss. Besides that, high transmission rates are usually necessary for the transmission of videos at higher quality.

In all figures in this work, we show the performance of the aforementioned metrics under different network data rates for different combinations of η and τ.

The delivery ratios achieved with the use of RLC is shown in Figure 1(a). The block size η influences the performance



(a) Random Linear Coding

(b) XOR-based Coding

Fig. 1. Delivery Ratio - Different graphs for different block sizes η

of RLC when it is used depending on the underlying frame loss related to different network data rates. At lower data rates, where frame loss in the unicast solution is by itself smaller, larger blocks have a better performance. However, for higher data rates where frame loss is bigger, we can see that smaller blocks outperform larger ones. The reason for this is that when insufficient packets are received to decode a block (number of packets received is less than η), the number of non-decoded packets is smaller with smaller blocks. The likelihood of receiving an insufficient number of packets to decode is smaller with larger blocks when frame loss is low; this is because there are more packets that form the redundancy (e.g. for $\tau = 20\%$ the number of additional packets is 1 when $\eta = 5$ while it is 4 when $\eta = 20$); so, in the case of eventual losses, the chances of receiving an additional packet are higher. The issue is inverted with higher frame losses; out of all sent packets, it is necessary to receive η packets within $\eta \times (1 + \tau)$ packets transmitted consecutively. In this case, it is more challenging to decode a block with larger values of η.

The main concept is that the importance of a single packet is inversely proportional to the value of η. At lower frame loss rates it is more beneficial to decrease the relevance of eventual lost packets while at higher frame loss rates it is better to increase the relevance of less frequently received packets.

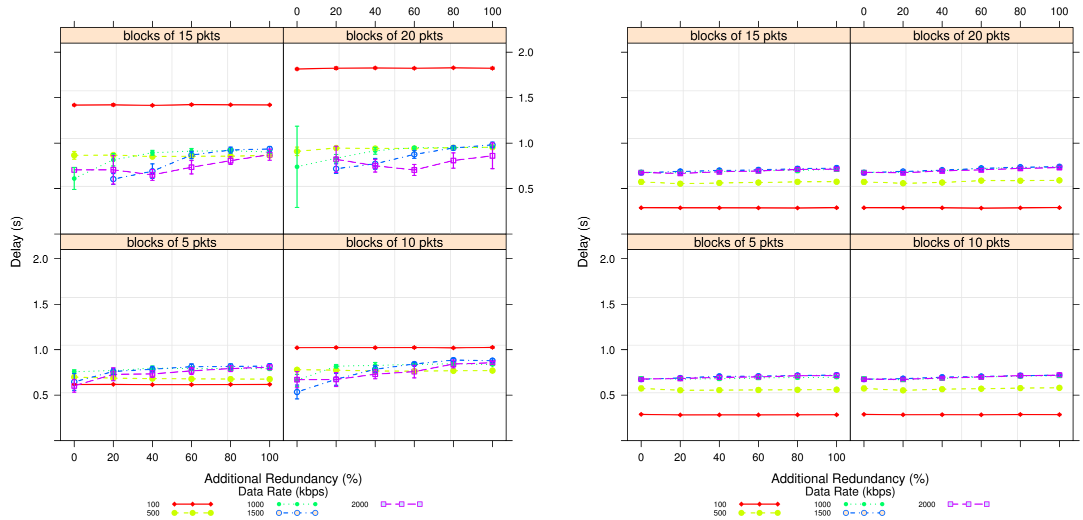
The use of XOR-based Coding has a different effect on the delivery ratios achieved, as shown in Figure 1(b). The impact of block size η with the use of XOR-based Coding is insignificant, which is different from its impact on RLC performance. Due to the degree distribution, the role of packets in terms of whether they are received or lost does not vary as it does when RLC is used. It is noticeable in Figure 1(b), that the initial additional redundancy of 20% leads to steep increases in the delivery ratio achieved. Furthermore, there is a correlation between the increases in delivery ratio and the degree of new extra packets. This is made evident by the less steep improvement in delivery ratios when τ is increased from 40% to 60%. Based on our degree distribution, the new packets above 50% additional redundancy have their degree initially higher than the expected degree 2 of the previous packets. Therefore, it can be understood that packets with greater degrees are not as useful as the ones with lower degrees.

The end-to-end delay when RLC is used can be observed in Figure 2(a). The need to await the decoding of a block in order to delivery video content to the application leads to higher delays. The block size η impacts end-to-end delay solely when data rates are low because in such cases the time for the transmission of η packets is greater than the average time spent transmitting a single packet from source to destination. At higher data rates, this time becomes insignificant and the impact of η is not noticeable.

Figure 2(b) shows the observed end-to-end delay achieved by the use of XOR-based Coding. The results were always under 1 second which complies with the requirement mentioned at the Introduction (up to 5 seconds). The only factor that had an impact on the delay experienced was the network data rate where higher data rates led to slightly longer delays. This behavior is explained by the fact that VIRTUS, the underlying unicast solution, suffers slightly higher delays under a congested scenario as delayed transmissions have a higher chance to be received [8].

The evaluation of the cost of using Erasure Coding was performed in terms of the number of transmissions. Due to lack of space, we do not show the results but when Erasure Coding is used (both with RLC and XOR-based Coding) follows the expected behavior and it is directly proportional to the amount of additional redundancy.

This thorough evaluation of the use of Erasure Coding has shown that the use of XOR-based Coding outperforms that of RLC. XOR-based Coding has achieved the necessary delivery ratios with smaller amounts of additional redundancy.



(a) Random Linear Coding

(b) XOR-Based Coding

Fig. 2. End-to-end Delay - Different graphs for different block sizes η

Although RLC offers a higher diversity of encoded data, its inability to offer partial recovery of a block when insufficient amount of packets are received for a full decoding causes, in many cases, an intensification of the impact of packet loss. Besides that, XOR-based Coding tackles the issue of a erasure network through erasure coding in a more suitable fashion as the original data is transmitted normally followed by encoded packets that assist in the recovery of lost packets.

IV. Final Remarks

We have thoroughly investigated the use of Erasure Coding through two distinct coding techniques: Random Linear Coding (RLC) and XOR-based Coding. These techniques differ in how entropy is used with the additional redundancy. We have investigated the peculiarities on the use of these techniques and how they perform towards the improvement of delivery ratio for video streaming over VANETs.

We have shown that Erasure Coding can be use to handling packet loss and increasing the successful reception of video content at the receiver. We have observed that there is a strong connection between the ideal block size and the level of packet loss. In scenarios where packet loss is not excessive, it was shown that larger blocks are more suitable; while, when packet loss is high, smaller blocks have obtained better performance. The reason for this is that the block size in RLC determines the relevance of individual packets; and, in scenarios with lower packet loss is better to decrease the relevance of eventual lost packets while in scenarios with higher packet loss it is more suitable to increase the relevance of received packets. XOR-based Coding performance was affected by the degree of encoded packets transmitted. It was noticeable that the use of encoded packets with smaller degrees incurred steeper increases on delivery ratio than when bigger degrees were used.

Finally, we have observed that XOR-based Coding has achieved higher delivery ratios than RLC at similar amounts of additional redundancy. XOR-based Coding follows a mechanism ideal to the use of Erasure Coding over VANET scenarios that are highly prone to packet loss. In this mechanism, the majority of received packets are translated into useful video content.

In the future, we want to evaluate the punctual use of Erasure Coding solely on more relevant data in video content. In this way, there will be not higher increases on the number of transmissions while improving the delivery ratio of crucial data (such as that of I-frames). We would also like to investigate the use of redundancy for video streaming dissemination.

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