A Quality Guaranteed Video Dissemination Protocol over Urban Vehicular Ad Hoc Networks

by

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Abstract

Video dissemination over Vehicular Ad Hoc Networks (VANETs) is an attractive technology which supports many novel applications. Hence, the merit of this thesis has twofold.

Firstly, we evaluate and compare three routing techniques and two error resilience techniques. We select a sender-based routing technique called SUV and compare it with the other two selected receiver-based routing techniques named REACT-DIS and CDS. The results, more specifically, show that the receiver-based solutions outperform the sender-based solution. In addition, only CDS method fulfils the general quality requirements as it is the best that reduces redundancy packets and covers the whole topology. The results also indicate that the video coding scheme, Interleaving, can fix the multiple consecutive packet losses and guarantee reliable video qualities over VANETs. Network Coding, however, fails to provide satisfactory video quality for urban scenarios. This study next combines the selected receiver-based routing techniques and the two error resilience techniques. We find the best combination is Interleaving over CDS.

Secondly, we design a quality guaranteed video dissemination protocol for urban VANETs scenarios. Based on our comparison result, our protocol selects the CDS and Interleaving as the routing and error resilient techniques. To fix the single packet loses caused by the topology's intermittent disconnection and collisions, we propose a store-carry-broadcast scheme for the nodes to re-transmit the local buffer saved packets. The results, when compared to the selected techniques and combinations, show that our proposed protocol is the most efficient one in terms of packet delivery, delay, overhead and video quality.

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Publications Related to Thesis

Conference Paper:

- Yang Li, Farahnaz Naeimipoor, and Azzedine Boukerche, Video dissemination protocols in urban vehicular ad hoc network: A performance evaluation study, in IEEE WCNC14 Track 3 (Mobile and Wireless Networks) (IEEE WCNC14 Track 3 : NET), Istanbul, Turkey, Apr. 2014.

Glossaries

ACK	Acknowledge	**AIFS**	Arbitration Inter-frame Space
B-Frame	Bi-directionally Predictive-coded Frame	**CCH**	Control Channel
CDS	Connected-Dominating-Set	**CSMA/CA**	Carrier Sense Multiple Access with Collision Avoidance
CTS	Clear To Send	**CW**	Contention Window
DCF	Distribute Coordination Function	**EC**	Erasure Coding
EDCA	Enhanced Distributed Channel Access	**EOS**	Expected Offset Calculation
FEC	Forward Error Correction	**FMO**	Flexible Macroblock Ordering
GoP	Group of Pictures	**GPS**	Global Positioning System
GRA	Grey Relational Analysis	**GRC**	Grey Relational Coefficient
HCCA	HCF Controlled Channel Access	**HCF**	Hybrid Coordination Function
Id	Identification	**I-Frame**	Intra-coded Frame
IL	Interleaving	**IL-CDS**	Interleaving over CDS
IL-RD	Interleaving over REACTDIS	**IN_CDS**	in the Connected-Domination Set
LC	Layer Coding	**MAC**	Media Access Control
MANETs	Mobile Ad Hoc Networks	**MDC**	Multiple Description Coding
MOS	Mean Opinion Score	**MPEG**	Moving Pictures Experts Group
NAV	Network Allocation Vector	**NC**	Network Coding
NC-CDS	Network Coding over CDS	**NC-RD**	Network Coding over REACTDIS
NOT_CDS	not in the Connected-Domination Set		

NS2	Network Simulator 2	**P-Frame**	Predictive Frame
PSNR	Peak Signal-to-Noise Ratio	**QGVD**	Quality Guaranteed Video Dissemination Protocol
QoE	Quality of Experience	**QoS**	Quality of Service
REACT-DIS	Reactive, Density-Aware and Timely Dissemination Protocol	**RLNC**	Random Linear Network Coding
RTS	Request To Send	**SCH**	Service Channels
SUMO	Simulation of Urban Mobility	**SUV**	Streaming Urban Video
TDMA	Time Division Multiple Access	**WAVE**	Wireless Access Vehicular Environment
VANETs	Vehicular Ad Hoc Networks	**XOR**	Exclusive-OR

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Chapter 1

Introduction

VANETs are new promising technologies which achieve intelligent vehicle to vehicle and vehicle to infrastructures communications. These communications improve traffic safety and efficiency. Video streaming over VANETs is an increasingly important and attractive service for the driver to make accurate decisions. In this chapter, video dissemination over VANETs is introduced. Moreover, the challenges caused by application demands and mobility patterns will also be discussed. Finally, the contributions and thesis organization are provided.

1.1 Background

It is commonly acknowledged that VANETs are promising technologies which enable vehicles to communicate with each other and with roadside infrastructures by using equipped wireless communication technologies. Figure 1.1 shows a simple vehicle to vehicle communication situation. VANETs mostly resemble the features of Mobile Ad Hoc Networks (MANETs) in their capabilities of self-organization, rapid deployment and adaptivity. Moreover, VANETs also have theirs own features: intensive vehicle movement, disconnection and vehicle high speed and mobility.

Enabling video dissemination applications over vehicles can provide more precise information for a wide variety of distinguished news broadcasting and advertising services such as safety, entertainment, and military and scientific applications [2]. With plain text sending approaches, we can only briefly provide information which may not be enough for the driver to make accurate decisions. However, video can provide every detail of the information. In addition, broadcasted video content can tolerate a certain amount

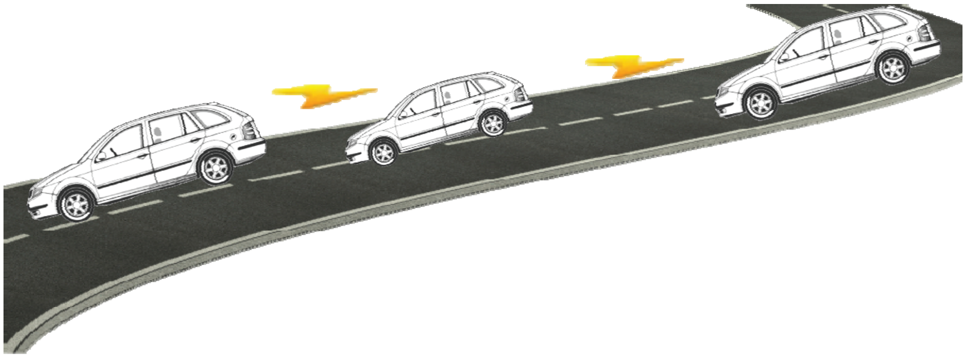

Figure 1.1: A Simple Scenario of VANETs

of packet loss ratio, while the sending of text needs a more reliable transmission scheme. VANETs also bring several advantages to video streaming over an ad hoc network such that:

- Resource limitation such as battery power and data storage is no longer a problem.
- Global positioning system (GPS) has been introduced for navigation. GPS can provide vehicles' geographic position information.
- Compared with nodes in MANETs, vehicle motion is restricted by the predetermined road geometry and becomes less random.

Quality of Service (QoS) has been widely used as a metric to assure video dissemination services [28], [44] and [50]. QoS includes typical parameters such as delivery ratio, bandwidth, delay and jitter. However, it lacks an important element in its ability to evaluate of video dissemination service: the human perception. Research has sufficiently shown that the subjective indicators, QoS, often fails to correlate with human perception [17]. Human perception of received video is the best judges of the user-centric indicators: Quality of Experience (QoE). A common way of subjectively characterizing QoE is by the mean opinion score (MOS), which maps out human impressions of distortion on a pre-defined quality scale. In this study, we consider the QoE performance as we evalute the MOS.

1.2 Problem Statement

Many researchers have proposed protocols to improve the quality of video streaming over VANETs. Researchers mainly focus on responding the following two challenges:

- The first challenge comes from the features of VANETs: high dynamic topology, disconnected platoons and complex scenarios. The speed and random direction of vehicles movements defined highly dynamic topology of VANETs. As vehicles move quickly, the links between them last for a short period of time. In addition, in the case of low density, the frequently disconnection of vehicles from the network is another obstacle to obtaining high-performance. In urban scenarios, street structures, uncertain vehicle density and roadside constructions also reduce the efficiency of protocols.

- Video content is constructed with large amounts of data and has strict QoS and QoE requirements like packet delivery ratio, throughput, delay, overhead, Peak Signal-to-Noise Ratio (PSNR), Mean Opinion Score (MOS), etc. These distinguishing requirements become another obstacle to design high-performance video streaming protocols. The worldwide networking technique leader Cisco has defined some of these requirements [47] for the exchange of video content. In the case of video streaming, a delay should not be higher than 4 to 5 seconds, while packet loss should not exceed 5%. In addition, [52] showed the PSNR requirement for video dissemination (see Table 3.1).As discussed before, MOS, which is usually calculated using PSNR, is useful parameter of QoE. From [12], if the MOS value of a received video is 4, it is perceptible and not annoying (as shown in Table 3.2).

1.3 Contribution

This thesis has two main contributions. In the first stage, this study aims to analyze the performance of current routing and error resilient techniques which adapt different approaches to cope with video streaming challenges in VANETs. The second stage of this thesis focuses on proposing a quality guaranteed video dissemination protocol for urban VANETs. The purpose of this is to resolve the aforementioned challenges in vehicular networks by providing high qualities broadcasted video with low latency values. The main contributions of this thesis are the following:

- To analyze the specific features and characteristics of video streaming techniques, we study and classify the existing related works of VANETs. These techniques concentrate on different stack layers to improve video streaming techniques. Comparison of the similar approaches are presented and summarized in an appropriate way.

- To get an impressive performance analysis of existing routing and error resilient solutions, a number of efficient and effective techniques have been selected to implement over urban VANETs scenarios. Their advantages and disadvantages are analysed.

- To meet application requirement, we design a quality guaranteed video dissemination protocol for urban VANETs scenarios. We combine the error resilient techniques and the routing techniques and find the best combination for our protocol in terms of video quality, latency and overhead.

- To solve the intermittent disconnection problem of VANETs and to fix the packet losses, we design a new store-carry-forward scheme for our protocol. Our protocol is able to detect video frame losses of neighbours and to help them by sending buffered packets.

- To provide a deep analysis of all selected approaches, combination of existing protocols and a proposed video dissemination protocol, all of them are implemented and evaluated with the Network Simulator 2 (NS2) and the Simulation of Urban Mobility (SUMO), which are employed as the network simulator and traffic simulator, respectively. Different reliable urban scenarios are used and the results of QoS and QoE are compared and discussed in detail.

1.4 Thesis Organisation

The remainder of this thesis is organized as follows:

- Chapter 2 surveys the various techniques used for video streaming over vehicular ad hoc network: i.e., coding and error resilient techniques for the application layer, routing methods and store-carry-forward for the network layer, and IEEE 802.11 and IEEE 802.11p based techniques in the data link layer. For each category, we discuss and compare the existing schemes and algorithms in detail. This chapter can serve as a starting point to research video streaming protocols over VANETs.

- Chapter 3 outlines the selected routing schemes. The schemes under study are one sender-based solution called Streaming Urban Video (SUV) [46], and two receiver-based solutions referred to as Connected-Dominating-Set (CDS) [54] and the Reactive, Density-Aware and Timely Dissemination Protocol (REACT-DIS) [42]. The simulation environment, video evaluation method and mobility model are explained. In addition, the QoS and QoE metrics used to evaluate all introduced protocols are shown.
- Chapter 4 introduces two types of error resilient techniques: Network Coding (NC) [1] and Interleaving (IL) [7]. For the purpose of investigating their performance, we implement and evaluate techniques using the same method as Chapter 3.
- Chapter 5 describes the simulations for combination involving selected receiver-based routing techniques and error resilient techniques. This chapter also discusses the performance evaluation results in detail. We also compare the distinct features of these protocols and select the best combination.
- Chapter 6 fully describes the design and implement of the proposed quality guaranteed video dissemination protocol that has been proposed in this study. First, the motivation of the proposed method is provided. After that, the store-carry-forward process as well as the whole protocol are described in detail. Last, the obtained creditable results are exhibited.
- Chapter 7 concludes with the main contributions of this thesis and shortly provides the future work.

Chapter 2

Related Works

Streaming video content over VANETs is an attractive technology which supports many novel applications. Video streaming involves a large amount of data to be transmitted. It is hard to achieve good video quality and to also meet strict delay requirements. Many researchers have proposed protocols to achieve a better performance of video streaming. In this chapter, we provide a survey of video streaming for VANETs. As shown in Figure 2.1, we classify existing techniques into three major categories: techniques in the application layer such as video coding techniques and error resilient techniques, the optimizing routing and store-carry-forward techniques in the network layer, and media access control techniques for the data link layer. This survey gives an overview of challenges and solutions to the researchers and developers who are new to this area.

2.1 Techniques in Application Layer

Compared to the transportation of other type of files, video streaming over VANETs mainly suffers from the problem of resource constraints since the video files are usually larger than other file types. In addition, because of the highly dynamic topology of VANETs, unpredictable packet loss becomes another major challenge in meeting the stringent requirements of video streaming. A range of application solutions proposed to address the above-mentioned challenges are shown in this section.

2.1.1 Scalable Video Coding

The Moving Pictures Experts Group (MPEG) and the family of advanced video coding standards (H.26x) have defined the non-scalable video coding standard [56]. The com-

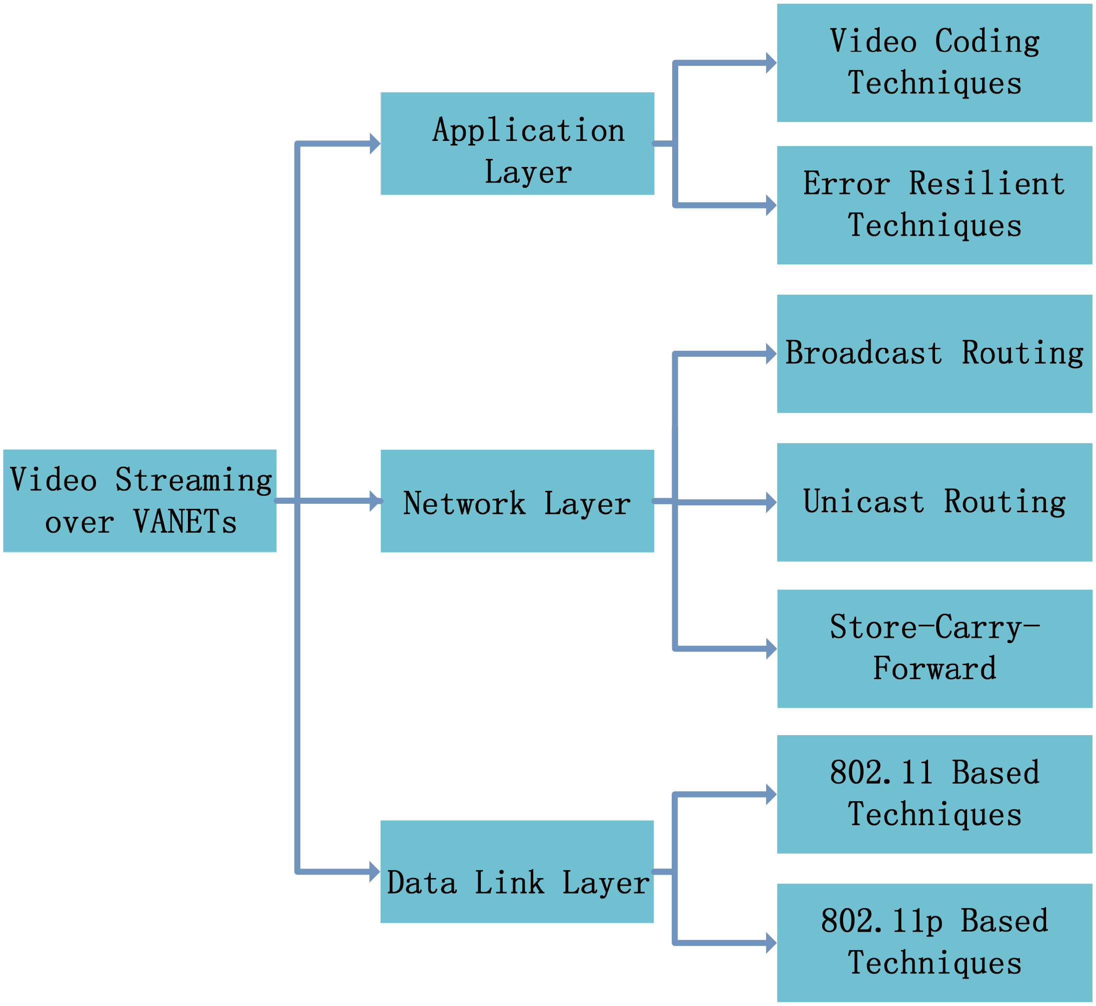

Figure 2.1: Classifications of Video Streaming over VANETs

pressed file contains 3 different frames: Intra-coded frame ($I\text{-}frame$), Predictive frame ($P\text{-}frame$) and Bi-directionally predictive-coded frame ($B\text{-}frame$). In general, all the video frames are divided into smaller groups called Groups of Pictures (GoP). In one GoP, $I\text{-}frame$ is self-contained and its compression rate is low. The compression rate of $P\text{-}frame$ is higher than the rate of $I\text{-}frame$. To encode and decode $P\text{-}frame$, the previous $I\text{-}frame$ and/or $P\text{-}frames$ in the same GoP are needed. The encoding and decoding of $B\text{-}frame$ is based on the previous and the following $I\text{-}frames$ and $P\text{-}frames$. The compressed rate of $B\text{-}frame$ is the highest compared to the other frames.

However, the limited scalability of non-scalable video coding restricts the ability of these videos to meet the stringent requirements of video streaming. Therefore, scalable video coding techniques are proposed and have been standardized as an extension of the H.264/AVC. The video bit stream is called scalable when parts of the stream can be removed and the remaining sub-stream represents the source content with an acceptable quality. There exist three scalability types, including temporal, spatial and quality scalability. Their basic concepts and developing tools are provided by H. Schwarz *et al.* in [45]. In [23], T. A. Le *et al.* provided an evaluation platform for scalable video coding. Moreover, in [20], C.-H. Ke designed an integrated scalable video coding transmission simulation evaluation framework. In general, there exist two approaches of Multi-stream Coding: Layer Coding and Multiple Description Coding.

Layer Coding

In Layer Coding (LC), the video content is usually encoded hierarchically into a base layer and one or more enhancement layers. As Figure 2.2 shows, the base layer is formed by $I\text{-}frame$ and $P\text{-}frame$, and the enhancement layer is formed by $B\text{-}frame$. The base layer is usually the most important layer and the enhancement layers are referenced to the base layer. When there are congestions, enhancement layer packets may be dropped to protect base layer packets. Decoding the base layer offers basic video quality, while decoding the base layer together with the enhancement layers provides further refined and smooth video. The enhancement layer(s) n become useless for the decoder if the lower layer $n-1$ is lost. The key impediment of utilizing LC for video streaming comes from designing reliable base layer protecting and the maximum number of enhanced layers providing schemes [26].

S. Mao *et al.* in [29] proposed a combination scheme between multi-path and LC to protect the base layer. Since the base layer packets are important, they are transported on a better path, which provides lower loss probability. If a packet is dropped from the

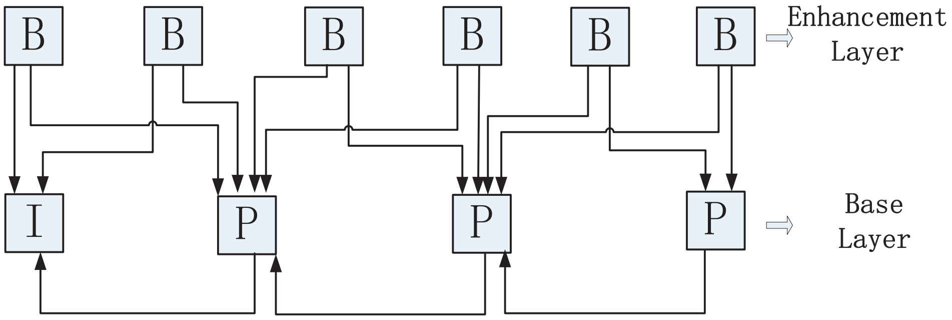

Figure 2.2: An Example of Layer Coding

base layer, it is retransmitted on the enhancement layer path. Therefore, the transmission of the enhancement layer packet is temporarily stopped.

In [41], A. Razzaq *et al.* also combine LC with multi-path routing. They calculate the network topology for appropriate path selection and rank the robustness of all available paths. They select different paths for different layers according to their importance. Therefore, the highest priority data is transmitted through the highest quality path and the lowest priority data is transmitted through the most unstable paths. Their results showed that the base layer received more correctly than other layers, as it received transmissions through the most reliable path.

Multiple Description Coding

In Multiple Description Coding (MDC) [11], the video frames are split into multiple sub-streams which are called descriptions. Unlike LC which suffers from strict layer importance, every received description in MDC is useful. As shown in Figure 2.3, decoding any description, the receiver gets a base quality. The more additional decodable descriptions are received, the better the video quality is. Descriptions in MDC may have the same importance in balanced MDC schemes or different importance level of in unbalanced MDC schemes.

N. N. Qadri *et al.* in [38] and [39] introduced spatial MDC to peer-to-peer video diffusion strategies over VANETs to exploit path diversity. They combined spatial Flexible

Table 2.1: The Comparison between Layer Coding and Multiple Description Coding

Techniques	Layer Coding	Multiple Description Coding
Resilience	Sensitive to transmission loss	More resilient to transmission loss; Robust even in the case of burst losses
Method for dropping packet	If it is required to packet, LC drops the last enhancement layer firstly; And the base layer must never be dropped	Any description could be dropped if required. But protecting one description is more effective than trying to protect descriptions randomly
Repair mechanism	The repair mechanisms are needed to guarantee the delivery of the base layer. The repair mechanisms unequally protect different layers.	The delivered quality is acceptable even at a high loss ratio
Delivery Ratio	Medium	High
Latency	Medium	Medium
Overhead	Medium	High
PSNR	Medium	High

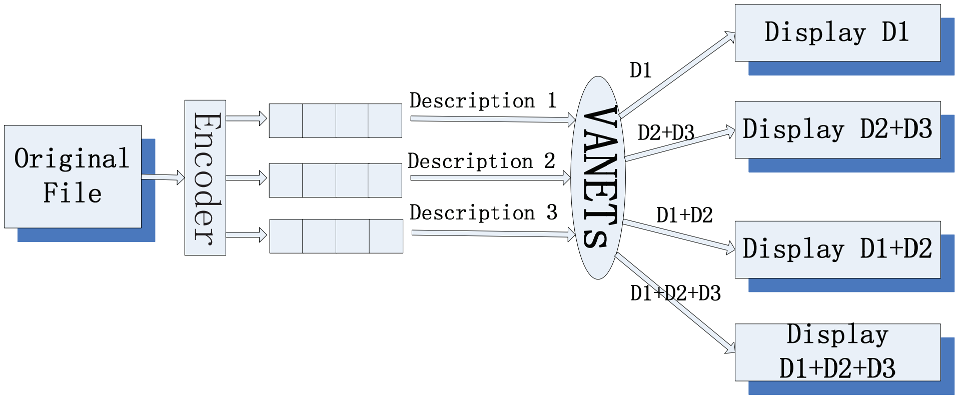

Figure 2.3: An Example of MDC

Macroblock Ordering (FMO) and MDC in their scheme by splitting each frame into two slices and forming two descriptions. Their scheme was shown to be considerable in the reduction of the impact of packet loss and to achieve robust video delivery quality. N. N. Qadri *et al.* in [39] also discussed the advantages of MDC layer independence.

As shown in Table 2.1, the main shortcoming of LC is the protection and recovery of the base layer. On the other hand, any description in MDC can be used to display. Therefore, the performance of MDC is better than the one of LC for video streaming.

2.1.2 Error Resilient

In non-scalable video coding, if an $I\text{-}frame$ or $P\text{-}frame$ is lost, all frames thereafter in the GoP become un-decodable. Therefore, a single packet loss can cause a huge impact on video quality at the receivers end. Simply forwarding redundant packets is an error recovery method in the application layer, but it is not always a good solution. Other reliable error resilient and concealment techniques, which are Forward Error Correction (FEC), MDC and IL, are mainly discussed in this section.

FEC

FEC is a technique used to add redundant information used to transmitted packets. The redundant information is used to help recover some amount of missing data at the receiver.

XOR (exclusive-OR) is one of the most popular FEC techniques which recently attracted a substantial amount of attention. XOR uses the function $F(a, b, ...)$ to compute the XOR results over the a, b, ... packets. An example of XOR is shown in Figure 2.4, where 100% of redundant packets are used. If the FEC stream is received correctly, every packet loss can be recovered. In [6], P. Bucciol *et al.* showed that XOR is a low complexity technique which is efficient to reduce the packet loss ratio without retransmission delays. In [24], A. Li also showed that XOR based FEC scheme allows for the recovery of missing data. M. Pasin *et al.* in [37] introduced Interleaving to spread adjacent packet losses. They use Interleaving to guarantee the loss rate reducing ability of XOR based FEC.

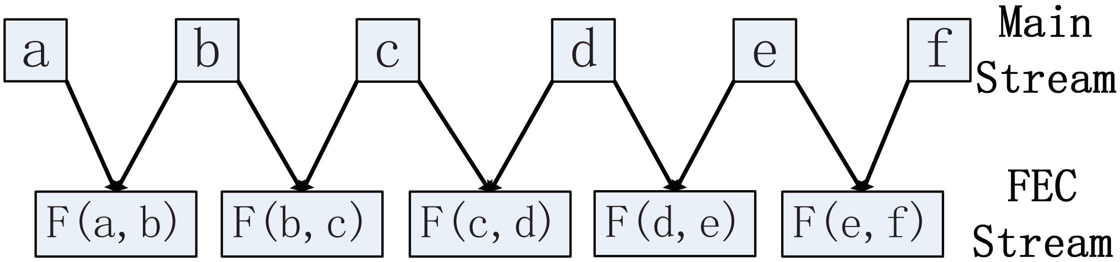

Figure 2.4: An Example of XOR

Another frequently used FEC technique is the so-called Erasure coding (EC) [53]. EC scheme encodes k packets into n ($n > k$) packets. At the receiver, the original k source packets can be recovered if it is provided any k of n packets [8]. In [51], Y. Wang *et al.* proposed a forwarding algorithm based on EC. Their results showed that EC spreads the responsibility of forwarding over many nodes while maintaining a constant overhead, which results in fewer cases of long delays. However,C. Rezende *et al.* in [43] showed that EC cannot enhance the overall efficiency of redundancy. It is necessary to receive a minimum amount of packets to recover the original packets.

NC [1] is also a well-known coding technique considered to be used for error resilience. NC essentially shares a common concept with EC: encode original video content into a stream of packets k and send n ($n > k$) packets to provide redundant packets. Receivers can recover the original content once they get k encoded packets. The main difference between them is that NC allows intermediate nodes to re-encode packets they have received. C. Rezende *et al.* in [42] used NC to provide an additional redundancy control aimed at handling packet loss ratio. They found that as a redundancy control method, NC fulfills video-streaming requirements while maintaining suitable levels in terms of latency and overhead. In [34] and [35], J.-S. Park *et al.* proposed a new approach. When a node receives a newly coded block packet, it sets up a timer for receiving the whole block. Upon the expiration of the timer, the node re-encodes and forwards the packets only if it has already received the full block.

A significant amount of researchers focuses on comparing EC and NC. J.-S. Park *et al.* in [36] explored the reliable and timely data dissemination service. They combined data mulling technique with EC and NC. They showed that the NC-based strategy outperforms EC-based strategy in terms of reducing data delivery delay. In [8], A. Fujimura *et al.* compared the reliability and efficiency of EC and NC. They showed that NC can achieve high reliability in terms of packet delivery ratio and packet transmission overhead. F. Naeimipoor *et al.* in [32] also showed that NC outperforms EC in terms of delivery ratio, latency and overhead. The overall similarities and differences are shown in Table 2.2, from which we can also conclude that NC outperforms EC in terms of video streaming over VANETs.

MDC

The MDC is also an error resilient technique which provides a certain amount of redundant packets among the descriptions. As shown in Figure 2.5, if a packet in one description is lost, the corresponding packets in the other descriptions may be available for the decoder. As the same data in the lost packet is contained in the corresponding packets, the video decoder decoded the video successfully. M. Kazemi *et al.* in [19] provided enough insight into existing MDC techniques and presented their abilities to tackle transmission losses.

Interleaving

The loss distribution is also a key parameter that determines the performance of video streaming. A method that can tackle this issue is IL. The main idea of IL is to spread out clustered packet losses as the several small losses degrade quality less than the original big gap. In [7], M. Claypool *et al.* proposed a video interleaving approach that reduces

Table 2.2: The Comparison between Network Coding and Erasure Coding

Techniques		Network Coding	Erasure Coding
Similarities	Function	Inject redundant packets into the network to recover the original packets without asking the source node for retransmission	
	Aim	Increase the reliability in erasure networks	
	Method	Encode the video into encoded packets	
Difference	Influence	Encoded packets come from everywhere in the whole network	end-to-end packets which are only from the source node
	Technique	Allow every intermediate node to encode packets	Only the source node encodes packets
Performance	Delivery Ratio	Medium	Low
	Latency	Medium	Low
	Overhead	Medium	High
	PSNR	Medium	Low

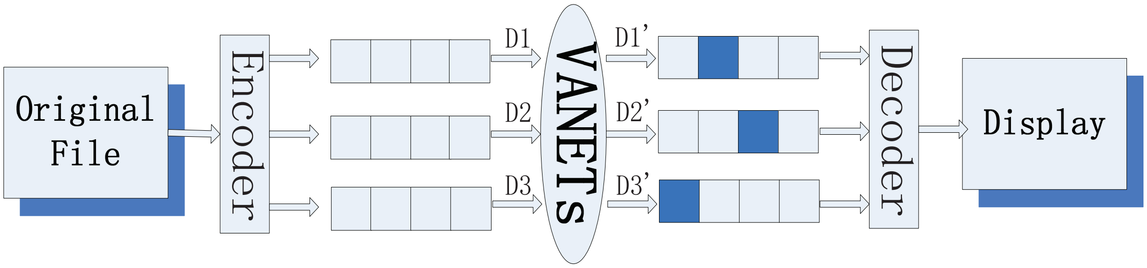

Figure 2.5: Using MDC as a Error Resilient Technique

the effects of frame losses. In their IL scheme, as shown in Figure 4.2, the sender first re-sequences video packets before transmission, and returns the data to the original sequence at the receivers. Their results showed that their algorithm can improve the perceptual quality of the received video.

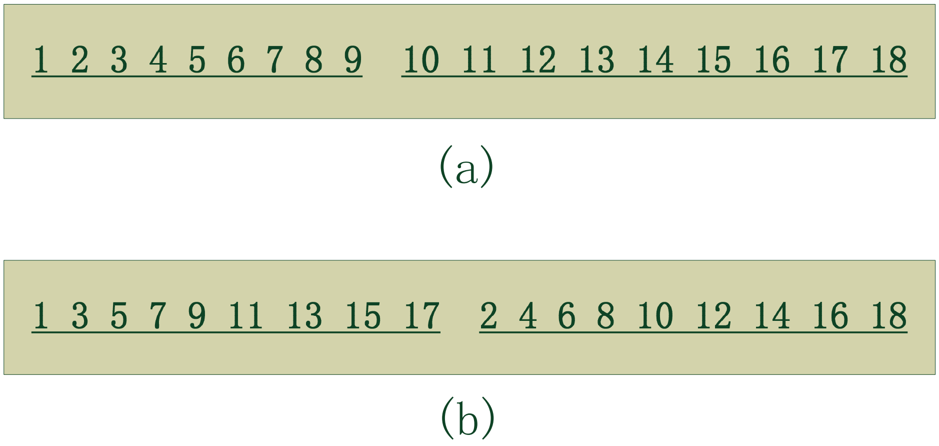

Figure 2.6: An Example of Mark's Interleaving

P. Bucciol *et al*. in [6] introduced a new IL method as shown in Figure 2.7. At the

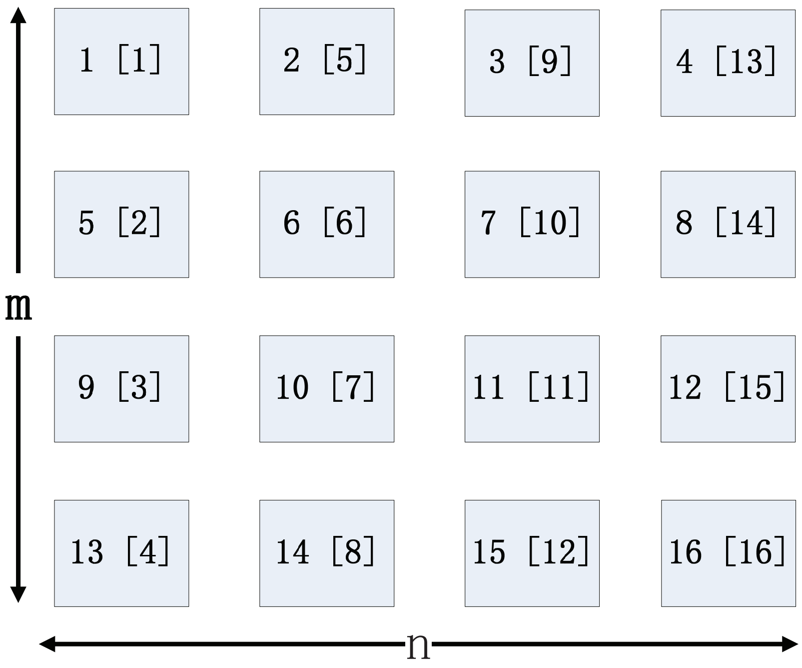

Figure 2.7: An Example of New Interleaving

sender, packets are first written in rows, and each row is a block of n packets. Then the packets are switched as the transpose of a matrix: eg., *packet*$[i][j]$ is switched with *packet*$[j][i]$. In the case where *packet* 1 2 3 are lost, the loss bursts are converted into *packet* 1 5 9 as separated losses. They also combined Interleaving with FEC techniques and showed that the combinations are validated to improve the video transmission quality. Moreover, M. Pasin *et al.* in [37] used the same Interleaving method.

In this section, we discussed the error resilient techniques for video streaming over VANETs. Based on their characteristics, we derive Table 2.3, where we illustrate the advantages and disadvantages of each technique discussed earlier, according to the requirements of applications.

2.2 Techniques in Network Layer

The network layer is responsible for routing through intermediate nodes. Routing for video streaming is used to establish and maintain paths to guarantee that the video is delivered with an acceptable quality. A significant number of video streaming based studies in VANETs are devoted to routing approaches. Generally, routing protocols can

Table 2.3: Comparison of the Error Resilient Techniques

Technique	Main Idea	Advantages	Disadvantages
FEC	Add redundant packets to the video stream.	If the losses are within the correcting capability, the whole video can be recovered.	1. Adding redundant packets increases the overhead and the chance of collisions. 2. If the loss rate is beyond the correcting capability, no lost data can be recovered.
MDC	Use the content of other descriptions to display.	As long as one or more descriptions arrive at the receiver, the video can be displayed with certain quality.	Video get from other description has a lower fidelity.
Interleaving	Spread the burst packet losses.	Increase the quality of received video without influencing the delivery ratio, latency and overhead.	Fail to fix the single packet losses.

be divided into three categories: broadcasting, multicasting (geo-casting) and unicasting. Multicast is defined by delivering packets to a group of destination nodes simultaneously by multi-hop transmission from the source. Geocast routing is basically a location-based multicast. In geocast, the source node delivers packets to a specific geographic region. Several multicast solutions have been proposed for MANETs [4] [3]. But they are not fit for VANETs because of their intensive vehicle movement and disconnected platoons. Unfortunately, there is no work which investigates video geocast and multicast routing over VANETs. This part surveys different routing techniques of the other two categories. Due to collisions and disconnection of the network, there are lots of packet losses. A popular technique to fixing these loss errors is store-carry-forward, which is also surveyed in this part.

2.2.1 Routing Techniques for Broadcast

Broadcast protocol is utilized for when a source vehicle need to send messages to all other vehicles in the network. The easiest way to implement broadcast is by flooding in which each node re-broadcasts every newly-received message. After the broadcast storm problem of flooding was mentioned in [33], many video dissemination protocols developed for VANETs include efficient routing methods. The first one is a probabilistic approach: Gossiping [57], in which each node forwards a message with some probability of reducing the overhead and collisions of flooding. Some deterministic approaches were then designed. In this part two deterministic techniques (1) sender-based routing (2) receiver-based routing are discussed.

Sender Based Routing

Generally, in a sender-based scheme, the source node is responsible to assign one or more potential re-broadcasting nodes.

In [46], F. Soldo *et al.* proposed an efficient solution called SUV for vehicular communications. In SUV, each node divide its surrounding space into four identical equal sectors. One of them is for the source node (parent node), and the other three sectors are for selecting relay nodes (children nodes). As shown in Figure 2.8, based on the position of parent node, current nodes select one child node for each part. Only the nodes which are selected as children nodes re-broadcast received messages. In [25], we have shown that SUV failed to meet the strict quality requirements of video dissemination over urban VANETs. SUV could reduce broadcast collisions, but it failed to disseminate enough video content to the receivers. In [27], G. Maia *et al.* proposed a protocol called U-HyDi which employs both the sender-based and receiver-based broadcast mechanism. It is a hybrid approach that tackles the broadcast storm problem under dense networks. In the sender-based approach, the sender chooses the farthest neighbor from both the sender and the origin to rebroadcast the message. It is based on the idea that the farthest node has a greater probability of covering more new area.

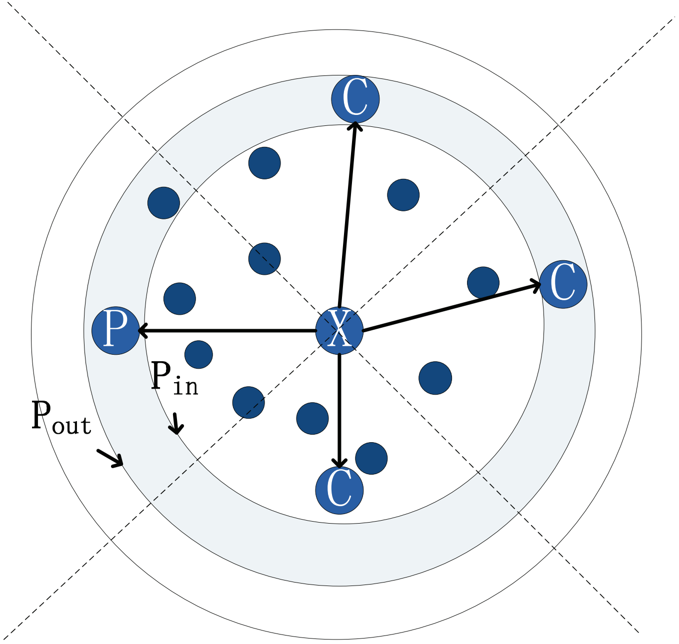

Figure 2.8: An Example of SUV [46]

Receiver Based Routing

Under a sender-based approach, if the selected node failed to receive the message from the sender, the message is not re-broadcasted to it anymore. Therefore, a sender-based approach may result in a poor delivery ratio. Receiver-based approaches are mostly reactive and reliable compared with sender-based approaches. According to [55], the receiver-based scheme outperforms the sender-based scheme in terms of collisions and PSNR and is more suitable for video dissemination over VANETs.

As mentioned before, U-HyDi also has a receiver-based approach in [27]. The approach works as follow: whenever a node a receives a message for the first time, and it is not the selected as a relaying node, the node a then schedules a timer to broadcast the message. Before the timer expires, if the node a receives a duplicate message, it cancels the rebroadcast. Otherwise, it rebroadcasts the packet considering that the assigned relay node failed to receive this packet. By combining receiver-based approach and sender-based approach, U-HyDi becomes a suitable solution which meets strict delivery requirements of video dissemination over VANETs.

In [28], G. Maia *et al*. designed a receiver-based broadcast routing mechanism called VoV which combines geographical approach and timer-based approach. When a vehicle in VoV receives a new message, it first decides whether a rebroadcast is required or determines that all of its neighbours have already been covered by the broadcast. If the latter situation happens, rebroadcasting becomes unnecessary. Otherwise, the vehicle determines its waiting time for rebroadcasting. Their results showed that VoV is an efficient scheme for video dissemination in diverse traffic conditions.

In [42], C. Rezende *et al*. designed REACT-DIS which is a receiver-based routing approach which can avoid high packet delay and transmission overhead. After it receives a packet, node in REACT-DIS overhears and calculates the duplicate messages for a period of time. The amount of duplicate messages is used to estimate local density and to make the decision of retransmission. The chance of a node becoming a relay node decreases exponentially with the amount of overheard retransmissions. In addition, the relay node forwards the received packet with the probability p which is also dependent on the number of overheard copies.

2.2.2 Routing Techniques for Unicast

Unicast routing is a fundamental operation for a vehicle to construct a source-to-destination routing in VANETs.

Single-Path

In single-path routing, only a single route is used between a source and destination node. C. Rezende *et al*. in [44] presented a feasible receiver-based single-path video unicast solution called VIRTUS. VIRTUS selects relaying nodes based on current and future locations of vehicles. Bayesian state estimation is used for location tracking. Upon receiving the packet, nodes inside the forwarding zone of the source node schedule timers based on the waiting time calculated. When the timer expires, a node considers itself a relaying node only if it has not overheard any transmission by the same packet of a node closer to the destination. This node continues to be a relay node within a calculated time interval called *reservation time*. From the performance analysis, VIRTUS is believed to outperform other baseline solutions.

Multi-path

In multipath routing, there exists more than one route between the source node and the destination node. The advantage of multipath routing is that it can provide load balancing, fault-tolerance, and higher aggregate bandwidth. Therefore, it can be used to provide reliable video streaming services for the dynamic and unpredictable VANETs.

As an enhanced version of VIRTUS [44], LIAITHON [50] introduced multipath for video streaming over VANETs. A special route coupling prevention scheme was developed in LIAITHON. To select two optical paths, LIAITHON divides the whole area of both the source node and the destination node into two sides. A packet can only be transmitted by nodes on the same side. This ensures the two chosen paths are always on the opposite sides. If a node is inside the intermediary area, it cancels the retransmission timer. LIAITHON also has a special waiting time calculation scheme. The waiting time calculation scheme contains the geographic advance, link stability and a proportion which measures the degree of closeness to the optimal path. The performance results showed that LIAITHON is a node-disjoint multipath solution and it outperforms VIRTUS.

In [41], A. Razzaq *et al.* designed a video multicast technique for LC. They transmitted the base layer data through the highest quality path and enhancement layer data through the least stable path. The source node uses Grey Relational Analysis (GRA) [18] to calculate the robustness of all available paths and ranks them according to their robustness. GRA classifies the parameters into categories: smaller-the-best and larger-the-best. Smaller-the-best means smaller values are preferable, in terms of delay and jitter. And throughput belongs to larger-the-best. The authors then calculate the Grey Relational Coefficient (GRC) of different paths base on the values of smaller-the-best and larger-the-best parameters, and rank the available paths according to the GRC values.

Based on the previous discussion of broadcast and unicast video streaming techniques over VANETs, this study summarizes the performance of proposed techniques for video streaming in terms of QoS parameters. Table 2.4 is an insight into performance comparison. We can conclude that the receiver-based scheme outperforms the sender-based scheme in terms of collisions and PSNR, and multi-path is more suitable than single-path for video dissemination over VANETs.

2.2.3 Store Carry Forward

A big challenge of video streaming over VANETs is that the packet loss which is due to collisions and the intermittent network connection. Store-carry-forward concept is proposed to respond to this challenge [28], [34] and [41]. In [13], M. Guo *el al.* used such an approach to transmit video data in a partitioned network environment.

G. Maia *et al.* in [28] designed a store-carry-forward mechanism to increase the video delivery capability of VoV. In VoV, considering that nodes may fail to receive the packets in the primary dissemination process, nodes in VoV stores the new message in the local buffer for a period. Besides, during the period of the message, all the buffered received message information is also transmitted. If a neighbour finds out that the vehicle does not have some already disseminated messages, it will forward them to the vehicle. To avoid retransmitting collisions, VoV uses a broadcast suppression mechanism. Nodes in VoV calculate the actual waiting times before forwarding the message. Vehicles closer to the source node have a lower waiting time than vehicles farther away. When the nearest vehicle finally forwards the message, other nodes overhearing this transmission will cancel the transmissions to avoid collisions.

As discussed before, a key impediment of designing a reliable NC-based protocol is that NC discards the packets if it fails to recover the whole block. In [35], J.-S. Park *et al.*

Category	BROADCAST		UNICAST	
Techniques	sender-based	receiver-based	single-path	multi-path
Delivery Ratio	Low	High	Medium	High
Latency	Low	Medium	Low	Low
Overhead	Low	Medium	Low	Medium
PSNR	Low	High	Medium	High

Table 2.4: Comparison of Routing Techniques

used store-carry-forward scheme for error correction. When the timer expires, if a node in NCDD realizes that it needs more packets to decode, it sends a help request. Neighbours who have recovered the block will respond to this request and transmit another coded packet. In this way, the sender has a higher chance to collect more coded packets.

In [41], A. Razzaq *et al*. selected some nearby nodes along the transmission path for recoding their received packets and storing them in buffers for unit periods of time. They select the nearby nodes which have free resources to store packets. For the buffered packets, the nearby nodes use XOR-based network coding and store resultant packets in the buffer to provide as many packets as they can. These network coded packets were impediment to nodes for recovering lost packets. The results showed that their network coding based store-carry-forward scheme can help to reduce error rate.

2.3 Techniques in Link Layer

A significant number of video streaming studies in VANETs are focused on Link Layer, especially on the sub-layer: Media Access Control (MAC) [5], [46], [28] and [16]. The MAC layer determines which node is given access to the medium. In this section, IEEE 802.11 standard and protocols are discussed.

2.3.1 IEEE 802.11

IEEE 802.11 [9] is a wireless communication standard which employs the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). The Distribute Coordination Function (DCF) is responsible of CSMA/CA. Before transmission, a node first checks the medium state by determining whether another node is transmitting or not. If the state is *idle* for a period, the node can transmit. Otherwise, it needs to wait for a period of time before listening again. The length of the waiting time is chosen within a Contention Window (CW).

To minimize the risk of frame collisions, 802.11 uses Request to Send/Clear to Send/Acknowledge (RTS/CTS/ACK) packets exchanging method. If the medium is *idle*, the node sends a RTS packet, and the receiver sends a CTS packet after receiving the RTS packet. All the neighbors receiving the RTS and/or CTS packets do not transmit for the duration of the main transmission. An ACK is transmitted to confirm the successful transmission. The waiting time of neighbors is set based on Network Allocation Vector (NAV) which is extracted from RTS/CTS/ACK packets. NAV in IEEE

802.11 acts as a virtual carrier sense by predicting when the medium will be busy.

In [5], M. A. Bonucceli *el al.* proposed a solution that applies frame skipping and transcoding together with frame rate reduction techniques over IEEE 802.11 based vehicular networks. The main idea of the proposed frame skipping and transcoding is that each sender node monitors the channel access delay for a video frame transmission. When it detects that the delay will exceed the maximum admitted value, the sender skips the frame to save bandwidth, and the next frame will be transcoded. If skipping occurs on two consecutive frames, the sender assumes that the network is congested, and decides to reduce the frame rate. Their system achieves a better video quality and saves bandwidth compared to IEEE 802.11 standard protocol based networks.

F. Soldo *el al.* in [46] designed a Time Division Multiple Access (TDMA) scheme to minimize the chance of collisions in adjacent areas. Their TDMA scheme is based on graph coloring techniques. A c-coloring of a graph G is a mapping of node V into {1,...,c} such that every node has a different color than its neighbors. Their TDMA scheme uses the colors to represent the slots of nodes. Within a time frame, nodes in the same color transmit packets at the same slot. As shown in Figure 2.9, *node* 1 has a different color (slot) from its neighbours (*node* 0, *node* 2 and *node* 6). When *node a* transmit packets, its neighbour are waiting for their slots. Therefore, collisions are avoided.

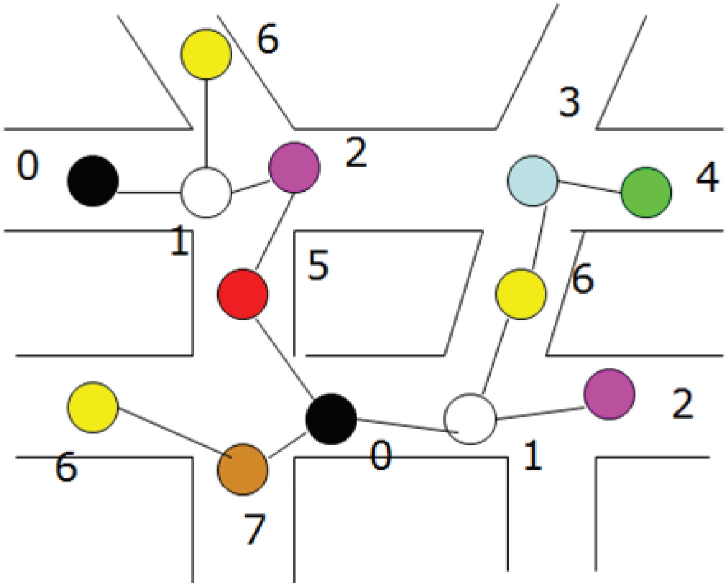

Figure 2.9: An Example of TDMA

2.3.2 IEEE 802.11p (WAVE)

The Wireless Access Vehicular Environment (WAVE) is a standard of VANETs set by the IEEE 1609 Working Group [48]. WAVE is also referred to IEEE 802.11p. The 802.11p Standard uses the medium access procedure of IEEE 802.11e: the Hybrid Coordination Function (HCF). Within the HCF, there are two methods of channel access: HCF Controlled Channel Access (HCCA) and Enhanced Distributed Channel Access (EDCA). The 802.11p follows the EDCA scheme.

The EDCA allows messages with a higher priority to have a higher chance of being transmitted. The Arbitration Inter-frame Space (AIFS), in addition to the contention CW size, is shorten for higher priority messages. CW has two parameters: CWmin, the minimum Contention Window size; and CWmax, the maximum Contention Window size. In addition, EDCA also provides contention-free access to the channel for a period called a Transmit Opportunity.

The channels of DSRC are divided into six Service Channels (SCH) and one Control Channel (CCH). SCH are used to exchange non-safety and long stream data and CCH is reserved for communication coordination and safety message delivery. Since the standard does not mandate the use of several antennas, it contains a channel hopping scheme in which every 50ms the transceiver is switched between CCH and SCH. If the MAC is currently under the CCH, it buffers the messages from the layer above to be transmitted until the SCH becomes active. On the other hand, beacon messages are also buffered if the SCH is currently active.

G. Maia *et al*. in [28] argued that at the beginning of the SCH, all the vehicles transmit all buffered messages with maximum throughput. This causes severe video packet losses as a significant amount of packets is transmitted at the same time. They proposed a rate control mechanism that guarantees that video messages are sent down to the MAC layer only when the SCH is active. This is to avoid the synchronization at the beginning of every SCH cycle, and thus to overcome the collision problem.

C.-W. Hsu *et al*. in [16] suggested that the initial setting value of CWmin will affect the performance. In order to adapt the CWmin according to the number of vehicles, they proposed an adaptive contention window mechanism called AOS to reduce contention and collision based on the modified initial value of CWmin. AOS approach is based on two important functions: the Contestant Estimation (CE) and Expected Offset Calculation (EOS). CE is used to make an estimate of the potential number of neighbors. EOS is in charge of picking the offset slots based on the estimated number of neighbors which is calculated by CE. As the result shows, AOS has a lower packet loss probability compared to other fixed contention window solutions. F. Naeimipoor *et al.* in [31] introduced AOS to their hybrid video dissemination solution to avoid network congestion over high way scenarios. Their results showed that it is a reliable technique for broadcasting high quality videos over VANETs.

2.4 Summary

In this chapter, we primarily provided a classification of the techniques that weres recently proposed to address challenges involved in video streaming over VANETs. Various ways to enhance the quality of received video for vehicles in VANETs using concepts like video coding, error resilience, routing, store- carry-forward and CW control and etc., are discussed here. This chapter provides a broad insight into the research done in video streaming over VANETs.

Chapter 3

Routing Schemes

To the best of our knowledge, only a very small number of researches are devoted to evaluating the performance of routing schemes for video dissemination protocols over VANETs [10] and [55]. In [10], Ghafoor et al. reviewed recent routing protocols for unicast, broadcast and geocast approaches. Their quality based study provides a clear insight to current video dissemination techniques. In [55], Fei Xie et al. compare the sender based forwarding and receiver based forwarding schemes in a highway scenario. They also used real video data and NS2. To analyze the ability to avoid broadcast storms, the performance of selected routing protocols is studied in part.

3.1 Streaming Urban Video (SUV)

With simple algorithms like Flooding, each node acts as both a transmitter and a receiver. Whenever a node receives packets, it forwards them immediately. This kind of scheme introduces some disadvantages for video dissemination, such as packet collision and packet latency. The protocol outlined in [46] shows that SUV can reduce these disadvantages and can maximize the network coverage area by selecting a subset of nodes as the relay nodes. In SUV, each vehicle periodically broadcasts hello messages, which carry sender Identification (Id). Vehicles can maintain an updated list of their 1-hop neighbours. SUV has a sender-based node selection scheme with the definition of *parent node* and *children nodes*. If *node X* receives a packet from *node P*, then *node P* becomes its *parent node*. In SUV, only the relay nodes selected by a *parent node* in turn select their relay nodes (*children nodes*) and forward packets.

There is no method of selecting *children nodes* for the source node in [46]. In this

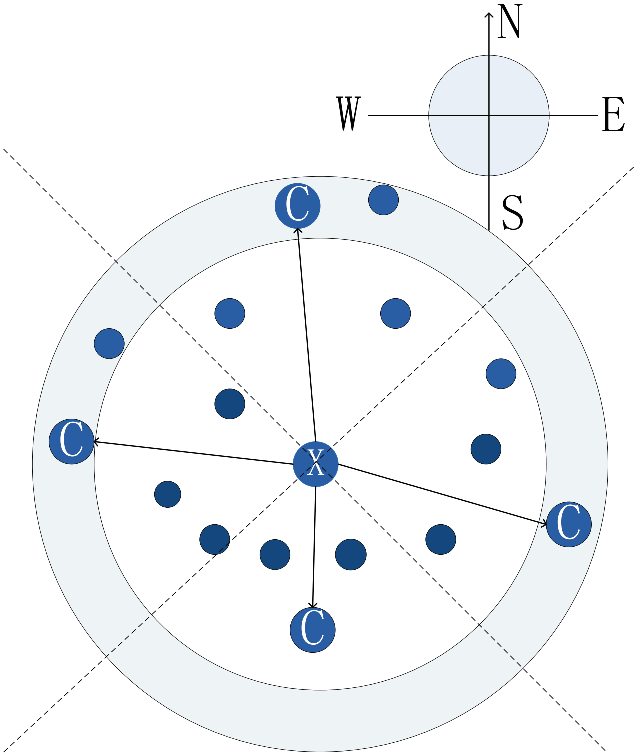

Figure 3.1: Node Selection Scheme of SUV Source Node

study, we propose a method as shown in Figure 3.1, the source node divides its surrounding space into four identical sectors and each sector covers 90 degree. It then selects one *child node* for each sector $(N, E, S \text{ } and \text{ } W)$ according to the following procedure:

- It first sets a shaded ring which is from $2/3R$ to R of its surrounding space, where R is the radio range of nodes.
- It then selects the neighbor node which is near the *bisectrix* of this region and in the shaded ring, for instance, the *children nodes* in region W and N.
- Otherwise, it selects the node in the shaded area as shown in region E.
- If no node can be found, it selects the node closer to $2/3R$ like region S.
- The sector is declared as *unschedulable* if no node is selected.

If *node* X receives a packet from *node* P and it is one of *node* P's *children nodes*, X therefore selects three *children nodes* from sectors S, N and E (since the sector W is taken up by P as its parent node), and relays the packet. According to *node* P's position, *node* X defines a preferred range for choosing *children nodes*. As shown in

Figure 3.2, the shaded ring is defined by P_{in} and P_{out} which are $1/6R$ away from *node* P. *Node* X selects its *child node* for each sector based on the following procedure:

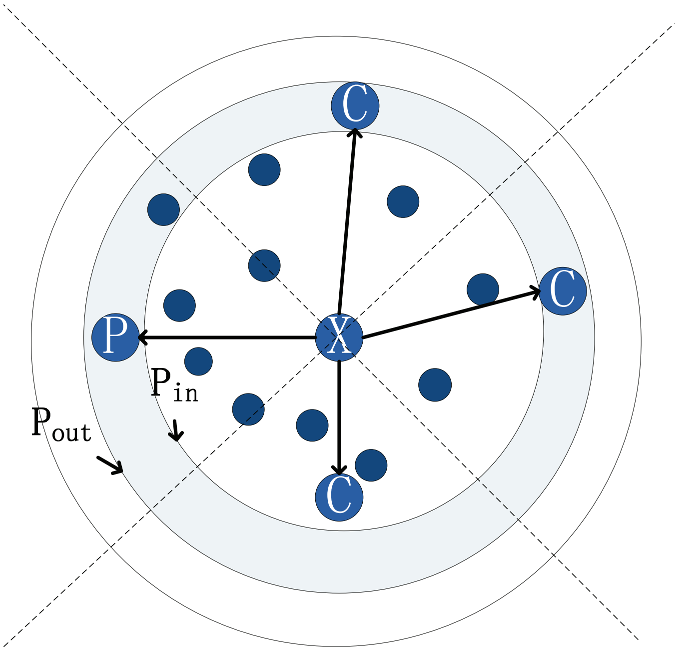

Figure 3.2: Node Selection Scheme of SUV Intermediate Node [46]

- It selects the node neighbour near the *bisectrix* of this region and in the preferred range, such as in the region N.
- If there is no such node, it selects the node in the preferred range such as that of region E.
- If nothing to select, it selects the node closer to the preferred range and near the *bisectrix* as shown in sector S.
- If there is still no node selected, this sector is declared as *unschedulable*.

By this sender-based manner, SUV can reduce the collision by limiting the transmission of duplicated packets.

Algorithm 1: CHECK INTERMEDIATE NODE for *Node A*

```
Initialize:
$N$: the one-hop neighbour set of node $A$;
$R$: the communication range;
for each neighbour node $m$ in the neighbour set do
    for each neighbour node $n$ in the neighbour set do
        $D$ = Distance between $m$ and $n$;
        if $D > R$ then
            return true;
return false;
```

Algorithm 2: CHECK INTERGATEWAY NODE for *Node A*

```
Initialize:
$N$: the one-hop neighbour set of node $A$;
$R$: the communication range;
if CHECK INTERMEDIATE NODE use Algorithm 1 then
    for each neighbour node $B$ in the neighbour set do
        $if\ Covered$ = true;
        for each neighbour node $i$ in the neighbour set do
            $D$ = Distance between node $B$ and node $i$;
            $K_a$ = the key of node $A$;
            $K_b$ = the key of node $B$;
            if $D > R$ or $K_a > K_b$ then
                $if\ Covered$ = false;
    if $if\ Covered$ then
        return false;
return true;
```

Algorithm 3: CHECK GATEWAY NODE for *Node A*

```
Initialize:
$N$: the one-hop neighbour set of node $A$;
$R$: the communication range;
if CHECK INTERGATEWAY NODE use Algorithm 2 then
    for each neighbour node $B$ in the neighbour set do
        for each neighbour node $C$ in the neighbour set do
            $if\ Covered$ = true;
            for each neighbour node $i$ in the neighbour set do
                $D_b$ = Distance between node $B$ and node $i$;
                $D_c$ = Distance between node $C$ and node $i$;
                $K_a$ = the key of node $A$;
                $K_b$ = the key of node $B$;
                $K_c$ = the key of node $C$;
                if $D_b > R$ and $D_c > R$ or $K_a > K_b$ or $K_a > K_c$ then
                    $if\ Covered$ = false;
        if $if\ Covered$ then
            return false;
return true;
```

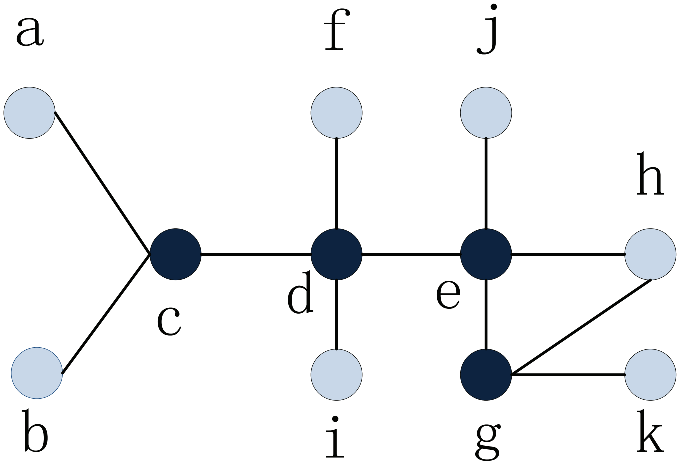

Figure 3.3: An Example of CDS

3.2 Connected Dominating Set (CDS)

In order to evaluate the performance of SUV, a receiver-based node selection scheme called the CDS [54], is implemented in this study. CDS is a virtual backbone of the whole topology which can quickly adapt to network topological changes. In order to reduce collision and overhead, only nodes in CDS forward the packets. To calculate the set, nodes in CDS also broadcast hello messages periodically as in SUV. Let $G(V, E)$ be the graph of vehicular topology, where V represents a respective vehicle and E is the connectivity between them. V_{CDS} is the required subset. The definition of V_{CDS} is as follows: any node $v \in V$ is either in V_{CDS} or has at least one neighbour belonging to V_{CDS} and $G(V_{CDS}, E_{CDS})$ is connected. Therefore, when a packet is rebroadcasted by the nodes in CDS, the whole network can be covered. Figure 3.3 is a simple example, in which the CDS is {c, d, e, g}.

Unfortunately, identifying the minimum CDS is a NP-hard problem [54]. This study employs the heuristic method used in [54]:

- If a node has two unconnected neighbours, it becomes an *intermediate node*. The judging method is shown in in Algorithm 1.
- If an *intermediate node* is not covered by any node, it is known as an *intergateway node*. *Node A* is covered by *node B* means that each neighbour of node A is a neighbour of *node B*; and, $key(A) < key(B)$ (as shown in Algorithm 2).
- An *intergateway node* that is not covered by any pair of connected nodes, becomes a *gateway node*. *Node A* is covered by two connected nodes (*node B* and *node C*). This means that each neighbour of *node A* is either the neighbour of *node B* or the neighbour of *node C* (or both); and, $key(A) < key(B)$ and $key(A) < key(C)$ (line 24 - 39 in Algorithm 3).

The *gateway nodes* are collectively known as the Connected Domination Set.

3.3 Reactive, Density-Aware and Timely Dissemination Protocol (REACT-DIS)

As mentioned before, the drawback of SUV and CDS is that broadcasting hello messages periodically causes high overhead. Therefore, REACT-DIS [42] which is reactive to constant topology changes without relying on hello messages is selected in this study.

As shown in Figure 3.4, the authors in [42] define a procedure for nodes in REACT-DIS so that these nodes may assess whether to forward the received packets when they are idle. Nodes wait t seconds before making the decision. The value of t is chosen within the defined waiting time range $[\ \tau,\ \Gamma\ ]$ (τ and Γ are the minimum and maximum waiting time respectively) base on the distances between the receiving nodes as shown in Equation 3.1.

$$t = \tau + \left[\left(1 - \frac{\delta(n_s, n_r)}{R} \right) \times (\Gamma - \tau) \right] \tag{3.1}$$

In Equation 3.1, R is the radio range and $\delta(n_s, n_r)$ is the distance between the sender node n_s and receiver node n_r. Before t expires, nodes observe and count the number of duplicate packets c and the number of total received packets s during the waiting period. When t seconds run out, the nodes then forward the packets with a probability ρ which follows Equation 3.2.

$$\rho = \frac{1}{\gamma^{\lfloor c/s \rfloor}} \tag{3.2}$$

In Equation 3.2, γ is the defined forwarding probability reducing factor. Through this manner, REACT-DIS performs as a reactive and density-aware protocol, because the relay of packets depends on node distance to the last hop sender, the quantities of specific forwarded packets and the density of its region.

As repeated competitions are unnecessary and lead to excessive average delay, nodes in REACT-DIS continue to be suitable as relay nodes for a predefined amount of time ϕ. Therefore, within this period, any received packet is immediately forwarded by the relay nodes to make sure that REACT-DIS performs in a timely manner. After this period, the relay nodes revert to an idle status. However, during the period ϕ, an excessive number of nodes may consider themselves as suitable relay nodes. Authors in [42] propose a ϱ-probability function (as Equation 3.3 shows) for each node to decide whether or not to forward a specific packet. The value k is the preferred number of relay nodes per hop and in this study we chose four. Λ is the relay node forwarding probability reducing factor.

$$\varrho = \begin{cases} 1.0 & if\ c \leq k \\ 1.0 - [(c-k) \times \Lambda] & if\ c > k \end{cases} \tag{3.3}$$

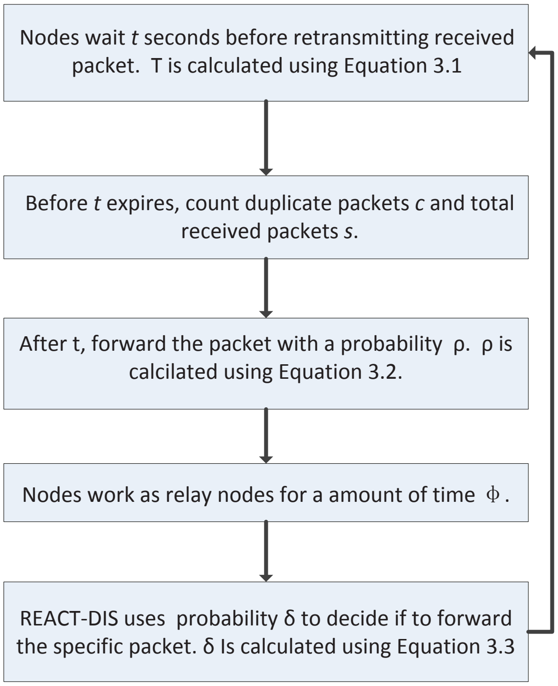

Figure 3.4: the Procedure of REACT-DIS

3.4 Simulation Settings

Compared to broadcasting other types of data, the quality of video content is sensitive to data rate. Therefore, the data rate is selected in our implementation to provide different quality level measurements. We have 100 *kbps* (kilobits per second), 200 *kbps*, 500 *kbps*, 1000 *kbps* and 1600 *kbps*. To determine whether the protocols meet the general video dissemination requirements, we evaluate the following metrics:

- **Packet loss ratio:** is the primary measure of promising protocols [47]. The definition used in this work is the average percentage of video message lost by vehicles. And it is calculated as:

  $$packet\ loss\ ratio = 1 - the\ delivery\ ratio \tag{3.4}$$

  .

- **Delay:** the average time it takes for messages to travel from a source vehicle to other vehicles. In [47], Cisco defines some requirements for sending video content. In the case of video dissemination, average delay should be less than 4-5 seconds.

- **Network overhead:** the total number of video messages transmitted by all vehicles. It is chosen to evaluate the cost and to estimate the feasibility of the selected protocols. A high overhead indicates that lots of duplicated messages are retransmitted.

- **PSNR:** is the most commonly used QoS parameter to measure the quality of reconstructed video. According to [52], PSNR uses a value to reflect the video quality as shown in Table 3.1.

- **MOS:** according to [49], PSNR reflection of perceived video quality is limited and QoS often fails to correlate with human perception. Therefore, MOS, as a common way of subjectively characterizing QoE, is used in this work to describe the human impression of video quality. MOS is on a scale from 5 (best) to 1 (worst) as shown in Table 3.2 [12].

PSNR Value	Video Quality
over 40 dB	an excellent video quality
30-40dB	the quality of video is very good
20-30db	the quality is quite poor
less than 20dB	the video is unacceptable

Table 3.1: PSNR and Video Quality

In order to evaluate the selected routing protocols, we implement them using NS2 [30]. This simulator is the most widely used and represents with high fidelity real wireless communication. The simulation parameter is set according to Table 3.3. For the video dissemination protocols, hello messages are generated with a frequency of 2 Hz. To increase the accuracy of our simulation, we use SUMO 0.16.0 [22] traffic simulator to build 20 urban scenarios in Ottawa and to produce the vehicles' movements. A map of the urban area of Ottawa (as shown in Figure 3.5) is obtained from OpenStreetMap [14].

Scale	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible, not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

Table 3.2: ITU-R Quality and Scale Quality

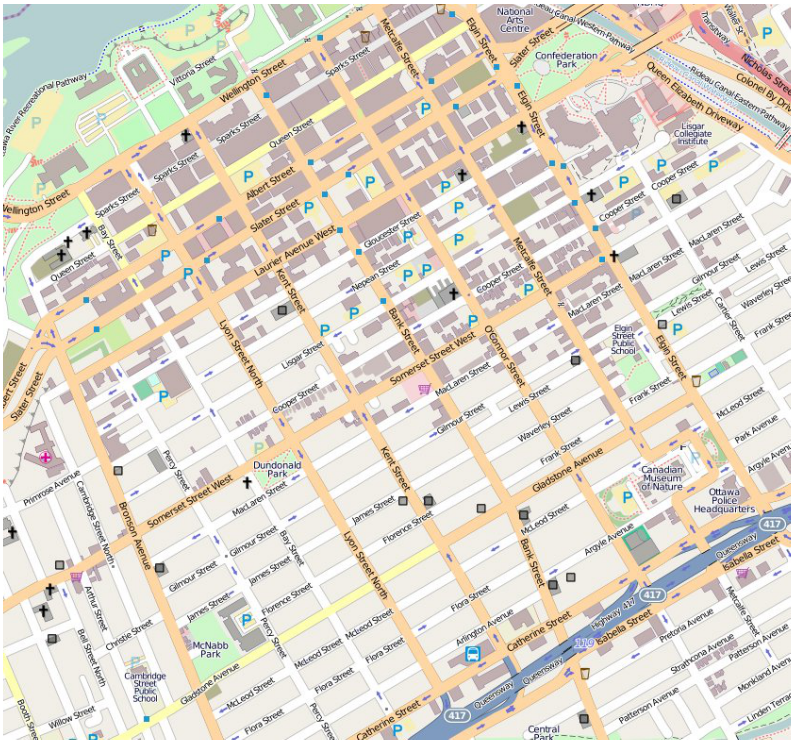

Figure 3.5: Urban Area of Ottawa

Parameter	Value
Radio Propagation Model	TwoRayGround
MAC Layer	IEEE 802.11
Radio Range	300m
Antenna	OmniAntenna
Number of Nodes	436 - 456
Simulation Time	300 seconds
Area Size	$2305m \times 2337m$
Source Location (x, y)	(954.96, 1249.70)
Vehicle Speed	0 - 27.78 m/s

Table 3.3: Simulation Parameters

The video transmitted in our simulations is the well-known video which is named akiyo_cif [21]. It is a Motion Picture Expert Group (MPEG) video with a resolution of 360 × 486 and contains 300 frames. The video is encoded at 30 frames/sec (fps), and is divided into the payload of 1,000 bytes of 353 different packets. Evalvid [21], a video quality evaluation tool-set, is chosen to evaluate the performance of selected video dissemination protocols. EvalVid uses the received files to compare with original video trace file then provides latency, delievery ratio, PSNR, MOS and etc for receiver nodes. In this paper, packet loss ratio, average delay, PSNR and MOS are measured by Evalvid. Packet overhead is obtained by counting the total number of broadcasted video packets. In order to obtain statistical measurement results and to generate graphs, R [40] is selected. Each plotted point in the graphs is an average result of 20 trials and confidence intervals are calculated using Student's t-distribution with a confidence level of 95%.

As discussed before, there are five configurations in REACT-DIS which greatly influence its performance. Table 3.4 summarizes the values we use, and these values are provided by [42] in the sense of providing best performance.

Parameter	Value
τ: minimum waiting time	0.10s
Γ: maximum waiting time	0.25s
γ: forwarding probability reducing factor	10
ϕ: the time a node remains as a relay node	5s
Λ: the relay node forwarding probability reducing factor	0.10

Table 3.4: REACT-DIS Parameters

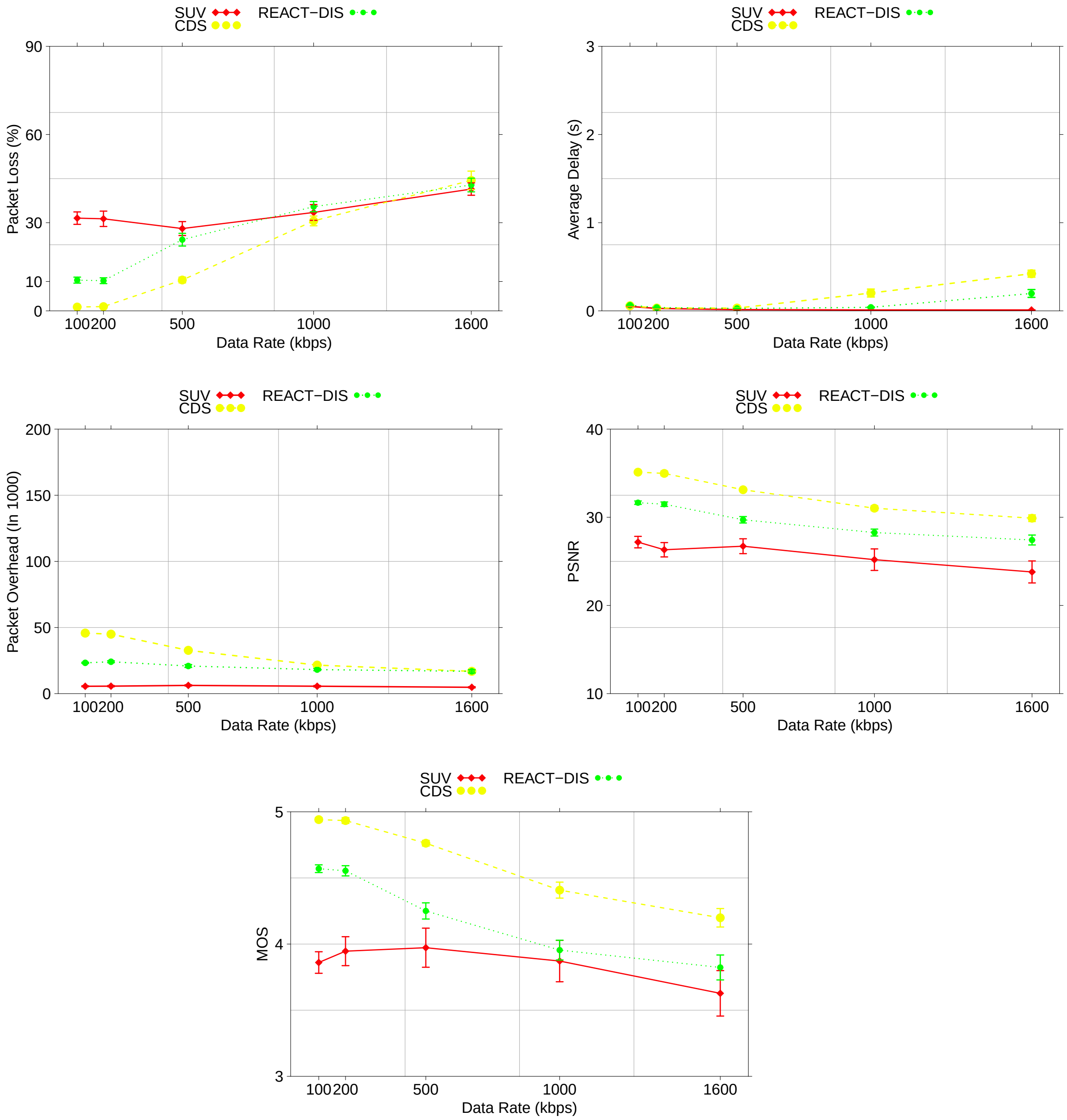

Figure 3.6: Performance Comparison of Routing Protocols

3.5 Routing Scheme Performances Evaluation

Figure 3.6 shows the result of evaluated routing protocols under the mentioned simulation regimes. We conclude that only CDS fulfils the basic PSNR and MOS requirements of video dissemination and is suitable for urban VANETs. Moreover, the receiver-based node selection solutions (REACT-DIS and CDS) outperform the sender-based solution (SUV). These results verify the conclude of the study in [55] in which receiver-based solutions are proofed to be more reactive to constant topology changes than sender-based solutions.

However, SUV has its own advantage. From the simulation results shown in Figure 3.6, we can easily conclude that SUV is the most efficient scheme to avoid broadcast storms and unnecessary retransmissions. It provides the smallest packet overhead values and the shortest delay values. Unlike the other two schemes, the packet loss ratios of SUV do not change significantly when the data rate increases. Its selection scheme can effectively reduce the redundant packets transmissions and avoid collisions. As its latency and overhead are much smaller than the ones of CDS and REACT-DIS, SUV is suitable for low-delay or low-overhead requirement and high packet loss ratio tolerant video streaming applications. In terms of providing good video quality, however, we have to conclude that SUV provides the worst performance. Its PSNR values are less than $30dB$, and moreover, its MOS values are below 4. These results indicate that the disseminated video qualities are unacceptable and SUV is not fit for video dissemination over urban VANETs. Therefore, it is not selected for the studies of following parts.

When taking into account video quality, PSNR and MOS become the main considered metrics. Therefore, the receiver-based schemes CDS and REACT-DIS are more considerable than the sender-based SUV as they provide better video qualities for receivers. For a low data rate, the quality of broadcasted video is good as their PSNR values are all over $30dB$ and their MOS values are all over 4. Therefore, they both provide excellent video dissemination for Urban VANETs scenarios in a low data rate. However, the PSNR and MOS values of REACT-DIS decrease sharply and achieve less than the minimum requirements (PSNR $\geq 30\ dB$ and MOS ≥ 3) for high data rates (1000kbps, 1600kbps). Therefore, between CDS and REACT-DIS, the former is better in terms providing reliable video quality for vehicular network. As CDS and REACT-DIS are both considerable for video dissemination, they are selected for Section 5 to analyze the performance of the combination between error resilient techniques and routing techniques.

Chapter 4

Error Resilient Schemes

Error resilient schemes are the error recovery method used to deal with the packet losses. In this section we select two error resilient schemes: 1) the video coding technique Interleaving which is usually used to conceal errors; 2) the frame coding technique Network Coding which is good at providing redundant packets.

4.1 Interleaving

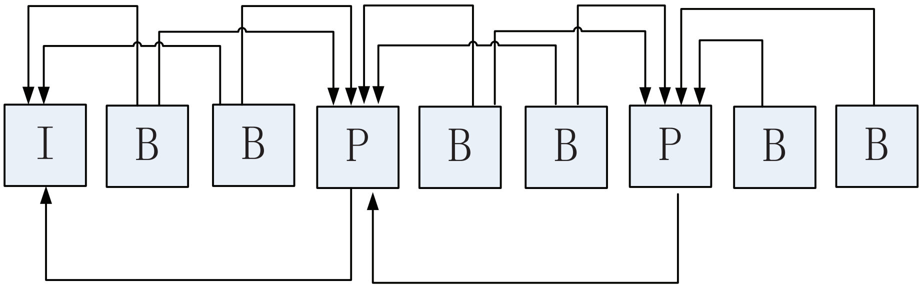

Figure 4.1: MPEG Frames Dependency Relationship

Before transmitting the video content, the file is compressed by MPEG. The compressed file contains 3 different frames: $I\text{-}frame$, $P\text{-}frame$ and $B\text{-}frame$. Figure 4.1 shows the dependency relationships within different frames, and we use I, P and B to present $I\text{-}frame$, $P\text{-}frame$ and $B\text{-}frame$ sequentially. $I\text{-}frame$ is self-contained. Its compressed rate is lower than $P\text{-}frame$ and $B\text{-}frame$. To encode and decode $P\text{-}frame$, the previous *I-frame* and/or *P-frames* in the same Group of Pictures (GOP) are needed. The compression rate of *P-frame* is higher than the rate of *I-frame*. The encoding and decoding of *B-frame* is based on the previous and the following *I-frames* and *P-frames*. The compressed rate of *B-frame* is the highest rate of all three. Therefore, if an *I-frame* or *P-frame* is lost, all frames thereafter in the GoP become undecodable.

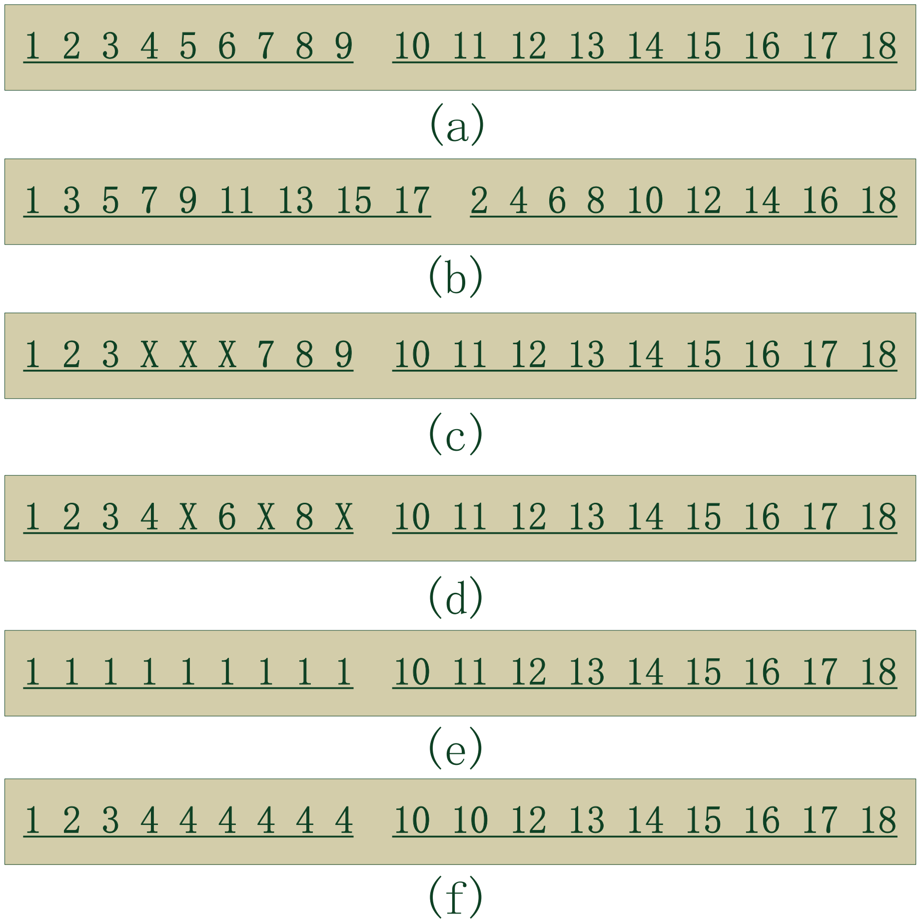

Figure 4.2: An Example of Interleaving

As discussed before, PSNR and MOS are useful parameters to measure the reconstruction quality of a transmitted video. They are used as the major performance metrics in this work. Data rate is the factor which has the greatest influence on disseminated video performance. Yali et al. in [7] proposes a video coding method called IL which can conceal the cluster packet losses and thus maintain acceptable PSNR and MOS values when the data rate increases. The design of IL is based on the definition of repletion error-recovery scheme, in which a lost frame is recovered by repeating the previous consecutive one. The main idea is to spread out the big gap losses of received video stream into several small gaps. At the sender, the frames from k GOPs are interleaved based on the specific distance k. Next, the packets are broadcasted as the method in Flooding. After arriving at the receiver, the frames are reconstructed according to their original order. Figure 4.2 (a) is the original order of 2 GOPs, and the size of each GOP is 9. Figure 4.2 (b) shows an example of Interleaving in which the distance k equals to 2.

In the event that the 4th, 5th and 6th frames are lost during the broadcast, the reconstructed stream as shown in Figure 4.2 (c) is compiled with a big gap without IL. As Figure 4.2 (d) shows, the IL scheme distribute the big gap in to 5th, 7th and 9th. And after decoded, the big undecodable gap (in Figure 4.2 (e)) is broken down into two small gaps as shown in Figure 4.2 (f). Therefore, after using the repletion error-recovery scheme, the PSNR of IL will be higher.

4.2 Network Coding

Recently, the frame coding technique, NC, has been frequently studied as a promising technique to provide redundant packets [42] [34]. NC proposed by Ahlswede et al. [1] can provide robust performance and high throughput in wired or wireless networks. In NC, assume *node p* receives two packets: *packet a* and *packet b*. Instead of forwarding them sequentially, *node p* combines them using a function f and forwards packet $f(a, b)$ to improve network throughput. In this study, we evaluate Random Linear Network Coding (RLNC) [15]. As shown in Figure 4.3, RLNC divides the packets into different blocks and each block contains a certain number of frames. Let $p_0, p_1, p_2, \ldots, p_k$ be k packets in the same block. Each block has a unique block Id, B_{id}, which is equal to the Id of the first packet p_0 of the block. The source node encodes the packets using the following function:

$$C(B_{id}, \eta) = \sum_{i=1}^{\eta} e_i P_{(i-1+B_{id})} \tag{4.1}$$

where the coded block of the source node owns η ($\eta \geq k$) packets and the coefficient e_i is randomly selected from a finite field F. In order to decode the encoded block at the receiver end, the coefficient $E = e_0, \ldots, e_\eta$, block id B_{id}, and block size k, are stored in the header of the coded packet. If a node receives any k of η sent packets can recover the original block by multiplying the received coded block with the inverted coefficients as shown in Equation 4.2.

$$P = E^{-1}C \tag{4.2}$$

After decoding, the receiver selects a new set of coefficients randomly, and broadcasts the new encoded packets.

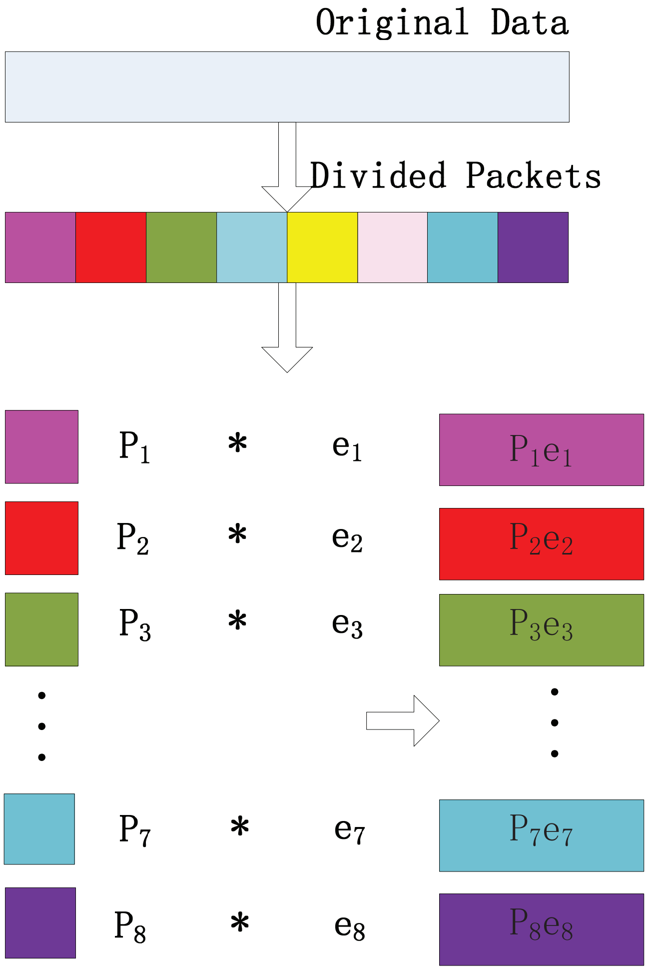

Figure 4.3: Random Linear Network Coding

This study also implements the a RLNC-based protocol CodeCast [34]. The author in [34] modified the Network Coding scheme as follow: at the time of receiving the first packet of a new block, receivers schedule a long enough timer for receiving the entire block. In this case, if a receiver gets the full block when the timer runs out, it decodes the received coded block and encode a new block for broadcast. In CodeCast, no redundancy packets are encoded, and each coded block owns η packets which means η equals to k in Equation 4.1.

There is a deterministic node selection scheme in [34]. The scheme defines two parameters: *dist* and *vldd*. Each node keeps a local variable *dist*, which indicates the hop distance from the source node and copies the value to the header of the forwarding packets. When a node receives a new packet, it recalculates its *dist* value as one plus the maximum *dist* value found in the packet header. In addition, a neighbour that maintains a larger *dist* value is considered as a *downstream node*. If the node has received a previous-block coded packet with a *vldd* set from its downstream nodes, its *vldd* is set and transmitted with its forwarding packets. If a node receives at least one packet with a *vldd* set transmitted by any downstream node while processing *prunecount* packets, it arrives in the selected subgraph. Otherwise, it is pruned from the topology for *sleeptimout* seconds. The values of *prunecount* and *sleeptimout* used are 15 and 0.08 respectively.

In CodeCast, if the receiver fails to receive the whole block, it broadcasts a help message to direct its neighbours which have received the same block successfully to broadcast its required packets. However, when the node density or data rate is high, CodeCast results in an significant number of broadcast storms. Broadcasting help packet and response packets also cause more packet collisions and worse results. Therefore, this study modifies CodeCast by removing the help scheme, and call the modified version CodeCast*. The comparison of their results is shown in next part.

4.3 Error Resilient Scheme Performances Evaluation

The evaluation of the performance of the selected error resilient schemes follows the same configuration setup as the one used in Section 3. The scenarios and the video are the same as well. The same statistical methodology is used. Evalvid is used for measuring the same selected metrics in this simulation work.

4.3.1 Interleaving

To evaluate the ability of IL to improve the quality of disseminated video, we will compare the performance of Interleaving with Flooding; Using the same broadcast scheme,

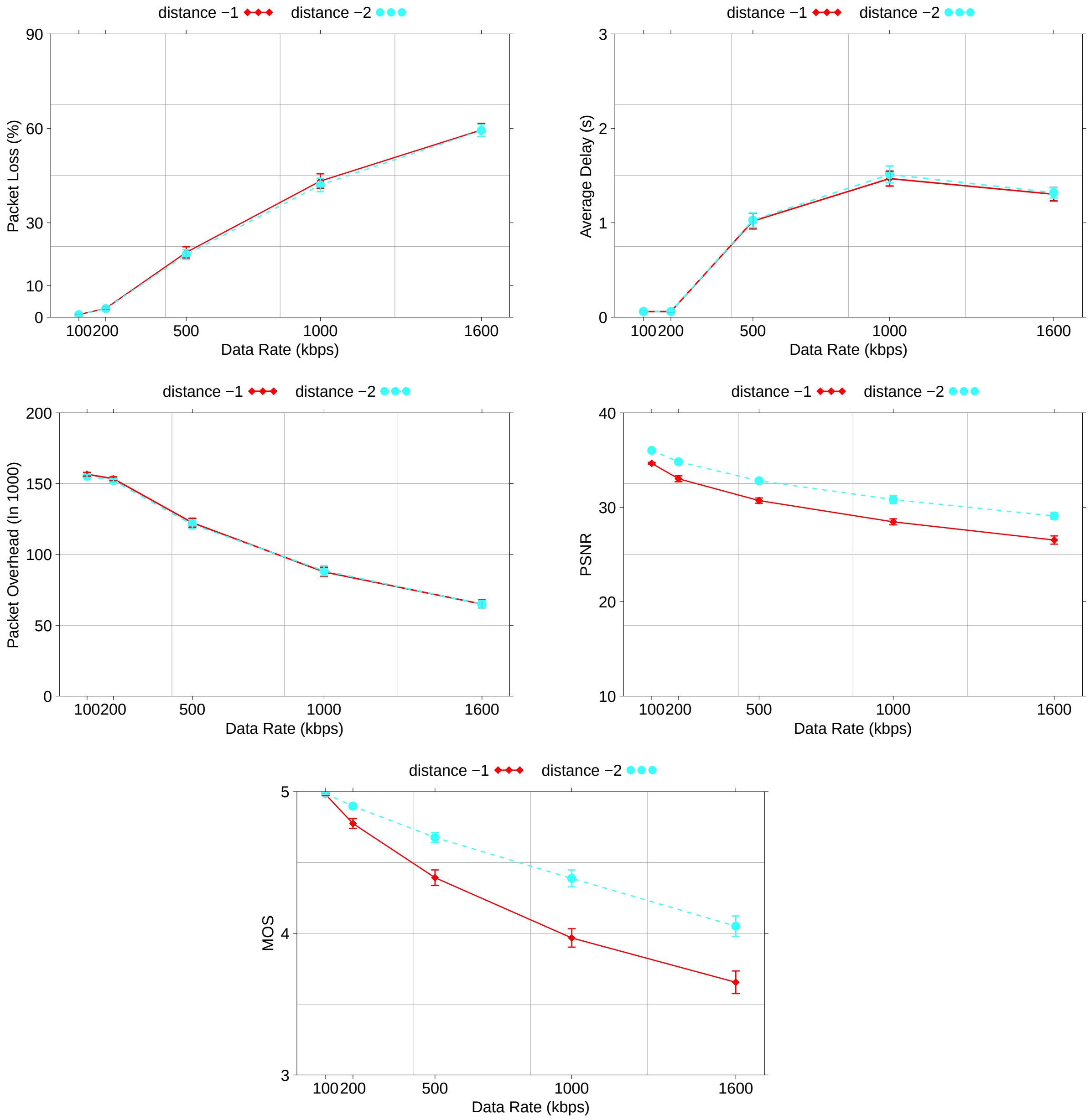

Figure 4.4: Performance Comparison of Different Distances of Interleaving

Flooding is the same as IL which use 1 as distance. Authors in [7] have indicated that increasing the interleaving distance is not helpful to the performance of the IL algorithm. Therefore, in this study, we use 2 as the interleaving distance. Figure 4.4 shows the performance of IL with *distance* 1 and *distance* 2.

As shown in Figure 4.4, IL has no influence on packet loss ratio, average delay or overhead as *distance* 1 and *distance* 2 have almost the same values. This is because Interleaving just changes the sequence of packets, and it keeps the packet size and the total number of packets. In 4.4, we can easily conclude that by spreading out a large loss gap into small gaps, Interleaving could improve the performance of video dissemination. We can also achieve the collusion that the PSNR values of *distance* 2 are higher than the ones of *distance* 1. In addition, as shown in Figure 4.4, the MOS values of *distance* 2 are higher than for those of 4 which indicate that the video qualities are between good and excellent. Therefore, Interleaving is a reliable scheme which conceals the errors and provides higher quality videos.

4.3.2 Network Coding

As the results show in Figure 4.5, both CodeCast and CodeCast* have high packet loss ratios and small PSNR values. The packet loss ratios increase sharply in accordance with the data rate. In particular, with a high data rate (1600 *kbps*), their PSNRs are less than $25dB$. This happens in Network Coding because the receiver either recovers the whole block after receiving k linear independent packets, or otherwise, completely discards the whole block as unrecoverable. When the data rate is high, lots of blocks in Network Coding can not be recovered because of high packet loss ratio. It is also worth noticing that with a high data rate, CodeCast transmits plenty of help messages and response packets, and it triggers a considerable amount of broadcast storms. On the other hand, deleting *help scheme* of CodeCast* to reduce broadcast storms, has enabled a higher chance of receiving the whole block. Therefore, it has smaller packet loss ratios, higher PSNRs, shorter latencies and larger overheads. It has better performance compared to the original CodeCast. Therefore, CodeCast* is selected for the comparisons in the following parts. In order to evaluate the performance of RLNC, we combine it with two routing techniques and get $Network\ Coding\ over\ REACT-DIS$ ($NC-RD$), $Network\ Coding\ over\ CDS$ ($NC-CDS$). Figure 4.6 shows the combination idea. In the application layer, we use RLNC to encode the packets, and then transmit the encoded packets over the scenarios using CDS and REACT-DIS sequentially.

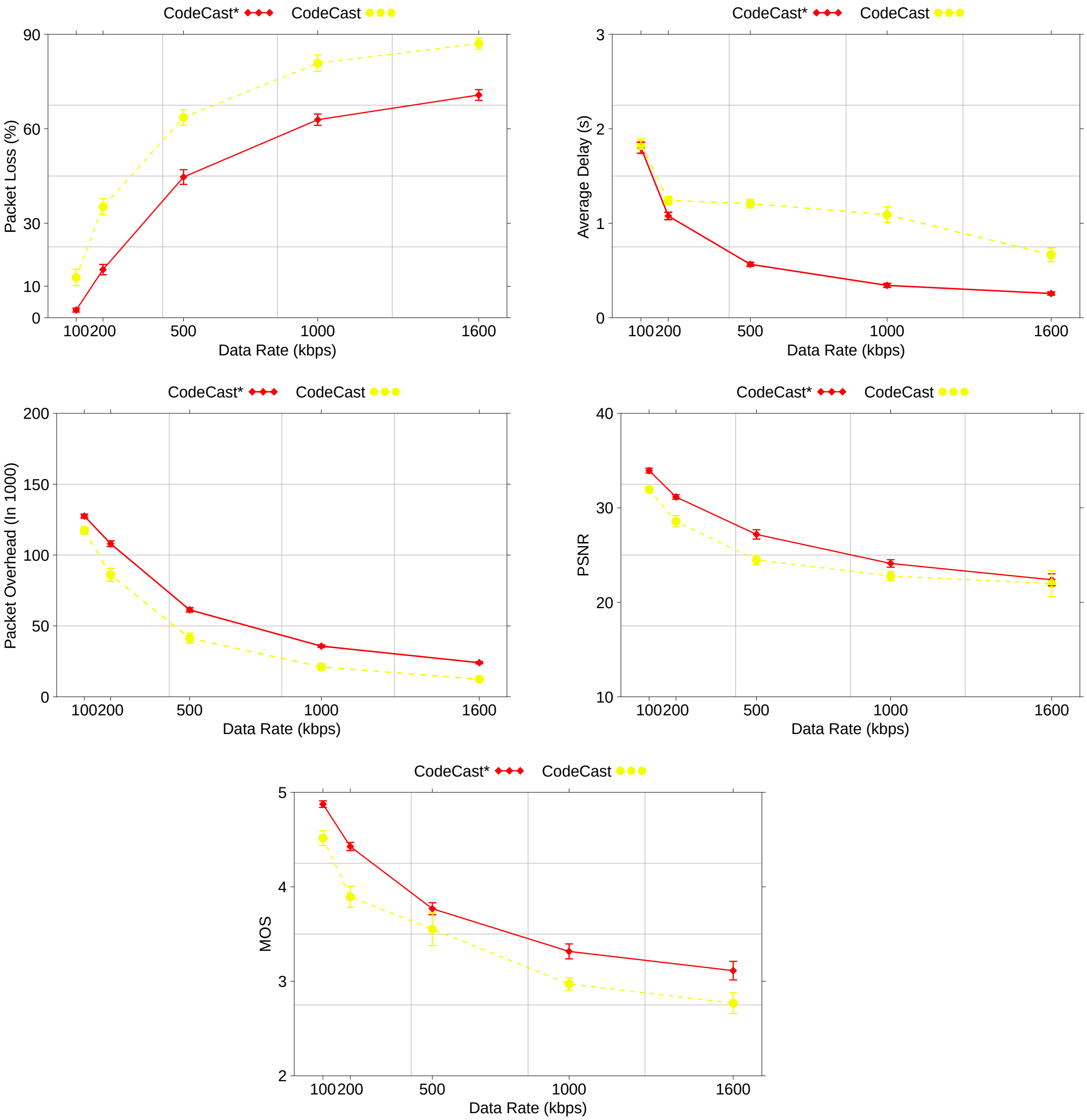

Figure 4.5: Performance Comparison of Network Coding-Based and Original Routing Protocols

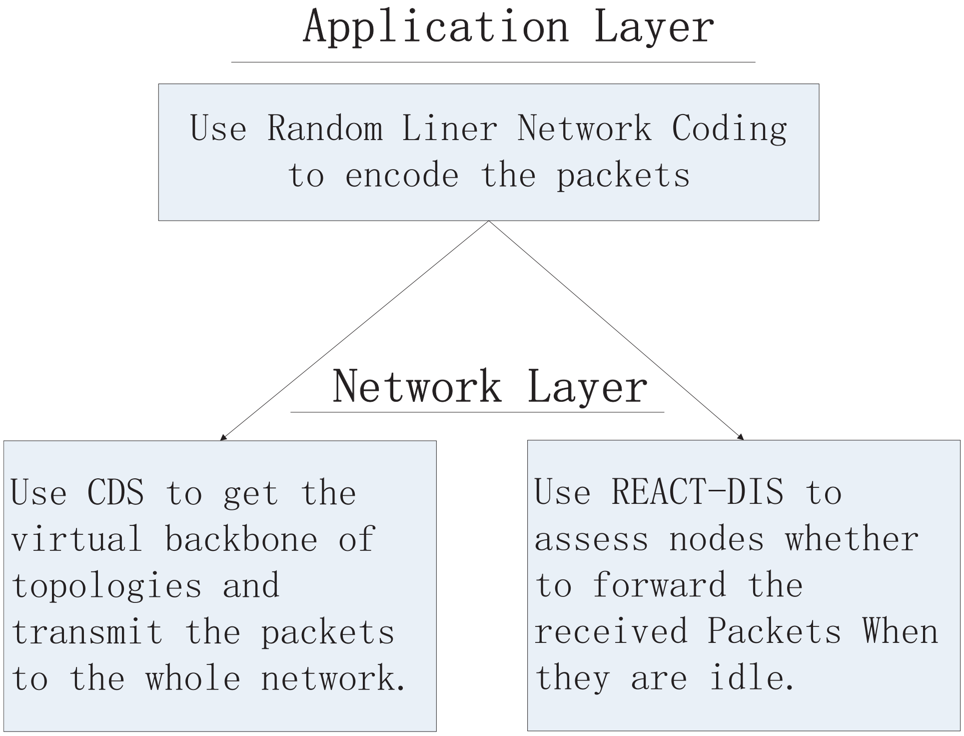

Figure 4.6: Network Coding over Selected Routing Schemes

Protocol	k	η	*receiving timer*
CodeCast	8	8	yes: $k \times datarate$
RD-NC	8	12	no
CDS-NC	8	8	no

Table 4.1: Coding Parameters of Network Coding based Protocols

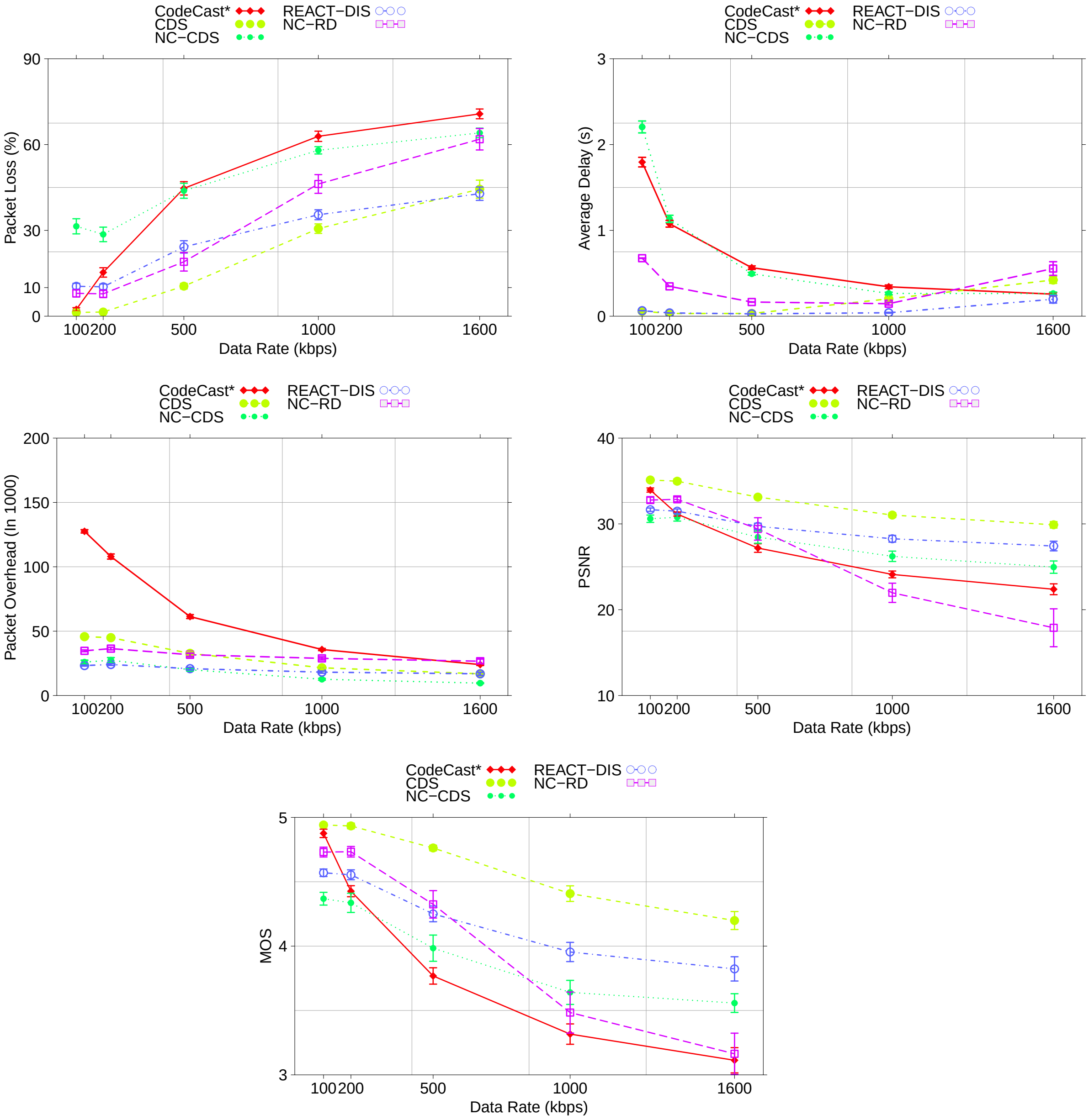

Figure 4.7: Performance Comparison of Network Coding-Based and Original Routing Protocols

Table 4.1 shows the parameters we use for Equation 4.1. We are using 50% additional redundancy for NC-RD. As shown in Figure 4.7, CodeCast provides the worst performance overall the protocols. It results in the most packet loss ratio, average delay and overhead. Moreover, it has the lowest PSNR and MOS values. In CodeCast*, nodes only re-encode and forward the blocks which are received at the right time. Regardless of the node selection schemes, it forwards less packets than the NC in NC-RD and NC-CDS. However, it shows highest overhead of all. That is to say NC-RD and NC-CDS have better node selection scheme than CodeCast*. Their node selection schemes could reduce more redundant packets.

In NC, the receiver either recovers the whole block after receiving k linear independent packets, or otherwise, completely discards it as unrecoverable. With the same forwarding scheme as Flooding, even if nodes in NC receive packets, they discard a lot as they can not decode them. Therefore, Network Coding-based Flooding performs worse than Flooding in terms of delivery ratio, PSNR and MOS when the parameters in Equation 4.1 meet that η equals to k. In addition, the major factor of delay of NC is the necessary waiting time needed to receive the whole block in order to decode and encode a new block for later retransmission. This also leads to longer end-to-end delays for NC-based protocols.

From Figure 4.7, we can first easily conclude that NC-CDS performs worse than CDS in every metric. In NC-CDS, the source node contains same number η of frames in each block as it encoded k. There are no redundancy packets for the receivers to recover the blocks. Therefore, its worse performance is not unexpected. On the other hand, as shown in Table 4.1, in NC-RD, NC is implemented with 50% redundancy packets. However, it performs worse than REACT-DIS. Its 50% redundancy retransmission scheme causes the well-known broadcast storm problem and results in severer packet losses, higher delay and higher overhead.

Chapter 5

Error Resilient Schemes over Selected Routing Schemes

5.1 Combination Scheme

To get an in-depth look into the performance of routing schemes and error resilient schemes, we combine Interleaving with two routing techniques and get $Interleaving$ $over\ REACT-DIS$ ($IL-RD$) and $Interleaving\ over\ CDS$ ($IL-CDS$). As shown in Figure 5.1, the video content is first interleaved using the method mentioned in Chapter 4. After that, the streaming is disseminated over urban VANETs scenarios using the two receiver-based routing solutions described in Chapter 3. Finally, the received packets are reorganized to the original sequence at the receivers. To select the best one of all the combination between error resilient techniques and routing techniques, this study also compares RD-Interleaving and CDS-Interleaving to Network Coding-based protocols (CodeCast*, NC-CDS and NC-RD) proposed in Chapter 4.

5.2 Combination Schemes Performance Evalution

As mentioned in Section 4.3.1, Interleaving has no influence on packet loss ratio, average delay and overhead. The main benefit of Interleaving is to guarantee tolerable video quality, acceptable PSNR and MOS values when the data rate increases. As discussed in Section 4.3.2, Network Coding discards the packets if it has failed to recover the whole block and waits to receive the whole block in order to decode it. The Network Coding-based protocols, therefore, provide worse performance compared to the protocols without

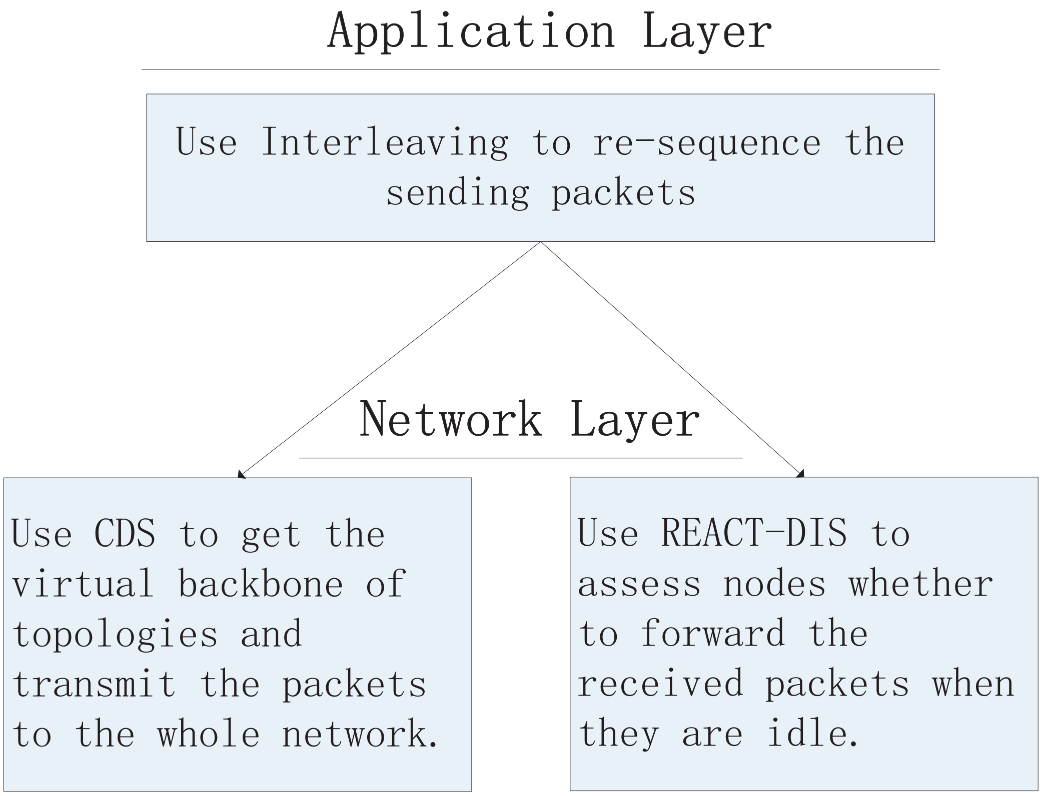

Figure 5.1: Interleaving over Selected Routing Schemes

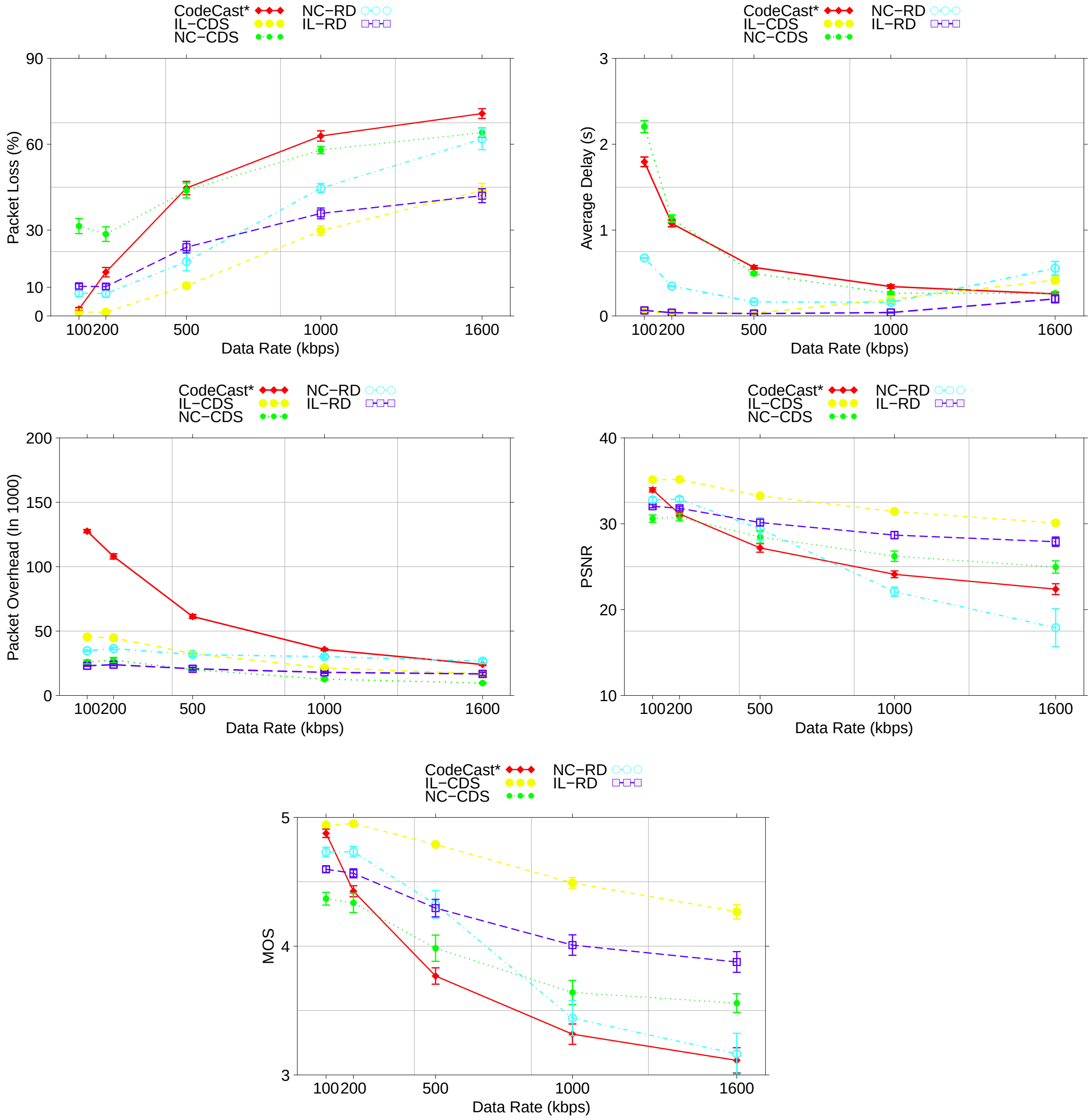

Figure 5.2: Performance Comparison of Error Resilient Protocols

the Network Coding mechanism such as CDS and REACT-DIS. Therefore, the protocols with Interleaving scheme perform better than the protocols with Network Coding scheme.

The performance results are shown in Figure 5.2. CodeCast*, NC-CDS and NC-RD have higher packet loss ratios, smaller PSNR and MOS values compared to the protocols with the IL mechanism such as IL-CDS and IL-RD. In terms of average delay, solutions using Network Coding obviously have shorter delays as the data rate increases. In Network Coding based solutions, the delay of waiting for receiving the whole block decreases as the data rate increases. On the other hand, the delays of other protocols increase with the increasing a date rate. This happens because the chance of broadcast storms increases accordingly with the increase of the data rate. The broadcast storms impede the nodes from receiving packets. What's more, the delays are less than 2 seconds ,and therefore, they are still in the acceptable range [47]. We also observe that CodeCast* has largest packet overheads. This means that the routing approaches (CDS and REACT-DIS) could reduce the cost of broadcasting.

Within the IL-based protocols, IL-CDS outperforms IL-RD. It is also the only one that guarantees the acceptable quality of video dissemination. As mentioned in Table 3.1, if a PSNR value is larger than $30dB$, the quality of video is very good [52]. IL-RD can fulfil this requirement only in a low data rate (100kbps, 200kbps and 500kbps). IL-CDS can also meet this requirement for a higher data rate. From [49], the score 4 of a video MOS indicates that the quality is perceptible but not annoying and 5 means the video is imperceptible. IL-RD has the ability to provide a good to excellent video when the data rate is not larger than 1000kbps. When the data rate increases to 1600Kbps, the video MOS score, however, decreases to less than 4 which means the disseminated video is slightly annoying. But as shown in Figure 5.2(e), IL-CDS makes sure that the video scores are larger than 4 and that the video quality is more than perceptible but not annoying. Altogether, IL-CDS is the best out of all the combined protocols.

Chapter 6

A Quality Guaranteed Video Dissemination Protocol

In this section, we design a quality guaranteed video dissemination protocol called QGVD. It is based on the conclusions of the previous parts: 1) the best routing scheme is CDS; 2) IL is a better error resilient scheme comparing with NC; 3) IL-CDS is the best combination. We introduced CDS and IL as our main routing and error resilient solutions for QGVD. Realizing that the topology of urban VANETs is intermittently disconnected, and Interleaving can not fix single packet loss, we designed a store-carry-broadcast scheme to handle this problem. The integral solution is shown in Algorithm 4 and Figure 6.1.

6.1 Streaming Process of QGVD

As described in Part 3, CDS is the best one of the three selected routing schemes. However, its repeatedly computing the connected-domination set is unnecessary and a waste of time. As shown in Algorithm 1 to 3, the computing of each node takes a lot of CPU because it traverse the neighbour list once in a loop and there are up to three loops. We, therefore, add a parameter *status* to each node in QGVD. Each node can exist in three status: in the Connected-Domination Set (IN_CDS), not in the Connected-Domination Set (NOT_CDS) and $IDLE$. The *status* of each node is initialized as $IDLE$ (line 4). Each time a node need to forward a packet, it check its *status*. If its *status* is $IDLE$, it calls Algorithm 1 to 3 to compute if it is in CDS (Line 14-15 in Algorithm 4 and line 13-14 in Algorithm 5). If the *status* is IN_CDS, it forwards the packet, or else

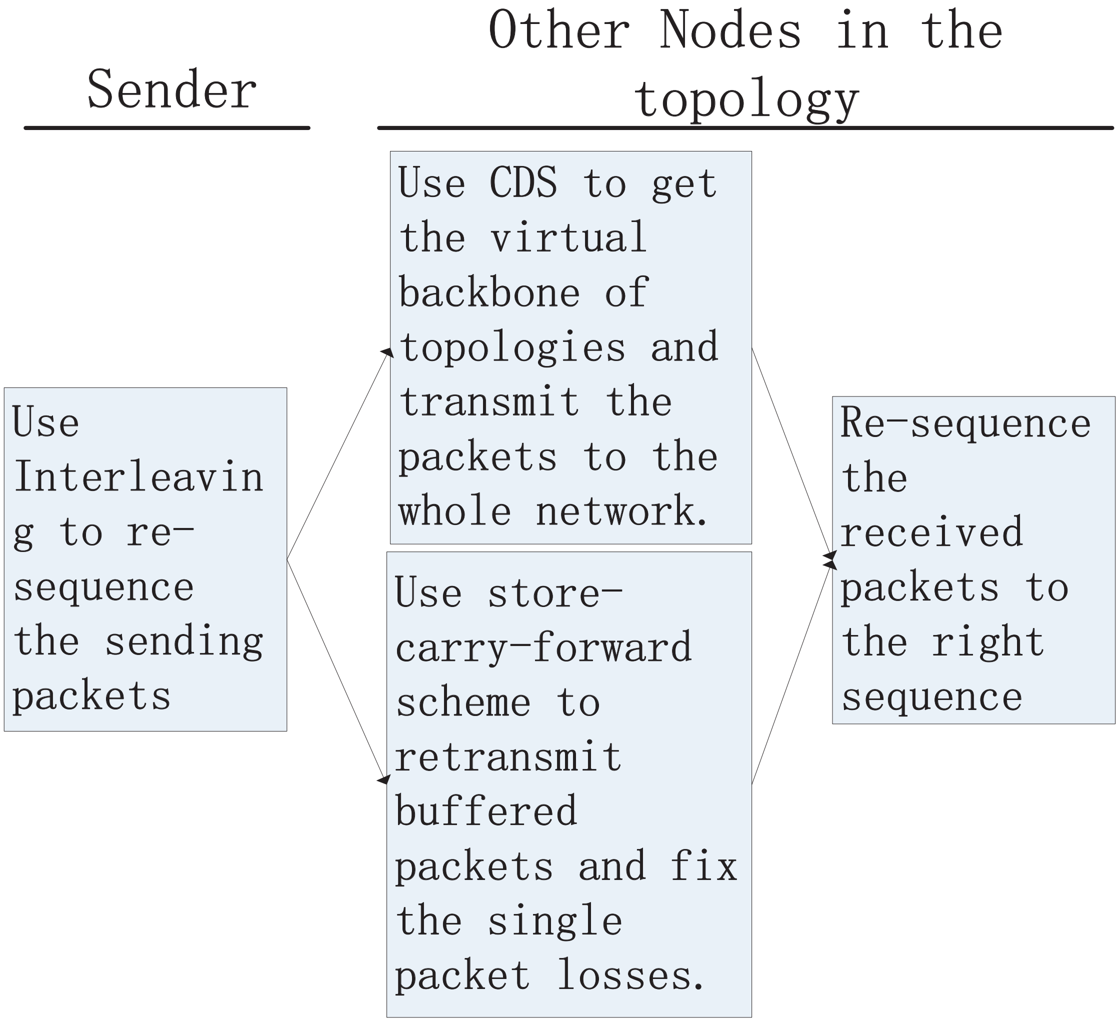

Figure 6.1: The Quality Guaranteed Video Dissemination Protocol

it does nothing (Line 15-16 Algorithm 4 and line 15-16 in Algorithm 5). Every-time the *status* is computed, it is kept for α second (Line 17-23 in Algorithm 4). After α seconds, the *status* is changed back to $IDLE$ automatically (Line 29-30 in Algorithm 4).

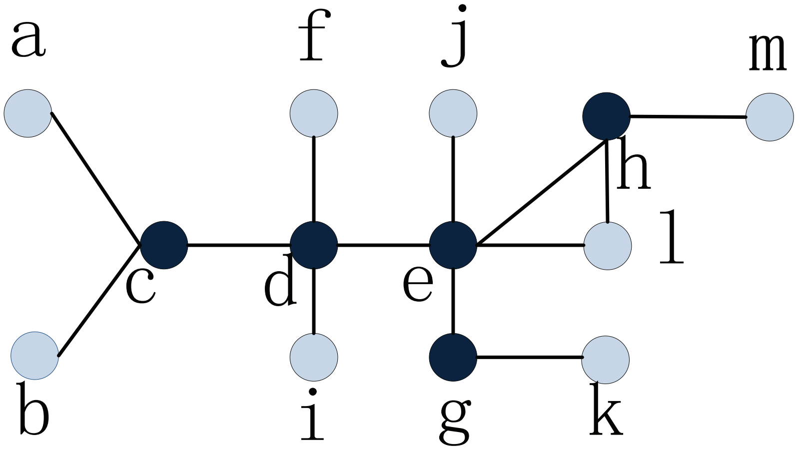

Figure 6.2: An Example of Unnecessary Retransmission

In Chapter 4, we analysed two error resilient schemes: Interleaving and Network Coding. Interleaving has the ability to increase the disseminated video quality without influencing packet loss ratio, delay and overhead. On the other hand, Network Coding impacts the performance in terms of delivery ratio and delay. What's more, Interleaving over CDS is the best of the combinations and is the only one that can provide good to excellent videos (Chapter 5). We, therefore, select IL-CDS for QGVD in this part as the routing and error resilient schemes. The sender re-sequences the frames based on specified Interleaving distance before broadcasting. The receivers return these frames back to the original sequence (Line 5 in Algorithm 4).

CDS could cover the topology and guarantee high data delivery. However, some message loss may still happen due to the fact that VANETs are naturally intermittently connected networks and nodes may fail to receive some of the frames from the initial dissemination process. Interleaving is able to address the problem of multiple consecutive packet losses and provide high quality video content. But IL becomes less effective when single loss occurs in video streaming. With this in mind, we design a store-carry-broadcast scheme in this part to help to retransmit the single lost frames due to collisions and intermittent disconnections.

In many store-carry-broadcast scheme, help packets are retransmitted immediately after the neighbour nodes decide to broadcast redundant packets [35]. However, as shown in Figure 6.2, *node e* receives *packet* 1 from *node d* and receives *hello message* from *node l*. The *hello message* indicates that *node l* has not received *packet* 1. *Node e* then retransmits *packet* 1 to help its neighbour *node l*. However, *node l* may receive *packet* 1 from *node h* later. Therefore, the retransmission of *node e* is unnecessary. As shown in Algorithm 5, in our scheme, when a vehicle receives a frame for the first time, it schedules a timer to store the message in a local buffer for further retransmission. To avoid unnecessary retransmissions, the timer is handled after a long enough duration: $\beta \times$ *data period* seconds (line 11-12 in Algorithm 4). *data period* is calculated using Equation 6.1. If nodes keep received packets forever for the retransmission schemes, we could provide a higher delivery ratio. However, endless numbers of retransmission also cause unacceptable overhead and broadcast storms. To avoid these problems, the packets are valid in the local buffer in $\gamma \times$ *data period* seconds. After this period, the packets are deleted from the buffer (Line 26 to 28 in Algorithm 4).

$$data\ period = 1000(byte) * 8(bits/byte) * \frac{1}{data\ rate(kbps)} \tag{6.1}$$

In QGVD, hello message is also used as in CDS to verify the neighbour list and to compute connected-domination set. We add the received packets information in hello message to notify neighbours concerning all the messages it has received so far. Therefore, when *node a* receives hello message from its neighbour, it compares its local packet-received list to the received information. Once it detects a packet that it has but its neighbour does not, *node a* triggers its packet retransmission scheme (Line 6-10 in Algorithm 4).

In the packet retransmission scheme, *node a* first calculates the distance from its neighbour in need (Line 5 in Algorithm 5). To avoid redundant retransmissions, only if a is $\frac{1}{3}R$ to $\frac{2}{3}R$ far from its neighbour will it retransmit the packet (Line 6 in Algorithm 5) and the needed packet is saved in the local buffer (Line 7 in Algorithm 5). We select nodes $\frac{1}{3}R$ to $\frac{2}{3}R$ because the nearer nodes are in the same situation with the help-needing node, so they also have a high probability that missed the packet. For the nodes in $\frac{2}{3}R$ to R, they may be too far to make sure the help message can be received successfully. If there is scheduled a retransmitting timer for this packet, *node a* reschedule the timer by cutting off half of remained time. On the other hand, if there exists such a timer, *node a* then schedules a retransmission and triggers it after the distance D divides $\frac{1}{2}R \times$ *data period* seconds (Line 8-11 in Algorithm 5). Therefore, the nearest node in the range of $\frac{1}{3}$R to $\frac{2}{3}R$

Algorithm 4: Pseudo-code of QGVD

```
Initialize:
$N$: the one-hop neighbour set of this node;
$R$: the communication range;
status: $IDEL$;
Before dissemination: Coding the video file with Interleaving. After
dissemination: Arrange the video packets into right sequence.
Event: RECEIVE HELLO MESSAGE
update its neighbour list $N$;
for each message m in the list of received messages do
    if m is not acknowledged in hello message then
        use Algorithm 5 to retransmit packet m;
Event: RECEIVE NEW PACKET
Schedule SAVE PACKET timer which expires after $\beta \times$ data period seconds;
if status = $IDEL$ then
    CHECK STATUS;
if status = $IN\_CDS$ then
    forward packet;
Event: CHECK STATUS
compute CDS use Algorithm 1 to Algorithm 3;
if in CDS then
    status = $IN\_CDS$;
else
    status = $NOT\_CDS$;
Schedule INITIALIZE STATUS timer which expires after $\alpha$ seconds;
Event: SAVE PACKET
copy the packet to the saving list;
Schedule DELETE PACKET timer which expires after $\gamma \times$ data period second;
Event: DELETE PACKET
delete the packet from the saving list;
Event: INITIALIZE STATUS
status = $IDEL$;
```

Algorithm 5: Pseudo-code of Store-Carry-Forward Scheme

```
Event: RECEIVE DUPLICATE PACKET
if packet m has a retransmitting timer then
    cancel the retransmitting timer;
Event: RETRANSMIT PACKET m
D = distance between the neighbour and this node;
if $\frac{2}{3}R > D > \frac{1}{3}R$ then
    if packet m is in the saved list then
        if packet m has a retransmitted timer then
            reschedule retransmit timer for m to expire after $reminder\ time/2$ seconds;
        else
            schedule retransmit timer for m which expires after $data\ period \times \frac{D}{frac12R}$ seconds;
Event: RETRANSMIT TIMER EXPIRES
if $status = IDEL$ then
    CHECK STATUS;
if $status = IN\_CDS$ then
    forward packet;
```

have the lowest waiting times to rebroadcast. Before the retransmitting timer expires, if the same packet is received, the timer (Line 2-3 in Algorithm 5). Therefore, when the nearest neighbour finally forwards the message, it suppresses the retransmissions of other involved nodes to avoid of redundant retransmissions.

6.2 Simulation Setting

The performance evaluation of QGVD follows the configuration is also the same as the one used in Section 3. The scenarios, evaluation method and statistical methodology are the same as well. Table 6.1 shows the QGVD parameters used in this part. We set α as 0.5 because the hello messages are sent with a frequency of $2Hz$. β and γ are set 20 which is a large enough number to avoid unnecessary retransmissions.

Parameter	Value	Meaning
α	0.5	the statuses of node is kept for 0.5 seconds
β	20	the new received packet start to be valid for retransmission when it is received for 20 × *data period* seconds
γ	20	the saved packet can be retransmitted for 20 × *data period* seconds

Table 6.1: Parameters of QGVD

6.3 Performance Evaluation

One significant observation is that IL-CDS and QGVD have similar frame losses at 100 kbps as IL-CDS is good enough for a low data rate. QGVD repair technique starts to improve the quality of the video in the higher data rate (500kbps) and achieves about 13 percent improvements at the 1600kbps data rate. The explanation for this is that the store-carry-forward schemes employed in QGVD performs as a recovery mechanism for packet losses caused by collisions or network disconnection. With a high data rate, a whole bunch of packets are lost in IL-CDS because of the increasing of broadcast storm probability. If a node fails to receive the first disseminated packet from its neighbour *node a*, other neighbours using store-carry-forward scheme will deliver it later.

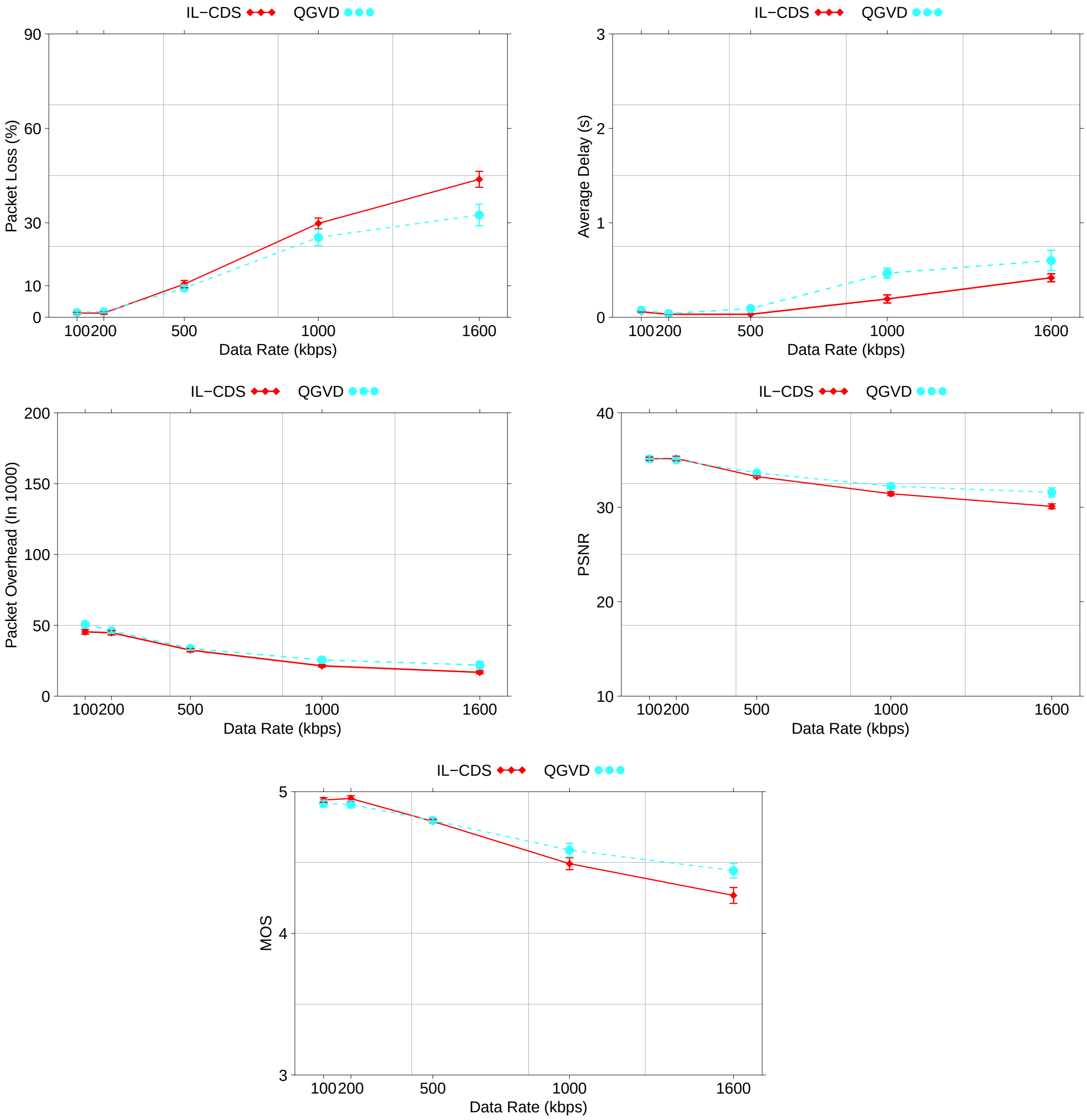

Figure 6.3: Performance Comparison between QGVD and IL-CDS

In terms of average delay, as shown in Figure 6.3, QGVD has longer delays as the data rate arrives at 500kbps or higher. The major factor in this the difference is that store-carry-forward works after packet is received for $\beta \times$ *data period* seconds. For the packets that received by the store-carry-forward scheme, their delays are added by at least $\beta \times$ *data period* seconds. Therefore, the average delays of QGVD are longer. With the increase in a data rate (5000kbps to 1600kbps), the gap between delays reaches a peak and then decreases. This is because the waiting duration for saving received packets are is based on the data rate. When the data rate increases, the waiting duration decreases. And therefore, the average delay gap between QGVD and IL-CDS decreases. The delays of QGVD are less than 1 second, and therefore, they are still in the acceptable range.

Figure 6.3 shows the overhead of QGVD and IL-CDS. We can observe that QGVD transmits slightly more packets than IL-CDS because it has the store-carry-forward retransmission scheme. This is also an indication that the scheme, which maintains the packet delivery, works without transmitting too much redundancy packets. The local buffered deleting packet scheme is efficient and timely. What's more, because the nearest node in range helps first and other nodes cancel the scheduled retransmission when they overhear the help message, our store-carry-forward scheme is efficient.

As mentioned before, if the PSNR is larger than 30 dB, the quality of video is very good. For low data rates (100kbps and 200kbps), the PSNRs of IL-CDS and QGVD are all larger than 30 dB. Their performance has no obvious difference. However, when the data rate increases, the PSNR value of IL-CDS falls much faster than that of QGVD. For a high data rate (1600kbps), the PSNR value of IL-CDS goes to 30 dB. However, QGVD, with the single fixed packet loss, has a higher value and the quality of its disseminated video is better.

Finally, Figure 6.3 also shows the comparison of MOS values between IL-CDS and QGVD. As a video quality metric, the MOS values have the same trends as the values of PSNR. Providing higher MOS values in high data rates, QGVD provides more reliable video qualities for video dissemination over Urban scenarios. Therefore, it is a more viable solution for video applications over VANETs.

Chapter 7

Conclusion

In this study, we proposed a guaranteed quality video dissemination protocol for urban VANETs. We first compared three routing protocols and discovered that only CDS meets the strict quality requirements of video dissemination over urban VANETs. We then evaluated the two error resilient approaches, and our research result indicated that Network Coding fails to provide satisfactory video quality for Urban VANETs. This work also revealed that the simple error resilient scheme Interleaving is a necessary scheme to fulfil crucial video quality requirements. In addition, this study examined the performance of error resilient approaches over two of the routing approaches. We selected the best combination, Interleaving over CDS, for our QGVD protocol. Because of the existence of intermittent disconnection and collisions in VANETs, we designed a store-carry-broadcast scheme for QGVD to fix single packet losses. Our QoS and MOS results showed that our QGVD is suitable for video dissemination over urban VANETs with both low and high data rates.

Even though this thesis provides a deep evaluation with different perspectives, there still exist some other open issues that can be pursued to evaluate the performance of existing techniques and propose video dissemination protocols. The first consideration is the evaluation metrics. As mentioned before, PSNR has been widely criticized because of its performance. Structure Similarity is a reliable method for measuring the similarity between two images. In the future, this work will extend to include it to provide better performance evaluation work and to proof that QGVD is fit for video dissemination over urban VANETs.

Bibliography

[1] R. Ahlswede, N. Cai, S.-Y.R. Li, and R.W. Yeung. Network information flow. *Information Theory, IEEE Transactions on*, 46(4):1204–1216, 2000.

[2] M. Asefi. Quality-driven cross-layer protocols for video streaming over vehicular ad-hoc networks. 2011.

[3] A. Asif, U.T. Nguyen, and G. Xu. Scalable video multicast over manets. In *Multimedia Signal Processing, 2006 IEEE 8th Workshop on*, pages 403–408. IEEE, 2006.

[4] O.S. Badarneh and M. Kadoch. Multiple description video multicast in manets. In *Performance Computing and Communications Conference (IPCCC), 2010 IEEE 29th International*, pages 222–230. IEEE, 2010.

[5] M.A. Bonuccelli, G. Giunta, F. Lonetti, and F. Martelli. Real-time video transmission in vehicular networks. In *2007 Mobile Networking for Vehicular Environments*, pages 115–120. IEEE, 2007.

[6] P. Bucciol, J.L. Zechinelli-Martini, and G. Vargas-Solar. Optimized transmission of loss tolerant information streams for real-time vehicle-to-vehicle communications. In *Computer Science (ENC), 2009 Mexican International Conference on*, pages 142–145. IEEE, 2009.

[7] M. Claypool and Y. Zhu. Using interleaving to ameliorate the effects of packet loss in a video stream. In *Proceedings of the 23rd International Conference on Distributed Computing Systems*, ICDCSW '03, pages 508–, Washington, DC, USA, 2003. IEEE Computer Society.

[8] A. Fujimura, S.Y. Oh, and M. Gerla. Network coding vs. erasure coding: Reliable multicast in ad hoc networks. In *Military Communications Conference, 2008. MILCOM 2008. IEEE*, pages 1–7. IEEE, 2008.

[9] M. Gast. *802.11 wireless networks: The definitive guide*. Southeast University Press, 2006.

[10] K. Ghafoor and K.A. Bakar. Inter-vehicle communication protocols for multimedia transmission. *International MultiConfrance of Engineering and Computer Scientists*, 2010.

[11] V.K Goyal. Multiple description coding: Compression meets the network. *Signal Processing Magazine, IEEE*, 18(5):74–93, 2001.

[12] J. Gross, J. Klaue, H. Karl, and A. Wolisz. Cross-layer optimization of ofdm transmission systems for mpeg-4 video streaming. *Computer Communications*, 27:1044–1055, 2004.

[13] M. Guo, M.H. Ammar, and E.W. Zegura. V3: A vehicle-to-vehicle live video streaming architecture. *Pervasive and Mobile Computing*, 1(4):404–424, 2005.

[14] M. Haklay and P. Weber. Openstreetmap: User-generated street maps. *Pervasive Computing, IEEE*, 7(4):12–18, 2008.

[15] T. Ho, M. Medard, R. Koetter, D.R. Karger, M. Effros, J. Shi, and B. Leong. A random linear network coding approach to multicast. *Information Theory, IEEE Transactions on*, 52(10):4413–4430, 2006.

[16] C.-W. Hsu, C.-H. Hsu, and H.-R. Tseng. Mac channel congestion control mechanism in ieee 802.11 p/wave vehicle networks. In *Vehicular Technology Conference (VTC Fall), 2011 IEEE*, pages 1–5. IEEE, 2011.

[17] R. Jain. Quality of experience. *IEEE MultiMedia*, 11(1):96–95, January 2004.

[18] E. Kayacan, B. Ulutas, and O. Kaynak. Grey system theory-based models in time series prediction. *Expert Systems with Applications*, 37(2):1784–1789, 2010.

[19] M. Kazemi, S. Shirmohammadi, and K.H. Sadeghi. A review of multiple description coding techniques for error-resilient video delivery. *Multimedia Systems*, pages 1–27.

[20] C.-H. Ke. myevalsvc-an integrated simulation framework for evaluation of h. 264/svc transmission. *KSII Transactions on Internet and Information Systems (TIIS)*, 6(1):377–392, 2012.

[21] J. Klaue, B. Rathke, and A. Wolisz. Evalvid - a framework for video transmission and quality evaluation. In *In Proc. of the 13th International Conference on Modelling Techniques and Tools for Computer Performance Evaluation*, pages 255–272, 2003.

[22] D. Krajzewicz, J. Erdmann, M. Behrisch, and L. Bieker. Recent development and applications of SUMO - Simulation of Urban MObility. *International Journal On Advances in Systems and Measurements*, 5(3&4):128–138, December 2012.

[23] T.A. Le, H. Nguyen, and H. Zhang. Evalsvcan evaluation platform for scalable video coding transmission. In *Consumer Electronics (ISCE), 2010 IEEE 14th International Symposium on*, pages 1–6. IEEE, 2010.

[24] A. Li. Rtp payload format for generic forward error correction. 2007.

[25] Y. Li, F. Naeimipoor, and A. Boukerche. Video dissemination protocols in urban vehicular ad hoc network: A performance evaluation study. In *IEEE WCNC'14 Track 3 (Mobile and Wireless Networks) (IEEE WCNC'14 Track 3 : NET)*, Istanbul, Turkey, April 2014.

[26] M. Lindeberg, S. Kristiansen, T. Plagemann, and V. Goebel. Challenges and techniques for video streaming over mobile ad hoc networks. *Multimedia Systems*, 17(1):51–82, 2011.

[27] G. Maia, A. Boukerche, A.L. Aquino, A.C. Viana, and A.A.F. Loureiro. A data dissemination protocol for urban vehicular ad hoc networks with extreme traffic conditions. In *Communications (ICC), 2013 IEEE International Conference on*, pages 5997–6001. IEEE, 2013.

[28] G. Maia, C. Rezende, L.A. Villas, A. Boukerche, A.C. Viana, L.A. Andre, and A.L. Antonio. Traffic aware video dissemination over vehicular ad hoc networks. In *16th ACM MSWiM 2013*, pages 419–426.

[29] S. Mao, S. Lin, S.S. Panwar, Y. Wang, and E. Celebi. Video transport over ad hoc networks: Multistream coding with multipath transport. *Selected Areas in Communications, IEEE Journal on*, 21(10):1721–1737, 2003.

[30] S. Mccanne, S. Floyd, and K. Fall. ns2 (network simulator 2). http://www-nrg.ee.lbl.gov/ns/.

[31] F. Naeimipoor. *Video Streaming and Multimedia Broadcasting Over Vehicular Ad Hoc Networks*. University of Ottawa, 2001.

[32] F. Naeimipoor, C. Rezende, and A. Boukerche. Performance evaluation of video dissemination protocols over vehicular networks. In *Local Computer Networks Workshops (LCN Workshops), 2012 IEEE 37th Conference on*, pages 694–701, 2012.

[33] S.-Y. Ni, Y.-C. Tseng, Y.-S. Chen, and J.-P. Sheu. The broadcast storm problem in a mobile ad hoc network. In *Proceedings of the 5th annual ACM/IEEE international conference on Mobile computing and networking*, MobiCom '99, pages 151–162, New York, NY, USA, 1999. ACM.

[34] J.-S. PARK, M. GERLA, D.S. Lun, Y. Yi, and M. Medard. Codecast: a network-coding-based ad hoc multicast protocol. *Wireless Communications, IEEE*, 13(5):76–81, 2006.

[35] J.S. Park, U. Lee, S.Y. Oh, M. GERLA, and D.S. LUN. Emergency related video streaming in vanet using network coding. In *Proceedings of the 3rd international workshop on Vehicular ad hoc networks*, pages 102–103. ACM, 2006.

[36] J.S. Park, U. Lee, S.Y. Oh, M. GERLA, D.S. LUN, W.W. RO, and J. Park. Delay analysis of car-to-car reliable data delivery strategies based on data mulling with network coding. *IEICE transactions on information and systems*, 91(10):2524–2527, 2008.

[37] M. Pasin, M. Petracca, P. Bucciol, A. Servetti, and J.C. De Martin. Error resilient real-time multimedia streaming over vehicular networks. *Vehicle Systems and Safety, Dallas, TX, USA*, 2009.

[38] N.N. Qadri, M. Fleury, M. Altaf, B.R. Rofoee, and M. Ghanbari. Resilient p2p multimedia exchange in a vanet. In *Wireless Days (WD), 2009 2nd IFIP*, pages 1–6. IEEE, 2009.

[39] N.N. Qadri, M. Fleury, M. Altaf, B.R. Rofoee, and M. Ghanbari. Multi-source video streaming in a wireless vehicular ad hoc network. *Communications, IET*, 4(11):1300–1311, 2010.

[40] R Development Core Team. *R: A language and environment for statistical computing*. R Foundation for Statistical Computing, Vienna, Austria, 2009. ISBN 3-900051-07-0.

[41] A. Razzaq and A. Mehaoua. Video transport over vanets: Multi-stream coding with multi-path and network coding. In *Proceedings of the 2010 IEEE 35th Conference on Local Computer Networks*, LCN '10, pages 32–39, Washington, DC, USA, 2010. IEEE Computer Society.

[42] C. Rezende, R. W. Pazzi, and A. Boukerche. A reactive solution with a redundancy-based error correction mechanism for video dissemination over vehicular ad hoc networks. In *the 15th ACM international conference on Modeling, analysis and simulation of wireless and mobile systems*, MSWiM, pages 343–352. ACM, 2012.

[43] C. Rezende, R.W. Pazzi, A. Boukerche, and S. Nikoletseas. The effect of redundancy on video broadcasting in vehicular networks. In *Communications (ICC), 2011 IEEE International Conference on*, pages 1–5. IEEE, 2011.

[44] C. Rezende, H.S. Ramos, R.W. Pazzi, A. Boukerche, A.C. Frery, and A.A.F. Loureiro. Virtus: A resilient location-aware video unicast scheme for vehicular networks. In *Communications (ICC), 2012 IEEE International Conference on*, pages 698–702. IEEE, 2012.

[45] H. Schwarz, D. Marpe, and T. Wiegand. Overview of the scalable video coding extension of the h. 264/avc standard. *Circuits and Systems for Video Technology, IEEE Transactions on*, 17(9):1103–1120, 2007.

[46] F. Soldo, C. Casetti, C. Chiasserini, and P.A. Chaparro. Video streaming distribution in vanets. *Parallel and Distributed Systems, IEEE Transactions on*, 22(7):1085–1091, 2011.

[47] T. Szigeti and C. Hattingh. *End-to-End QoS Network Design: Quality of Service in LANs, WANs, and VPNs (Networking Technology).* Cisco Press, 2004.

[48] R. Uzcategui and G. Acosta-Marum. Wave: a tutorial. *Communications Magazine, IEEE*, 47(5):126–133, 2009.

[49] D. Vatolin, A. Parshin, O. Petrov, and A. Titarenko. Msu subjective comparison of modern video codecs. *CS MSU Graphics & Media Lab Video Group (January 2006)*, 2006.

[50] R. Wang, C. Rezende, S.R. Heitor, R.W. Pazzi, A. Boukerche, and A.A.F. Loureiro. Liaithon: A location-aware multipath video streaming scheme for urban vehicu-

lar networks. *2012 IEEE Symposium on Computers and Communications (ISCC)*, 0:000436–000441, 2012.

[51] Y. Wang, S. Jain, M. Martonosi, and K. Fall. Erasure-coding based routing for opportunistic networks. In *Proceedings of the 2005 ACM SIGCOMM workshop on Delay-tolerant networking*, pages 229–236. ACM, 2005.

[52] Y. Wang, Y. Zhang, and J. Ostermann. *Video Processing and Communications*. Prentice Hall PTR, Upper Saddle River, NJ, USA, 1st edition, 2001.

[53] H. Weatherspoon and J.D. Kubiatowicz. Erasure coding vs. replication: A quantitative comparison. In *Peer-to-Peer Systems*, pages 328–337. Springer, 2002.

[54] J. Wu and H. Li. A dominating-set-based routing scheme in ad hoc wireless networks. *Telecommunication Systems*, 18(1-3):13–36, 2001.

[55] F. Xie, K.A. Hua, W Wang, and Y.H. Ho. Performance study of live video streaming over highway vehicular ad hoc networks. In *VTC-2007 Fall*, pages 2121–2125. IEEE, 2007.

[56] Y. Zhu, M. Claypool, and Y. Liu. Using redundancy and interleaving to ameliorate the effects of packet loss in a video stream.

[57] Z.J.Haas, J.Y. Halpern, and L. Li. Gossip-based ad hoc routing. *IEEE/ACM Trans. Netw.*, 14(3):479–491, June 2006.