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Improving QoS of VANET using Network Coding

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ABSTRACT

Vehicular Ad hoc Networks -VANET as a sub class of Mobile Ad hoc Networks -MANET provides a wireless communication among vehicles and vehicle to road side equipment. VANET allows vehicles to form a self-organized network without the need for permanent infrastructure. With high number of nodes and mobility, ensuring the Quality of Service- QoS in VANET is a challenging task. QoS is essential to improve the communication efficiency in vehicular networks. Thus a study of QoS in VANET is useful as a fundamental for constructing an effective vehicular network. In this paper, we propose Network coding Technique to improve Bandwidth utilization on VANET. When two sources are involved in broadcasting in the same area and at same time , the relay will make use of Network coding to reduce the Bandwidth consumption. While receiving the packet, a relay has to decide whether to send the packet directly to reduce the delay or use Network coding for effective Bandwidth utilization. In order to make trade off, we introduce two kind of protocols named Buffer Size Control Scheme -BSCS and Time Control Scheme -TCS. By this two protocols, we aim to reduce the delay that is experienced by each packet and achieving better bandwidth utilization.

***Keywords*--** VANET, Network Coding, BSCS, TCS, QoS

I. INTRODUCTION

Network Coding allows the nodes to combine the packets and make linear combination of the data before forwarding it to the next node instead of simply storing and forwarding. In Network Coding, the routers and switches have the capability of encoding of the incoming packets and transmitting them to the neighbor nodes. If N packets in the source node are encoded together, one innovative packet is formed. The receiver needs a minimum of N such innovative coded packets to be able to decode all the N packets sent by the source node successfully.

NC refers to the concept of coding multiple packets into a single coded packet to optimise throughput. This technique enables us to transmit multiple packets in a single transmission; hence transmitting more data for the same number of transmissions or requiring less transmissions to send the same amount of data, when compared to traditional routing mechanisms.NC increases network throughput to the maximum throughput as specified by the min-cut max-flow theorem[2].

The theorem states that the maximum throughput achievable in a network is equal to the minimum removed links that will disable any data flow between the source and destination nodes. The theorem is depicted in figure 1: In this scenario, source node S wants to send packets a and b to destination nodes X and Y . The minimum cut between source node S and destination nodes X and Y that will disable communication, is two. Therefore the maximum throughput obtainable in this network is two. After a full round of transmissions, three packets reached the destination nodes. Two packets were sent. This translates to a suboptimal throughput of 1.5 (3/2).

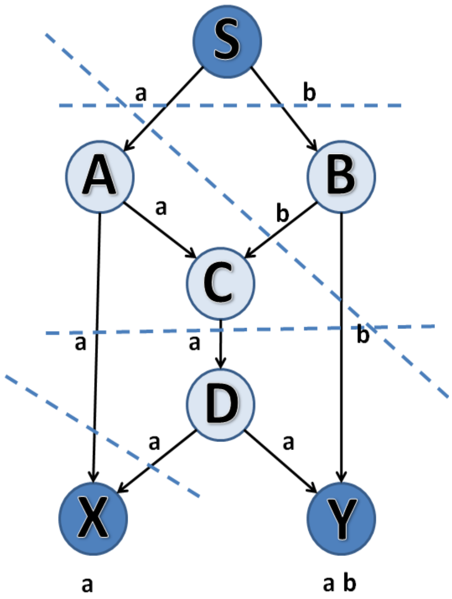

Figure.2 Min – cut Max- flow Theorem

When NC is implemented, the throughput increases to two, as depicted in figure 2.

In this scenario, packets a and b are coded at node C. The coded packet is sent in a single transmission to node D, and delivered to the destination nodes X and Y in the next transmission. The destination nodes then use the packets they have overheard from previous transmissions in order to decode the coded packet. In this scenario, four packets have reached the destination nodes, and it translates to two (4/2), which is the maximum obtainable throughput of the network. The following example explains the concept in more detail.

A single coded packet is created by means of XORing individual packets with each other. A coded packet is decoded by being XORed with a sufficient number of different individual packets it's comprised of. The set of single packets used to decode the coded packet will determine the retrieved packet obtained from the coded packet. Please refer to figure 3. Packet (1) and (2) respectively represents packets a and b from the previous example, depicted in figure 2. The coded packet comprised of the two packets is represented by packet (3). By XORing either of packets (1) or (2) with the coded packet (3) will decode the coded packet and deliver the other packet as illustrated in (4) and (5).

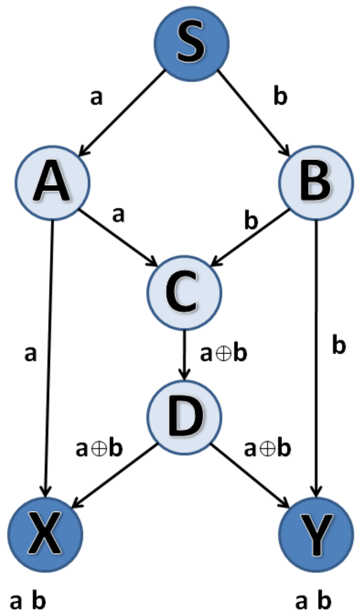

Figure.3 Network with Network Coding

The pipe line encoding aims to reduce the coding delay as well as to further improve the throughput [3]. Normally, packets are not sent from an application all at once, in pipeline coding, we relax the limitation of waiting for all data packets of a generation to be received from the Application. The coding module is implemented at the network layer and it does encoding, decoding, and broadcasting.

When the source receives a data packet from the transport layer, it first stores the data in the generation buffer. Thereafter, based on the coding redundancy, a number of coded packets are generated and sent out. For each generated coded packet, the source first randomly generates the coding coefficients. Secondly, it checks the generation buffer. For a missing packet, which has not arrived yet, the corresponding coefficient is set to zero. Finally, the source encodes the data packets based on this encoding vector, and sends it to the lower layer.

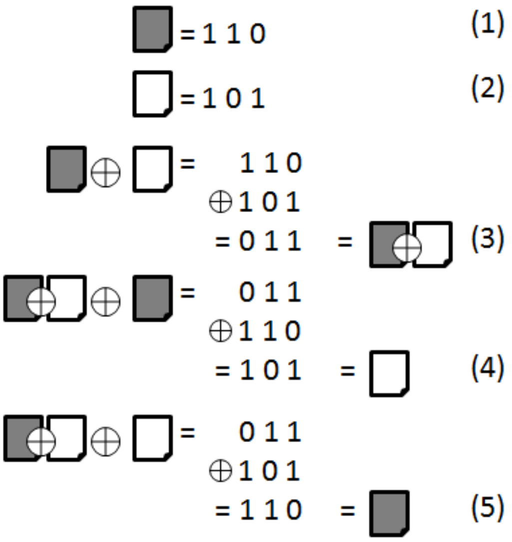

Figure.4Network Coding in detail

A destination first examines whether the received coded packet is innovative or not. An innovative packet will be stored in the generation buffer. Afterward, the decoding module invokes Gaussian Elimination routines and attempts to decode data packets. After decoding, newly decoded data packets will be stored into another decoding buffer and then delivered to the upper layer. In particular, the pipeline coding is used to protect the data from degradation beyond packet loss.

II. BACK GROUND

To illustrate how our proposal will work in such scenario, let us consider the following example. Assume we have two sources, source1 which is a restaurant advertising its services including the new offers and the menu, while the other source, source2, is a gas station advertising the opening hours and the prices. The two sources source1 and source2 are located N hop's distance from each other on the same road, and each source is willing to disseminate its data in a certain area which is the shared distance between them. Thus, the packets will need to be relayed at intermediate nodes to cover all the area (see Fig. 5).

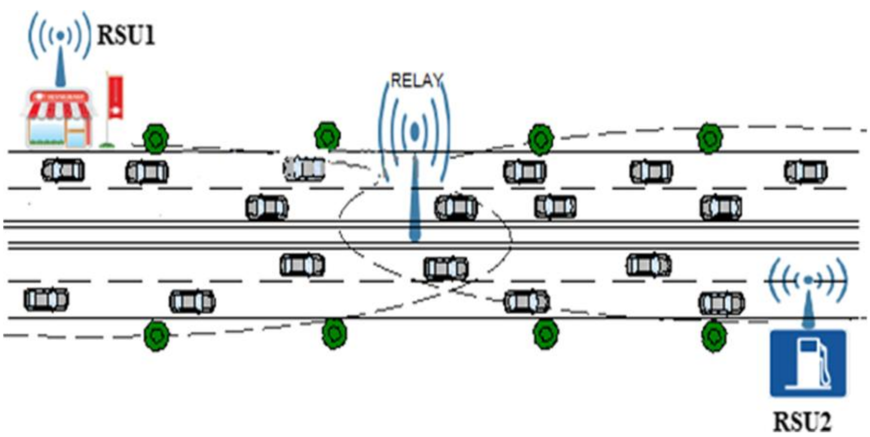

Figure. 5 Network Scenario

In order to disseminate one packet from source1,the packet will be relayed at intermediate nodes; thus, the packet will be broadcasted from source1 and then relayed which make two broadcasting events; the same will happen for a packet to be disseminated by source2. Thus, disseminating one packet from source1 and one packet from source2 will cost four broadcasting events. Now, let us consider the same scenario using the network coding technique; in this case, after source1 and source2 broadcast their packets, the relay will encode the two packets into one packet and relay it. So, the network coding technique will cost three broadcasting events and traditional broadcasting takes four.

Since the vehicles that are located one hop away from source1 already have a copy of the packet disseminated by source1, and the vehicles that are located one hop away from source2 have a copy of the packet that is disseminated by source2, both groups of vehicles will be able to retrieve and decode the received encoded packet.

When we find how much of the bandwidth can be saved in case of the network coding, let us say we have A packets to be disseminated from source1 and B packets to be disseminated from source2. In the conventional broadcasting case, the relay will need to relay A+B packets; if we assume that each packet will consume $\frac{1}{T}$ of the bandwidth, a total of $\frac{A+B}{T}$ of the bandwidth will be consumed. Assuming the network coding, each packet from source1 will be combined with a packet from source2; therefore, only a number of Max (A or B) packets will be rebroadcasted. This will consume $\frac{\text{Max (A or B)}}{T}$ of the bandwidth. Therefore, the improvement in the overall throughput will be $\frac{\text{Max (A or B)}}{A+B}$. However, if there is the same number of packets from both directions, then up to 50 % of the consumed bandwidth can be saved when we use the network coding technique. The intermediate nodes that act as relays could be either stationary nodes, where they could be RSUs attached to the light poles on the road, or they could be mobile nodes, and in this case, they might be any vehicle in the radio range of the source.

2.1 Buffer Size Control Scheme - BSCS

This scheme aims to control the imposed delay by controlling the size of the buffer. If the utilization of the queue increases, the average delay per packet will also increase. So in this scheme, the relay will decide to queue the packet based on the number of the packets in the queue when the packet arrived. In other words, the probability of p queuing a packet is proportional to the size of that queue when the packet arrived.

$$P = \frac{1}{\text{size of Queue}} \qquad (1)$$

However, doing so might cause a newly arrived packet to be relayed while the older packets are still waiting for coding opportunity in the queue. This will cause the packets to be disseminated out of order and might cause the queued packets to be disseminated too late for the application to consider them. So, instead, the relay will actually queue the new packet with probability p calculated as above, while queue the packet and release the head of the queue with probability 1 − p. Figure.6 explains the steps the relay will follow when receiving a new packet.

Assuming the data rate for both source1 and source2 in Fig.5 are Poisson distribution and equal to λ1 and λ2, respectively, and the queue is a M/M/1 model. Then the arrival rate at the queue λq is

$$\lambda q = \lambda 2\big(1 - P(0)\big) + \lambda 2 \text{ X } P \text{ X } P(0) \qquad (2)$$

Where P(0) is the probability of the queue to be empty. However, since every packet arrived from source1 will cause a packet from the queue to be released, the departure rate μq of the queue is

$$\mu q = \lambda 1 + \lambda 2(1-P(0))(1-p) \qquad (3)$$

The average queuing delay Dq can be expressed as follows:

$$D_q = \frac{1}{\mu q - \lambda q} \qquad (4)$$

However, the average delay per packet from source2 (e.g., the slower source)

$$D_{dir} = \frac{\lambda q}{\lambda 2} \text{ x } D_q \qquad (5)$$

Where λq λ2 represents the probability of a packet to be queued.

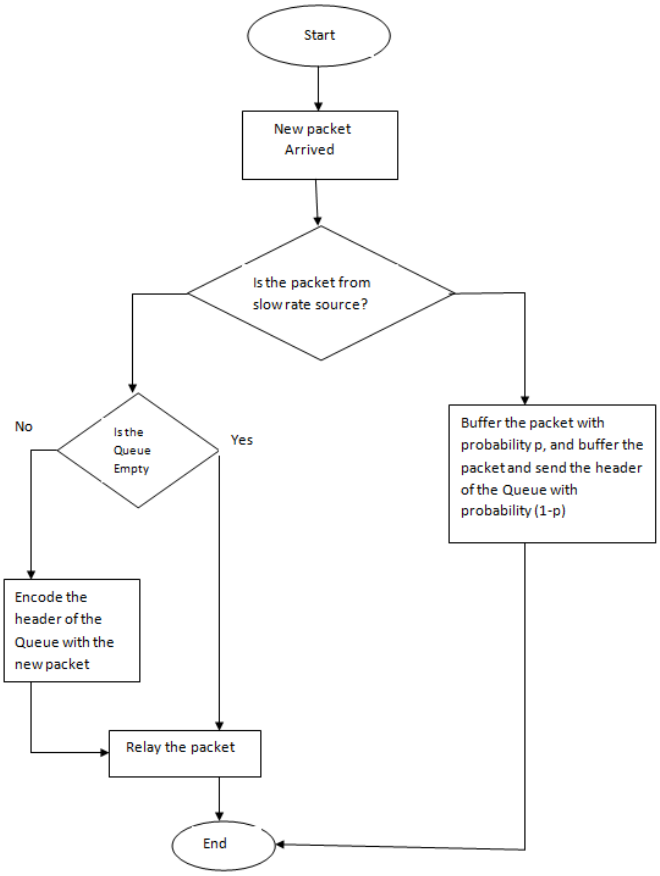

Figure.6 BSCS Flow Chart

2.2 Time-Limited Scheme –TLS

In this method, the relay will put all the packets that come from the slow source; however, the relay sets a delay limit to each packet. It means that the packet will not going to be buffered more than a certain time Tmax. As soon as, a relay receives a packet from the slow source, it will be queued directly, but sets a timer and after Tmax, if the packet is still in the queue, then it should be sent immediately without encoding. Figure 7 explains the procedure executed by the relay in TLS.

2.3 TCS Delay Analysis

Considering the same assumption of the previous scheme, as the relay will queue all the packets from the slow source, the arrival rate of the queue will be λ2, while the departure rate remains only λ1. However, the packet will not stay longer than Tmax in the queue, by considering this, the average queue delay can be expressed as follows:

$$D_q = \{Tmax, \frac{1}{\lambda1-\lambda2}, \lambda1- \lambda2 <= \frac{1}{Tmax} \qquad (6)$$

In this case, all the packets from source2 will be immediately buffered in the queue, while the packets from sources will be sent directly, either encoded if the queue is non-empty or as native packet if the queue is empty.

III. PURE NETWORK CODING SCHEME

If P = 1, then equation 2,3 & 4 becomes

$$\lambda q = \lambda 2$$
$$\mu q = \lambda 1$$
$$D_q = \frac{1}{\lambda1-\lambda2} \qquad (7)$$

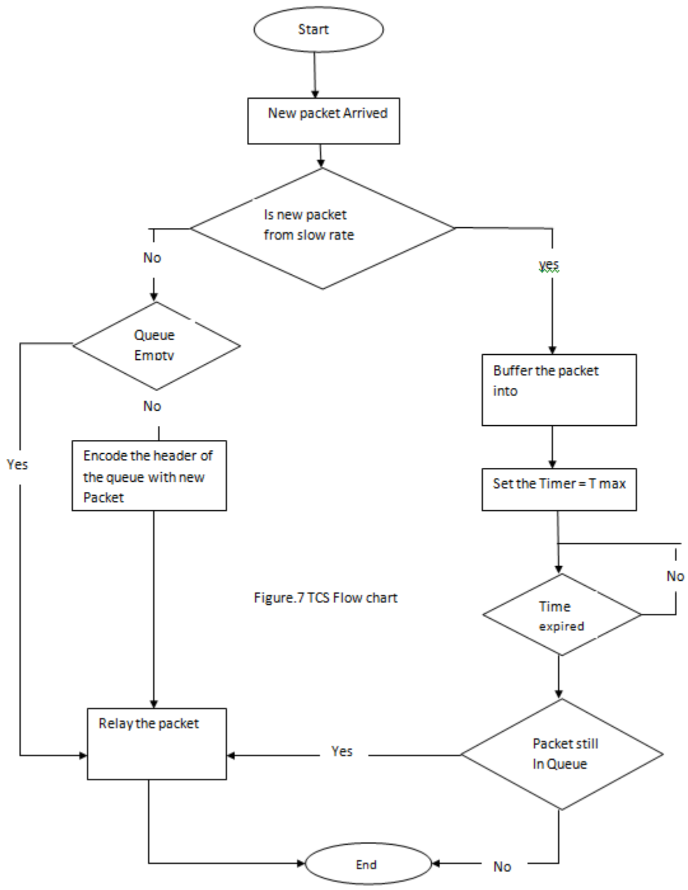

Figure.7 TCS Flow chart

3.1 Stationary Relay Scenario

Our first scenario is when all the relays over the area of interest are stationary RSUs that are installed on the light poles. In this case, the RSUs are placed with a radio range distance between them.

3.2 Mobile Relay Scenario

Here, we consider the vehicles which have to cooperate with each other in order to disseminate the data between the sources. However, the main challenge in our protocol will be how to select the vehicle that should act as a relay.

In order to increase the coding opportunities, we have the same node to work as relay during all the periods of the transmission. However, certain classes of clustering and/or broadcasting algorithms might be suitable to implement our protocol [4 - 6]. In those classes, the header of the cluster will always be responsible for all the rebroadcasting operations.

We adopt the relay selection method that is described in [7]. In this, the road area is divided into virtual segments or grids, with the role of the relay is assigned to one vehicle in each segment, and when the current relay leaves the segment, it will hand over all its buffered data to a new relay. In other words, the same relay will continue to relay the packets until it leaves the segment and hand its data to the new relay.

When the width of the segment will be the width of the two directions of the road, that is, the two directions of the road will have one relay. Based on the vehicles velocity and their location from the end of the segment, the vehicle that will spend more time in the segment will be selected as the relay for the corresponding segment.

IV. SIMULATION SETUP AND NUMERICAL RESULTS

To simulate a scenario such as the one that is described in Figure.5 a network simulator NS3 is Experimented using IEEE 802.11p Table 1 shows the simulation parameter setting for our protocol. We set the two sources to randomly generate the packets, and we use the Poisson distribution to calculate the interval between the generated packets. We set $\lambda 2$ to be 1 packet/sec; however, $\lambda 1$ is varied throughout the simulation. The speeds of the vehicles were set to be between 36 and 54 km/h. Initially, 250 vehicles were distributed into a road area of 4000 m in both road directions.

The simulation runs for 20 s before the sources started sending their data. Tmax is set to be 0.3 for the TCS, while the segment size is 250 m in the case of mobile relay scheme However, for the stationary relay scenario, three relays were placed with a distance of 500 m between them.

Simulation parameter	value
Radio Range	600 m
MAC Protocol	802.11p(WAVE)
Area of interest	2000 m
Packet size	1000 bytes
Physical layer setting	Modulation type Data Rate Channel BW
	OFDM 6Mbps 10Mhz

Table.1 The Simulation Parameters

Our Objective is to to achieve better bandwidth utilization while controlling the imposed delay. We consider the following performance metrics to proof performance improvements through our protocol.

4.1 The hop delay

This is the average time the packet from the buffered traffic might spend at each hop. Figure 8 shows the average hop delay for the packets that is sent by source2Figure 8a represents the stationary relay scenario. In the symmetric flows case (e.g., $\lambda1/ \lambda2 = 1$), pure network coding will lead to unbounded delay while both BSCS and TCS schemes shows considerable amount of control on the imposed delay. On the other hand, for both pure network coding and BSCS scheme, as the ratio between the two traffic (e.g., $\lambda1/ \lambda2$) increases, the experienced delay decreases, e.g., when $\lambda1/ \lambda2 = 1.5$, the delay was 1 and 2s for BSCS and pure NC, respectively, while when $\lambda1/\lambda2 = 2.5$, the delay was 0.75 s for both schemes.

That is reasonable as the fact that $\lambda1$ is assumed to be the departure rate of the queue, and as the departure rate of the queue increases, the average delay in the queue decreases. It is worth to mention here that BSCS outperforms the pure network coding mainly when the two traffics are symmetric, or the ratio between them is low. For the case of the mobile relay, the dynamic nature of the relay has affected the stability of the network, where the network behaviour was difficult to predict; however for the BSCS and TCS schemes, the performance was better than the pure network coding scenario.

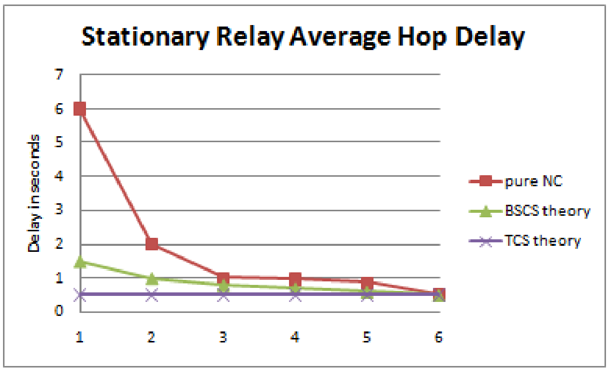

Figure.8.(a) Average hop Delay

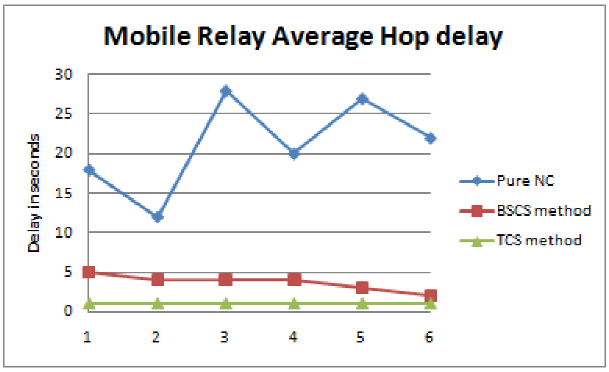

Figure.8(b) Average hop Delay

4.2 The Throughput Improvement

In Fig. 9 shows the improvement in the network throughput. Although, pure network coding achieves up to almost 50 % in the throughput improvement when ($\lambda1 = \lambda2$), yet it must be considered that, this performance costs the network unbounded delay Thirty-eight percent improvement in the bandwidth usage has been recorded for the BSCS scheme when $\lambda1 = \lambda2$ when the relay is stationary. This is to be considered a good performance, since the BSCS scheme decreases the imposed delay. However, as the ratio between λ1 and λ2 increases, the performance of both pure network coding and BSCS become more similar.

The increment in the ratio between the λ1 and λ2 causes a decrement in the improvement of the throughput. In the TCS scheme, the improvement in the throughput remains constant despite the ratio between λ1 and λ2, which refers to the fixed coding opportunity the protocol gives to the packet despite the rate of the data.

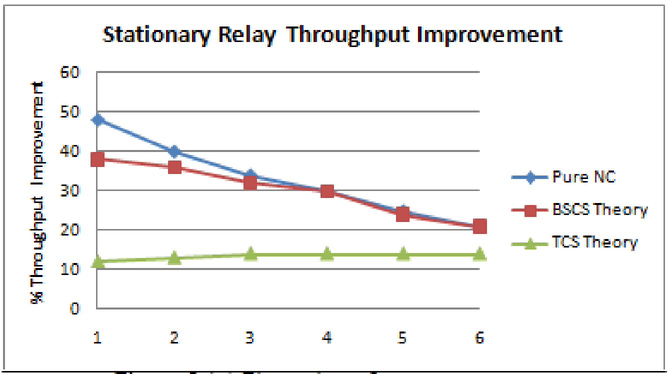

Figure.9 (a) Throughput Improvement

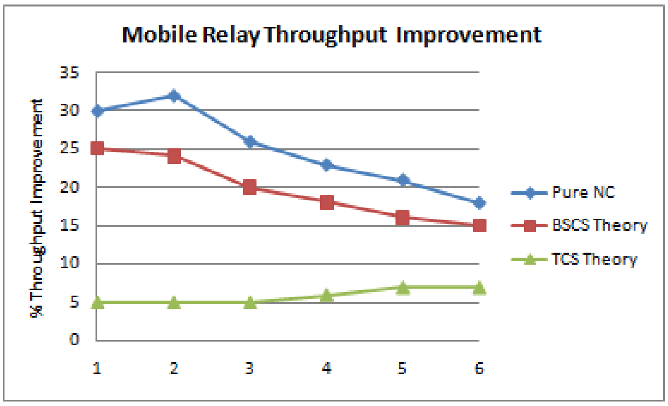

Figure.9 (b) Throughput Improvement

V. CONCLUSIONS

In this paper, we consider VANET non-safety applications, where two sources are disseminating the data in a common area of interest. To reduce the network congestion, we are using the same relay to rebroadcast the packets from both resources. And for better utilization of the bandwidth, we propose the network coding technique to be used by the relay when broadcasting the packets. By this method, we record up to 38 % improvement in the throughput. In order to control the imposed delay, two versions of our protocols have been described. We have tested both BSCS and TCS protocols under both stationary and mobile relay scenarios and showed a noticeable decrease in the imposed delay with respect to the pure network coding scenario. We are upgrading our mobile relay selection methodology to improve the performance of the protocols in the fully dynamic environment.

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