Multi-hop Broadcasting in VANETs Integrating Intra-flow and Inter-flow Network Coding

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***Abstract*—Multi-hop data dissemination in vehicular ad hoc networks (VANETs) is very important for the realization of collision avoidance systems and other many interesting applications. However, designing an efficient data dissemination protocol in VANETs has been a challenging issue due to vehicle movements, limited wireless resources and lossy characteristics of wireless communication. In this paper, we propose a protocol which employs intra-flow and inter-flow network coding to reduce the protocol overhead as compared to traditional protocols. The protocol also can improve the packet reception probability at the receiver nodes by using the network coding. Therefore, the protocol can provide a lightweight and reliable solution for data dissemination in VANETs. We use theoretical analysis and computer simulations to show the advantage of the proposed protocol over other existing alternatives.**

I. Introduction

Vehicular ad hoc networks (VANETs) have been attracting interest for their potential roles in intelligent transport systems. In VANETs, a multi-hop broadcast protocol is required when the transmission range is not enough to cover all intended receivers. Many applications including collision warning system and other value-added applications require a lightweight and reliable multi-hop broadcast protocol as the network layer communication protocol. However, due to the various vehicle densities, vehicle movement, limited bandwidth of wireless communications, it is difficult to provide a high dissemination ratio and low end-to-end delay. An efficient forwarding mechanism is required to eliminate unnecessary transmissions. The mechanism should be able to provide a high packet dissemination ratio and low overhead in various scenarios including high density networks and lossy networks. There have been many works [1]–[10] considering the multi-hop broadcasting issue. However, when the number of traffic flows is large, the protocol overhead is very huge. This results in high end-to-end delay. Moreover, retransmissions are not seriously considered in these works. Since there is no acknowledgment for the broadcast data frames at the MAC layer, an efficient loss detection and retransmission mechanism should be considered at the network layer. In this paper, we propose a protocol which can significantly reduce the number of required transmissions in comparison to existing works.

In order to provide a high packet dissemination ratio and low end-to-end delay, it is important to reduce the redundant broadcasts. Many protocols have been proposed to handle this issue. These protocols can be classified into two categories: (i) sender-oriented (deterministic) protocols, and (ii) receiver-oriented (non-deterministic) protocols. In the receiver-oriented protocols [1]–[4], upon a packet reception, each node determines whether to forward or not by using an autonomous approach. Therefore, it is difficult for the receiver-based protocols to entirely eliminate the redundant broadcasts.

In the sender-oriented protocols, each sender node specifies the relay nodes. Many protocols [5], [6], [8] do not take account of channel fading in the relay node selection. Some protocols [6], [9] utilize retransmissions to improve reliability. However, these protocols [6], [9] require acknowledgments from all receivers, resulting in a high overhead and collision probability. The end-to-end delay of [7] needs improvement. In all these sender-oriented protocols [5]–[7], the relay nodes are different for different broadcast data flows. This is inefficient in terms of MAC layer contention time, especially when the number of data flows is large. A backbone-based broadcast protocol (BBBR) is proposed in [10]. Although BBBR can improve the MAC layer contention efficiency by using the same backbone nodes to forward data packets coming from different traffic flows, the number of required transmissions remains the same. The retransmission issue is not discussed in [10].

In this paper, we propose a protocol which can significantly reduce the number of required transmissions as compared to existing broadcast protocols. The protocol achieves this by employing intra-flow and inter-flow network coding jointly. The protocol uses backbone vehicles (the same as [10]) to disseminate broadcast messages. The sender nodes (including the source node and other forwarder nodes) conduct network coding. Since different traffic flows use the same backbone nodes, the network coding approach is efficient in most scenarios. By using inter-flow network coding the protocol can significantly reduce the number of required transmissions of data packets. This can significantly reduce the end-to-end delay. The protocol also provides a lightweight retransmission mechanism by using intra-flow network coding.

In section II, we give a detailed description of the proposed protocol. Next, we present theoretical analysis in section III, and present simulation results in section IV. Finally, we present our conclusions in section V.

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II. PROPOSED PROTOCOL

A. Assumptions

Each node knows its position information, velocity information and antenna height. Each node transmits this information by using hello messages. The road width is considered to be negligible as compared with the radio range. All vehicles have the same transceiver and transmit with the same power. The average transmission range is assumed to be known by all vehicles. Each sender node needs to know a packet should be delivered to whether the forward direction or backward direction (using the current node as the reference node).

B. Backbone-based data dissemination

The proposed protocol, selects backbone vehicles to relay data using the same method as Ref. [10]. Backbone nodes use network coding to encode the packets before transmissions (this will be explained in the next Subsection). As shown in Fig. 1, upon reception of a data message, a node forwards the message if the node is a backbone vehicle. The backbone vehicles are updated periodically (with the same interval as hello messages) based on the topology information acquired from the received hello messages. Backbone vehicles are selected in a distributed manner from the neighborhood. The vehicle velocity, vehicle density on the driving direction and antenna height are considered in the backbone node selection by using a fuzzy logic algorithm [11] to combine these constraints. The backbone selection algorithm ensures the generation of a reliably connected vehicle backbone.

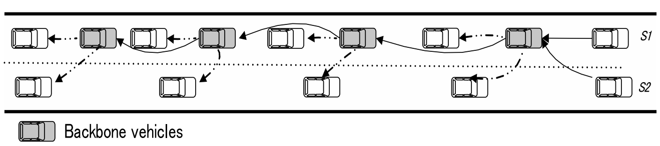

Fig. 1. Data dissemination using backbone vehicles.

C. Inter-flow network coding

The backbone-based forwarding makes the inter-flow network coding possible. Now we explain how to encode two packets coming from two different directions (inter-flow network coding). As show in Fig. 2, upon reception of a_i from S_1 and b_j from B_2, node B_1 encodes a_i and b_j, and transmits $a_i + b_j$. Upon reception of $a_i + b_j$, node S_1 and B_2 can successfully decode the packet and retrieve the original packets b_j and a_i respectively. For traditional protocols, node B_1 has to send two packets a_i and b_j. By using the network coding, the proposed protocol can save one transmission for each 4 transmissions.

D. Intra-flow network coding

Since there is no MAC layer acknowledgment for broadcast frames, packet losses at the relay node could happen. Therefore, we have to consider a lightweight retransmission mechanism. Since there are multiple intended receivers for

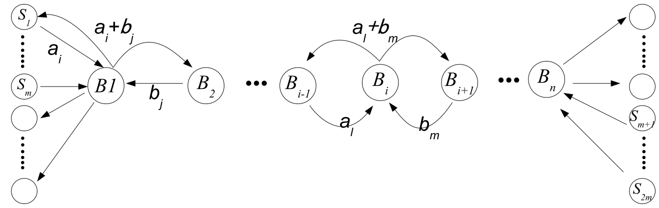

Fig. 2. Inter-flow network coding at backbone vehicles.

broadcast applications, the aim of the protocol is to disseminate data packets to all these receivers. However, different nodes could experience different channel conditions. If we use a simple retransmission mechanism, the retransmission overhead could be very large. This is because different nodes could lose different packets and therefore the sender node has to retransmit all these lost packets. In order to handle this problem, we use network coding to reduce the number of transmissions when a packet loss occurs.

In the proposed protocol, the sender node processes network coding-based on a batch of two packets (we say these two packets belong to the same generation). The source node uses network coding to encode 2 consecutive native packets (which bound for the same direction) to get 2 encoded packets, and transmits the encoded packets. As shown in Fig. 3, B_{i-1} encodes a_l and a_{l+1} to get two encoded packets, $a_l + a_{l+1}$ and $a_l + 2a_{l+1}$. When $a_l + a_{l+1}$ is lost at the next relay node B_i, and $a_l + 2a_{l+1}$ is lost at node M, the proposed protocol only needs to retransmit one packet which is $2a_l + 3a_{l+1}$. After reception of the packet, B_i and M can retrieve the original packet a_l and a_{l+1}. For the traditional approach (without network coding), two retransmissions (a_l and a_{l+1}) are required. Therefore, the proposed protocol can significantly reduce the number of retransmissions.

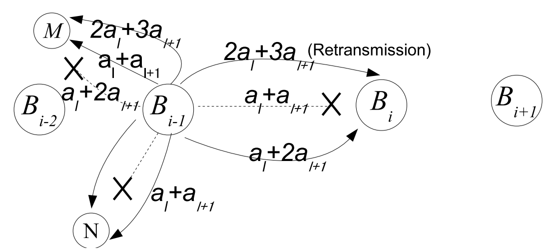

Fig. 3. Inter-network coding at backbone vehicles.

For implementation simplicity, we use deterministic coding coefficients. The coding vectors used in the protocol are

$$Y = \begin{pmatrix} y_1 \\ y_2 \\ y_3 \\ y_4 \end{pmatrix} = \begin{pmatrix} 1 & 1 \\ 1 & 2 \\ 2 & 3 \\ 3 & 5 \end{pmatrix} \begin{pmatrix} a \\ b \end{pmatrix} = \begin{pmatrix} a+b \\ a+2b \\ 2a+3b \\ 3a+5b \end{pmatrix}. \quad (1)$$

The benefit of using these coding vectors is that they are mutually independent. We can transform from any two encoded packets to other two encoded packets. This is important for performing retransmissions (this feature is used in Algorithm 1 of Subsection II-F). Each node can retrieve the original packets upon the reception of any two encoded packets.

E. Integration of intra-flow and inter-flow network coding: dissemination direction-aware network coding

In the proposed protocol, data packets are forwarded by the backbone vehicles. We use an approach which conducts different network coding processing depending on dissemination directions of the packets. In the proposed protocol, network coding is conducted based on a batch of two packets. If the two packets come from different directions (which means the dissemination directions are different), the proposed protocol uses the inter-flow network coding approach as described in Subsection II-C. When two packets are required to transmit to the same direction, the proposed protocol encodes these two packets in order to improve the packet dissemination ratio.

As shown in Fig. 4, upon reception of $a_l + a_{l+1}$, $a_l + 2a_{l+1}$, $b_l + b_{l+1}$ and $b_l + 2b_{l+1}$, B_i can retrieve the original packets a_l, a_{l+1}, b_l and b_{l+1}. Node B_i first encodes a_l and b_l to get $X = a_l + b_l$, and encodes a_{l+1} and b_{l+1} to get $Y = a_{l+1} + b_{l+1}$. After that node B_i transmits $X + Y$ and $X + 2Y$ which can be used to retrieve the original packets, a_l, a_{l+1}, b_l and b_{l+1}.

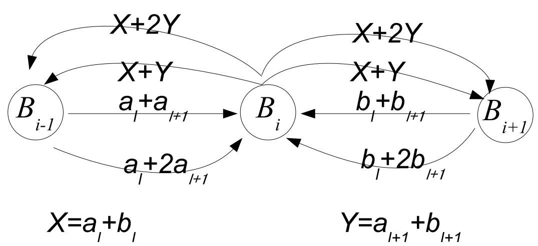

Fig. 4. Integratioin of intra-flow and inter-flow network coding.

F. Data encoding and forwarding algorithm

Each node maintains two different send buffers, the forward direction buffer and the backward direction buffer. The forward direction buffer is used to maintain the packets which are required to disseminate to the forward direction (using the current node as the reference node). The backward direction buffer is used to maintain the packets which are required to disseminate to the backward direction. When there are packets to send, a node does the actions as shown in Algorithm 1. Upon reception of a packet, a node does the actions as shown in Algorithm 2.

Algorithm 1 Actions at each node when the node has packets to transmit

```
if (the current node detects a packet loss) then
    if (the packet was sent before using encoded transmissions) then
        Transform the two encoded packets (which were sent before) to get a new encoded packet, and transmit the new encoded packet.
    else
        Get the lost packet and transmit the packet.
    end if
else
    if (there are packets in both the forward direction buffer and backward direction buffer) then
        Get one packet from the forward direction buffer and one packet from the backward direction buffer.
        Encode the two packets and transmit (Using inter-flow network coding as described in Subsection II-C).
    else
        if (there are two packets in the forward (or backward direction) direction) then
            Encode the two packets to get an encoded packet and transmit (Using intra-flow network coding as described in Subsection II-D).
        else
            Transmit the packet.
        end if
    end if
end if
Maintain the sent packet for a short period of time for future use.
```

Algorithm 2 Actions at each node upon reception of a packet

```
if (the packet is an encoded packet) then
    if (the packet is the first packet in this generation) then
        Maintain the packet for a short period of time (in order to retrieve the original packets, when the next corresponding packet is received)
    else
        Decode the encoded packets to retrieve the original packets.
    end if
else
    if (the packet can be used with a previously received packet to retrieve the original packets) then
        Use with the previously received packet to retrieve the original packets
    end if
end if
Maintain the packet for a short period of time for future use.
```

III. Theoretical analysis

A. Number of transmissions

By using inter-flow network coding, the proposed protocol can significantly reduce the number of transmissions. As shown in Fig. 5, we assume packets $\{a_1 \dots a_z\}$ are broadcasted from the left to right ($B_{i-1} \rightarrow B_{i+1}$), and packets $\{b_1 \dots b_z\}$ are broadcasted from the right to left ($B_{i+1} \rightarrow B_{i-1}$). In order to transmit a_1 (from B_{i-1}) to B_{i+1}, the traditional approach (without network coding) requires 4 transmissions. However, by using inter-flow network coding, the proposed protocol can do it in 3 transmissions. Therefore, when the traffic flows are evenly distributed in the network, we can reduce the number of transmissions by 25%.

For the proposed protocol, after exchanging data messages for a while, each backbone node would have one packet to send from right to left (b_m), and one packet from left to right (a_l). In this case, the backbone node only needs to send $a_l + b_m$. After convergence, every backbone node (excluding the source node) only needs to send the encoded packet. As a result, the proposed protocol can reduce the number of transmissions by near 50%.

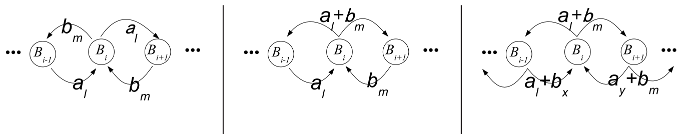

Fig. 5. Data forwarding schemes (left: Traditional, middle: proposed protocol at the beginning, right: proposed protocol after convergence)

B. Reception probability at a non-relay node: the benefit of using intra-flow network coding

We analyze the packet reception probability of two consecutive packets at a receiver node. We assume the link loss probability is p_l. For the proposed protocol, a receiver node can retrieve the original packet when the node receives two encoded packets. With a retransmission (there would be 3 transmissions, 2 original packets and 1 retransmitted packet), the probability the receiver node can retrieve the original packets is

$$\hat{P}_1 = 1 - p_l^3 - \binom{3}{1} p_l^2 (1 - p_l). \tag{2}$$

For the traditional approach, the reception probability at a receiver node is

$$P_1 = 1 - p_l^3 - \binom{3}{1} p_l^2 (1 - p_l) - \frac{1}{3}\binom{3}{2} p_l (1 - p_l)^2. \tag{3}$$

If we retransmit twice (there would be 4 transmissions, 2 original packets and 2 retransmitted packets), the reception probability for the proposed protocol is

$$\hat{P}_2 = 1 - p_l^4 - \binom{4}{1} p_l^3 (1 - p_l). \tag{4}$$

For the traditional approach, the reception probability at a receiver node is

$$P_2 = (1 - p_l^2)^2. \tag{5}$$

Based on above analysis, Fig. 6 shows the comparison for various packet loss probabilities. We can observe that the proposed protocol can significantly improve the packet reception probability by using the network coding approach.

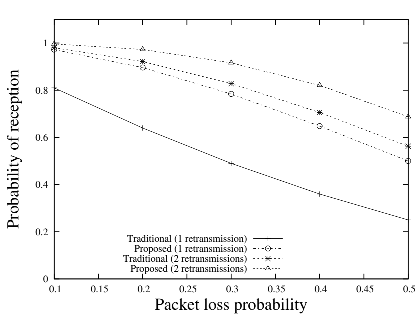

Fig. 6. Reception probability at a non-relay node for various packet loss probabilities.

C. Data dissemination latency

Data dissemination latency is affected by the contention delay and propagation delay for the data packet dissemination. Since the proposed protocol can significantly reduce the number of sender nodes by using the backbone-based data dissemination, the contention delay is very small. Based on equations (Eq.(4) and Eq.(10)) given in Ref. [10], the channel busy incurred delay can be caculated as

$$\tau = \left(1 - (1 - \frac{1}{\mathrm{CW}+1})^N\right) \cdot \frac{\mathrm{CW}+1}{2} \cdot T +$$
$$(1 - \frac{1}{\mathrm{CW}+1})^N \left(\sum_{j=1}^{\frac{\mathrm{CW}+1}{2}-1} \sum_{i=1}^{j} \binom{j}{i} p^i \right) \cdot \Theta(N) \cdot T \tag{6}$$

where CW is the contention window size, N is the number of sender nodes in the sensing range, and T is the time required for transmitting a data frame. As shown in Fig. 7, since the number of sender nodes is limited, the proposed protocol can significantly reduce the channel busy time in comparison to the traditional approach.

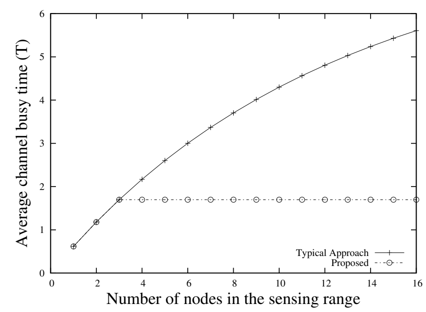

Fig. 7. Channel busy time for various numbers of nodes in the sensing range.

As explained in Subsection III-A, the proposed protocol also can reduce the number of transmissions by 25%. In a saturated network, after convergence, the proposed protocol can reduce the number of transmission by 50%. This is significant especially when the number of hops is large. When the number of traffic flow is m, the delay required to disseminate 1 packet from each source node to entire network is as shown in Fig. 8. The advantage of the proposed protocol is very significant especially when the number of hops is large.

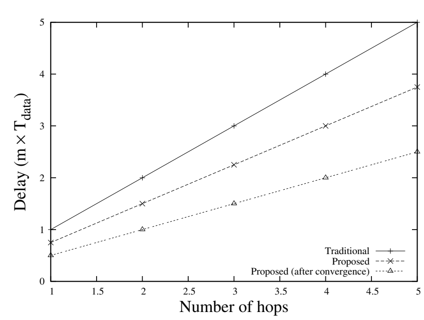

Fig. 8. Delay for various numbers of hops.

IV. Simulation Results

We used ns-2.34 [12] to conduct simulations (see Table I). In the Freeway simulation, we used a freeway which had two lanes in each direction [13]. The maximum allowable vehicle velocity was 40m/s. We evaluated the protocol's performance in various numbers of nodes. In the street scenarios [14], [15], the maximal vehicle velocity was 18m/s. The street consisted of 5 horizontal streets and 5 vertical streets and every street had one lane in each direction. The distance between any two neighboring intersections was 400 m. We generated scenarios with various numbers of broadcast source nodes (traffic flows). Nakagami propagation model was used to simulate the channel fading (see Table II). We used these parameter values because they model a realistic wireless channel of VANETs [16]. In the simulation, 10% of vehicles were with higher antennas. The links involving the higher antenna vehicles were set to provide 10% higher reception probability. The proposed protocol was compared with Flooding, Weighted p-persistence [1], Enhanced MPR Broadcast [6] and BBBR [10]. The error bars indicate the 95% confidence intervals.

TABLE I
SIMULATION ENVIRONMENT

	Freeway scenario	Street Scenario
Topology	2000m, 4lanes	1700m × 1700m
Number of nodes	100 to 600	619
Mobility generation	Ref. [13]	SUMO + TraNS
Number of sources	4	5 to 55
Number of packets	400 packets (512bytes) at each source	
Data rate	5 packets per sec	1 packet per sec
MAC	IEEE 802.11 MAC (2Mbps)	
Simulation time	500 s	

TABLE II
PARAMETERS OF NAKAGAMI MODEL: FREEWAY (STREET)

gamma0_	gamma1_	gamma2_	d0_gamma_	d1_gamma_
1.9 (2.0)	3.8 (2.0)	3.8 (2.0)	200 (200)	500 (500)
m0_	m1_	m2_	d0_m_	d1_m_
1.5 (1)	0.75 (1)	0.75 (1)	80 (80)	200 (200)

A. Number of messages

Fig. 9 shows the number of messages per data packet for various numbers of nodes. This performance metric is calculated as the number of messages generated (including both ACK messages and data messages transmitted by all nodes in the network) divided by the number of data packets generated by the source nodes. The Flooding and Weighted p-persistence scheme incur a large number redundant transmissions. The Enhanced MPR Broadcast shows the largest number of broadcasts due to the ACK messages and retransmissions. The proposed protocol and BBBR can significantly reduce the number by using backbone nodes to disseminate data packets. The advantage of the proposed protocol over BBBR is due to the inter-flow network coding which can significantly reduce the number of required data transmissions. This advantage is more significant when the number of source nodes is large (see Fig. 10).

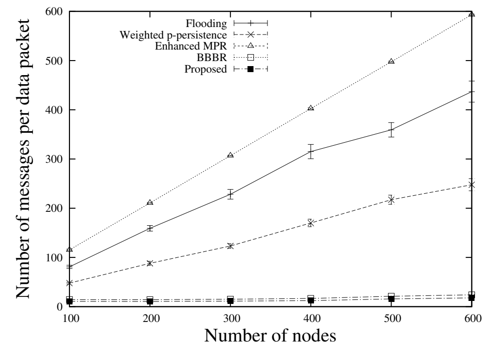

Fig. 9. Number of messages per data packet in Freeway scenario.

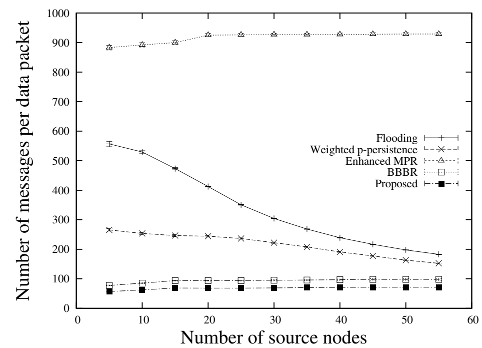

Fig. 10. Number of messages per data packet in Street scenario.

B. Packet dissemination ratio

Fig. 11 and Fig. 12 show the packet dissemination ratio comparison in Freeway scenario and Street scenario respectively. The proposed protocol shows the best performance due to the joint inter-flow and intra-flow network coding. The intra-flow network coding can provide a high packet reception probability at a receiver node. The inter-flow network coding can reduce the collision probability by reducing the number of required transmissions, which also contributes to the high packet dissemination ratio.

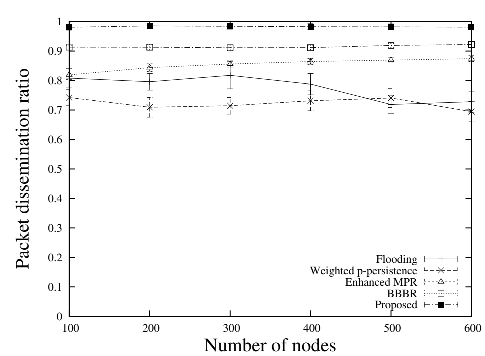

Fig. 11. Packet dissemination ratio in Freeway scenario.

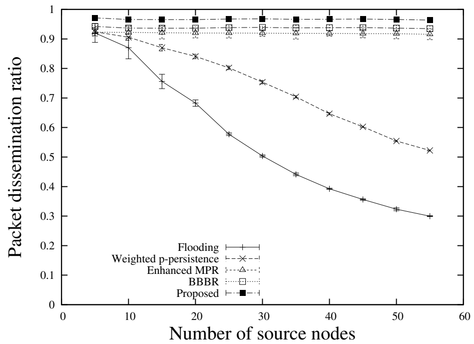

Fig. 12. Packet dissemination ratio in Street scenario.

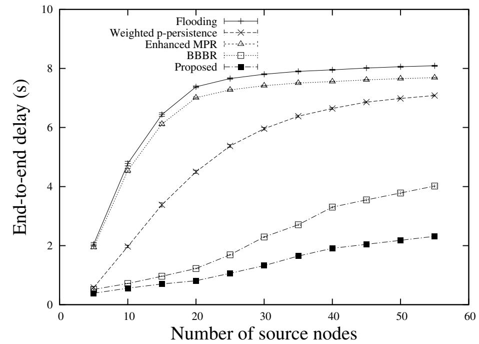

Fig. 14. End-to-end delay in Street scenario.

C. End-to-end delay

Fig. 13 shows the average end-to-end delay for various numbers of nodes. Due to the joint network coding approach, the proposed protocol can significantly reduce the number of data transmissions (or retransmissions), resulting in shorter MAC layer contention time at each node and lower end-to-end delay. The advantage of the proposed protocol over BBBR becomes more notable when the number of nodes increases. This is mainly because the proposed protocol can reduce the number of transmissions significantly by using the inter-flow network coding. As shown in Fig. 14, the advantage is very significant when the number of source nodes is large. This is because that protocol overhead has a significant effect on the end-to-end delay when the data rate is high.

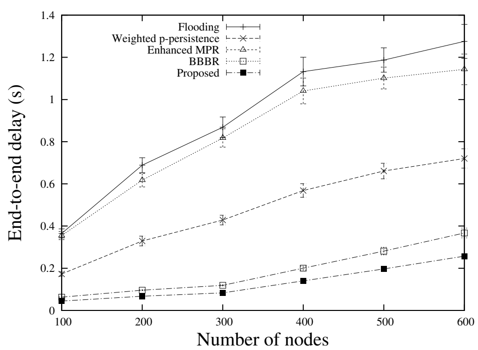

Fig. 13. End-to-end delay in Freeway scenario.

V. Conclusions

We proposed a network coding-based broadcast protocol for vehicular ad hoc networks. The protocol employs a joint inter-flow and intra-flow network coding approach. By using the inter-flow network coding, the protocol can significantly reduce the number of transmissions. The intra-flow network coding can improve the packet dissemination ratio. Therefore, the proposed protocol can provide a lightweight and reliable solution for data dissemination in VANETs. We used theoretical analysis and simulation results to show the advantage of the proposed protocol over existing alternatives.

Acknowledgment

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