Network Coding Based Anycasting In Vehicular Ad-Hoc Network

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Network Coding Based Anycasting In Vehicular Ad-Hoc Network

Major Project

Submitted in partial fulfillment of the requirements

For the degree of

Master of Technology in Computer Science and Engineering

By

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Declaration

I, Ankit Patel, Reg.No. 09MCES06 , give undertaking that the Major Project entitled "Network Coding based Anycasting in VANET" submitted by me, towards the partial fulfillment of the requirements for the degree of Master of Technology in Computer Science and Engineering of Nirma University, Ahmedabad, is the original work carried out by me and I give assurance that no attempt of plagiarism has been made. I understand that in the event of any similarity found subsequently with any published work or any dissertation work elsewhere; it will result in severe disciplinary action.

Ankit Patel (09MCES06)

Date: 15-5-12

Place: Institute of Technology, Nirma University

Certificate

This is to certify that the Major Project entitled "Network Coding Based Anycasting in Vehicular Ad-Hoc Network" submitted by Ankit Patel (09MCES06), towards the partial fulfillment of the requirements for the degree of Master of Technology in Computer Science and Engineering of Nirma University, Ahmedabad is the record of work carried out by him under my supervision and guidance. In my opinion, the submitted work has reached a level required for being accepted for examination. The results embodied in this major project, to the best of my knowledge, haven't been submitted to any other university or institution for award of any degree or diploma.

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Abstract

Vehicular Ad-Hoc Network (VANET) are sparse ad hoc networks in which no contemporaneous path exists between source and destination most of the time which increase delivery and decrease delivery ratio. Mobile Ad hoc Network routing protocols such as AODV, DSR etc. fail in such scenario because they try to find end-to-end path before data transmission which is not exist in VANET. So different routing strategy are require in VANET which follow 'store-carry-forward' paradigm in which two nodes exchange messages with each other only when they come into contact. Many comfort applications in VANET requires anycast service which allows a node to send a message to at least one, and preferably only one, of the members in a group. In traditional network, relay node or router simply forward the information packets destined to other node. In network coding, source node or intermediate node or router allows to combine number of packets it has received or generated into one or several outgoing packets. In network coding the successful reception of information does not depend on receiving specific packets but on receiving sufficient number of independent packets. So reliability is one of the issue in network coding. So we use network coding with mulit generation mixing in which packets are grouped into generations and generations are grouped into mixing set. Packets of particular generation in particular mixing set is mixed with the packets of all previous generations. So encoded packets of later generations are having knowledge of packets of previous generations and hence increase reliability. We propose Anycast routing protocol for VANET which uses 'Network coding with mulit generation mixing' to improve the performace. By using simulation we compared the performance of proposed protocol in terms of delay, delivery ratio and throughput with the same protocol using network coding and using conventional scheme. Simulation results suggest, our protocol achieves significantly less delay, higher delivery ratio and higher throughput compared to network coding based scheme and conventional scheme.

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With immense pleasure, I would like to present this report on the dissertation work related to "Network Coding Based Anycasting in Vehicular Ad-Hod Network".

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- Ankit Patel (09MCES06)

Abbreviation Notation

VANET .... Vehicular Ad-Hoc Network
NC .... Network Coding
LNC .... Linear Network Coding
RLNC .... Random Linear Network Coding
ACK .... Acknowledgement
PNCO .... Partial Network Coding
AF .... Amplify-and-Forward
DF .... Decode-and-Forward
ONC .... Opportunistic Network Coding
UE .... User Equipment
BS .... Base Station
ARQ .... Automatic Repeat Request
G-by-G .... Generation-by-Generation
MGM .... Multi Generation Mixing
SVC .... Scalable Video Coding
GF .... Galois Field
MSS .... Mixing Set Size
GenSize .... Generation Size

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Chapter 1

Motivation

1.1 Motivation

Vehicular Adhoc Network (VANET) is an emerging flavor of Mobile Adhoc Networks (MANET) to improve intelligent inter Vehicle and Vehicle to Vehicle communication without any fixed infrastructure [1]. Millions of people around the world die every year in car accidents and many more are injured. Implementations of safety information such as speed limits and road conditions are used in many parts of the world but still more work is required. VANET should, collect and distribute safety information to massively reduce the number of accidents by warning drivers about the danger before they actually face it. Two main applications of vehicular Adhoc networks are Safety Applications and Comfort Applications. First category improves safety levels of passengers via Inter Vehicle Communication or Vehicle to Vehicle Communication Some common examples of this application are: signal violation warning, road condition warning, intersection coordination and emergency warning systems. Second kind of applications improves passengers comfort level through optimized route to destination, traffic information, weather information, gas station or restaurant location and price information are good examples of comfort applications. In both classes of applications data messages or control messages should be continuously exchanged between mobile nodes or vehicles.

VANET will form the biggest ad hoc network ever implemented, therefore issues of stability, reliability and scalability are of concern. VANET therefore is not an architectural network and not an ad hoc network but a combination of both [2], this unique characteristic combined with high speed nodes complicates the design of the network. As these networks have no fixed communication structure and may vary heavily due to which routing of data packets through VANETS is very crucial.However, due to dynamic network topology, frequent disconnected networks, varying communication conditions and hard delay constraints VANETS can be distinguished from other kinds of Adhoc networks.

There are number of different applications where the network is sparse and experiences frequent and long disconnection. As an example, consider a traffic management system in a city where vehicles are network nodes which generate and forward vehicular traffic data through other vehicles. There may never be contemporaneous path between source and destination through other vehicles. Many VANET applications need anycast service. For example, vehicle on road may send the packet requiring optimal route to destination, traffic information, weather information, gas station or restaurant location to one of the server on road side, it is necessary to transmit information from a server to a vehicle or vehicle may transmit information packet regarding accident to one of the server(ambulance or emergency service providers). However, traditional anycast methods proposed for the Internet or mobile ad hoc networks are not suitable for VANET, due to the challenge of frequent network partitions. Data transmissions suffer from large end-to-end delays along the tree because of the repeated partitions due to frequent disconnections. Also the traditional approaches may fail to deliver a message when the possibility of link unavailability becomes high. To increase chance of delivery and to reduce delivery delay, routing approaches in VANET make multiple copies of a packet in the network. However communication overhead and buffer occupancy increases as we increase number of copies per packet. If we can reduce number of copies per packet without impacting the performance, this overhead can be reduced.

In Network coding(NC), instead of forwarding packets as it is, nodes may recombine two or more input packets into one or more output packets[3]. The successful reception of information does not depend on receiving specific packets but on receiving sufficient number of independent packets. It is illustrated by the following intuitive example[4] shown in Figure 1.1. In the example, packets x and y are to be sent from node 1 to node 2 using multipath as the links are lossy. In Figure 1.1(a), routers R1 and R2 encode incoming packets into same number of outgoing packets while in Figure 1.1(b), packets are transmitted as it is. Please note that number of transmissions and number of losses in both the cases are same and so number of packets received at sink are also same. But in the second case, only packet x is received successfully while in the first case, with high probability, both the packets x and y will be received successfully. i.e., network coding is more effective in delivering packets successfully. Further, multiple copies of a packet increase buffer requirement of nodes in the net-

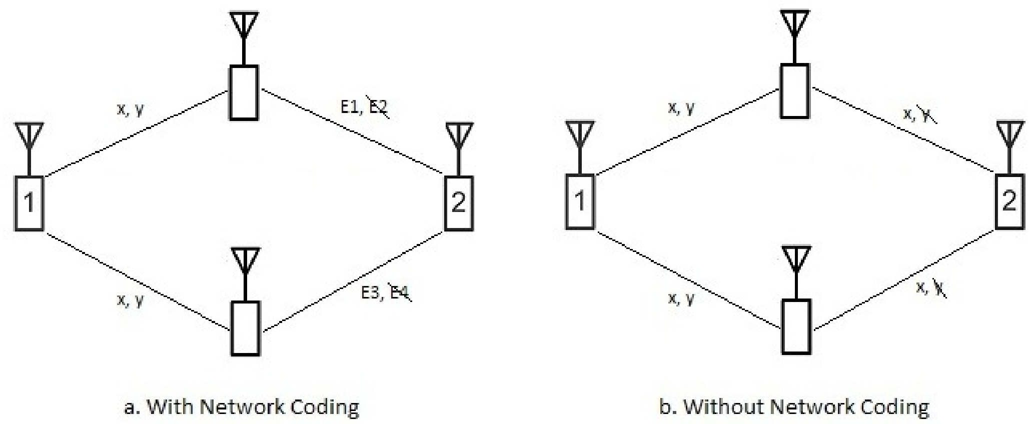

Figure 1.1: Example showing benefit of network coding in lossy networks.

work. So, efficient buffer management is very crucial. Network coding can use limited buffer more efficiently because instead of dropping packets, it can reduce number of packets by combining existing packets.

1.2 Thesis Outline

In Chapter 2, literature survey of related work and identified open issues are presented. In Chapter 3, problem definition is given. We present our protocol in Chapter 4, In Chapter 5 Implementation detail of G-By-G Network coding and Network Coding with Multi-Generation Mixing is given. We present and discuss simulation results in Chapter 6, In Chapter 7 Conclusion and future work are elaborated.

Chapter 2

Literature Survey

2.1 Network Coding

There are two types of network coding, **Linear Network Coding (LNC)** and **Random Linear Netwokr Coding (RLNC)**. In traditional network, relay node or router simply forward the information packets destined to other node. In LNC, source node or intermediate node or router allows to combine number of packets it has received or generated into one or several outgoing packets, where addition and multiplication are performed over the field GF_2^s[5]. Linear combination is not concatenation, if we linearly combine packets of length L, the resulting encoded packet also has size L.

In LNC , meaningful coefficients should be used for encoding and decoding of packets. LNC requires central authority to control generation of this meaningful coefficient. Algorithms employed for this should be centralized. But in wireless networks due to node's mobility and heterogeneity of network distributed approaches are suitable. So RLNC[6]suggests the random generation of the encoding coefficient. In wireless networks, channels have a bigger error rate, higher interference between channels, unknown network topology. So protocols used in wireless networks should be optimized for above conditions. In RLNC, each node generates its own coding coefficient for each encoded packet. Also coefficients are sent to the destination in the packet header. So, the destination can decode the packet without knowing network topology or encoding rules, even if the topology is not fixed.

Number of successful transmission was measured for two cases in[7] , with and without random linear network coding. Simulation results indicates, there is an increase in number of successful transmissions for distance greater than 500 distance units in the case of random linear network coding.

For broadcasting there are no ACK to confirm the reception and we do not have mechanism to avoid congestion which decreases throughput. Results were collected for broadcasting scenario using network coding. Using random linear network coding the optimum result is obtained for a network with 150 nodes and congestion coefficient 0.026 (i.e. 4/150, where 146 nodes receive original packets out of 150 nodes, so 4 nodes with congestion). In RLNC, with increase in number of nodes, congestion decreases. In RLNC, for multicast scenario, probability that RLNC is valid is at least $(1-d/q)^n$, where d is group size i.e. number of destination nodes, q=field size and n is number of links[8].

In LNC or RLNC there is higher packet delay as in that we have to delay the transmission of already arrived packets until additional packets have been collected. In opportunistic network coding, instead of selecting a particular node to be the next hop forwarder, nodes in the network coordinate with each other to select a multiple nodes which can potentially be served as next-hop forwarder. From multiple nodes, the node which is closest to the destination will forward the packet and other will drop the packets. In this scheme coordination amongst the nodes is required.

2.1.1 Encoding and Decoding of packets in Network Coding

In RLNC packets are encoded and decoded as follows:

Encoding

Original Packets : M1,...Mn

Encoded Packets :

$$Xi = \sum giMi \tag{2.1}$$

where summation is over i=1 to n and gi=g1,...,gn are randomly generated coefficients

Forwarding:Encoding already encoded packets

Set of encoded packets :(g1,X1),...,(gn,Xn)

New encoded packets : (g1',X1'),...,(gn',Xn') where

$$Xi' = \sum gi'Xi \tag{2.2}$$

where summation is over i=1 to n and gi'=g1',...,gn' are randomly chosen coefficints

Forwared coefficent with packets are : (h1,...,hn)where

$$hi = \sum gi'gi \tag{2.3}$$

where summation is over i=1 to n

Decoding

Set of received packets : (h1,X1'),...,(hn,Xn')

System of n lienar equations

$$Xi' = \sum hiMi \tag{2.4}$$

where summation is over i=1 to n with Mis as unknown

2.1.2 Variant of Network Coding

In this section, the survey of various variant of Network Coding is presented with their results.

- **Partial Network Coding(PNCO) with Opportunistic Routing**

  In this scheme,source node breaks the information into n blocks of k packets and randomly mixes packets of same block before forwarding[9]. When sender receives ACK of previously sent block from receiver, it sends next block. ACK

will allow sender to send next block and hence reduces the delay. Sender also calculates the delivery probability on each link with expected cost metric and sends the forwarder list in the packet header. When forwarding nodes receive dependent packet it drops it. If it receives independent packets then packets are encoded again by using forwarders coefficient. When destination receives the encoded packets, it decodes them and gets the original packets. In above scenario, windowing scheme is used to stop retransmission of already sent packet by sender as well as to allow sender to send next packet.
Resulsts shows that compare to path routing protocol, PNCO with opportunistic routing result in an increase in the achieved throughput. Over 90 percent of PNCO flows have throughput greater than 50 packets/sec. but path routing protocol is about 40 percent. Network coding can result in higher per-packet delay. The average delay of PNCO is higher, compared to path routing protocol. But it is reduced about 50 percent when compared to conventional network coding.

- **Cooperative Network Coding**

  In **Cooperative Network Coding** with cooperative communication different nodes collaboratively forward the information packets to exploit the spatial diversity[10]. Participating nodes will be determined through upper layer protocols. Nodes in cooperative domain synchronized through synchronization technique e.g. GPS. There are two kinds of protocols for the forwarding nodes. i) Amplify-and-forward (AF) ii) decode-and-forward (DF). In AF mode, relay nodes transmit the received signal after some power normalization to amplify the information. In DF, relay nodes decode the received signal and then transmit it with its own encoding scheme to forward clean information, which reduces transmission rate. Here the information is exchanged in two phase. In phase I source transmits information to relays and in phase II relays will broadcast the combined signals of different sources to destination or another relay nodes.

Results in [10] shows that in one relay system Decode-and-Forward outperforms the Amplify-and-Forward. However, in two relay the Amplify-and-Forward is better than Decode-and-Forward.

Network coding allows mixing of various data packets. But in this scheme, packet has to wait to be coded with other packet until other packet arrives. It increase delay and loss rate.

- **Opportunistic Network Coding (ONC)**
  It is the approach in which whether packet is transmitted with or without network coding is decided by the status of the buffer's queue at a node[11]. With network coding, number of packets transmitted by relay will decrease and hence increase the power efficiency of relay node. While in physical network coding, relay nodes are not decoding the received signal but it simply amplifies and broadcasts the received signal and hence complexity of the relay node increased. This scheme is suitable for stationary traffic flow. For real time traffic, packet has to wait to network coded with other packet (delay). Also for finite buffer, packet loss rate will increased. To overcome these limitations, ONC can be used. When the probability of sending packets without encoding is fixed to be 0 then it reduces to conventional network coding. So conventional network coding is one of the cases of ONC.
  Although the in term of delay and packet loss conventional network coding is not optimal. For ONC there is a delay-power tradeoff. Simulation results shows that ONC achieves lower delay compared to conventional network coding.

- In wireless digital broadcasting applications , base station (BS) broadcasts information to a user terminals (User equipment-UE) through wireless broadcasting channels. A received packet at User Equipments (UE) is either error-free or discarded as erroneous. UE will request BS to retransmit discarded packets which is automatic repeat request (ARQ) error control protocol. This strategy becomes inefficient as the number of UEs increases or number of packets increases. To improve system efficiency, use of network coding during the retransmission phase was suggested in [12].

There are N information blocks sent by BS to $M >= 2$ UE. So there are M BS to UE block erasure channels are assumed. After the transmission of N information blocks to M UEs, each UE feeds back the indices of the lost or erased blocks. An error matrix E is generated by BS to record the block erasure status reported by UEs. The size of matrix E is MxN where ei,j =1 if the jth block is erased otherwise ei,j = 0.

In retransmission phase the set of erased blocks is divided into subsets such that at least one erased block per UE is in any particular subset. The erased blocks in a subset are encoded into one encoded block for retransmission. In ARQ each erased block is retransmitted separately. Ex. Error matrix E for M=2 and N=6 is as shown below.

$$E = \begin{pmatrix} 1 & 0 & 0 & 1 & 0 & 1 \\ 0 & 1 & 1 & 0 & 0 & 1 \end{pmatrix}$$

As shown above with ARQ, blocks 1,2,3,4 and 6 retransmitted separately. With network coding only three blocks are retransmitted 1 xor 2, 3 xor 4, and 6. Now UE1 has received blocks 2, 3 and 5 correctly and UE2 has received blocks 1, 4 and 5 correctly in original broadcast. Now, in retransmission, UEs can retrieve the remaining blocks through simple modulo 2 additions.

Performance of proposed scheme is compared against traditional ARQ. In simulation in [12] the impact of N (Number of blocks), M (No of User Equipments) and erasure probability pi on the normalized overhead is considered two scenario, i) by keeping identical erasure rate on all links and erasure rate on link1,$p1 > pi$, ii) identical erasure rate on all other links. Proposed scheme can asymptotically achieve the lower bound on normalized overhead when the numbers of information blocks are sufficiently large.

2.2 Generation-by-Generation(G-by-G) RLNC

For all practical purposes, the size of the matrices with which network coding operates has to be limited. This is straightforward to achieve for deterministic network codes, but more difficult with random network coding. So for RLNC packets are grouped into generations[13]. Here the size of generation i.e. number of packets in one generation is fixed. Packets of same generation are encoded with each other as shown in Figure 2.1[14]. Upon receiving encoded packets, intermediate node makes generations of received encoded packets destined to same destination node and encodes the packets of same generation using its own encoding vectors. Intermediate node sends effective coefficient generated using its own coefficients generated and received from sender along with encoded packets. This new effective coefficients received by receiver will help it to decode original packets sent by sender.

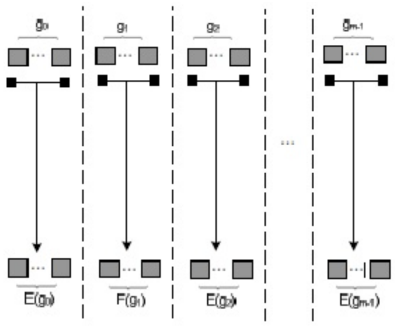

Figure 2.1: G-by-G Network Coding, generations are encoded separately

As the number of originating packets (n) in a network for given destination increases, the amount of memory needed to store coefficients of encoded packets increases because these coefficients are to be remembered till a node receives at least n packets to decode all original packets. Further, till at least n encoded packets are not received, most of the n original packets can not be decoded. Thus delivery delay of a packet increases with n. To reduce memory requirement and the delay, originating packets can be grouped together into so called ‘generation’. Now all the nodes in the network will encode packets of the same generation. If the generation size is k$(k \ll n)$, whenever the receiver receives at least k packets of a generation, it will able to decode k original packets and the corresponding k encoded packets can be discarded, freeing the memory and the delay will also be reduced as receiver will have to now wait only for k encoded packets to be able to decode. But as packets from different generations can not be 'mixed', mixing opportunity reduces as k reduces.

Performance of Network Coding can be improved by increasing generation size (k)[14]. But on the other hand, increasing generation size after some threshold, increases overhead. In G-by-G Network coding where generation size is k, sender S generates atleast k encoded packets from k input packets of one generation. Here each generation is encoded and decoded separately of other. In G-by-G Network coding losses are expensive as the partial reception of encoded packets of same generation means a complete loss of that generation. To overcome the problem of losses, redundant packets can be sent with generation. Redundant packets enhance the reliability of communication. But note that in G-by-G Network coding, extra packets sent with generation, protects that generation only.

2.3 Network coding with Multi Generation Mixing(MGM)

Network Coding with Multi-generation mixing(MGM) is a RLNC approach which improves the performance without increasing buffer size. In MGM mixing set of size m generations can be coded together. A new set of generation packet is mixed with previously transmitted generations. Results show that MGM reduces overhead for a recovery of packets. In MGM, N packets are grouped into generations where the size of each generation is k packets. Each generation is assigned a sequence number from 0 to N/k. In MGM generations are grouped into mixing sets where the size of mixing set is m generations. Each mixing set has an index M. Generation i belongs to mixing set with index M=i/m. Each generation in mixing set has a position index. Position index (l) of generation i in a mixing set of size m is i mod m. G-by-G Network coding is a special case of MGM where m=1. When node sends a packet belonging to generation i with position index l in mixing set, that node encode all packets that are associated with the generations of same mixing set and have the position indices less than or equal to l as shown in Figure 2.2[15]. Size of encoding vector depends on the number of packets encoded together at sender node. Packet in generation with position index l have the size of encoding vector is (l+1)k. So sender will generate (l+1)k independent packets. Computation overhead is incurred at in-

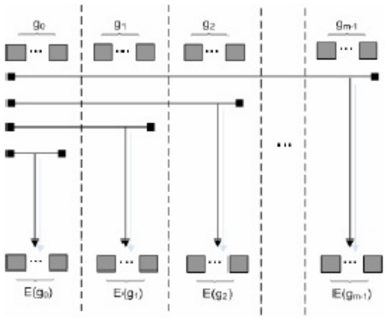

Figure 2.2: Network Coding with MGM, each generation is encoded with previous generations in mixing set.

termediate node to check the usefulness of received packets and at receiver node to decode received packets[16]. In g-by-g Network coding computation are performed on packets within the generation so it is fixed due to fixed generation size. But in MGM encoding/decoding is performed on packets belonging to at least one generation in mixing set and so computational overhead is not fixed. In MGM in case generation is unrecoverable due to the reception of insufficient encodings, it is still possible to recover that generation collectively as a subset of mixing set generations. Packets received with generation of higher position indices have information from generations of lower position indices and hence contribute in recovery of unrecovered generations of lower position indices in the same mixing set.

As shown in Table I[16] for mixing set size m=2, overall number of useful packets received of generations g0 and g1 is greater than 2k which is sufficient for collective decoding of two generations.

Mixing set size	Generation position index	Condition for generation gurenteed delivery
M=1	G0	ko^{+}
M=2	G0	ko^{+} ko^{-},$(ko+k1)^{+}$
	G1	ko^{+},$k1^{+}$ ko^{-},$(ko+k1)^{+}$

Table I: Guaranteed delivery conditions for the generations of size k within mixing sets of sizes m=1 & 2.

Enhancing reliability of communicating different groups of sender packets is QoS requirement of many applications where, there is systematic grouping of sender packets such that different groups have varying importance i.e. Scalable Video Coding (SVC), where video is encoded in layers base layer and one or more enhancement layers. MGM supports priority transmission by providing enhanced reliability for delivering different groups of sender packets. Due to the way, encoding/decoding is done in MGM; it provides different level of protection for mixing set generations. In other words different generations in mixing set can be considered as different layers of priority as shown in Figure 2.3[17]. Each layer has priority value depending on the generation's position index in a mixing set. Redundant encoded packets enhance

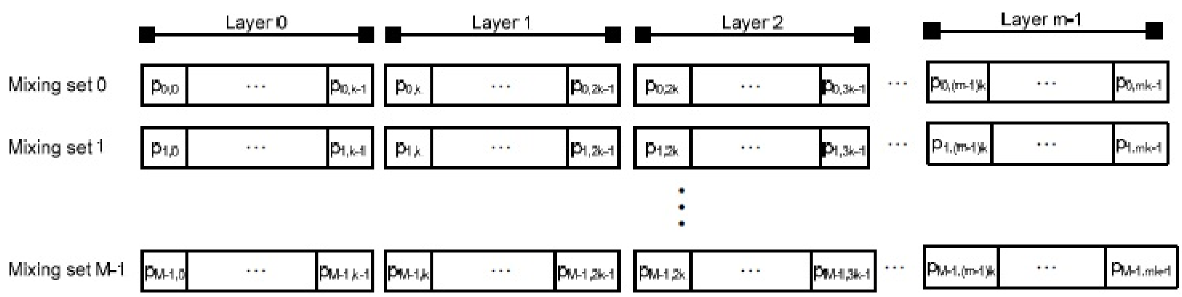

Figure 2.3: Generation's partitioning with MGM into different layers of priority. Mixing set size is m, generation size is k.

the reliability of communication. With MGM extra packets protects all generations with lower position indices. In MGM there are different options for sending extra packets. One option is distribute the packets over all generations of mixing set. Another option is to send extra encodings with the last generation of mixing set. So, extra encodings protects all mixing set generations. In Network Coding with MGM goal is to enhance decodable rates in situation where losses prevent efficient propagation of sender packets. MGM allows the cooperative decoding among the different generations of a mixing set which enhances decodablity. Results in Table II,III and IV[16] shows that MGM with redundancy transmitted with the last generation of the mixing set achieves best decodable rate than distributed redundancy. At the same time MGM with distributed redundancy achieves best decodable rate than G-by-G Network coding. It has been also observed that increasing the size of mixing set improves decodable rate because it increases number of opportunities where a generation can be decoded.

Generation Size(k)	Distributed Redundancy			Redundancy sent with last generation		
	G0,m=1	G0,m=2	G1,m=2	G0,m=1	g0,m=2	g1,m=2
10	89	95	85	88.4	92	91.5
20	92	97	90	91.5	95.2	95.2
30	94	98	93	93.8	97.2	97.2
40	95.1	99	94	95.2	98.5	98.5
50	96	99.1	95.9	96.2	99	99
Average decodable rate	93.22	97.62	91.58	93.02	96.38	96.28

Table II: Decodable rates (in percentage) achieved over different generation size of mixing sets of sizes m=1 and 2 when the redundancy is sent with each generation and redundancy is sent with last packet. Packet loss rate is 0.1.

As shown in Table II and III in the case when redundancy is sent with each generation and when m=2 or 3 highest decodable rate is achieved for first generation (i.e. g0) compare to others for any generation size. It can also be observed that decodable rate is decreasing from first generation to last generation for both m=2 and 3. The reason is first generation of mixing set is protected by the second and second is protected by third and so on. Last generation in the mixing set is not protected by any other mixing set generations. So last generation is delivered with lowest decodable. It can also be noted that the decodable rates achieved for the last generation of MGM, m=2 or 3 is close to that for the single generation of m=1 especially for larger generation size.

When redundancy is sent with the last generation in the mixing set, very close decodable rates are achieved for each generations compare to the case of distributed redundancy. At the same time the decodable rates achieved with MGM is higher than that achieved with traditional generation based Network Coding (m=1). By sending redundancy with the last generation in the mixing set, redundant packets protect all mixing set generations and hence the overall mixing set decodable rate is enhanced. There is an advantage in sending redundancy with the last generation in

Generation size(k)	Distributed Redundancy				Redundancy sent with last generation			
	G0 m=0	G0 m=3	G1 m=3	G2 m=3	G0 m=1	g0 m=3	g1 m=3	g2 m=3
10	89	97.5	89.9	83.5	88.9	94.1	92.8	93.9
20	91.9	99	94.9	89	91.8	97.3	97.3	97.3
30	93.9	99.5	97.1	92.5	93.9	98.8	98.8	98.8
40	95.7	99.9	98.2	94.5	95.2	99.5	99.5	99.5
50	96.1	100	99	96	96.5	99.8	99.8	99.8
Average decodable rate	93.32	99.18	95.82	91.1	93.26	97.9	97.64	97.86

Table III: Decodable rates (in percentage) achieved over different generation size of mixing sets of sizes m=1 and 3 when the redundancy is sent with each generation and redundancy is sent with last packet. Packet loss rate is 0.1.

the mixing set but it will increase the delay for generation recovery because an unrecovered generation with lower position index needs to wait for sufficient number of encodings that most likely will be received with the last generation in the mixing set. Table II and III also shows the average decodable rates for generations in mixing sets of sizes m=1, 2 and 3. There is an improvement in decodable rate when increasing the mixing set size for both the case distributed redundancy and redundancy sent with last generation.

As shown in Table IV for different packet loss rate, achievable average decodable rate is higher for MGM compared to G-by-G Network Coding (m=1). For both scenario distribute redundancy and redundancy sent with last generation average decodable rate is close to each other for m=2 and 3. But compare to distributed redundancy scenario, average decodable rate is high for the scenario where redundancy is sent with last generation.

MGM can be applied in networks communicating scalable video contents[18]. By applying MGM on scalable video the goal is to prioritize the transmission of video layers to improve decodable rates and hence enhance recovered video quality.

Packet loss rate		Distributed Redundancy		Redundancy sent with last generation	
	m=1	m=2	m=3	m=2	m=3
0.08	97	98	97.5	99	99.5
0.1	93	95	95.1	96	98
0.12	83	84	84.9	86	88
0.14	77.5	78	78.1	80	83

Table IV: Average decodable rates (in percentage) achieved over different packet loss rates. Mixing sets sizes m=1,2 and 3.

2.4 Galois Field Arithmetic

The operators in conventional arithmetic (divide, multiply, add, subtract etc.) deal with infinite or unbounded numbers. Computer hardware uses binary arithmetic where integer results are stored in registers, bytes or words of a finite size. Dealing with the overflows caused by very large calculations is a considerable problem if the result is to be used in bit by bit error detection.

Reed Solomon codes are created by the manipulation of finite group of numbers called a 'Galois Field'. GF(256) is a field consisting of the every integer in the range 0 to 255 arranged in a particular order. If you could devise an arithmetic where the result of each operation produces another number in the field the overflow issues could be avoided. The generation (ordering) of the field is key. e.g. a simple monotonic series from 0 to 255 is a finite field but modulo 255 arithmetic fails commutative tests i.e. certain operations will not reverse.

A Galois field gf(p) is the element 0 followed by the (p-1)succeeding powers of α : 0,1,α, α^1, α^2,...,α^{p-1}

Extending the gf(2) field used in binary arithmetic (and CRC calculation) to 256 elements that fit nicely in a computer byte: gf(2^8) = gf(256). Substituting the primitive element $\alpha = 2$ in the galois field it becomes 0, 1, 2, 4, 8, 16, and so on. This series is straightforward until elements greater than 127 are created. Doubling element values 128, 129, ..., 254 will violate the range by producing a result greater than 255. Some way must be devised to "fold" the results back into the finite field range without duplicating existing elements (this lets modulo 255 arithmetic out). This requires an irreducible primitive polynomial. "Irreducible" means it cannot be factored into smaller polynomials over the field. In our implementation irreducible polynomial 285 is used.

The Galois arithmetic operations for GF(256) must be implemented as follows:

The **ADDITION** and **SUBTRACTION** of two numbers are both implemented by the bitwise exclusive-or of the two numbers as follows.

```
function addition(a,b)
begin
  result:=a xor b;
end;

function subtraction(a,b)
begin
  result:=a xor b;
end;
```

The **Multiplication** of two numbers are implemented by modulo operation of the two numbers as follows.

```
function multiplication(a,b)
begin
  result:={a(x) * b(x)} mod p(x);
end;
```

The **Division** of two numbers is a multiplication of dividend and multiplicative inverse of divisor. Ex. $5 \div 2 = 5 *$ multiplicative inverse(2)

```
function multi_inverse(a)
begin
  result:=b such that [{a(x) * b(x)} mod p(x)] = 1;
end;
```

2.5 Anycasting in wireless networks

Anycast is a service that allows a node to send a message to at least one, and preferably only one, of the members in a group. The idea behind anycast is that a client wants to send packets to any one of several possible servers offering a particular service or application but does not really care any specific one. Anycast can be used to implement resource discovery mechanisms which are powerful buildings block for many distributed systems, including file sharing etc. It can also be used to implement load balancing, robustness against breakdown.

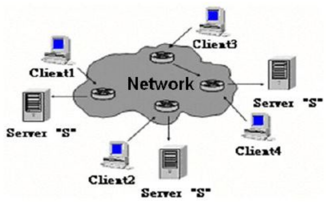

Figure 2.4: Anycast Packet Flow
[21]

when a sender node searches for the receiver in ad-hoc wireless networks, it relies on a reliable broadcast mechanism to discover the services. Broadcast is very simple and has the benefit of reliability but it may produce a high overhead in the network due to the broadcast and reply storm problems. Besides, when the number of nodes or services increases in the networks, the clients need to flood the request packets to the service providers via a longer path. Thus, the possibility of packet collision increases and the delivery ratio decreases. In order to solve above problems and to reduce the amount of request/ reply packets, anycasting scheme for ad-hoc wireless networks can be used. In this scheme when a client node sends out a request message, the client is responded by its nearest or best server. Therefore, the anycasting scheme is able to reduce the control overhead and is suitable for large-scale ad-hoc wireless networks.

Types of Anycast According to the view point of range anycast can be implemented by two ways: Subnet Anycast and Global Anycast. In subnet Anycast all anycast responders are in the same subnet as shown in figure 2.5[22]. In the subnet anycast, the anycast packet is transmitted to edge router R1 by the existing unicast routing, and the edge router selects the correspondent anycast responder. The edge router can use a Neighbor Discovery mechanism to select the correspondent anycast responder. In Global Anycast, anycast responders are widely distributed across the

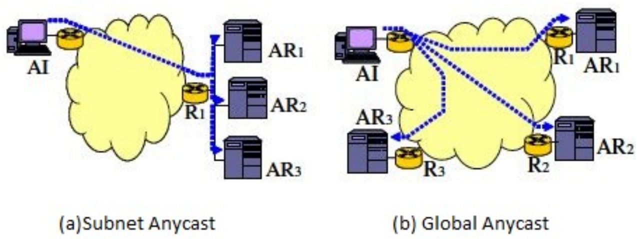

Figure 2.5: Types of anycast

Internet as shown in figure 2.5[22]. A global anycast is more difficult to achieve than the subnet anycast because the anycast responders are not in a range that single router can manage. However, a global anycast can provide a wider anycast service.

According to the view point of the layer anycast can be implemented at three layers: MAC layer Anycast[23], Network layer Anycast (IP Anycast) [24] and Application layer Anycast[25]. When anycast is realized in the network layer, there is the advantage that the anycast functions can be added to existing applications without editing the source codes so in this thesis anycasting is going to be implemented at network layer.

Anycast is a service that allows a node to send a message to at least one, and preferably only one, of the members in a group. The idea behind anycast is that a client wants to send packets to any one of several possible servers offering a particular service or application but does not really care any specific one. Anycast can be used to implement resource discovery mechanisms which are powerful buildings block for many distributed systems, including file sharing etc.

Due to the unpredictability of network connectivity and delay, and limited buffer, anycast in VANETs is a quite unique and challenging problem. It requires both re-definition of anycast semantics and new routing algorithms. In anycast, during the routing, both the path to a destination group member and the destination of the anycast message can be changed dynamically according to current mobile device movement situation, correspondingly enlarging the delivery delay.

But these approaches cannot be applicable to VANETs since for anycast routing it cannot assume the connectivity is guaranteed, and the uncertainty of both the path to a destination group member and the destination of the anycast message during the routing makes the problem more challenging. One of the challenges in designing an anycast routing protocol is to maintain the group membership efficiently. Due to the long delivery delay in VANETs, group membership may already change during the delivery of a message, introducing ambiguity in anycast semantics. For example, if a message is intended for group G with member a, b, and c when it is sent, when it arrives, the membership of group G may change to a, d, and e. The anycast routing scheme has to deal with such membership dynamics.

Chapter 3

Problem Definition

To develop a network coding based multi-copy protocol to anycast information in VANET.

3.1 Assumptions

- No prior knowledge about the network is available to the protocol.
- The network is Vehicular ad hoc Network. Nodes(vehicles) in the network move randomly and there is no correlation in movement. So heuristic approach about events is not representative of future events.
- The network is not overlay VANET, i.e., there is no underlying network. So, the protocol does not have any information which is available to overlay VANET through underlying network.

3.2 Objective

- To achieve better 'percentile delay' (time taken to deliver given percentage of total packets) and throughput than network coding based protocol and protocol using conventional scheme.

3.3 Intended Outcomes

- Protocol design.
- Simulation results comparing the proposed protocol with protocol using network coding and protocol using conventional scheme.

Chapter 4

The Protocol:Anycasting in VANET

4.1 Overview

- In VANET, end-end connectivity between vehicle is not present most of the time. It is due to low density of the vehicles or due to speeds of the vehicles.In such a network, conventional approach of finding route towards destination before data transmission is not feasible. So, in our protocol, data transfer takes place whenever two nodes come into communication range of each other. So we consider it as an **Opportunistic Network**.

- In VANET, no prior information about the network topology or connection pattern is available, as in mobile ad hoc networks most of the time. Single copy schemes generally rely heavily on such information and these schemes have very high delivery delay and low delivery ratio. Further, Multi-copy schemes are very robust. So, we choose our protocol to be a **Multi-copy protocol**.

- In conventional routing schemes for VANET, whenever a transmission opportunity arrives, ideally, a node should forward packets such that the destination node gets all the required packets without getting any redundant packet. On the other hand, in the network coding based protocol, a node can transmit any of the coded packets since all of them can contribute the same to the eventual delivery of all data packets to the destination with high probability. So, our protocol uses **Network Coding with Multi Generation Mixing** to exploit these. For mixing packets, linear coding is sufficient to achieve nearly optimal performance. The coefficients for encoding can be decided by some central entity or can be generated independently by each node. As, we want our scheme to be completely distributed because of the obvious advantages, we use the later type of scheme. Random Linear Coding is one such scheme in which coefficients are randomly decided by each node and it does not require deterministic codes or detail coordination between nodes which increases overhead. RLNC does require coefficients to be carried with the packet but it does not significantly add to overhead.

- Our protocol can run on top of the semantic models of Anycst in VANET and it will not hamper the performance of the protocol as the protocol's forwarding policy is independent of who the intended destination is. Rather, in the situations where intended destinations change frequently, the proposed protocol has an edge over the schemes which try to find out routes to destinations because such schemes will need frequent updating of the routes while the proposed protocol will not require such updating.

- For efficient buffer usage, copies of already delivered packets have to be purged from the buffer of the node in the network. In our protocol, anti-packet is sent only when sufficient number of encoded packets of particular mixing set are received. But, generally, as the size of anti-packets is very small compared to data packets amount of overhead is still very low.

- There are the two configurable parameters in our protocol:

  – Generation Size : It denotes the size of the generation. For the reasons explained earlier, packets are grouped into generations.

- Mixing Set Size : It denotes the size of the mixing set. Various generations are grouped into mixing set and only the packets of L generation can be mixed with all packets of previous generations of same mixing set.

- The network parameter of interest is Meeting rate. Meeting rate of the network is the average number of times a node meets with any other node in the network per second. It depends on the area within which the network nodes move, their velocities and communication range and the mobility pattern.

- As mentioned in the objectives, the performance parameters of interest are percentile delay and delivery ratio.

4.2 Protocol Description

In this section, we describe working of our protocol. Data packets are grouped into generations and generations are grouped into mixing set. Nodes store independent packets along with their coefficients according to RLNC scheme.
Below is the Pseudocode for Anycasting Protocol:

```
sendproc(packet)
    - call encoding(packet)
    for each encoded packets generated by encoding()
        - create a packet
        - set des_id to Grp_id (G(A))
        - send packet when neighbor comes in contact
    end for
end sendproc
```

```
recvproc(packet)
     if received packet is redundant
         discard the packet
     else
         if des_id of incoming packet belongs to G(A)
            - store packet in buffer
            - call calc_rank(M)
            - call decoding(packet) if rank is sufficient
            - create anti packet with des_id=src_id of
            received packet and Grp_id=G(A)
            - call sendproc(anti packet)
         else
            - call forwardproc(packet)
         end if
     end if
end recvproc
```

Main components of our protocol used in above protocol are explained below.

- Every node in the network transmits 'HELLO' packet at fixed interval called as **'HELLO Timer'**. Whenever a node receives HELLO packet from another node, it adds that node into its neighbor list if that node is not in the list already. If any node does not receive any HELLO packet for the interval equal to three times 'HELLO Timer' from its neighbor node, that node is purged from the neighbor list. This interval is called as **'Neighbor Purge Timer'**.

- The initiator node sends encoded data packets to the neighbor till it is within communication range. The Mixing Set of which next encoded packet is to be forwarded, along with its coefficients, is decided based on following forwarding policy:

    - All the mixing sets are grouped into three classes. First class consists of the mixing sets for which the node is the source. Second class is of the mixing sets for which the neighbor is the destination node. Finally, the remaining mixing sets are in the third class.

- When the sufficient number of packets of particular mixing set or particular generation are received at the destination node, it decodes original packets.

Chapter 5

Implementation

In this chapter implementation details are given for both G-By-G Network Coding and Network Coding with MGM.

Implementaion focuses on how packets are devided into mixing set and generations, mixing of the packets and decoding of the packets if sufficient packets are received and how operations are performed over finite field during mixing and decoding of the packets. To achieve these four components have been coded as mentioned below.

- Encoding
- Decoding
- Rank
- Gallois Field opertaion's library

5.1 G-By-G Network Coding

The functionality provided by the implementation is represented using following example.

Let node has received the packets to send to destination. Packets are 1,2,3,4. Assume that the generation size is 4. So all four packets are grouped into one generation. Now encoding function will mix all this four packets by using some randomly generated vectors and make four independent copies of mixed packets as shown below. Vector of information is stored in M and generated vectors are stored in G.

$$M = \begin{bmatrix} 1 \\ 2 \\ 3 \\ 4 \end{bmatrix} \quad G = \begin{pmatrix} 83 & 91 & 151 & 109 \\ 14 & 203 & 121 & 177 \\ 246 & 246 & 63 & 243 \\ 0 & 234 & 173 & 202 \end{pmatrix}$$

Mixed packets are derived from G and M as shown below and stored in E.

$$E = \begin{bmatrix} 1 \\ 2 \\ 3 \\ 4 \end{bmatrix} \begin{pmatrix} 83 & 91 & 151 & 109 \\ 14 & 203 & 121 & 177 \\ 246 & 246 & 63 & 243 \\ 0 & 234 & 173 & 202 \end{pmatrix} = \begin{bmatrix} 232 \\ 240 \\ 173 \\ 44 \end{bmatrix}$$

Now E and G reached to intermediate node. Intermediate node will again encode these received encoded packets with its own coefficient and generate new coefficient and new encoded packets as shown below.

$$\begin{bmatrix} 232 \\ 240 \\ 173 \\ 44 \end{bmatrix} \begin{pmatrix} 76 & 201 & 166 & 110 \\ 35 & 109 & 224 & 50 \\ 87 & 75 & 12 & 168 \\ 239 & 125 & 226 & 165 \end{pmatrix} = \begin{bmatrix} 38 \\ 71 \\ 140 \\ 39 \end{bmatrix}$$

Effective coefficients are determined from received coefficients and newly derived coefficients as shown below.

$$\begin{pmatrix} 76 & 201 & 166 & 110 \\ 35 & 109 & 224 & 50 \\ 87 & 75 & 12 & 168 \\ 239 & 125 & 226 & 165 \end{pmatrix} \begin{pmatrix} 83 & 91 & 151 & 109 \\ 14 & 203 & 121 & 177 \\ 246 & 246 & 63 & 243 \\ 0 & 234 & 173 & 202 \end{pmatrix} = \begin{pmatrix} 51 & 162 & 108 & 62 \\ 13 & 201 & 19 & 60 \\ 93 & 195 & 42 & 13 \\ 100 & 216 & 0 & 181 \end{pmatrix}$$

At destination node effective coefficients and encoded packets are received from intermediate node as shown below.

$$\begin{bmatrix} x \\ y \\ z \\ w \end{bmatrix} \begin{pmatrix} 51 & 162 & 108 & 62 \\ 13 & 201 & 19 & 60 \\ 93 & 195 & 42 & 13 \\ 100 & 216 & 0 & 181 \end{pmatrix} = \begin{bmatrix} 38 \\ 71 \\ 140 \\ 39 \end{bmatrix}$$

Above is system of four linear equation. So from it native packets i.e. x,y,z,w can be decoded if sufficient rank is achieved for received packets.

5.2 Network Coding with MGM

I take same packets as shown in previous section i.e. 1,2 3,4. Now with MGM packets are divided into generations and generations are divided into mixing sets. Assume generation size is 2 and mixing set size is 2. So two generations are derived per mixing set and one mixing set is derived for these four packets.

Mixing set 1

Generation 1 : 1,2

Generation 2 : 3,4

In mixing set1 for generation 1 mixing of packets is done as follows.

$$\begin{bmatrix} 1 \\ 2 \end{bmatrix} \begin{pmatrix} 254 & 42 \\ 168 & 28 \end{pmatrix} = \begin{bmatrix} 170 \\ 144 \end{bmatrix}$$

In mixing set1 for generation 2 mixing of packets is done as follows.

$$\begin{bmatrix} 1 \\ 2 \\ 3 \\ 4 \end{bmatrix} \begin{pmatrix} 83 & 91 & 151 & 109 \\ 14 & 203 & 121 & 177 \\ 246 & 246 & 63 & 243 \\ 0 & 234 & 173 & 202 \end{pmatrix} = \begin{bmatrix} 232 \\ 240 \\ 173 \\ 44 \end{bmatrix}$$

Now at receiver for mixing set 1, out of 6 assume 4 encoded packets are received with their coefficient from both the generations i.e. 170,232,173 and 44. So it becomes system of four linear equations as shown below.

$$\begin{bmatrix} x \\ y \\ z \\ w \end{bmatrix} \begin{pmatrix} 254 & 42 & 0 & 0 \\ 83 & 91 & 151 & 109 \\ 246 & 246 & 63 & 243 \\ 0 & 234 & 173 & 202 \end{pmatrix} = \begin{bmatrix} 232 \\ 240 \\ 173 \\ 44 \end{bmatrix}$$

So from it native packets can be decoded by any one of the method given in next section if sufficient rank is achieved i.e. all four packets are independent of each other.

5.3 Observations

In the case of MGM from four linear equations native packets of both the generations can be determined. Here in same mixing set generation with lower position index is protected by generation with higher position index.

While in the case of G-By-G Network Coding if sufficient packets of particular generations are not received then whole generation should be retransmitted.

For both G-By-G and MGM, to decode the encoded packets and to determine native packets four techniques have been reviewed in terms of their time complexity for vector of size 10x10. Name of these techniques with observations are as below.

- **Cramer's rule** This method is quite general but involves a lot of labour when the number of equations exceeds 4. For 10x10 system, it requires about 70,000,000 multiplications. So cramer's rule is not at all suitable for large system.

- **Matrix inversion method** Although this method is quite general, yet it is not suitable for large system since the evaluation of A^{-1} by cofactor becomes very cumbersome.
- **Gauss elimination method** This method can be preferred for large system as for system of 10 equations, it requires about 333 multiplications.
- **Guass-Jordan method** For a system of 10 equations, this method requires 500 multiplications. Though this method appears to be easier but requires 50% more operations than Gauss elimination method.
- **Factorization method or Dolittle's method** This method is superior to Gauss elimination method and is often used for the solution of linear systems and for finding the rank of matrix. For a system of 10 equations, it requires about 110 multiplications. So in implementation of both MGM and G-By-G Network coding Factorization method is considered for both to determine the rank of matrix and to determine native packets.

Chapter 6

Results and Discussion

We have simulated the proposed protocol in NS2 simulator. The network contains 20 to 40 wireless nodes which move Randomly. The average speed of a node is 30 m/s. The communication range of a node is 100 m. Meeting rate is changed by varying field area of the network. There are 3 to 11 randomly placed sink nodes. Source node is sending packets to one of the sink node and sink node is sending anti packet of each received encoded packets. For network coding with multi generation mixing, packets are grouped in the generations and various generations are grouped into mixing sets. For encoding of packets coefficients are chosen randomly and addition and multiplication operations are done over the finite field F_{2^8}.

6.1 Simulation Results

Our performance parameter of interest are packet delivery ratio, delivery delay and throughput. The protocol parameters are Mixing Set Size (MSS) and Generation Size (GenSize). The network parameter of interest is meeting rate. We compare our protocol with the protocol using network coding and the protocol using conventional scheme.

We have simulated the protocol by setting MSS=1 and GenSize = 1 in our protocol, which effectively disables the network coding, we will call it ‘conventional scheme’, by setting MSS=1 and varying GenSize, which disables network coding with multi generation mixing, we will call it 'network coding'. We also compare packet delivery ratio, delivery delay and throughput of conventional scheme, protocol using network coding and our protocol with different meeting rate.

Once average speed of the nodes in the network achieves steady state, source node generate given number of data packets which are grouped into generations and generations are grouped into mixing sets. Number of mixing set and number of generations in each mixing set depend on size of MSS and GenSize. We run the simulation to measure block delivery delay for sufficient time period such that all the mixing sets are received by intended destination node from group of sink nodes. Anti-packets are generated for each received encoded packet.

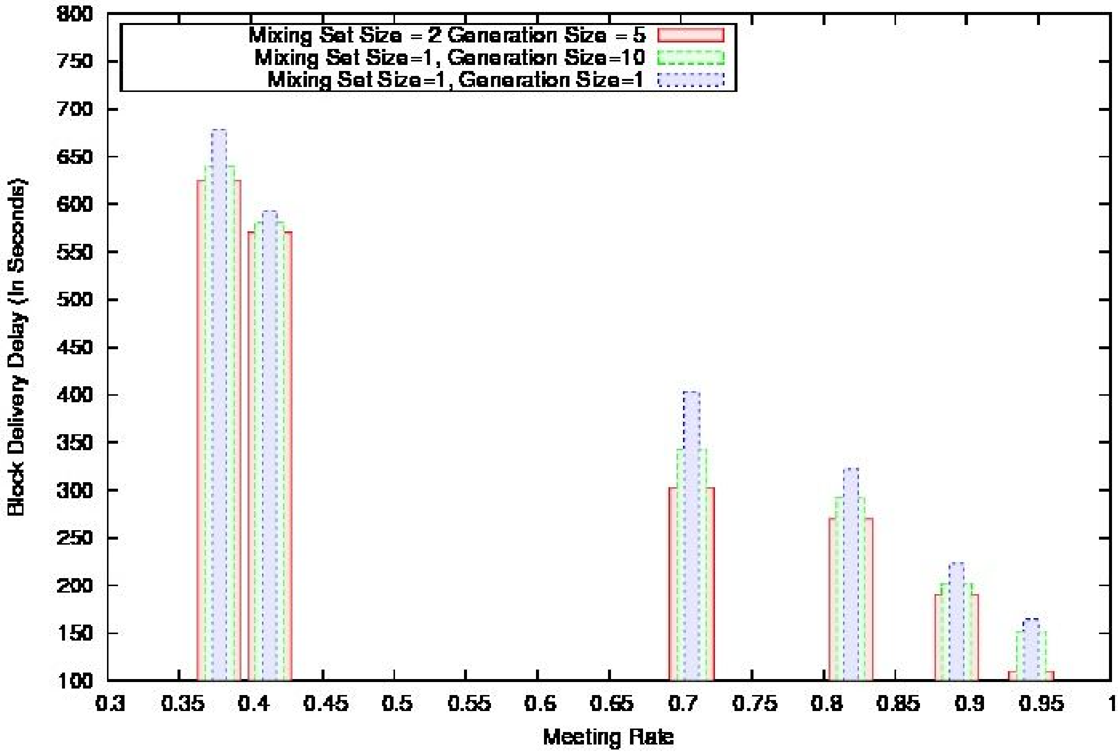

Figure 6.1: Block Delivery Delay v/s Meeting Rate

We observe that for lower meeting rate the chances of meeting a node with other node decrease and hence number of packets to be forwarded by a node is less. As shown in figure 6.1 as meeting rate increase delay to deliver all sufficient packets of particular mixing set is decrease for all three cases. We also compare the delay of our protocol when using network coding with MGM with our protocol when using network coding and conventional scheme. As shown in figure 6.1 network coding with MGM is taking less delay to deliver packets of particular mixing set compare to other two schemes.

Figure 6.2 shows delay to deliver sufficient packets of particular mixing set with respect to Mixing Set Size for different Generation Size and meeting rate = 0.708. As

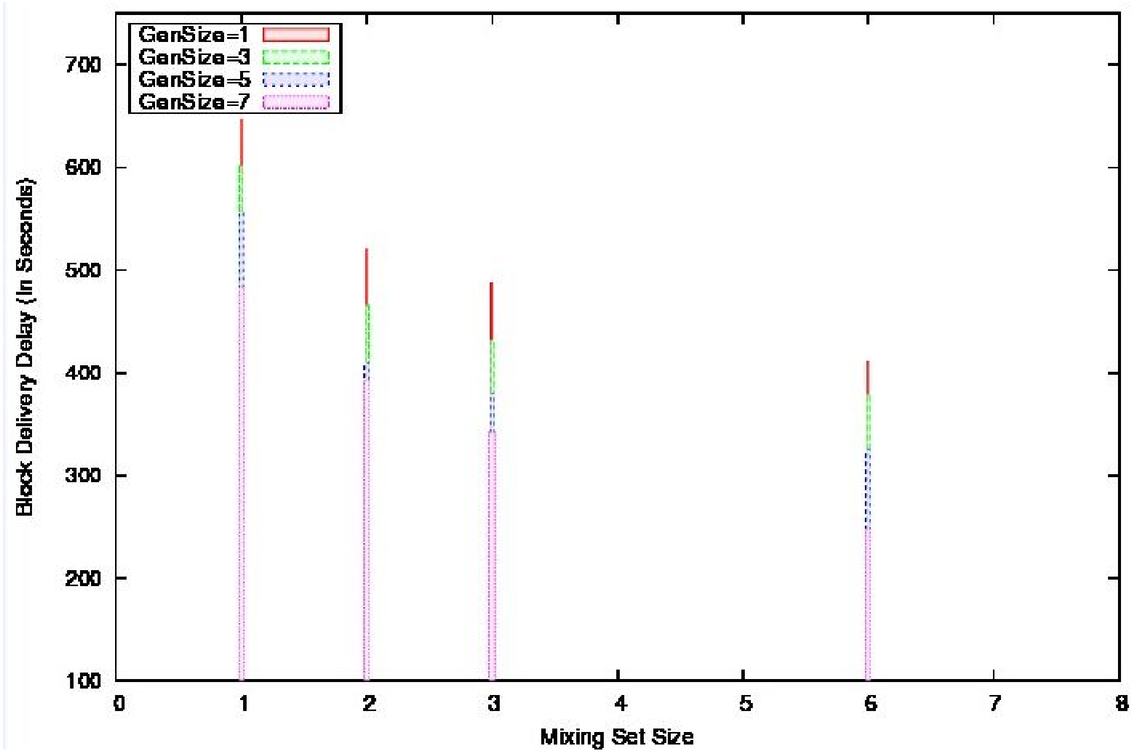

Figure 6.2: Block Delivery Delay v/s Mixing Set Size

evident from figure, with increase in mixing set size delay decreases significantly for different generation size. The reason behind this is as mixing set size is increase with increase in generation size more number of packets are mixed with each other as we have discussed earlier, and probability of delivering some of that packets increased. Compare to other two schemes block delivery delay in protocol with network coding with MGM is less.

We also vary the number of sink nodes from 3 to 11 and placed them randomly. We measure the packet delivery ratio for different number of sink nodes for higher meeting rate as shown in figure 6.3. As the number of sink nodes increase probability of delivering packets to any one node from all sink node increase. After delivery destination node can collect the encoded packets from any other node and hence destination node is receiving more number of packets. So packet delivery ratio increased for higher number of nodes. Protocol using network coding with MGM outperforms other two schemes.

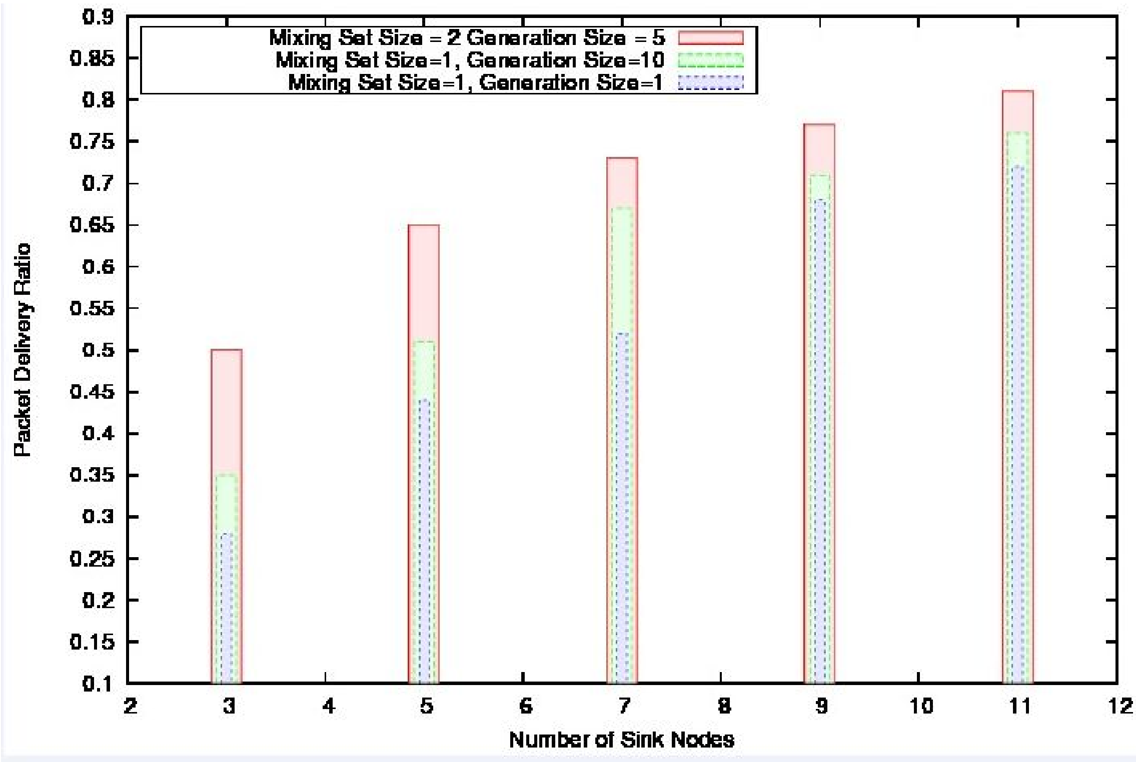

Figure 6.3: Packet delivery ratio v/s Number of sink nodes

As previously we have discussed as meeting rate increase probability of delivering more number packets increase. As shown in graph in figure 6.4 as meeting rate is increased average number of packets are delivered is more in all three cases. But compare to protocol using conventional scheme and protocol using network coding gain in delivery is more in the case of protocol using network coding with MGM.

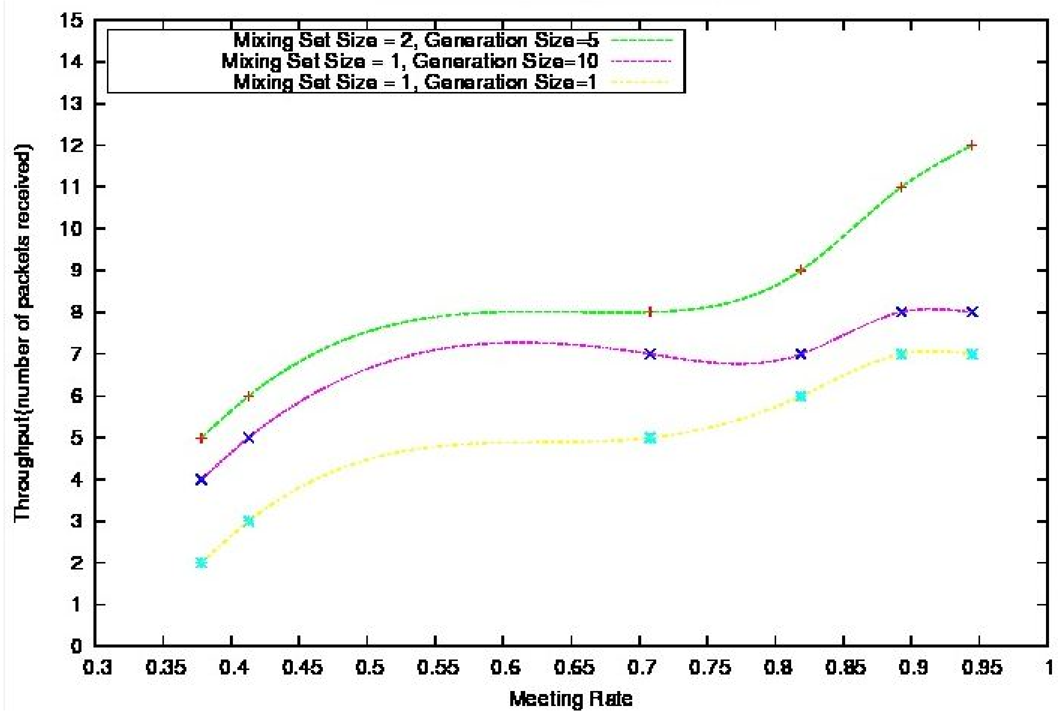

Figure 6.4: Throughput v/s meeting rate

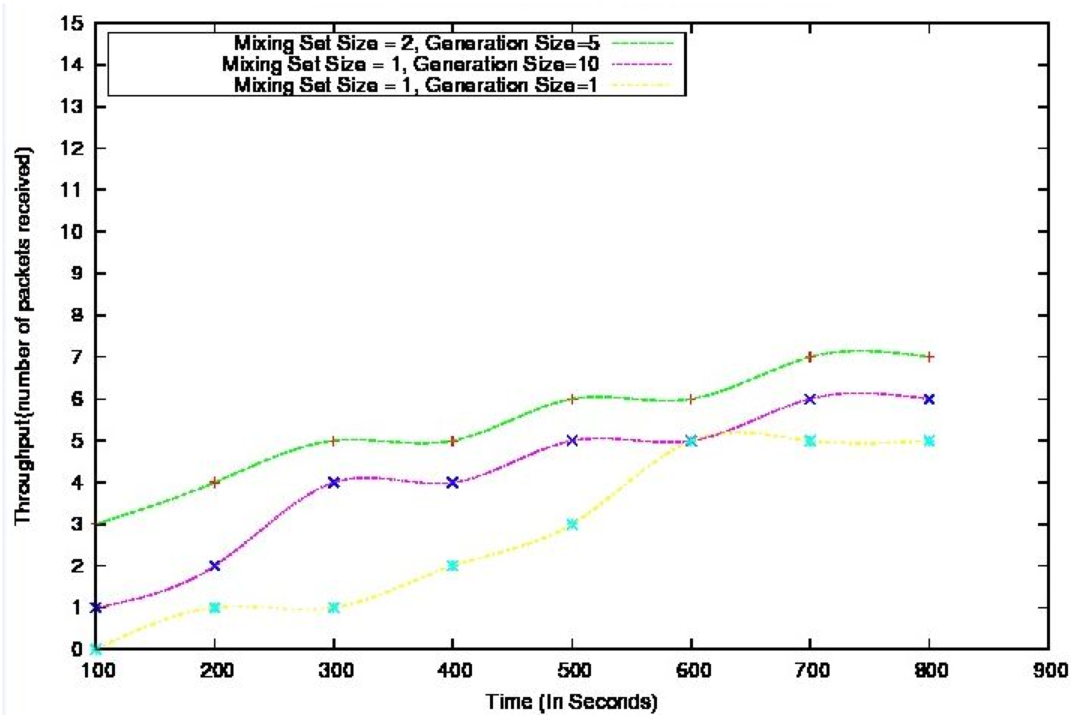

Figure 6.5: Throughput v/s Time

As shown in figure 6.5, once the network reaches steady state, gain in successful delivery of packets of our protocol compared to other two schemes is higher. Please note that the result is for meeting rate = 0.945. The reason for this behavior is as follows:As in the case of network coding with MGM, packets of current generation are mixed with the packets of all previous generation of particular mixing set. So as we have previously discusses average decidable rate is high in the case of protocol using network coding with MGM compare to protocol using other two schemes. Also in network coding the successful reception of information does not depend on receiving specific packets but on receiving sufficient number of independent packets. Probability of the same is high in the case of protocol using network coding with MGM.

Chapter 7

Conclusion

Due to dynamic network topology and frequent disconnected networks, VANET requires different routing strategy than other Ad-Hoc networks. Many VANET applications need anycast service. To improve reliability without impacting performance, we used network coding with multi generation mixing.
Major contribution of our work are as follows:

- Implemented a basic framework of G-by-G Network coding and Network Coding with MGM.
- Protocol design to anycast in VANET with network coding with multi generation mixing.
- comparison of conventional scheme and the scheme using network coding with our protocol through simulation.

Summary of major findings of our work are as under:

- Simulation results prove that the protocol reduces delay to deliver sufficient packets required to decode the mixing set or particular generation in mixing set.
- The protocol outperforms conventional scheme and scheme using network coding for delivery ratio greater than 10% to 15% and 20% to 25% respectively.

- Improvement of the protocol over conventional scheme and scheme using network coding has been observed for delivery ratio as number of sink nodes increase.
- Compare to conventional scheme and scheme using network coding throughput is high in our protocol.

7.1 Future Work

We found protocol using network coding with multi generation mixing outperforms conventional scheme and scheme using network coding in terms of delivery delay, delivery ratio and throughput. In the case of finite buffer to utilize buffer efficiently we intend to introduce purging scheme. To improve chance of delivery, we intend to use multi copy scheme but in this case to control the number of copies and to improve efficiency in terms of buffer and bandwidth (energy) usage we use mechanism like binary Spray and Wait as suggested in[27]. We intend to find optimal mixing set size and generation size as a function of meeting rate, delivery delay and delivery ratio empirically and analytically.

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