MPEG-4 Streaming using Network Coding in Wireless Network

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Major Project

Submitted in partial fulfillment of the requirements

For the degree of

Master of Technology in Computer Science and Engineering

By

Rahul Shrimali
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May 2013

Declaration

I, Rahul Shrimali, Reg.No. 10MCES06 , give undertaking that the Major Project entitled "MPEG-4 Streaming using Network Coding in Wireless Network" 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.

Rahul Shrimali (10MCES06)

Date: 18-05-2013

Place: Institute of Technology, Nirma University

Certificate

This is to certify that the Major Project entitled "MPEG-4 Streaming using Network Coding in Wireless Network" submitted by Rahul Shrimali (10MCES06), 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

Transmitting delay-sensitive video streaming over wireless network is becoming increasingly popular. However, the transmission of real-time video streaming is very challenging because of the time-varying and unreliable wireless channels, video content characteristics, limited bandwidth, dynamic topology, heterogeneous and distributed environment and high packet loss rate because of wireless interference and channel fading. Because of such open issues present in any wireless network, it is very difficult to meet the requirement of audio and video streaming applications such as low delay, low packet loss, jitter control etc. In such scenarios of networks where real time data is being streamed from server node to client node, Network Coding and its variants can be used by such nodes to meet different requirements. Network coding simply allows to change the role of such nodes from traditional routing or store and forward to encode the data packets. Encoding process includes mathematical operations on data packets.This thesis work has considered number of theoretical and practical scenarios where network coding or it's variant applied on multimedia traffic with the aim to improve performance and to provide protection against packet losses. This thesis work has mainly focused on the performance enhancement of MPEG-4 traffic over wireless network using Random Linear Network Coding with Multi Generation Mixing (MGM). Using Multi Generation Mixing, packets of greater importance has got more protection, less loss, more reconstruction and recovery of real time data.This thesis work also intended to generate real time MPEG4 traffic (I,B,P frames) using Evalvid utility to avoid the use of video traffic model. Results of simulation shows increasing Packet Delivery Ratio and decreasing jitter during Mpeg4 streaming.

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- Rahul B. Shrimali(10MCES06)

Abbreviations

NC .... Network Coding
LNC .... Linear Network Coding
RLNC .... Random Linear Network Coding
PNCO .... Partial Network coding with Opportunistic Routing
ONC .... Opportunistic Network Coding
CNC .... Cooperative Network Coding
G-by-G .... Generation by Generation Network coding
UNC .... Unstructured Network Coding
SNC .... Structured Network Coding
JPEG .... Joint Photography Expert Group
MPEG .... Moving Picture Experts Group
GoP .... Group of Pictures
VOP .... Video Object Plane
AF .... Amplify-and-Forward
DF .... Decode-and-Forward
ONC .... Opportunistic Network Coding
MGM .... Multi Generation Mixing
SVC .... Scalable Video Coding
GF .... Galois Field
I frame .... Intra coded frame
B frame .... Bidirectional frame
P frame .... Predictive frames

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Chapter 1

Introduction

General Overview

In any environment (wired and wireless), the transmission of the real time audio and video traffic using multimedia streaming software for VOIP, Video conferencing etc is too much complex and time consuming. In all Wireless Networks, the available channel capacity fails to meet the requirements of multimedia transmission, which results in packets loss, delay and degrade the quality of transmitted multimedia data. Running such types of applications on unguided environment is always bandwidth hungry and prone to loss of data because of 2Ls. First is Lossy Environment (Wireless Environment), second is Lossy Compression (MPEG-4). In addition to that, it is also necessary to have successful transmission of all desired data to target, which is not possible all the time because of 2Ls as discussed above. To overcome with these 2Ls and to enhance the performance of multimedia transmission, Network Coding is the best solution. In Network Coding technique, it allows a node to combine data (packets) received from different input links, encode received data and send encoded data on its output links (instead of just storing and forwarding). In this thesis work, a variant of Network Coding that is Random Linear Network Coding (RLNC) with Multi Generation Mixing (MGM) is employed for MPEG-4 video traffic o achieve performance improvement.

1.1 Project Definition

To enhance performance during MPEG-4 Streaming using Network Coding in Wireless Network

1.1.1 Objective of the Work

To enhance the performance of MPEG-4 streaming in wireless networks using Generation by Generation and Multi Generation Mixing (MGM) based network coding.

1.1.2 Expected Outcome of the Work

MPEG-4 Streaming with G by G and MGM based network coding will outperform in terms of jitter control, increased packet delivery ratio, decreased packet loss.

1.2 Scope of the Work

In wireless networks, the available bandwidth is very limited which fails to meet the requirements of MPEG-4 streaming. This results in to packets loss, delay and degrade the quality of transmitted multimedia data. Survey and results in this area shows that network coding can improve reliability, per-packet loss ratio with substantial decrease in per-packet delay in wireless network. In addition to that, it is very necessary to perform reconstruction of audio/video packets in multi hop communication environment to increase the quality of streaming.
So, this work is intended to improve performance during MPEG-4 streaming for any wireless network.

1.3 Motivation of the Work

Multimedia traffic is always compressed in MPEG 4 format contain frames such as I , B and P, where I-Frame contains data based on which it is referenced by P Frames (gen- really more than one) and B Frames (generally more than one).So, considering the importance of I-Frame in multimedia traffic, MGM based Network Coding can be applied to provide protection to packets of I Frame.

1.4 Thesis Outline

The rest of the thesis is organized as follow:

In **Chapter 2**, *Literature Survey*, describes Network Coding, RLNC- variant of Network Coding, Survey on different types of Network Coding schemes applied over Multimedia traffic,Concept of Single Generation and Multi Generation Mixing of packets.

In **Chapter 3**, *Survey on Multimedia Systems*, describes various compression standards,various MPEG frame formats, concept of GoPs and VOP etc.

In **Chapter 4**, *Problem Identification and Proposed Solution* describes the problem identified after survey and Solution for identified problem.

In **Chapter 5**, *Implementation strategy*, describe various steps to implement proposed schemes, Real Time Video traffic generation using Evalvid utility.

In **Chapter 6**, *Simulation system* ,describes Network Simulator 2.

In **Chapter 7**, *Results and Analysis*, shows results of simulations and findings.

In **Chapter 8**, *Conclusion and Future work*, describes conclusion of this work and future work.

In **Chapter 9**, *List of Publications*, List of Publications.

Chapter 2

Literature Survey

2.1 Introduction to Network Coding

2.1.1 Network Coding

Network Coding (NC) was first introduced in 1999 by R. W. Yeung and Z. Zhang as an alternative to routing. Serious works on using network coding began from 2003. Video frames can be divided into many small packets, each of them may route from different path to the destination. As a result, an intermediate node can combine received packets and apply simple XoR operation on them. This is shown in Fig. 2.1

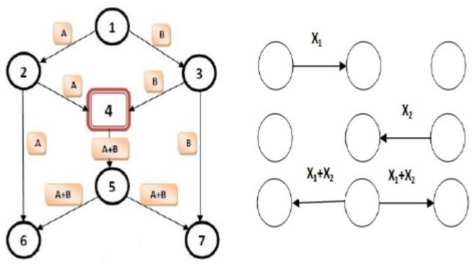


Figure 2.1: Efficiency of Network Coding in lossy networks

In this figure, the source node 1 delivers two bits A and B to nodes 2 and 3, respectively. Node 4 applies XoR operation on both bits A and B and sends one bit A+B instead of two bits. This not only reduces the number of transmission, but also lets nodes 6 and 7 extract required bits B and A sooner than usual by applying XoR operation on (A, A+B) and (B, A+B), respectively. In fact, end-to-end delay between source and destinations decreases which leads to higher video quality, because receivers can have their required video frames earlier than usual. Moreover, according to Fig. 2.1, network coding decreases energy consumption in wireless nodes which is very important, especially for small gadgets such as mobile phones [9].

Network coding approaches have their own pros and cons. XoR-based, as illustrated in Fig. 2.1, is the simplest approach which applies XoR operation on the received video segments or blocks. By using network coding, a video frame divides into some segments and each segment future is divided into many k-byte blocks. Usually, the sizes of segments and blocks within a frame and segment are the same, respectively. As per paper [9] XoR-based approach is topology dependent. Random Network Coding (RNC) addresses this problem. In fact, a node can apply this approach on different blocks of a segment.

2.1.2 Random Linear Network Coding (RLNC)

There are two types of network coding, **Linear Network Coding (LNC)** and **Random Linear Network 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 [1] 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 [6] with and without random linear network coding. Simulation results shows that there is an increase in number of successful transmissions for distance greater than 500 distance units in the case of random linear network coding.

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 among the nodes is required.

2.1.3 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 coefficients

Forwarded coefficient 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 linear equations

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

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

2.1.4 Other variants of Network Coding

In this section, survey, observations and results noted for other variants of network coding is discussed.

Partial Network coding with Opportunistic Routing(PNCO)

Observations: [10]

- In this scheme,Source node breaks the information into 'n' blocks of 'k' packets and randomly mixes packets of same block before forwarding.

- ACK based scheme where sender sends the next packet only after receiving the ACK of previously sent data, which reduces the delay.

- From sender side, for each and every link, the delivery probability is calculated with expected cost matrics.
- Once encoding is done, sender sends the forwarders list in the packet header.
- Up on receiving dependent packet, Forwarders drops the packet and Up on receiving independent packet, Forwarders encodes this packet again by using coefficients.
- When destination receives encoded packet, it decoded them and get the original packet.
- To avoid retransmission of same packet as well as to send next packet,Windowing scheme is used.
- Results shows that compare to path routing protocol, PNCO with opportunistic routing results in higher 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.

Opportunistic Network Coding(ONC)

Observations: [11]

- Status of the buffer's queue at a node decides whether packet is transmitted with network coding or without network coding.
- For real time traffic i.e MPEG-4 Transmission, 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 possible cases of ONC.

- For delay and packet loss, conventional network coding is not optimal. For ONC there is a delay-power trade off. Simulation results shows that this scheme achieves lower delay compared to conventional network coding.

Cooperative Network Coding(CNC)

Observations: [12]

- cooperative and collaborative packet transfer by different nodes is found to exploit spatial diversity in this scheme.

- Nodes are synchronized through synchronization technique e.g. GPS for co-operation.

- 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.

- In CNC, 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 paper [12] 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 here drawback is that, packet has to wait to be coded with other packet until other packet arrives which results in to more delay and loss rate.

2.2 Single Generation Mixing based Network Coding

It is also referred as Generation-by-Generation(G-by-G) mixing of packets during network coding.During network coding, it is very important to limit the size of matrices. 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 2.2 [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 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.

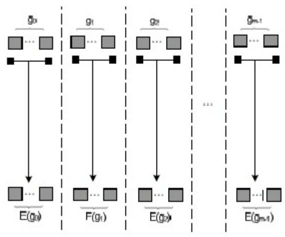


Figure 2.2: Single Generation Mixing in lossy networks, generations are encoded separately

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¡¡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 at least 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 2.3 [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 intermediate node to check the usefulness of received packets and at receiver node to decode received packets [16]. In Single generation mixing based network coding computation are

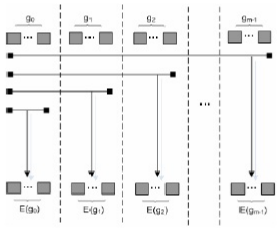


Figure 2.3: MGM based Network Coding, each generation is encoded with previous generations in mixing set

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 and 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 2.4 [18]. Each layer has priority value depending on the generation's position index in a mixing set. Redundant encoded packets enhance the reliability of communication.

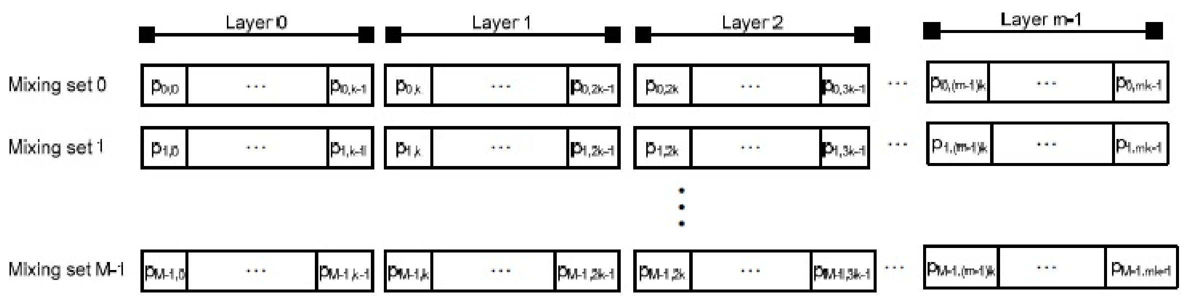


Figure 2.4: Generation's partitioning with MGM into different layers of priority. Mixing set size is m, generation size is k.

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 [20] 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 [20] 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 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.

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.

Table II and III [20] 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 [19]. 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 [20].

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 [20].

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. 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;
```

Chapter 3

Survey on Multimedia Systems

3.1 Need of Multimedia System

In the recent years, video streaming over wireless networks has become very popular. Users can join the network easily and move from one point to another with no restriction. On the other hand, there are some problems in providing smooth video playback in these nodes due to time-varying channels, obstacles and low upload and download bandwidth, especially in gadgets such as mobile phones. Although it is possible to degrade the side effects of these problems by using some efficient video compression techniques, better and more efficient solutions are required to cope with them. Multimedia transmission includes multiple forms of media which includes audio, video,text etc. Real time multimedia data requires the large storage capacity and higher bandwidth of network to meet the critical need of real time MPEG-4 streaming applications such as...

- Jitter control
- Low delivery ratio,High latency,Low packet loss
- Per packet delay, Recovery or Reconstruction of lost packet against 2Ls factors i.e. Lossy compression and Lossy network

3.2 Compression Standards in Multimedia Systems

Compression is the process of reducing the size of multimedia data without changing their actual meaning. The compression can be classified as below:

- Loss less Compression:- In Loss less compression technique, the original representation can be exactly recovered. No loss of single bit of data is there in this technique. This technique is used for the compression of text file or compression of database or for bit level image or picture. WAV is an example.

- Lossy Compression:-In Lossy compression, there is some loss of data. So data can be encoded into a form that takes up a relatively small amount of space. In this technique the original representation can not be recovered perfectly. For images and video one can use lossy compression because of redundant data. MP3 is an example of a lossy audio codec.

MPEG-1, MPEG-2, MPEG-4, Vorbis, DivX, are codecs, used to encode and decode (or Compress and Decompress) various types of data such as sound and video files. Container format contains one or several streams already encoded by codecs (AVI, Ogg, MOV, ASF, MP4, etc. are container formats). These extensions indicate the video file format. Essentially, videos are packaged up into encapsulation containers, or wrapper formats, that contain all the information needed to present video. Videos have a lot of different information that can be "wrapped" up into these containers including the video stream, audio stream, metadata, subtitles, chapter-information, synchronization information, etc. [23] [24].

There are main three compression standards available for multimedia as given below. [23]

- JPEG
- MPEG
- H.261/263/264

In my thesis work, I mainly focus on MPEG for real time video streaming.
MPEG Standars: MPEG is the video compression standard for the moving object. Video compression refers to reducing the quality of data used to represent the video images. Video data contains spatial and temporal redundancy. Similarities can thus be encoded by finding the differences within a frame (spatial domain) and between frames (temporal domain). Spatial encoding is performed by taking advantage of the fact that the human eye is unable to distinguish small differences in color as easily as it can changes in brightness and so very similar areas of color can be encoded in a way similar to jpeg images. With temporal compression only the changes from one frame to the next are encoded as often a large number of the pixels will be the same on a series of frames. This is called as a motion estimation process. In video compression one need to remove the temporal redundancy. This is done by temporal redundancy technique. MPEG do redundancy reduction by using three types of frames as indicated below:[23] [24]

- I-frame : Intra frame(pictures)
- P-frame : Predicted frame(pictures)
- B-frame : Bidirectional prediction frame (pictures)

MPEG algorithm considers the balance between the interframe and intraframe coding. MPEG algorithm works mainly on two basic techniques. First technique is block based motion compensation or motion estimation for reduction of the temporal redundancy (inter-frame) and second technique is transform domain, DCT-based compression for reduction of spatial redundancy (intra-frame). MPEG is most popular codec for video compression. **Compression ratio achieved by MPEG is about 50:1**. There are different versions of MPEG as discussed below: [23] [24]
MPEG-1: This version defines a group of audio and video coding and compression standards defined by MPEG (Moving Picture Experts Group). MPEG-1 video is used by the Video CD format and less commonly by the DVD-video format.

MPEG-2: This standard is for the coding of moving pictures and associated audio information. It defines a combination of both lossy video compression and lossy audio compression methods which permit storage and transmission of movies using currently available storage media and transmission bandwidth. MPEG-2 transport stream is commonly used in broadcast applications. MPEG-2 systems also defines Program Stream container format, which is designed for reasonably reliable media such as disks and used in DVD and SVCD standards.

MPEG-4: MPEG-4 is based on the segmentation of the different object. It divides the video into the different objects. A specific video object and scene background can be individually defined and allows a separate coding of each object comprising a scene. It is a standard used primarily to compress audio and video digital data. MPEG-4 Part 2(MPEG-4 Visual), is a video compression technology developed by MPEG Simple Profile is used where low bit rate and low resolution are mandated by conditions like network bandwidth, device size etc. Examples are mobile phones, some low end video conferencing systems,electronic surveillance systems etc. MPEG-4 Part 10 (H.264/AVC) , compresses video more efficiently than earlier MPEG codec. It also possesses more exibility, which allows it to accommodate applications in a wide variety of environments. It is currently the most popular video codec. It is used in Blu-ray Disc, YouTube, iTunes, and many nations' broadcasting and other video related applications MPEG-4 provides the following functionalities:

- Video communications art lower bit rate
- More efficient than over MPEG-2 s far as coding is concerned
- Error resilience to enable robust transmission
- Ability to interact with the audio-visual scene generated at the receiver

  Video stream encoded in mpeg 4 format is nothing but a sequence of Video Object Plane (Similar to Group of pictures - GoP).

3.3 Group of Pictures(GoP) and Video Object Plane (VOP)

3.3.1 Group of Pictures(GoP)

In video coding, a group of pictures, or GOP structure, specifies the order in which intra and inter-frames are arranged. The GOP is a group of successive pictures within a coded video stream. Each coded video stream consists of successive GOPs. From the pictures contained in it, the visible frames are generated. A GOP can contain the following picture types: [23] [24]

- I-picture or I-frame (intra coded picture) - reference picture, which represents a fixed image and which is independent of other picture types. Each GOP begins with this type of picture.
- P-picture or P-frame (predictive coded picture) - contains motion-compensated difference information from the preceding I- or P-frame.
- B-picture or B-frame (bidirectionally predictive coded picture) - contains difference information from the preceding and following I- or P-frame within a GOP.
- D-picture or D-frame (DC direct coded picture) - serves the fast advance

A GOP always begins with an I-frame. Afterwards several P-frames follow, in each case with some frames distance. In the remaining gaps are B-frames. A few video codecs allow for more than one I-frame in a GOP. The I-frames contain the full image and do not require any additional information to reconstruct it. Therefore any errors within the GOP structure are corrected by the next I-frame, where B-frames can be referenced by other pictures in order to increase compression efficiency. The more I-frames the video stream has, the more editable it is. However, having more I-frames increases the stream size. In order to save bandwidth and disk space, videos prepared for internet broadcast often have only one I-frame per GOP. The GOP structure is often referred by two numbers, for example M=3, N=12. The 'M' the distance between two anchor frames (I or P). The 'N' indicates the distance between two full images (I-frames): it is the GOP length. For the example M=3 N=12, the GOP structure is **IBBPBBPBBPBBI**. Instead of the M parameter one can use the maximal count of B-frames between two consecutive anchor frames.

3.3.2 Video Object Plane (VOP)

MPEG-4 video streams can be divided into hierarchical layers. The lowest layer is the Video Object Plane (VOP) layer. It corresponds to a single frame of the video stream. VOP is used to represent rectangular-plane frames or arbitrary-shaped object plane. In object-based coding the video frames are defined in terms of layers of video object planes (VOP). Each video object plane is then a video frame of a specific object of interest to be coded, or to be interacted with. Figure 3.3.2.1 shows a video frame that is made of three VOPs.

Figure 3.1: Video Object Plan-Example
[25]

In this figure, the two objects of interest are the balloon and the aeroplane. They are represented with their video object planes of VOP1 and VOP2.[23] [24]

3.4 Survey on video streaming using network coding in wireless networks

A. Rate-distortion Optimized Network Coding for Cooperative Video Stream Repair in Wireless Peer-to-Peer Networks [22]

It presents the idea of applying NC in Wireless 802.11 Network configured in Ad hoc mode. It is applied on received packets by peers (or intermediate Base station). It deals with packet transmission over unreliable channel. In a rate distortion optimization the encoder focuses on the motion estimation to increase the quality of multimedia streams. So, it considers P frames (H.264) for the motion estimation. It improves video quality by recovering packets of P frames using NC with appropriately higher probabilities than less important ones (B frames). In video streaming a frame is correctly decoded if its packets are correctly received and its referenced frames are correctly received. So, for the motion compensation P frames are considered as important frames.

It presents two scenarios for P frames transmission as 1) Unstructured Network Coding and 2) Structured Network Coding. In UNC, using NC, each peer generate and transmit a packet using a linear combination of its set of MBMS (Multimedia Broadcast Multicast Service) native packets and its set of received NC packets. In UNC, if a peer receives fewer than T (where T denotes the set of all the packets in a GoP) innovative native or NC packets, then the peer cannot recover any of the native packets from the received NC packets. While in case of SNC, if a peer receives fewer than T innovative native or NC packets received then the peer can decode the packets. To apply SNC, the MBMS source packetizes the frames into multiple RTP packets and searches for the optimal NC structure. The MBMS source adds the NC structure in the header of each of packet. Peers exchange their packets according to the predetermined packet types and transmission probability. After receiving the packets which are attenuated by the cellular broadcast channel, peers in the network initiate the packet repairing process using their 802.11 Wireless interfaces. In both (UNC and SNC) scenarios Wireless Ad hoc peers cooperatively relay packets to each other to repair packet losses during MBMS broadcast.It assumes Wireless peers are having perfect state knowledge of their neighbors.

B.Secure Network Coding for Multi-Resolution Wireless Video Streaming [22]

It presents the idea of applying RLNC over Wireless Networks. It deals with the providing protection for Wireless multi resolution video streaming. It is designed for delay sensitive applications , while increasing the overall robustness to losses and failures, reducing scheduling problems and adding resilience and this is possible using NC. It presents the idea of SPOC (Secure Practical Network Coding), is a lightweight security scheme to provide confidentiality using RLNC. It provides a powerful way to exploit the inherent security of RLNC in order to reduce the number of cryptographic operations required for confidential communication. This is achieved by protecting (using locked coefficients). The source coefficients required to decode the linearly encoded data, while allowing relay nodes to run their NC operations on the substitute "unlocked" coefficients.

The source encrypts each line of matrix A (locked coefficients matrix) with the corresponding layer key. The source then generates an n n identity matrix I, which corresponds to the unlocked coefficients. The packets are composed by the header, which includes the locked and unlocked coefficients, and the payload. The receivers apply Gaussian elimination following the standard of RLNC over the unlocked coefficients. The locked coefficients are recovered by decrypting each line of the matrix with the corresponding key. The plaintext is then obtained by forward substitution. The source applies RLNC which results in reduction in packets loss over Wireless channels. The task of providing different video streaming of variable quality to a heterogeneous set of receivers with different subscription levels with the security of video streams with maintaining the fairness is an issue.

C.Video Centric Network Coding strategies for 4G Wireless Network [22]

It presents a novel NC based strategy for video transmission over 4G Wireless Networks.It uses different coding approaches such as Reed Solomon, Raptor code etc, adds feedback and device discovery to tailor the coding to the receiver and achieve better performance.

RLNC applied by the video server which allows recuperating from the erasure and the flow of files continues uninterrupted even when erasures happen. It proposes a TCP/NC new protocol, this protocol inserts a new layer of NC between TCP and IP. In this layer RLNC masks link losses from the TCP to avoid transmitting window back off again, the losses can be recuperated from the coded information. The scheduling algorithm uses a new ACK design in which packet is ACK'ed even if partially decoded. This approach results in higher goodput when the loss rate is high. This approach is more suitable for Wireless and peer to peer Internet. The proposed scheme is implemented on real time traffic (where retransmission of video packets are not desired) using feedback of previously transmitted packets and using device discovery. Feedback and device discovery provides information about resolution, rates are supported and levels of NC to minimize the number of retransmissions. It mainly focuses on better utilization of bandwidth resources for video content transmission and allocation of resources. In future, the strategy can be further enhanced with power management at the signal level using "softcast" which will improve the quality of video and will make the worst receiver of a multicast group dictates the maximum bit rate of group.

D. An Effective Transmission Scheduling Mechanism with Network Coding for Adaptive P2P Streaming [22]

It present a novel transmission scheduling mechanism called random layer selection with random push (RR), it based on intra-layer NC scheme for Scalable Video Coding

(SVC) over Peer to Peer Networks,. In RR, when a peer pushes a packet, it will Randomly choose a layer and then encode a packet using Random Network Code. SVC emerged as a highly attractive solution to solve the problem of various needs of users with different network conditions. In P2P streaming employing SVC, peers could degrade video quality to ensure smooth playback when bandwidth variations occur. The P2P transmission scheduling mechanism not only decides to deliver which part of the stream (at segment level), but also has to decide to transmit which layer of the part. As per the characteristic of SVC the data of the same segment are received at the same time in SVC streaming.

In RR, the decision of request layer depends on the current buffer level. The selected probability of each layer is very important and it decides the performance of the P2P system. The RR is more scalable to network fluctuations and can improve the latency and efficiently use the bandwidth.

Chapter 4

Problem Identification and Proposed Solution

4.1 Problem Identification

While watching live cricket match or streaming your favorite videos from YouTube. It is quite obvious that every one feel delay in video,variation in delay (jitter) and sometimes may be some part (may be video of milisec. or seconds) of live video may be skipped. This is due to packets loss,delay in delivering packets in a specified time interval and variation in delay. For any wireless network, bandwidth is limited so we cant not increase for particular transmission of real time video and/or audio data but packets loss can be decreased by applying good technique. To overcome these problems, MGM based Network Coding is one of the solutions to be applied.

4.2 Suggested Approach

- Suggested approach starts with generation of real time MPEG4 traffic (I,B,P frames) using Evalvid utility to avoid the use of video traffic model.

- For real time video data, I frame's priority should be highest than other frames as loss of single I frame results in to the loss of entire GOP because I frame is referenced by P frames and B frames. So, loss of I frame means loss of Group of Picture which introduces delay (breaks in the video sequence) and decreases the quality of video during streaming.

- To minimize such types of losses of I frames in MPEG-4 streaming, sufficient number of redundant encoded packets should be present in network so streaming client can easily reconstruct original video contents streamed by streaming server. This also helps in recovery of lost packets.

- As per first approach, we can go for Generation wise encoding which is also refereed as Single Generation Mixing. For each packets of I frames are placed together in one encoded packet and enough copies will be generated in order to recover original packets at the receiver side. P and B frames are encoded in different packets of different generations and similarly, enough copies will be generated in order to recover original packets at the receiver side. but again this can not satisfy the protection criteria of I frames as all I frames are independently are encoded together. In the figure 4.2.1, the approach with the generation wise encoding for GOP having length M=3 (Distance between two anchor frames. i.e.I or P) and N=12 (Distance between two full image i.e. I frames) is explained.

  But, This approach will not satisfy the requirement of protection of I-frames, as GEN-0 can be lost in wireless network and two GoPs can be lost.

- To overcome limitation of this approach, we can go for Multi Generation Mixing of video contents as shown in figure 4.2. In MGM, packets of I frame (I frame's Generation) are mixed with other packets of P and then packets of I and P are further mixed with packets of B frames (B frame's Generation) and enough copies will be generated in order to recover original packets at the receiver side. This Multi Generation Mixing of packets in this way provides robustness to

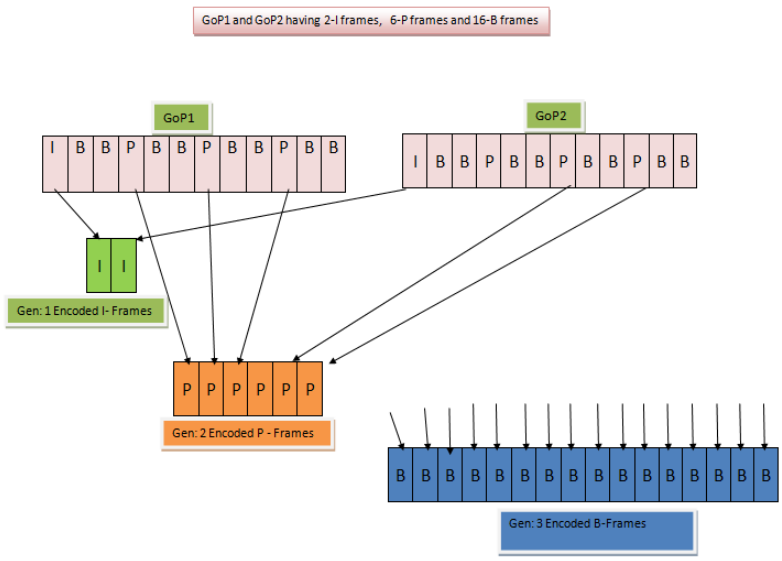


Figure 4.1: Generation wise encoding of Packets

packet transmission of I frame such that packet loss of I frames can be recovered with the little redundancy and this can improve the overall performance of streaming.

4.3 Suggested Implementation Flow

As per implementation, each sender encode real time video packets generated by Evalvid utility using RLNC with single generation wise and Multi Generation Mixing wise. At Application Layer each sender generates encoded data packets which is of type APPDATA. And using Application layer Agent and Sendmsg method, it sends this APPDATA to Transport layer. At Transport layer, its Agent (target) sets APPDATA in packet's payload and sends the packet to the Network Layer. At the receiver side, Network layer passes the received encoded packet to the Transport layer.

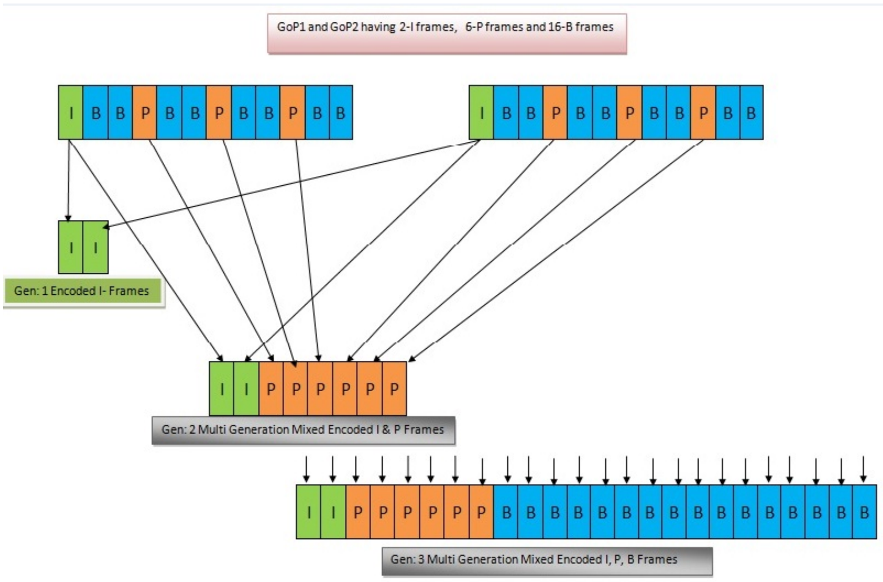


Figure 4.2: Multi Generation Mixing wise encoding of Packets

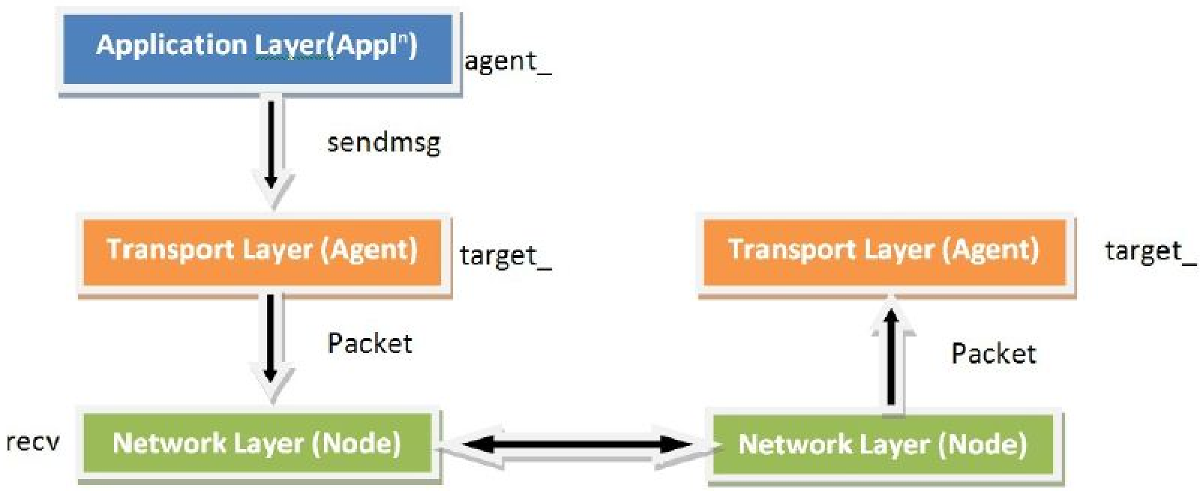


Figure 4.3: Implementation Flow

Chapter 5

Implementation Strategy

5.1 Steps to implement proposed schemes

This section of thesis describes the way to implement the proposed scheme to maintain the protection of I frames which are of higher priority in MPEG4 transmission. In addition to that, maintenance of other frames like P and B frames are also having different importance.

- For each Group of Pictures (GOP) packet or packets, I frames are placed together in one encoded packet and enough copies will be generated in order to recover original packets at the receiver side. P and B frames are encoded in different packets of different generations and similarly, enough copies will be generated in order to recover original packets at the receiver side. Then encoded packets of I frame (I frame's Generation) is mixed with other encoded packet of P and then mixed packet of I and P is further mixed with packet of B frames (B frame's Generation) and enough copies will be generated in order to recover original packets at the receiver side.

- In order to encode packets of different frames generation wise and MGM wise, this thesis work started with generating the real time mpeg4 traffic from raw video file (.yuv) using Evalvid utility. There is no any video traffic model is needed at all for video transmission and generation.

- After generating the real time traffic of mpeg4, the next task is to implement a protocol which can support such transmission at application layer in ns2.

- Simulation of RLNC with MGM wise and Generation wise encoded packets transmission under various scenarios for real time mpeg4 traffic will be done.

- This thesis work follows the suggested implementation flow and following figure shows the actual implementation flow.

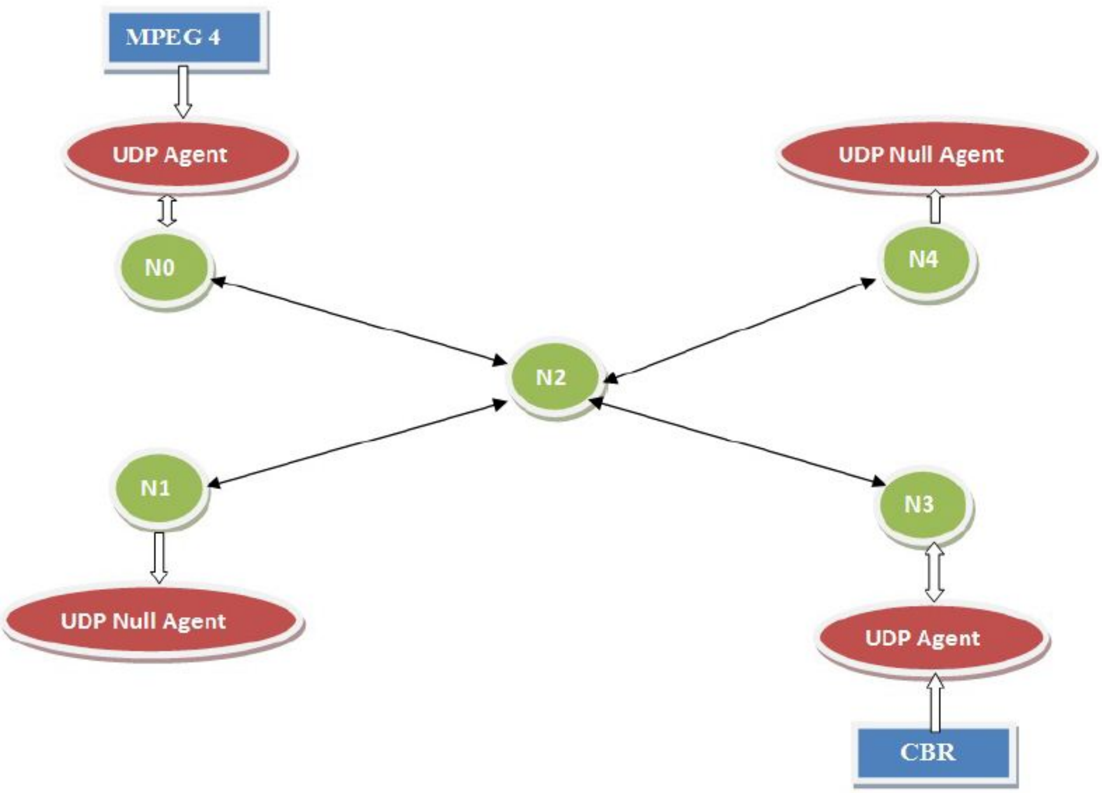


Figure 5.1: Actual Implementation Flow

5.2 MPEG4 Traffic Generation and Evaluation using Evalvid

The first need of this work is to generate the real time mpeg4 traffic as per our requirement (I, B, P frames) to implement this proposed scheme. Evalvid is selected for the said purpose which is further incorporated with NS 2 simulator.

5.2.1 Evalvid - A Video Quality Evaluation Tool

EvalVid is a framework and tool-set for evaluation of the quality of video transmitted over a real or simulated communication network. It is targeted for researchers who want to evaluate their network designs or setups in terms of user perceived video quality. Besides measuring QoS parameters of the underlying network, like loss rates, delays, and jitter, standard video quality metrics like PSNR and SSIM and a subjective video quality evaluation metric of the received video are provided. Currently H.264, MPEG-4 and H.263 are supported.[27] [28]

5.2.2 Steps for Real Time MPEG4 Traffic Generation

This entire section describes the different steps and commands to generate MPEG4 traffic in terms of I, B, and P frames which is prerequisite for this work.

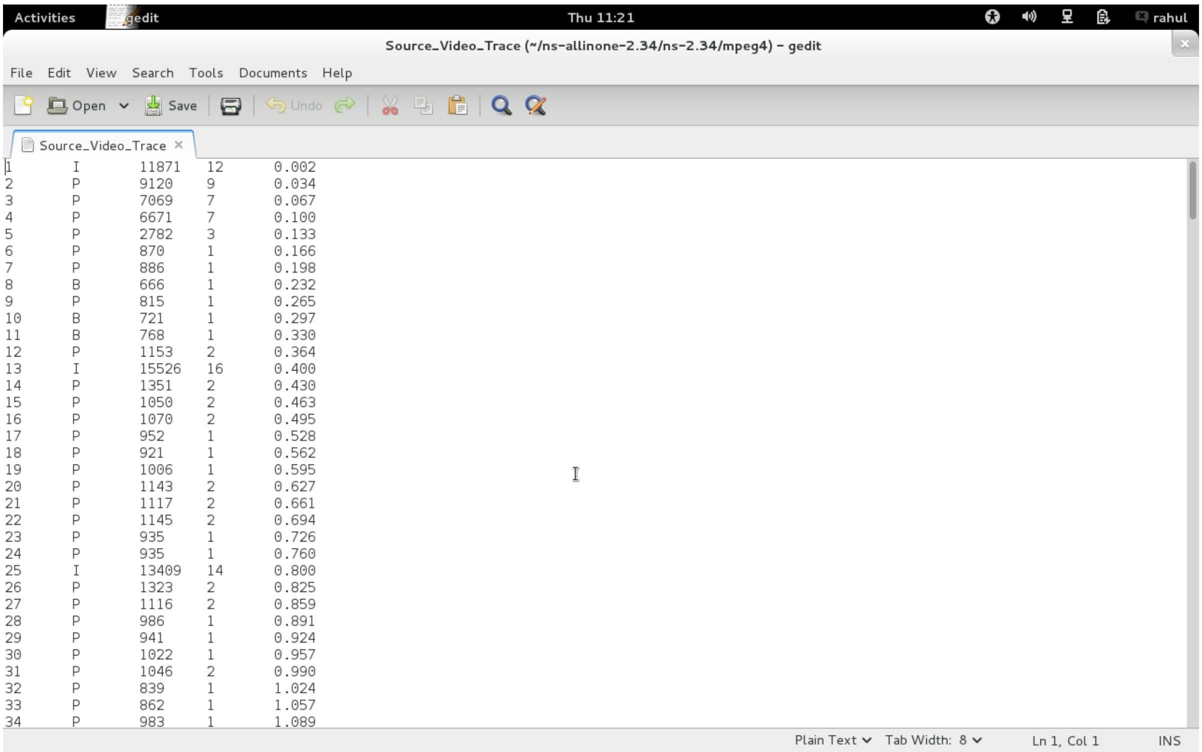
Step:1 Encode yuv file to m4v using
ffmpeg -s cif -r 30 -b 64000 -bt 3200 -g 30 -i sample.yuv -vcodec mpeg4 sample.m4v

Step:2 Convert the m4v file to mp4 using
MP4Box -hint -mtu 1024 -fps 30 -add sample.m4v sample.mp4

Step:3 Send a sample.mp4 file per RTP/UDP to a specified destination host. The output of mp4trace command will be needed later, so it should be redirected to a file.

mp4trace -f -s 222.1.2.4 12346 sample.mp4 >Source-Video-Trace

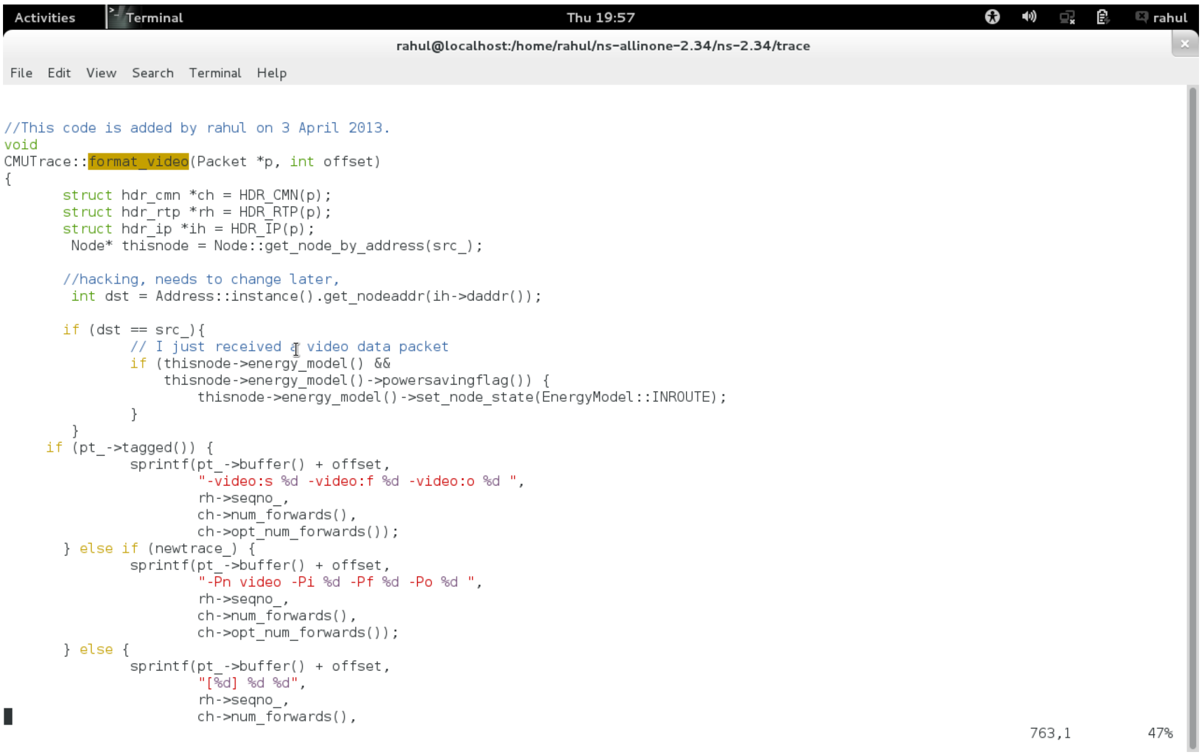
Now, MPEG4 traffic is generated in Source-Video-Trace file as shown in figure 5.2



1	I	11871	12	0.002
2	P	9120	9	0.034
3	P	7069	7	0.067
4	P	6671	7	0.100
5	P	2782	3	0.133
6	P	870	1	0.166
7	P	886	1	0.198
8	B	666	1	0.232
9	P	815	1	0.265
10	B	721	1	0.297
11	B	768	1	0.330
12	P	1153	2	0.364
13	I	15526	16	0.400
14	P	1351	2	0.430
15	P	1050	2	0.463
16	P	1070	2	0.495
17	P	952	1	0.528
18	P	921	1	0.562
19	P	1006	1	0.595
20	P	1143	2	0.627
21	P	1117	2	0.661
22	P	1145	2	0.694
23	P	935	1	0.726
24	P	935	1	0.760
25	I	13409	14	0.800
26	P	1323	2	0.825
27	P	1116	2	0.859
28	P	986	1	0.891
29	P	941	1	0.924
30	P	1022	1	0.957
31	P	1046	2	0.990
32	P	839	1	1.024
33	P	862	1	1.057
34	P	983	1	1.089

Figure 5.2: MPEG4 Traffic Generation using Evalvid utility

To have video traffic support and to display its trace in .tr file in NS-2.34, cmutrace.cc file of ns-2.34 is modified as shown in figure 5.3.



```
//This code is added by rahul on 3 April 2013.
void
CMUTrace::format_video(Packet *p, int offset)
{
       struct hdr_cmn *ch = HDR_CMN(p);
       struct hdr_rtp *rh = HDR_RTP(p);
       struct hdr_ip *ih = HDR_IP(p);
        Node* thisnode = Node::get_node_by_address(src_);

       //hacking, needs to change later,
        int dst = Address::instance().get_nodeaddr(ih->daddr());

       if (dst == src_){
               // I just received a video data packet
               if (thisnode->energy_model() &&
                   thisnode->energy_model()->powersavingflag()) {
                       thisnode->energy_model()->set_node_state(EnergyModel::INROUTE);
               }
        }
     if (pt_->tagged()) {
               sprintf(pt_->buffer() + offset,
                       "-video:s %d -video:f %d -video:o %d ",
                       rh->seqno_,
                       ch->num_forwards(),
                       ch->opt_num_forwards());
       } else if (newtrace_) {
               sprintf(pt_->buffer() + offset,
                       "-Pn video -Pi %d -Pf %d -Po %d ",
                       rh->seqno_,
                       ch->num_forwards(),
                       ch->opt_num_forwards());
       } else {
               sprintf(pt_->buffer() + offset,
                       "[%d] %d %d",
                       rh->seqno_,
                       ch->num_forwards(),
```

Figure 5.3: Adding function to support and display video traffic in cmutrace.cc

After adding the formatvideo() function in cmutrace.cc, Now trace file (.tr) will be able to show traces of video packets as shown in figure 5.4

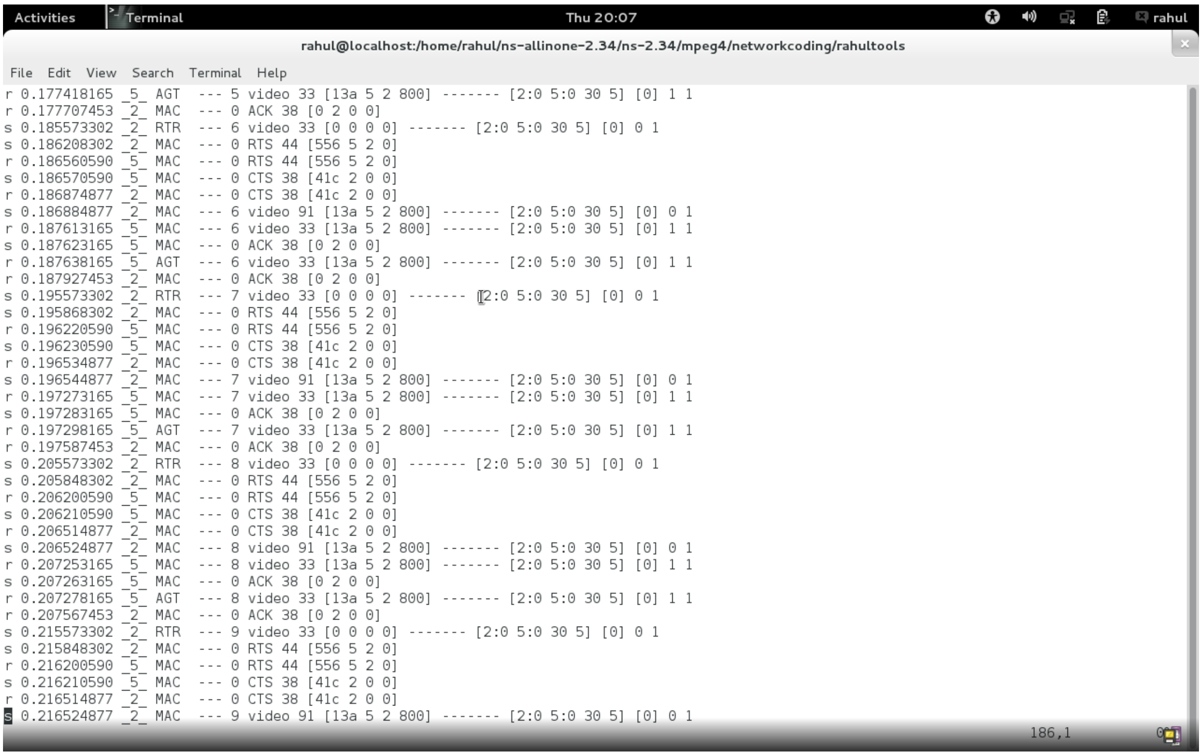


Figure 5.4: Video Packets displayed in trace file

In this work, Network coding is applied at Application layer. So, Application layer is made responsible to generate encoded packets as per suggested approaches. These encoded data packets is in the form of APPDATA which can be set by setting this APPDATA in packets's payload. For that, existing packet.h file of NS-2.34 is modified by adding member functions as shown in figure 5.5

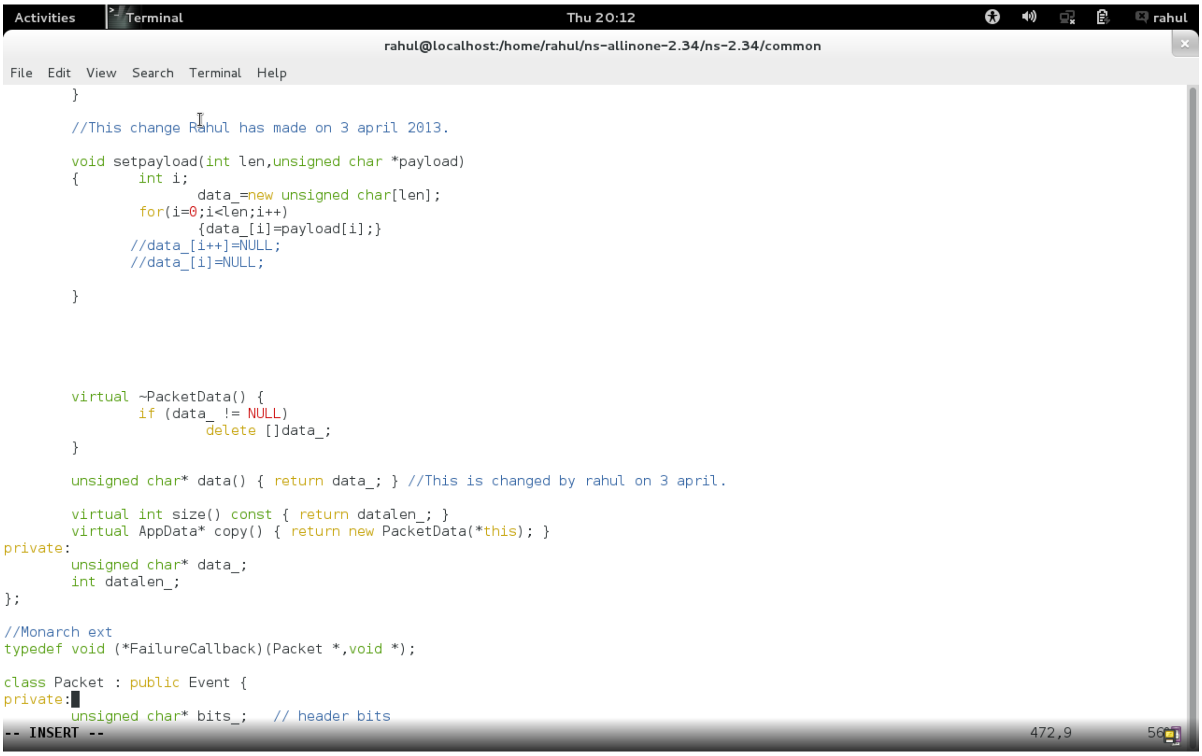


Figure 5.5: Modification of packet.h to set APPDATA in payload

5.2.3 LU Decomposition

The LU decomposition can be viewed as the matrix form of Gaussian elimination which is Numerical analysis method which used to solve linear equations. In RLNC, Packets are encoded and decoded using coefficients .Upon receiving encoded packets at the receiver side,it needs to decide how many packets are innovative (new) and how many packets are duplicate copies of already received packets? So this factorization method is used to find out exact number of innovative packets.

Chapter 6

Simulation System

6.1 Network Simulator

6.1.1 NS-2

NS-2 in its different versions is one of the most popular simulation environments for research. It has a hybrid approach to programming simulations with both C++ and an object-oriented version of Tcl scripting called OTcl. This duality can lead to confusion when not familiar with the system, but it proves to be very convenient once the user becomes acquainted with it. The modules are developed using C++, in order to provide higher simulation speeds by the use of compiled code.[30] [31]
C++ modules are configured and executed via OTcl scripts, which provide the description of the simulation environment and the configuration parameters for each module involved. This OTcl scripts are not compiled but interpreted by the ns software. This makes the set-up of simulations very easy and convenient to batch, as there is no compilation needed to run the scripts, and these contain all the required configuration parameters for the C++ modules.
This duality becomes critical when it comes to develop or modify modules. The modules have two parts: one programmed using C++ and other OTcl. This is required to provide the usability features previously mentioned. There is an All-in-One pack-

age available for most of the releases. These versions include the network simulator, network animator NAM and xGraph in the latest version available at the moment of the creation of the package. The installation is not quite straightforward if you are not using one of the systems supported out-of-the-box for that version, but is easy to find community-developed scripts to compile and install the software properly.The installation of extra modules may require additions and modifications in the configuration files in order to work, being usually simple and well documented. There is an extensive documentation for the network simulator and its modules, in addition to *.tcl example files provided in the distribution in order to both validate the installation of the simulator and learn how to script for the different areas of application of the simulator.

The disadvantage of ns-2 is mainly the limited scalability in terms of number of nodes being simulated, which is not a fixed limit, but it depends on the simulation parameters.This fact is related with the lack of memory management of ns-2: it may require multiple times the amount of memory than some of its alternatives for similar simulations. This is in part a consequence of the use of interpreted software (OTcl), which in 1989 when the ns project was born was a very convenient method to improve the simulation work flow. However, at present, when the compilation process is not time-consuming, it is considered an unnecessary legacy burden when conducting large simulations. Another important disadvantage has already been introduced when stated that ns-2 is in its different versions the most used software: not all modules are updated and valid for all the versions. There have been different points in the development where a number of modules stopped performing properly, so there is a considerable number of research executed with older versions as those are able to execute the modules required by the developers. Outdated versions lack general improvements and patches on different parts of the software which may influence the simulation results and their validity. [32] [33] [34]

Chapter 7

Results and Analysis

Simulation is carried out for various scenarios using NS2. In this work, RLNC with single Generation and MGM is applied over MPEG 4 traffic consisting of I,B and P frames. MPEG4 traffic is generated using Evalvid as discussed previously. This traffic is converted in to encoded packets single Generation wise and Multi Generation Mixing wise. Simulation is carried out for different number of mobile nodes which are generated using Random Way Point mobility model. TwoRayGround propagation model is selected for this work. For routing packets over wireless network, routing protocol AODV is used in all scenarios. Size of Droptail queue is kept 8000. All Results are generated and compared with respect to the simulation work done in [29]. In [29], Video Traffic Model is considered as the source of MPEG4 traffic during streaming which is not resulted into realistic results and observations. This artificial traffic is also not be the real time video files having different length and size during streaming. In this thesis work, Results are collected using the real time video traffic generation out of raw video files using Evalvid utility. Different video files having different size are also considered here which is not considered in [29].

7.1 Results

Graph in Figure 7.1 represents effect of changing network parameter **Number of Connections** on performance parameter **Packet Delivery Ratio** for MGM, G by G wise and Without Network Coding. Results with real time traffic are compared with results with artificial traffic shown in [29] . Here, simulation is carried out by changing number of connections (from 1 to 4 different connections). The size of MPEG4 file is 223 KB. For this simulation, queue length is fixed and it is 8000. For this simulation, simulation time is 560 seconds. Total 2 GOPs are considered. Redundant copies of packets are considered. Here it can be easily observed from the below graph that with increasing the No of Connections, PDR is decreasing due to packet loss because of limited buffer size. But, with limited avaialable buffers, PDR achieved using MGM approach is quite better than using G by G and without network coding.

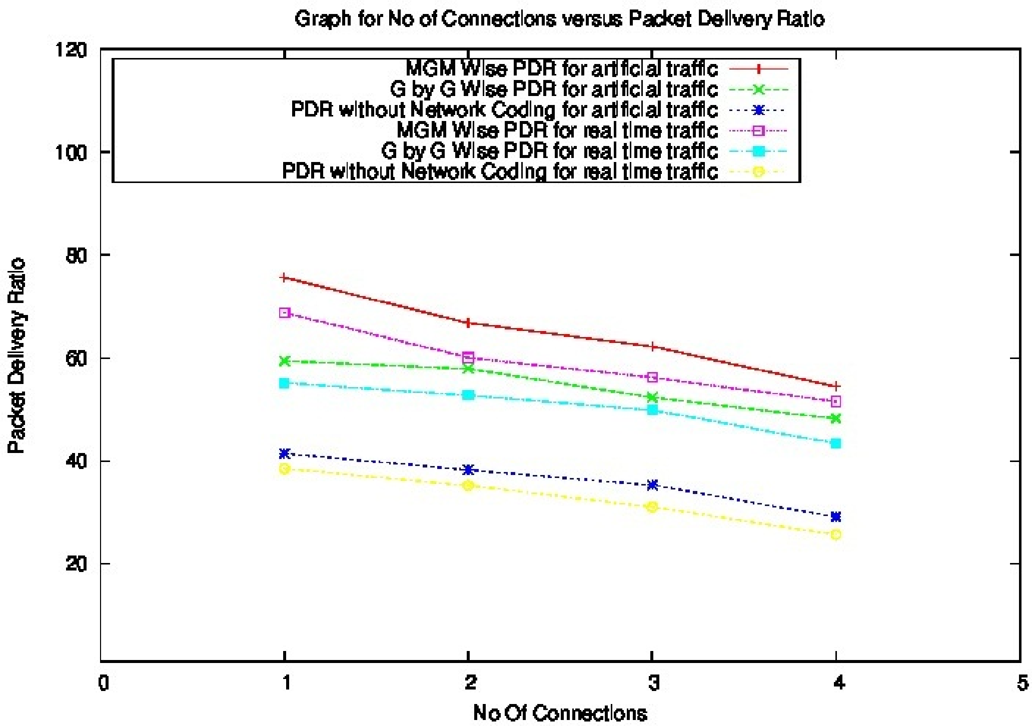


Figure 7.1: Graph for Number of Connections Vs Packet Delivery Ratio for MGM,G by G wise and Without Network Coding for Artificial and Real time traffic

Graph in Figure 7.2 represents effect of changing protocol parameter **Number Of GoPs** on performance parameter **Packet Delivery Ratio** for MGM and G by G wise. Results with real time traffic are compared with results with artificial traffic shown in [29] . Here, simulation is carried out by changing number of GoPs (from 2 to 6). For this simulation, queue length is fixed and it is 8000. The size of MPEG4 file is 223 KB. For this simulation, 3 connections are created. Simulation time is 560 seconds. Redundant copies of packets are considered. Here it can be easily observed from the below graph that mixing more and more number of GOPs is decreasing Packet Delivery Ratio. By mixing only 2 GOPs we can achieve better PDR in both MGM and G by G approach here. So, mixing of more than 2 GOPs is not preferable to achieve better PDR. In addition to that, MGM achieves better PDR as compared to G by G approach. So, to provide more protection to I frames and to recover packet loss, MGM approach is recommended here.

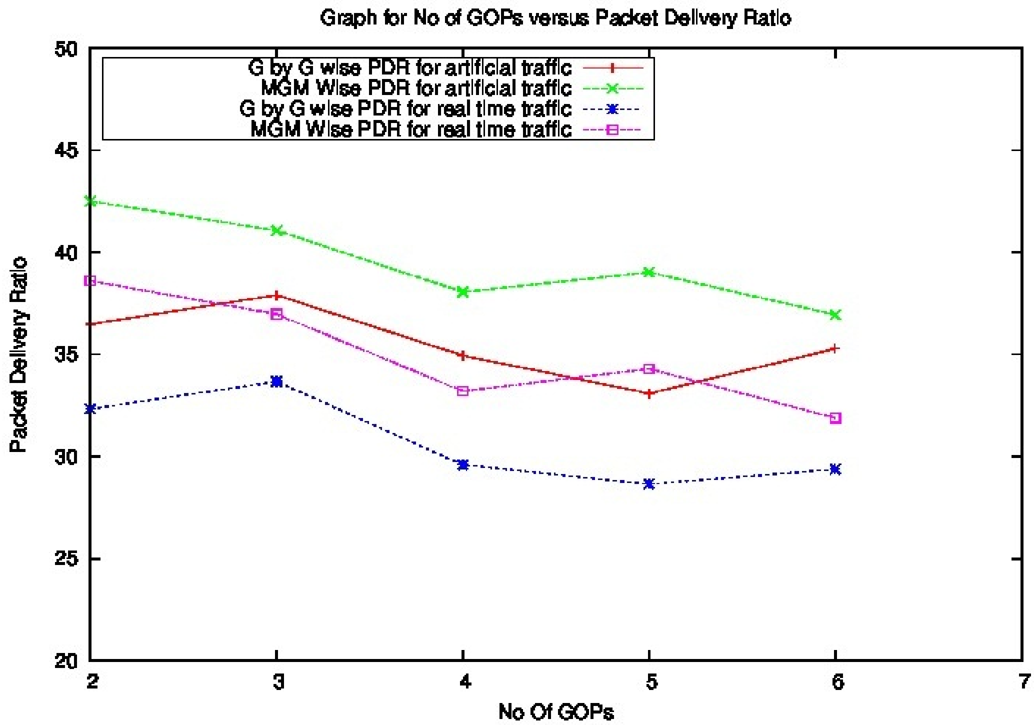


Figure 7.2: Graph for Number of GOPs Vs Packet Delivery Ratio for MGM and G by G wise for Artificial and Real time traffic

Graph in Figure 7.3 represents effect of changing protocol parameter **Number Of Copies** on performance parameter **Packet Delivery Ratio** for MGM and G by G wise. Here, simulation is carried out by changing number of redundant copies (from 1.0 to 2.0). For this simulation, queue length is fixed and it is 8000. For this simulation, simulation time is 560 seconds.3 traffic connections are created. Total 2 GOPs are considered.The size of MPEG4 file is 223 KB. Here we can easily observe and conclude from graph that increasing no of redundant copies does not have significant impact on PDR. Packet Delivery Ratio achieved for different number of redundant copies is similar for G by G and MGM approach for both artificial and real time traffic. So we can say that MGM requires the same no of redundant copies to achieve better PDR as compared to G by G.

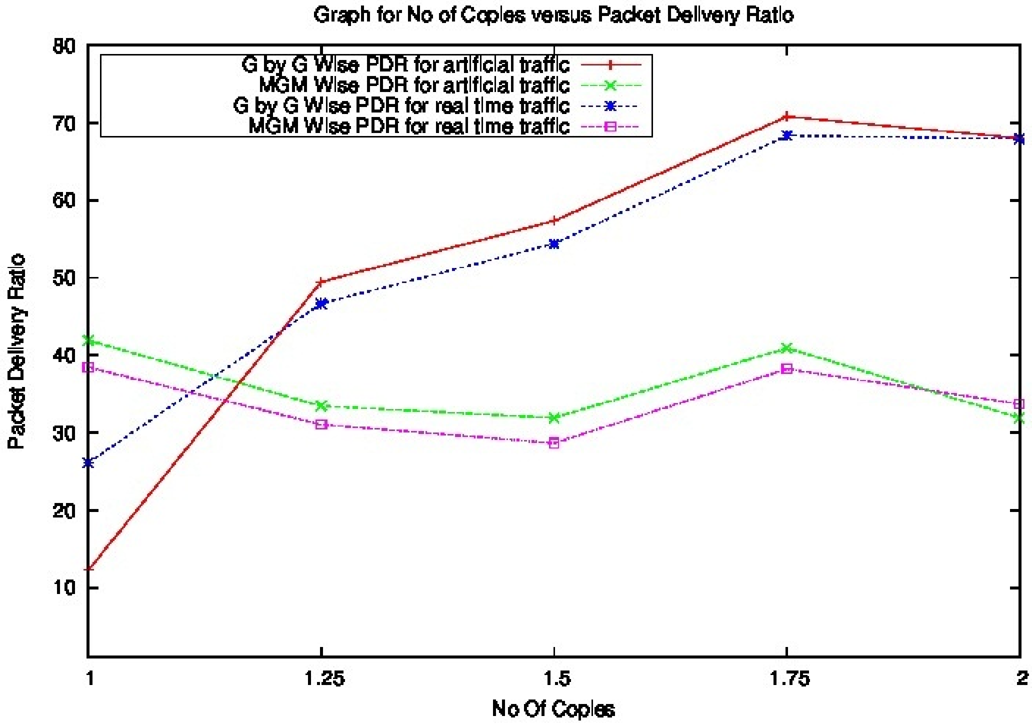


Figure 7.3: Graph for Number Of Copies vs Packet Delivery Ratio by forwarding different no of redundant copies for MGM and G by G for Artificial and Real time traffic

Graph in Figure 7.4 represents effect of changing protocol parameter **Number Of GoPs** on performance parameter **Block Delay** for MGM and Generation by Generation wise. Here, Simulation is carried out by changing number of GoPs(from 2 to 5). For this simulation, queue length is fixed and it is 8000. For this simulation, 3 connections are created. simulation time is 560 seconds. Redundant copies of packets are considered. The size of MPEG4 file is 223 KB. It is clear from the graph that the Block Delay for different values of GoPs, Results are similar for MGM and G by G wise for real time traffic.

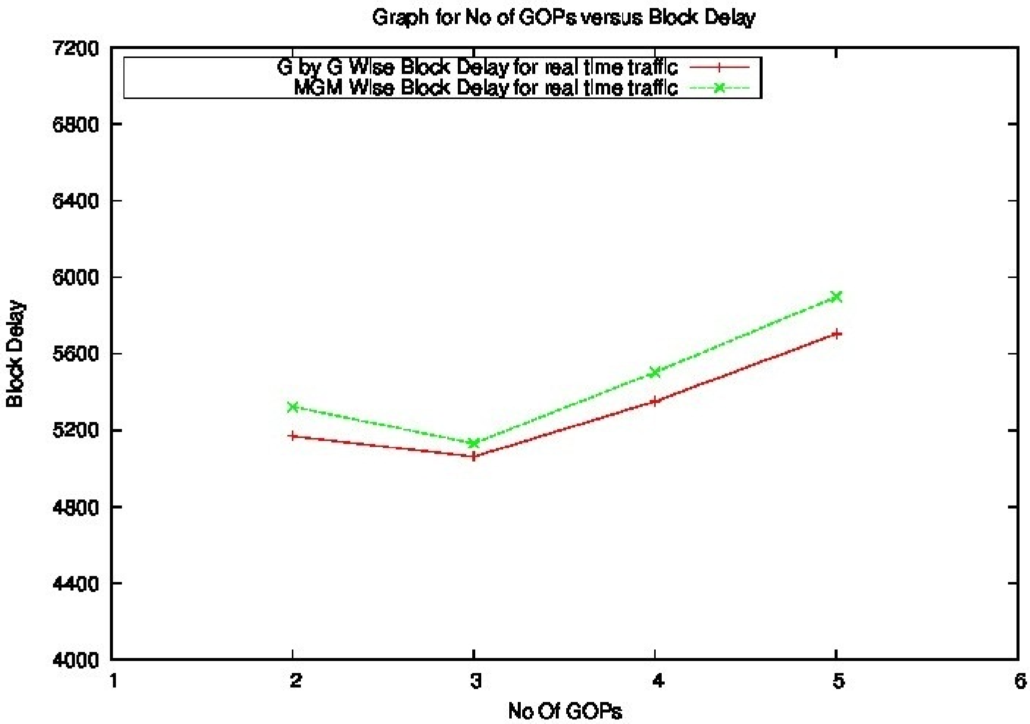


Figure 7.4: Graph for Number of GOPs vs Block Delay for MGM and G by G for Artificial and Real time traffic

Graph in Figure 7.5 represents effect of changing protocol parameter **Number Of Copies** on performance parameter **Block Delay** for MGM and Generation by Generation. Here, simulation is carried out by changing number of redundant copies(from 1.0 to 2.0). Queue length is fixed and it is 8000. 3 connections are created. Simulation time is 560 seconds. The size of MPEG4 file is 223 KB. Total 2 GOPs are considered. From the graph, it is clear that if we increase the no of redundant copies of packets for real time traffic, block delay is increasing in MGM approach as compared to G by G approach.

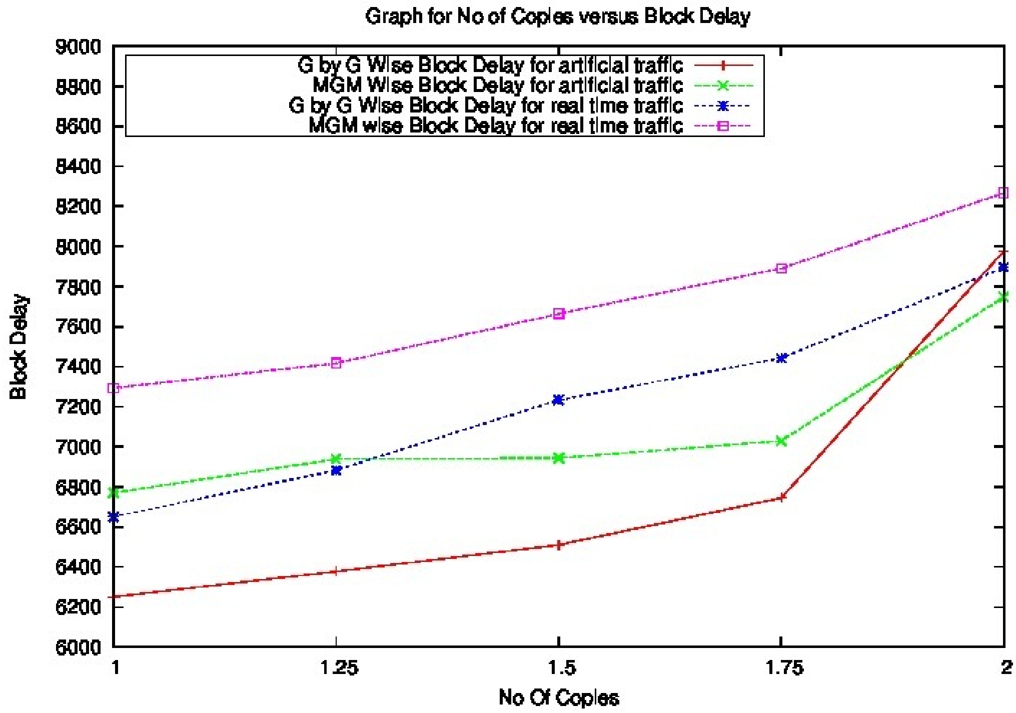


Figure 7.5: Graph for Number of Copies vs Block Delay for MGM and G by G Wise for Artificial and Real time traffic

Graph in Figure 7.6 shows the effect of changing network parameter **Number of Connections** on Performance parameter **Packet Delivery Ratio** for I,P, B frames for real time traffic. Simulation is carried out by changing number of connections (from 1 to 4 different connections). Queue length is fixed and it is 8000. The size of mpeg4 file is 223 KB. Redundant copies are considered. For this simulation, simulation time is 560 seconds. Total 2 GOPs are considered. It is clear from the graph that if we increase no of connections PDR is decreased due to packet loss. PDR achieved using network coding is better as compared to without network coding. PDR achieved using MGM is better that G by G and without network coding. PDR achieved for I frames is more as compared to P and B due to recovery of I-frames from IP and IPB packets which provides more protection to I frames.

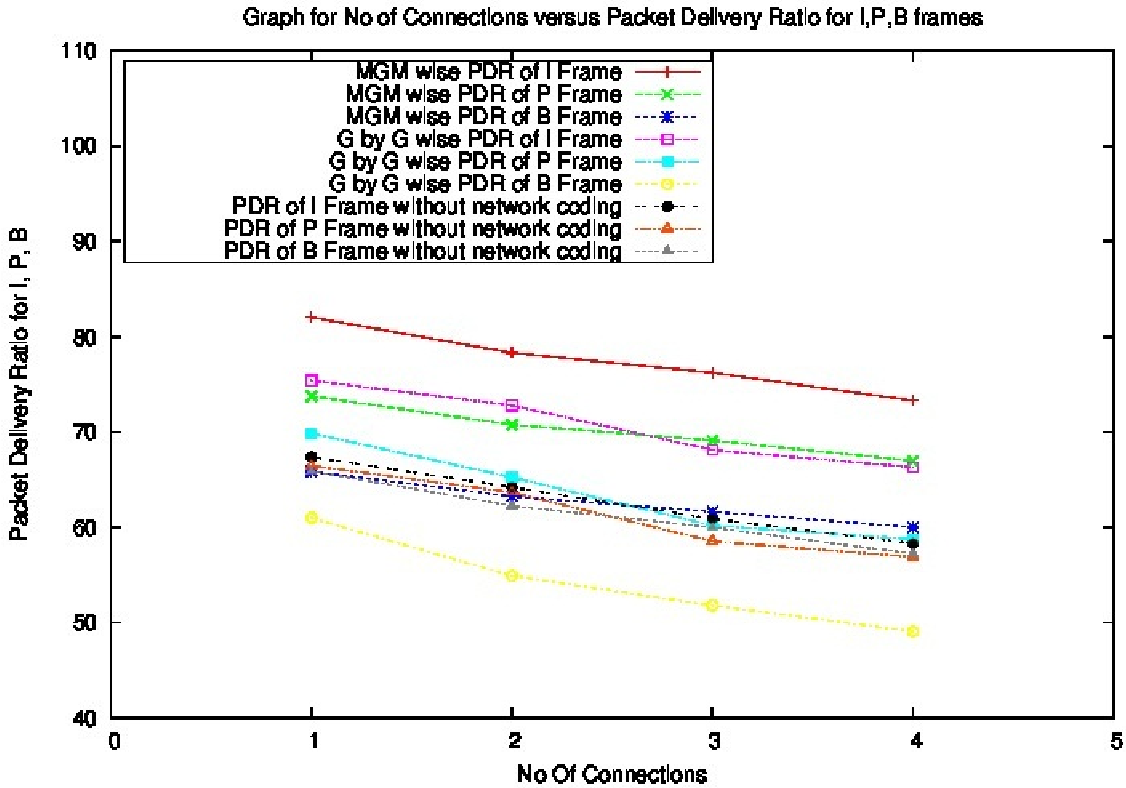


Figure 7.6: Graph for Number of Connections vs Packet Delivery Ratio for I,P,B frames using MGM, G by G and without network coding for Real time traffic

Graph in Figure 7.7 shows the effect of changing network parameter **File Size** on performance parameter **Packet Delay Variation** using MGM, G by G and without Network Coding for real time traffic. Simulation is carried out by taking various mpeg4 files of different size. For this simulation, queue length is fixed and it is 8000. Total 2 GOPs are considered. Redundant copies are considered. For this simulation, simulation time is 560 seconds. It is clear from the graph that PDV is better using network coding approaches as compared to without network coding approach. This lead to minimum buffer requirement if we use network coding based approach.

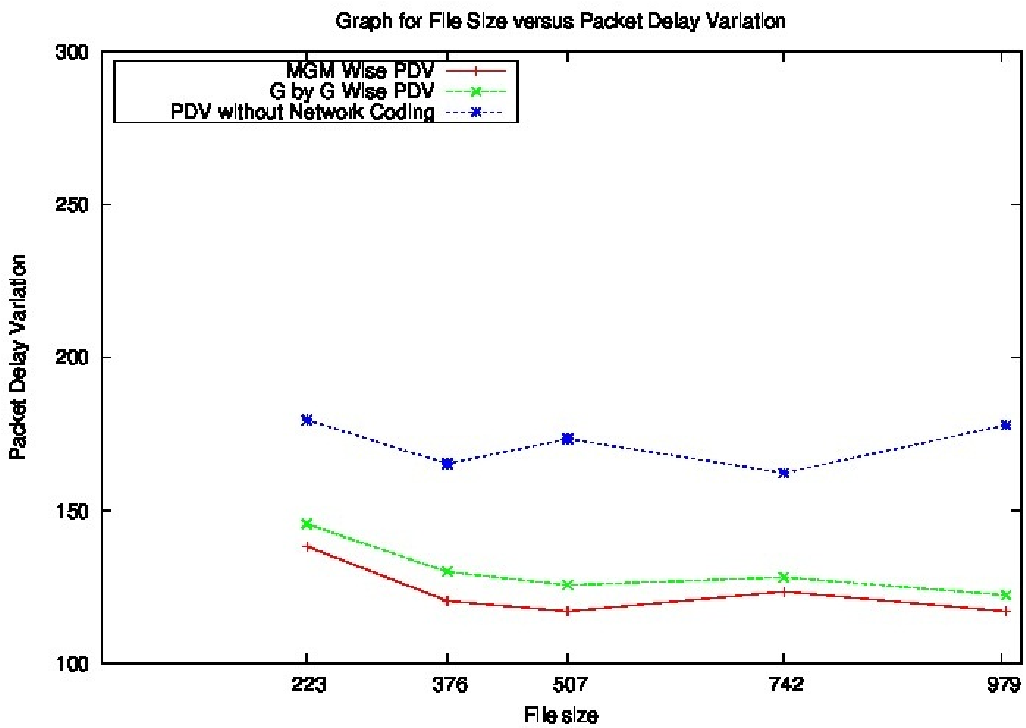


Figure 7.7: Graph for different File Size vs Packet Delay Variation using MGM, G by G and without network coding for real time traffic

Chapter 8

Conclusion and Future Work

8.1 Conclusion

Here we can conclude that using utility such as Evalvid it is possible to generate real time MPEG4 traffic. So, the need of Video traffic model is totally omitted for generation of video traffic frames as I,B and P at the specified frame rate. Mixing of different packets using Random Linear Network Coding with MGM increases the Packet Delivery Ratio. Increasing the No of Connections, PDR is decreasing due to packet loss because of limited buffer size. But, with limited avaialable buffers, PDR achieved using MGM approach is quite better than using G by G and without network coding. By mixing only 2 GOPs we can achieve better PDR in both MGM and G by G approach here. So, mixing of more than 2 GOPs is not preferable to achieve better PDR. PDR achieved for different number of redundant copies is similar for G by G and MGM approach for both artificial and real time traffic. So we can say that MGM requires the same no of redundant copies to achieve better PDR as compared to G by G. Block Delay for different values of GoPs, Results are similar for MGM and G by G wise for real time traffic. If we increase the no of redundant copies of packets for real time traffic, block delay is increasing in MGM approach as compared to G by G approach. PDR achieved for I frames is more as compared to P and B due to recovery of I-frames from IP and IPB packets which provides more protection to I frames. Packet Delay Variation is better using network coding approaches as compared to without network coding approach. This lead to minimum buffer requirement if we use network coding based approach.

8.2 Future Work

In this work, Suggested approaches are implemented and simulated in NS-2.34 simulator. No actual multimedia players and streaming softwares (like VLC player, Helix, Flash Media Players, Red5, Darwin Player) with their limitations are considered, so in the future it is an open research area to go for actual implementation of this simulated work to stream video traffic over wireless using Network Coding. In wireless networks, on receiving less number of encoded packets gives birth to partial decoding of packets or in future one can solve issue of partially decoded packets.

Chapter 9

List Of Publications

- Paper presented and published in '3rd International Conference on Current Trends in Technology,NUiCONE 2012,Organized by Institute of Technology, Nirma University, Ahmedabad, India, held during 06-08 DECEMBER, 2012. Paper Title 'A Survey on MPEG-4 Streaming using Network Coding in Wireless Networks'.

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