# Hybrid FEC/MDC Based Loss Recovery in Multimedia IP Networks

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***Abstract*—Losses due to network problems can result in jitter with streaming technologies, voice over IP, online gaming and videoconferencing. It will affect all other network applications to a certain degree. The uncorrected data loss leads to packet loss at the decoder. This packet loss is uncorrectable producing video and audio issues. In this paper we propose an agent based loss recovery technique for multimedia flows in IP networks. In our work we use relay nodes between ingress and egress nodes and develop a Hybrid FEC/MDC coding scheme in order to perform effective loss recovery while transmitting multimedia data between source and destination. By simulation results, we show that our proposed approach attains high throughput with reduced packet loss when compared with the existing technique.**

***Index Terms*—IP Networks, forward error control (FEC), multiple description coding (MDC), multimedia, loss recovery technique.**

## I. Introduction

Multimedia transmission is new and rapidly growing field which is concerned with all aspects of processing and manipulating multimedia data for transmission and storage. Fundamental issues in this area include data compression and coding, preprocessing (such as pre-filtering), interaction with physical transmission storage elements, and post-processing such as voice or video restoration [1].

The limitations of multimedia data transmission results in low Quality of service (QoS) that is offered to the end user. These constraints are concerned with the multimedia application that holds three main properties such as high data transmission requirement, sensitiveness to packet delays and packet loss tolerance. The above properties introduce new design challenges to the networking world as it is in fact difficult to combine guaranteed high bit rates and an acceptable packet loss ratio with low latency and jitter [2].

### *A. Multimedia Transmission in IP Networks*

Streaming services over IP networks are gaining momentum, and consumers show an increased interest in being able to play and enjoy their media wherever they are.

Deployment of high speed Internet access networks and continuous development of more efficient compression schemes for audio and video are two of many important factors enabling higher quality IP based multimedia services to end-users.

The recent success and large-scale deployment of portable media players indicate a consumer demand for portability requiring transparent delivery of media resources to end users irrespective of network access type, current network conditions or limited capabilities of the end-user's communication device [3].

In Multimedia over Internet protocol (MOIP) systems, one or several encoded video or audio data are grouped into a packet for the transmission through packet networks. The packet network for most MOIP systems operate based on RTP/UDP/IP, but they do not have any QoS control mechanism. Thus, packet losses could occur due to network congestion. Today, the underlying infrastructure of the Internet does not sufficiently support QoS guarantees. As a result, in the future users may have the capability to request specific end-to-end QoS even over the Internet, but this is not feasible today [4].

### *B. Losses in Multimedia Transmission*

Many of the loss recovery techniques have levels of effectiveness that are heavily dependent on the characteristics of packet loss in the network. There are three different packet loss profiles namely random loss, burst loss and real IP networks loss.

- **Random Loss -** In random loss, data packet losses occur randomly. If packet loss were entirely independent from one instant to the next, it could experience random loss. Random loss is a condition that is very favorable for many of the sender based loss recovery techniques due to the relatively rare occurrences of multiple consecutive lost packets with random loss.
- **Burst Loss -** Burst loss causes the loss of a packet persists for some period of time and therefore causes us to lose one or more subsequent consecutive packets.
- **Real Network Loss -** Every IP network has its own distinctive characteristics. Additionally, the behavior of any particular IP network changes, often significantly, from day to day and from time to time within a day. One segment of a network could be highly congested while another segment of the same network could be idle. These kinds of losses are known as network loss [5].

### *C. Loss Recovering Techniques*

The unreliable characteristics of a packet switched network causes packet loss or delayed packet arrival at the receiver during the data transmission. Such packet loss will cause speech signal dropped out which will in turn result in producing sound of poor quality. Hence, the recovery of the lost packets is very important [6]

Packet loss recovery techniques can be divided into

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twoclasses namely sender based and Receiver based techniques. The basic sender based mechanisms available to recover from packet loss are Automatic Repeat Request (ARQ), interleaving, Layered Coding (LC), Forward Error Correction (FEC) and multiple description coding (MDC). When losses occur that cannot be repaired with sender based schemes, receiver based error concealment schemes, such as insertion, interpolation and regeneration, produce a replacement for a lost packet. [7]

Some of the loss recovering techniques is as follows,

- ✓ **Automatic Repeat request (ARQ) -** ARQ is a typical acknowledgement based error recovery technique. In ARQ, lost data packets are retransmitted by the sender host. However, this retransmission mechanism is activated by receiving duplicate acknowledgement (ACK) packets or timer time-out, causing a large end to end delay. This large delay is not suitable for real-time applications such as video streaming and web conference.
- ✓ **Forward Error Correction (FEC) -** FEC is a well-known coding-based error recovery scheme. In FEC, redundant data is generated from original data, and a sender host transmits both the original and redundant data to a receiver host. When some part of the original data is lost, it can be recovered from the redundant data at the receiver host if the loss is below a pre-specified level. Since FEC needs no retransmission, it is suitable for real-time applications with stringent delay constraint such as video streaming. However, FEC does not work well against packet burst loss because the amount of redundant data has to be pre-determined with the estimate of the packet loss probability [8].
- ✓ **Hybrid Error Control (ARQ/FEC) -** A major difficulty when using FEC is to choose the right amount of redundancy in face of changing network conditions. Also, sending redundant data consumes additional bandwidth. In order to overcome this problem, ARQ and FEC can be used in combination
- ➢ **Hybrid ARQ Type II -** It does not send any redundant data with the first transmission, but send parity data when a retransmission is required. This approach is very bandwidth efficient for reliable multicast to a large number of receivers.
- ➢ **Hybrid ARQ Type I -** It immediately sends a certain amount of redundant data using FEC. If the loss rate obtained after reconstruction at the receiver is still too high, ARQ is used to retransmit. Using this approach it is possible to assure with a high probability that a large number of receivers obtain the data without retransmissions, which is attractive for real time audio-visual services [9].

**Retransmission Based Error Recovery** - It is the simplest technique to minimize the overall packet loss ratio in order to increase the quality of the applications. Retransmission can be also used for loss recovery in media applications, but the number of retransmissions is limited by the play-out buffer and the recent network delay. For the retransmission to be successful, retransmitted packet must arrive at the receiver in time for playback. To minimize the probability of wastefully retransmitted packets, a play-out buffer is usually set up at the receiver side to pre-fetch a certain amount of data before playback. The buffered data provides additional time to absorb the retransmission delay making the retransmission acceptable for one way pre-recorded and one way live media applications. In retransmission based schemes, upon detection of a gap, the receiver decides whether to send a negative acknowledgement (NACK) based on the current estimate of RTT and the play-out time of the missing packet [10].

- ✓ **Multiple Description Coding (MDC) -** MDC is potential enough to divide the information streams into multiple sub-streams and each sub-stream can be decoded without utilizing the details stored in neighboring sub-streams. Hence there is no necessity to rely on other sub-streams such as layered video coding. MDC requires more bandwidth utility owing to smaller video compression of the encoding process. [11].
- ✓ **Layered Coding (LC) -** A layered video encoding appropriate for internet applications has to face more requirements. For better utilization of the bandwidth, the compression operation should be enhanced. Moreover the computational difficulties of the codec must be low the real time applications. During the video conferencing applications, both encoding and decoding operations must be performed in real time whose latency should be low. While for streaming applications, non-real time encoding is performed that necessitates reasonable latency. [12].

In our previous paper [20], we have developed a QoS mapping framework to achieve scalability and end-to-end accuracy in QoS, using a Policy Agent (PA) in every DiffServ domain. This agent performs admission control decisions depending on a policy database.

Now as an extension to the previous work, we propose to develop an agent based loss recovery technique for multimedia flows in IP networks.

## II. Related Work

Le Martret et al. [13] have derived analytical expressions of performance metrics (efficiency, delay, PER) for a wide range of retransmission schemes such as Automatic Repeat request (ARQ) and Hybrid ARQ (HARQ) in the case of memory-less block fading channels encompassing new cross layer strategies. The novelty of their work is twofold (i.e.) their metrics were considered at the Network level and they have also introduced a new general framework which enabled them to derive analytically the considered performance metrics for most retransmission schemes, including recent cross layer strategies with the Network layer. In addition, they have also proposed a new general expression for the efficiency even valid when the incremental redundancy packets did not have the same length. These expressions allowed them to speed up notably their metrics computation.

Jiao Feng et al. [14] have proposed a channel adaptive FEC algorithm which balanced the trade-off between the QoS of video transmission and the bandwidth utilization ratio in wireless IP networks. Their algorithm could dynamically adjust to a suboptimal number of FEC redundant packets to cater to the time varying wireless channel. They have derived two analytical models, for the sake of obtaining the suboptimal amount of FEC redundant packets. One was the playable packet rate in MPEG video stream, another was the effective utilization ratio of FEC. Based on these analytical models they have calculated a suboptimal value of redundant packets, which make both the quality of video stream and the effective utilization ratio of FEC to approximate their maximum by predicting the quality of video stream and effective utilization ratio of FEC under different network conditions.

Ankit Bhamri et al. [15] have proposed two Smart Hybrid Automatic Repeat Request (SHARQ) schemes with incremental redundancy, that were developed for a dual hop network of two relays implementing cooperative communication. In their systems, the retransmissions could either be initiated at the relay nodes or at the source node and hence their HARQ protocol should be capable of dynamically deciding the node of retransmission. Their SHARQ schemes proposed were designed intelligently in a way that they took into account the presence of two relays and the benefit of using cooperative schemes. Basically the system developed by them was intended to provide the combined benefits of diversity gain from cooperative schemes and the throughput improvement from SHARQ in a best possible way.

Xunqi Yu et al. [16] have proposed a joint MDC and FEC coding approach for delay constrained applications on congested networks under some idealized modeling assumptions which were, nevertheless, sufficient to draw some qualitative conclusions. They have first modeled the delay constrained application using FEC and have also studied the overall efficacy of FEC in improving delay constrained applications in congested networks. They have also modeled the information transmission system jointly using a combination of FEC and MDC. They have showed that under selected range of operating parameters, there was an optimal FEC coding and MDC coding rate, which could achieve significantly improved end-to-end performance when compared with an SDC system.

Abdullah AlWehaibi et al. [17] have presented a new fair share policy (FSP) that utilized differentiated services to solve the problems of QoS and congestion control when reliable FEC multicast was adopted. They have also found that when the difference in packet processing time between IP and MPLS was high and when MPLS factor was small, IP multicast would perform less efficiently than MPLS in terms of the total packet delay. In addition to that they have showed that when using FEC/ARQ, there would be a slight increase in the total packet delay for all IP and MPLS sources compared to without using FEC/ARQ due to the increase in intrinsic arrival probabilities because of the FEC operation.

## III. Agent Based Loss Recovery Technique for Multimedia Flows

### A. *Forward Error Control (FEC)*

Forward error control (FEC) coding has often been proposed to combat network packet losses. FEC can help recover the lost packets through the use of redundant packets. However, from the network's perspective, the widespread use of FEC schemes by end nodes will increase the raw packet loss rate in a network. Moreover, the additional delay caused by FEC encoding and decoding also need to be considered, which is an issue particularly important for delay constrained applications, such as video telephony and video streaming [16].

FEC schemes have been proposed by many researchers to make applications more resilient to packet losses. FEC techniques rely on the transmission of redundant information from which lost packets can be recovered. This approach reduces the packet loss recovery time compared to ARQ schemes. The FEC encoder works on the sender side and generates a new block of n packets from a block of k data (or source) packets, where (n-k) FEC redundant (or parity) packets are transmitted. On the receiver side a FEC decoder recovers lost data packets using both received data and parity packets.

Some of the most interesting features of FEC schemes are the following:

- FEC encoding provides great advantages in terms of protection of data to losses over the network.
- No delay is introduced in the encoding phase. Data packets are sent over the network, buffered and used by the FEC codec.
- FEC packets can be easily discarded by clients that do not support FEC decoding. For example using RTP encapsulation, FEC RTP packets can be recognized by the RTP payload type and can be sent in two ways: together with data packets or over a different connection as an enhancement layer. In both cases it is very simple to discard FEC packets, in the first case RTP client should discard the packet with a payload type that does not recognize, in the later the connection is not opened at all.

The aspects that must be carefully considered are:

- FEC decoding process introduces some delay when data losses occur. The delay must fit application requirements.
- The FEC encoding/decoding introduces some overhead in computation that must be kept as low as possible. It is very important to use efficient codecs LDPC codecs are very satisfactory from this point of view.
- FEC redundancy can waste network resources if it is not tuned according to network conditions

The best way to introduce FEC without wasting network resources is to dynamically introduce redundancy according to the current link conditions [18].

Though there are many types of FEC we use Low-Density Parity-Check Convolutional Codes (LDPC-CCs) which is better suited to certain applications than block code counterparts (explained briefly in section III. C. 1). This is because LDPC-CCs are able to encode and decode arbitrary lengths of data without the need to fragment them into fixedsized blocks. Many packet switching networks, including those based on the Ethernet packet format, utilize a Protocol Data Unit (PDU) that can vary in size [19].

### B. Multiple Description Coding (MDC)

Another widely used technique to improve network information transmission is Multiple Description coding (MDC). Using this technique, each source sample is encoded by several encoders and each encoder generates a separate description of the source sample with some descriptions may be lost in the network. At the destination, the more descriptions received, the lower the distortion that can be achieved. Research on this subject has been focused on the achievable rate-distortion regions and specific coder designs for actual audio/video applications [16].

MDC has the capability to split the information stream in multiple sub streams, where each of the sub streams can be decoded without the information carried by the neighboring sub streams and therefore has no dependencies to other sub streams such as layered video coding. The advantages of MDC has been exploited for multi hop networks, Orthogonal Frequency Division Multiplexing (OFDM), Multiple Input Multiple Output (MIMO) systems, adhoc networks, Universal Mobile Telecommunications System (UMTS), Transport Control Protocol (TCP) and Content Delivery Networks (CDN). Unfortunately, the advantage of multiple description coding is achieved at the expense of higher bandwidth usage due to the smaller video compression of the encoding process [11].

### C. A Hybrid FEC/MDC Scheme

With the ultimate goal of any communication system to have an improved end-to-end performance, it becomes imperative to develop efficient error control scheme with incremental redundancy, which utilizes the cooperative schemes to maximum benefit and lead to a complete system. In a cooperative system of distributed relays, there is a need of Hybrid FEC/MDC scheme which is able to exploit the following benefits of cooperative system in addition to its inherent performance enhancing capability of reducing the system's block error rate.

- ✓ The cooperative system of distributed relays establishes end-to-end link in two phases, phase 1 being from source to relays (FEC) and phase 2 is from relays to destination (MDC), with phase 2 establishing the link even when just one relay decodes the signal.
- ✓ Our scheme should therefore be devised in a smart way which initiates retransmissions from source only when signal is decoded incorrectly at both the relays.
- ✓ In phase 2 (MDC) of cooperative system, error performance is expected to be better when both relays forward and exploit the macro-diversity. If the destination decodes the signal incorrectly, then two possibilities exist due to existence of cooperative relays: One is to have retransmission in phase 2 and the other is to have retransmission in phase 1 (if retransmissions in phase 1 are not exhausted).

Based on these possibilities, our scheme is based on the principal condition that source initiates retransmissions only when it receives Negative Acknowledgement (NACK) from both the relays. The reception of NACKs from the relay nodes can be facilitated by Coordinated Multipoint Reception in LTE-A. The source does not retransmit when it receives NACK from just one of the two relays in the system, it rather waits for ACK or NACK from the other relay and if it receives ACK from that relay, it does not retransmit. On the contrary, when the source receives ACK from both the relays, it automatically sets the retransmissions counter to maximum number so that phase 1 is shut for transmission of that particular packet.

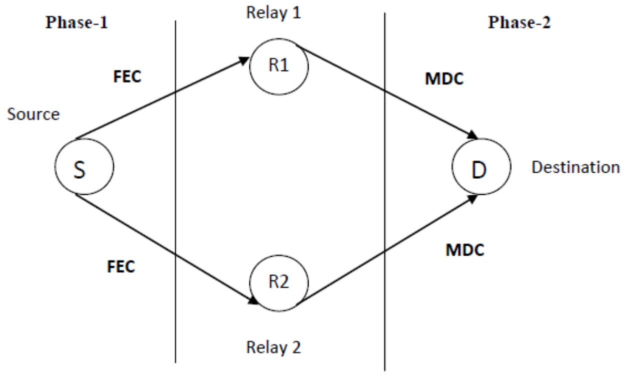


Fig. 1. Architecture

**Algorithm**

1. The source S sends the FEC encoded data to Relays (R1 and R2).

2. R1 and R2, then decodes the data.

3. If the data is decoded correctly both at R1 and R2, then go to step 5.

4. If the data is not correctly decoded at R1 or R2, then it sends NACK back to S.

4. 1. If S receives a NACK, it will retransmit the FEC encoded data again to R1 or R2.

5. R1 and R2 then send the MDC encoded data to Destination D.

6. If D decodes the data correctly, then the data transmission is success (go to step 8).

7. If D decodes the data incorrectly, then D sends a NACK back to R1 and R2.

7. 1. If R1 and R2 receive NACK, they will try to retransmit the MDC encoded data to D.

7. 2. If Retransmission is not successful, then

7.2.1. The Retrans_counter is incremented by 1.

7. 2. 2. If Retrans_counter > Retrans_thr, then

7. 2. 2. 1. R1 and R2 send NACK to S.

7. 2. 2. 2. If S receives NACK from R1 and R2, then

7. 2. 2. 2. 1. S will retransmit FEC encoded data to R1 and R2.

8. Stop.

**1. LDPC-CC**

The generator matrix for LDPCCCs is inherently lowered triangular and this simplifies the encoding process and reduces the encoding latency. LDPCCCs are defined by their infinite, but periodic, parity-check matrix H. For a rate1/2 code, an information sequence

$$U(0, t) = [U(0), U(1) \ldots U(t)] \tag{1}$$

is encoded into a sequence

$$V(0, t) = [V_1(0), V_2(0), V_1(1), V_2(1) \ldots V_1(t), V_2(t)] \tag{2}$$

**A. LDPC-CC Encoding**

We can determine the encoded sequence for a rate ½ systematic LDPC-CC as follows.

$$V_1(t) = U(t) \quad (3)$$

$$V_2(t) = \sum_{i=0}^{m_s} H_1^i(t)U(t-i) + \sum_{i=0}^{m_s} H_2^i(t)V_2U(t-i) \quad (4)$$

where we assume U(t) = 0, t < 0. Hence $V_2$ (t) is generated by an XOR operation upon a number of the $m_s$+1 most recent information bits and the ms most recent code bits. Therefore an LDPC-CC encoder can be constructed from delay lines, multiplexors and an XOR gate.

**B. LDPC-CC Decoding**

The systematic encoder output bit, U (t), and the parity bit, $V_2$ (t), are encoded using Binary Phase Shift Keying (BPSK) onto two independent channels at a rate of one code-symbol per Baud-period. Additive White Gaussian Noise (AWGN) is added such that, at time t, the two receive values are given by

$$r_u(t) = 1 - 2u(t) + n_u(t) \quad (5)$$

and

$$r_v(t) = 1 - 2V_2(t) + n_v(t) \quad (6)$$

We assume that the AWGN on the two channels is from identical distributions and that $n_u$ (t) and $n_v$ (t) are uncorrelated. The two receive values are passed to the decoder which produces an estimate of the transmitted information bit.

The LDPC-CC decoder can be constructed as the concatenation of multiple, identical units, called processors. Each processor implements the Belief-Propagation (BP) algorithm and consists of storage elements, a single check node and two variable nodes [19].

## IV. Simulation Results

*A. Simulation Model and Parameters*

In this section, we examine the performance of our agent based loss recovery technique for multimedia flows with an extensive simulation study based upon the ns-2 network simulator [21]. The topology used in our experiments is depicted in Fig- 2. As we can see from the Fig, we have three senders and three receivers connected by relays R1 and R2 through two routers IE1 and IE2.

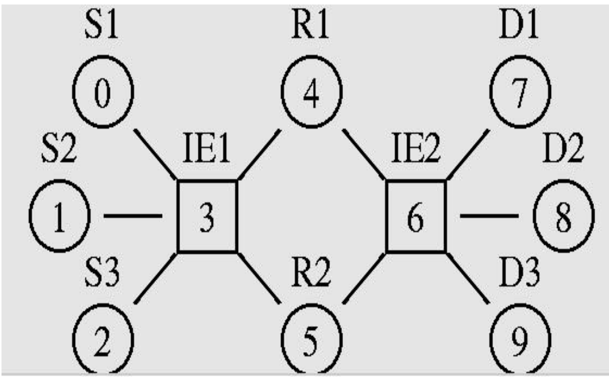


Fig. 2. Simulation topology

*B. Performance Metrics*

In our experiments, we measure the following metrics

- **Packet Loss -** It is the number of packets lost during transmission.
- **Throughput (in terms of packets and Mb/s) -** It is the number of packets received successfully.
- **Average end-to-end Delay:** The end-to-end delay is averaged over all surviving data packets from the sources to the destinations.

We compare our proposed LR (with Loss Recovery) scheme with the previous NLR (No Loss Recovery) scheme. The results are described in the next section.

*C. Results*

**A. Effect of Varying Rate**

In our first experiment, we vary the rates as 10Mb, 15Mb, 20Mb, 25Mb and 30Mb in order to calculate the packet loss, throughput (packets received) and average delay (Mbps). The results for the individual destination are given.

**A. 1. Packet Loss**

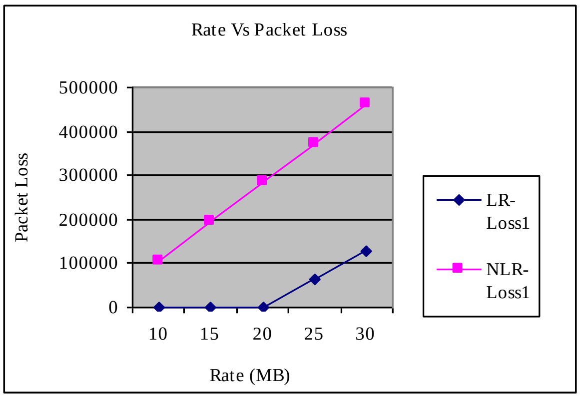


Fig. 3. Rate Vs packet loss

Fig: 3 shows the packet loss at the destination. From the Fig, we can see that the packet loss is high in the NLR scheme when compared with our LR scheme when varying the rates.

**A. 2. Throughput (Packets Received)**

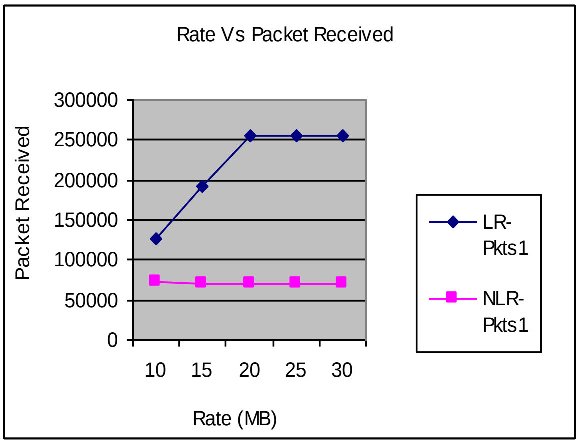


Fig. 4. Rate Vs packet received

Fig: 4 gives the throughput in packets for the destination by varying the rates. It shows that the throughput is more in the case of LR scheme when compared with NLR scheme.

**A. 3. Throughput (Mbps)**

Fig:5 gives the throughput in Mbps for the destination for varying rates. It shows that the throughput is more in LR scheme when compared to NLR scheme.

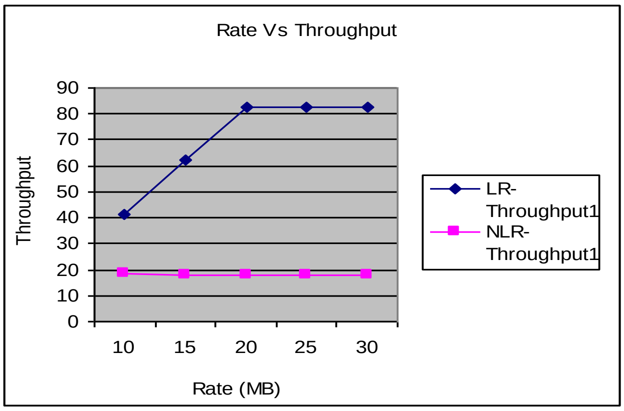


Fig. 5. Rate Vs throughput

### A. 4. Average Delay (s)

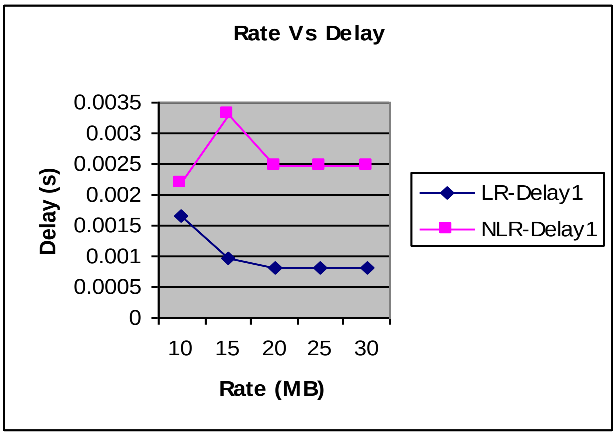


Fig. 6. Rate Vs average delay

Fig: 6 gives the average delay for the destination for varying rates. It shows that the delay is more in the case of NLR scheme when compared with LR scheme.

### B. Effect of Varying Simulation Time

In our second experiment, we vary the simulation time as 2, 4, 6, 10 seconds in order to calculate the packet loss, throughput (packets received) and throughput (Mbps). The results for the individual destinations are given.

### B. 1. Packet Loss

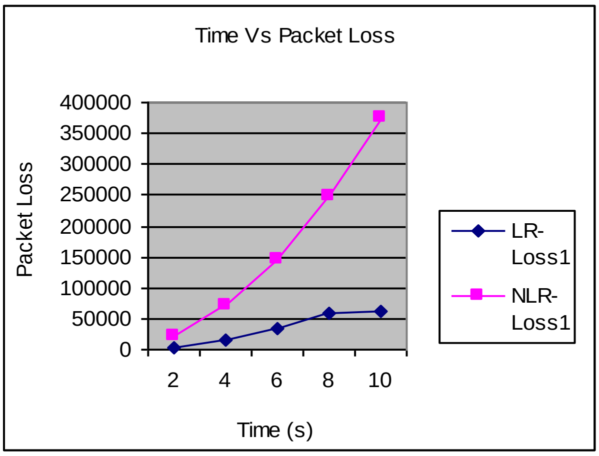


Fig. 7. Time Vs packet loss

Fig: 7 shows the packet loss at the destination. From the Fig, we can see that the packet loss is high in the NLR scheme when compared with our proposed LR scheme when varying the time.

### B. 2. Throughput (Packets Received)

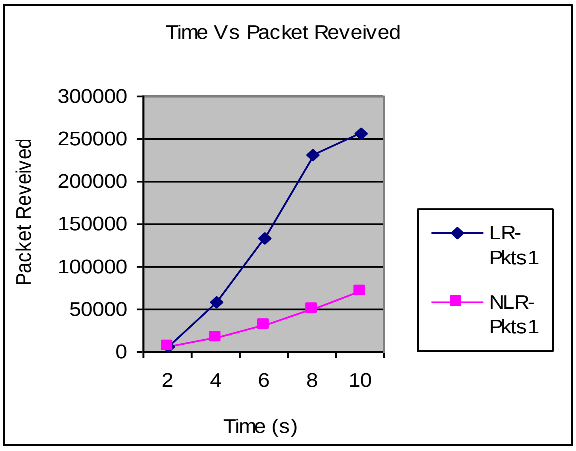


Fig. 8. Time Vs packet received

Fig: 8 gives the throughput in packets for the destinations by varying the time. It shows that the throughput is more in the case of LR scheme when compared with NLR scheme.

### B. 3. Throughput (Mbps)

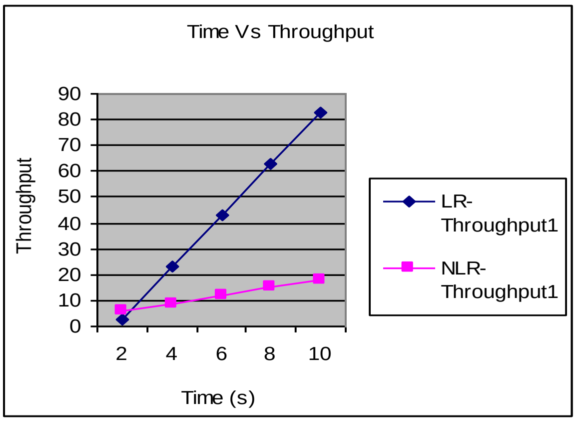


Fig. 9. Time Vs throughput

Fig: 9 gives the throughput in Mbps for the destinations for varying time. It shows that the throughput is more in the case of LR scheme when compared with NLR scheme.

## V. Conclusion

In this paper, we propose an agent based loss recovery technique for multimedia flows in IP networks. Our agent based loss recovery technique is carried out in two phases. In phase-1 (i.e.) from the source to the relay nodes, FEC encoding and decoding was performed. In phase-2 (i.e.) from the relay nodes to the destination, MDC encoding and decoding was performed. By simulation results, we have shown that our proposed approach attains high throughput with reduced packet loss when compared with our previous approach.

## References

[1] http://www.bu.edu/ece/research/information-sciences-and-systems/multimedia-transmission/

[2] Christos Bouras, Apostolos Gkamas, and Georgios Kioumourtzis, “Challenges in Cross Layer Adaptation for Multimedia Transmission,” *International Telecommunications, Networks and Systems (IADIS)*, 2007

[3] Odd Inge Hillestad, Bjornar Libak, and Andrew Perkis, “Performance Evaluation of Multimedia Services Over IP Networks,” *IEEE*

*International Conference on Multimedia and Expo (ICME 2005)*, Amsterdam, 6-6 July 2005.
[4] Mohd Farhan Ngatman, Md Asri Ngadi, and Johan M. Sharif, "Comprehensive Study of Transmission Techniques for Reducing Packet Loss and Delay in Multimedia over IP," *International Journal of Computer Science and Network Security (IJCSNS)*, vol. 8, no. 3, March, 2008
[5] Teck Kuen Chua and David C. Pheanis, "QoS Evaluation of Sender-Based Loss-Recovery Techniques for VoIP," *IEEE, Network*, vol. 20, Nov-Dec, 2006
[6] Wen Tsai Liao, Jeng Chun Chen, and Ming Syan Chen, "Adaptive Recovery Techniques for Real-Time Audio Streams," *Proceedings. IEEE, Twentieth Annual Joint Conference of the IEEE Computer and Communications Societies (INFOCOM 2001)*, Anchorage, AK, USA, pp. 22-26, April, 2001.
[7] Moo Young Kim and W. Bastiaan Kleijn, "Comparison of Transmitter-Based Packet-Loss Recovery Techniques for Voice Transmission," *INTERSPEECH - 8th International Conference on Spoken Language Processing (ICSLP)*, Jeju Island, Korea, pp. 4-8, October, 2004.
[8] Kenji Kirihara, Hiroyuki Masuyama, Shoji Kasahara, and Yutaka Takahashi, "FEC Recovery Performance for Video Streaming Services Based on H.264/SVC," *Recent Advances on Video Coding, Ed. Javier Del Ser Lorente, InTech*, ISBN 978-953-307-181-7, pp. 259-272, July 2011.
[9] Georg Carle and Ernst W. Biersack, "Survey of Error Recovery Techniques for IP based Audio-Visual Multicast Applications," *IEEE Network*, pp. 38-45, 1997
[10] Arpad Huszak and Sandor Imre, "*DCCP-based Multiple Retransmission Technique for Multimedia Streaming*," In the Proceedings of MoMM 2008, pp. 21-28, 2008
[11] Frank H. P. Fitzek, Basak Can, Ramjee Prasad, Marcos Katz, and D. S Park, "*Traffic Analysis of Multiple Description Coding of Video Services over IP Networks*," In the Proceedings of Wireless Personal Multimedia Communications, 2004.
[12] Mathias Johanson and Arne Lie, "*Layered Encoding and Transmission of Video in Heterogeneous Environments*," ACM, 2002
[13] C. Le Martret, A. Le Duc, P. Ciblat, and S. Marcille, "Analytical Performance derivation of Hybrid ARQ Schemes at IP layer," submitted for publication to IEEE Transactions on Communications, major revision in July 2011.
[14] Jiao Feng, Chi Xuefen, Peng Li, Wang Yining, Lin Guan, and Xingang Wang, "Adaptive FEC algorithm based on prediction of video quality and bandwidth utilization ratio", *IEEE International Conference on Advanced Information Networking and Applications (AINA '09)*, Bradford, pp. 26-29 May 2009.
[15] Ankit Bhamri, Florian Kaltenberger, Raymond Knopp, and Jyri Hamalainen, "Smart Hybrid-ARQ (SHARQ) for Cooperative Communication via Distributed Relays in LTE-Advanced," *IEEE 12th International Workshop on Signal Processing Advances in Wireless Communications (SPAWC)*, San Francisco, CA, pp. 26-29 June 2011.
[16] Xunqi Yu, James W. Modestino, and Dian Fan, "Joint Multiple Description Coding and FEC for Delay-Constrained Applications in Congested Networks," *IEEE Workshop on Information Theory (ITW '07)*, Tahoe City, CA, pp. 2-6 Sept. 2007.
[17] Abdullah AlWehaibi, Anjali Agarwal, Michael Kadoch, and Ahmed ElHakeem, "Hybrid FEC/ARQ Delay for DiffServ Over IP and MPLS Multicast Networks," *WSEAS Transactions on Circuits and Systems*, Issue 2, vol. 2, April, 2004.
[18] D. Melpignano, G. Convertino, and A. Vitali, "Performance Optimization for Multimedia Transmission in Wireless Home Networks," In *the Proceedings of 14th European Signal Processing Conference. European Signal Processing Conference (EUSIPCO)*, 2006
[19] Stephen Bates, Logan Gunthorpe, Ali Emre Pusane, Zhengang Chen, Kamil Zigangirov, and Daniel J. Costello Jr, "Decoders for Low-Density Parity-Check Convolutional Codes with Large Memory," *12th NASA Symposium on VLSI Design*, Coeur d'Alene, Idaho, USA, pp. 4-5, Oct., 2005.
[20] A.Veerabhadra Reddy and Dr. D. Sreenivasa Rao, "An Agent Based Approach for End-to-End QoSGuarantees in Multimedia IP networks," *(IJCSIS) International Journal of Computer Science and Information Security*, vol. 8, no. 4, July, 2010
[21] Network Simulator: www.isi.edu/nsnam/ns

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