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Optimizing Multimedia Transmission Through Throughput Delay Based Fragmentation and Priority Based Red Dropping

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Abstract: Market fusion towards different networking technology and multimedia technology witnessed a huge exigency on multimedia traffic support. Most of the applications such as video conferencing need multimedia transmission techniques to send the multimedia data from one end to another with enhanced efficiency, robustness and security. In several networks like voice network, sensor network providing delay guarantees to applications that deliver multimedia data in a strict timely fashion is mandatory to achieve good quality of service. Congestion occurrence in a network has to be resolved as it is the primary reason for loss of packet and delay which affects the quality of service in networks. In the proposed research a framework is designed for congestion control system that performs appropriate service differentiation delay and throughput based Fragmentation with proper Discarding Policy. A new family of both throughput and delay based fragmentation TDF with priority based RED dropping algorithm is proposed which reduces the packet loss as well as the delay and thereby attain quality of service in multimedia applications.

Key words: Quality of service, throughput and delay based fragmentation, priority based red, multimedia, congestion

INTRODUCTION

The demand for real time services is persistently increasing. IETF considers the requirement of QoS for loss of packet, throughput and delay as the main criteria to assess the performance of Internet Protocol (Floyd and Jacobson, 1993). The QoS should be able to handle jitter and delay sensitive applications, control the loss of packets during congestion and able to set the priorities of traffic. The parameters used to express QoS are Bandwidth which is the maximum rate in which the network can transmit data traffic, Latency, the maximum tolerable limit in delay that occurs during data transmission, Jitter, deviation among packet delay and packet Loss.

QoS is the network competence to provide a non default service to a subset of the aggregate traffic that entered the IP. The aggregation of subsets is done using Differentiated Services (Diff Serv) where the packets belonging to diverse flows are denoted in different ways to produce various classes. Packets in diverse classes receive different services. A network could be set to provide differentiated services by labelling each packet as high priority or low priority packet (Raouf *et al*., 2014). The traffic that is to be treated differently is identified through classification and marking. The methods of classification sort the packets into different traffic types by considering the profit factor of the packets and marking sets up a trust boundary. Tim Szigeti and Christina Hattingh classifiers after identifying the type of traffic that the packet is carrying provides preferential or deferential treatment based on the requirement.

The performance degrades with the rise in traffic which leads to congestion and as a result encounters packet loss. Packet loss affects packet delivery ratio which in turn affects multimedia packet orderliness. Similarly long delay affects timeliness, reduction of node throughput, dropping of overall throughput of the network and thereby affecting the quality of service in network. Hence, to achieve good QoS, effective congestion control scheme with sufficient bandwidth and Delay variation is needed.

Multimedia data traffic has diverse QoS requirements. For example transmission of Video data is tolerable to packet loss as a packet can be decoded partly even when all fragments are not received, but very sensitive to delay. Hence, congestion control mechanism for improving QoS in multimedia traffic takes into account several factors like

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balance between QoS parameters, deadline of multimedia packets, delay, packet loss, size of packets and priority.

The standard schemes for fragmenting packets usually shed a packet if all the corresponding fragments of it are not properly received .When the size of the fragment is kept very large the overhead bits for every packet is reduced and when the size of the fragment is kept very small transmission error rate is reduced (Kambhatla *et al*., 2012). As there is a trade off between the amount of overhead bits and transmission error best possible packet size is calculated to stay balanced. The algorithm Priority based RED is applied that discriminates the packets by considering both size and priority such that the probability of dropping multimedia packets is less compared with other traditional data packets. QoS assurance concerning bandwidth, delay and jitter is required for guaranteed flows. Two different types of QoS requirements are considered for guaranteed flows Bandwidth guarantees and Delay guarantees. The proposed system considers guaranteed flows where QoS parameters delay and throughput are considered and enhances the packet fragmentation algorithm by considering those two parameters to reduce the packet loss as well as the delay and attain guaranteed quality of service in multimedia applications.

Literature review: Static priority algorithm (Wang *et al*., 2004) schedule the packets based on QoS Parameters. Packets are given priorities and they are sorted based on that priority. Service differentiation is done depending upon the transmitting data type. Packet dropping is done based on buffer overflow. Sorting of packets is done based on their priority when congestion happens and packet with highest priority will be placed in the head of queue. Drops low priority packets in case of buffer overflow. It creates overhead due to sorting and leads to starvation of low priority packets. Buffer size should be kept optimum. Also consideration of deadline is not done.

Static priority with deadline considerations (Dag, 2007) integrates delay with classic SP (Static Priority) algorithm and improves the fairness compared to static priority algorithm. Sorting of packets is done based on both factors deadline and priority. It performs degree sorting. Service differentiation is done depending upon the transmitting data type. Packet dropping based on buffer overflow and deadline violations. However it considers only fixed priorities. Degree of sorting and buffer size should be kept optimum (i.e., if buffer size is less, overflow occurs and if buffer size is more, deadline violations occur).

Timeliness and QoS aware packet scheduling (Lien and Wun, 2009) algorithm improves the network throughput and fairness. Packets are assigned a profit function and forwarding is done based on that profit. Service differentiation is done with respect to the QoS Class whether streaming , Non real time or conversational. Packet dropping is done based on buffer overflow. However consideration of deadline is not done.

In streaming media over Internet there is a possibility of losing packets randomly. Network adapted selective frame dropping algorithm (Huo *et al*., 2007) addresses this problem. Before sending Group of Pictures the results of preceding hop is considered for predicting the playing window. If frames could not be played within playing window then they are dropped. This algorithm is presented with reduced computational complications and improved performance for real-time applications than other video adaptation methods. However Video traffic alone considered and fairness of other data is not considered.

Optimal prioritized packet fragmentation algorithm improves the quality of Pre encoded H.264 bit streams. Prioritization is done with respect to their CMSE (Cumulative mean square error) contribution towards video quality. Maximum goodput is attained with dropping of low priority frames. However consideration of deadline is not done and fairness of other data not considered.

In SBT, Size Based Treatment (Dimitriou and Tsaoussidis, 2008) packets of different sizes are assigned with different probability which is considered while dropping. SDP (Dimitriou *et al*., 2010) algorithm increases the quality on real time applications by distinguishing various flows like time sensitive and time insensitive by considering the size of the packet. Multimedia data corresponds to small size that experience less dropping probability. It considers size of the packet as important parameter for computing the dropping probability. Consideration of deadline is not done.

Unequal Loss Protection (ULP) scheme (Zhang and Peng, 2009) improves the received video quality as measured by PSNR for the environment that experiences high packet loss rate.Compared with Equal Loss Protection scheme this achieves higher PSNR with less redundancy for H.264/AVC video transmission.

In some applications of multimedia streaming, even a single packet drop may result in ineffective delivery of the entire sequence. This may be due to the interdependency among the packets. Proper buffer management algorithm (Scalosub *et al*., 2013) is designed to exhibit a model which acquires that interdependency among the packets and devise methods to discard a packet, thereby avoid goodput degradation. However consideration of deadline is not done.

Applications with diverse characteristics have diverse necessities. Hence equal treatment of their packets should not be done. For example traditional data flows are loss intolerant but delay tolerant where as real time data flows are time sensitive and delay intolerant. Moreover retransmission of lost packets for real time applications may not be practicable.

Various methods are used to perform Service Differentiation. Based on the transmitting data type say video or Email based , based on QoS class (conversational, Streaming, non real time), based on the CMSE values, based on deadline, based on size of packets.

Service differentiation could be implemented at the input port. Based on loss tolerance and real-time properties each input-queue provides a separate buffer for each of the service classes. Static priority scheme favors high priority traffic, whereas dynamic priority scheme provides fairness to all classes.

Less Impact Better Service (LIBS) provides service differentiation by considering the parameter delay caused by the packets. Non Congestive Queuing (NCQ) (Mamatas and Tsauossidis, 2009) gives priority to small packets and make use of service thresholds to limit the effect of delay on congestive applications. Both Less Impact Better Service scheme and NCQ comply with several users with various demands on both delay and throughput.

Non Congestive Queuing NCQ+ (Papastergiou *et al*., 2011), improved version of NCQ uses a prioritization scheme based on their impact on total delay. Few applications may require only little service time but when considering their total service time it may cause unnecessary delay in the queue. Those applications which are of tiny sizes and lesser transmission rate would not be part of the cause of congestion. NCQ+ supports those applications whose influence on other applications is not important. It promotes non congestive applications by effective scheduling and proper allocation of resources. It enhances the application performance as long as there is no violation of guaranteed services and other flows impact on the performance is insignificant.

Various congestion control algorithms like Drop Tail, Choke, RED, BLUE and REM were studied. Drop Tail, a simple algorithm that shed packets from end of the buffer if the buffer is full. Its decentralized nature makes it suit to heterogeneity. However the limitations of this algorithm are drop tail allows dominance of few flows over other flows that lead to starvation in the queue. Drop Tail signals congestion only when the queue has become full through packet dropping. Also fairness is not considered, no relative QoS. It doesn't suit for priority based transmission.

Choke algorithm is a simple stateless algorithm that shed packets randomly from the buffer if the buffer is full. The dropped packet and the newly arrived packet are compared to check whether they belong to the same flow. If so new packet is also dropped else assign some probability to that packet and add to the buffer. The probability is calculated similar to RED. However it doesn't work well when there is a large number of a flow than buffer space.

Blue Algorithm, an effective algorithm where packet loss gets reduced, provides data about the number of competing connections in the network containing shared links and ensures better link utilisation.

Random Exponential Marking Algorithm (REM) (Athuraliya *et al*., 2001) is another congestion control algorithm which also provides better link utilization, reduced packet loss and delay. The limitations are certain variable value has to be fixed and must be known in general.

RED algorithm (Floyed and Jocobson, 1993) not only minimizes the loss of packets and the queuing delay but also sustain good link utilization. Equation 1 computes average queue length using parameters minth and maxth which are considered to be two threshold values.

$$Q_{avg} = (1 - W_q) * Q_{avg} + W_q * Q \tag{1}$$

Where:

Q_{avg} = Average queue length
Q = The current queue length and
Wq = A weight parameter that takes value in the range (0- Wq - 1)

RED remains in any one of the three states based on the threshold value. It remains in normal state when the average length of the queue is lesser than the minimum threshold value (minth) and in this state it accepts all the arriving packets. It remains in the congestion control state when the average length of the queue is greater than the maximum threshold value (maxth) and in this state it discards all incoming packets. It remains in congestion avoidance state when the average length of the queue lies between minth and maxth .In this state it discards packets with a certain probability which is computed using several parameters including threshold values. The proposed system focuses on packet dropping in case of congestion by considering packet size, deadline and priority of the packets.

MATERIALS AND METHODS

In the proposed system the author focuses on size based differentiation and considers throughput and delay based packet fragmentation for optimal multimedia data delivery. The most distinctive indicator which also facilitates to obtain the type of application that created the packet is Packet size .We cannot simply consider the differentiation of packets as small and big in real life similar to binary classification as 0 and 1. Packets of size 150 and 500 byte are lesser when compared to 800 byte packet however treatment of those two packets should not be done equally. Hence packets should be differentiated relatively with the parameter size of the packets which are served by the router already. For example if a router serves packet of size 1000 byte then packet of size 250 is considered to be small for it .Similarly if a router serves 100B packets then packet of size 1000 bytes is considered to be large. Using static threshold value for differentiation would also result in undesirable behaviour. As the overhead bits for every packet gets reduced if the size of the fragment is kept very large and transmission error rate gets reduced if the size of the fragment is kept very small best possible packet size is calculated to stay poised with the above mentioned factors.

In the proposed system the size of the packet for which most of the packets got delivered in the destination with reduced delay is the best possible packet size which is computed using optimization techniques. The discarding policy is based on this packet size, deadline associated with the packet and packet priority.

TDF-Throughput and delay based fragmentation: Throughput is the quantity of data that have been successfully transferred from source to destination in a particular amount of time. Equation 2 gives the formula for obtaining throughput derived from (Rani and Suganthi, 2014) is given as:

$$T = \frac{nf + \frac{\min\left(f_1\left(x - n\times f\right)\left(s-h\right)\right)}{S}}{tot - data} \qquad (2)$$

Where:

T = The throughput
x = Channel rate/bandwidth
n = Percentage of other data
s = Optimal packet size
h = Header size of packet
tot-data = Total data (multimedia data + other data)
f = encoding rate of multimedia data

Delay, an important QoS parameter to assess the congestion level within the network is given by

$$D = d_p + d_q + d_t + d_{pp} \qquad (3)$$

Where:

d_p = the processing delay
d_q = Queuing delay
d_t = Transmission delay and
d_{pp} = Propagation delay

Transmission delay is the amount of time required to transmit the entire the packet's bits into the link. It depends on the length of the packet. Equation (4) denotes the transmission delay:

$$d_t = \frac{pkt - size}{x} \qquad (4)$$

Where:

d_t = The transmission delay
x = Channel rate and
pkt_size = The size of the arriving packet

Propagation delay is the time taken by the packet to propagate from the source to the destination It is determined by the distance from the sender to the receiver and does not depends upon the length of the packet and the rate of transmission . Equation (5) denotes the propagation delay:

$$d_{pp} = \frac{L}{P_s} \qquad (5)$$

Where:

L = The length of medium and
P_s = The propagation speed

Processing delay is the amount of time taken to process the header of the packet . When compared to other delay types the processing delay is considered to be least important and mainly ignored in common. Only in few systems where encryption techniques or modification of contents in the packet are processed, the processing delay can not be ignored as it is quite large .

The packets will be processed after its arrival into the router and then transmitted. If the processing speed of the router is slower than the rate of packet arrival then the router puts them into the queue until it can get around to transmit them. When the queue starts filling up delay a packet faces gets increased. The processing speed of the router depends upon the transmission rate. Queuing delay is the time a packet remains in a queue until it can be transmitted. It depends on several factors like load in the network, service disciplines followed in the network. Reducing queuing delay can be done by the appropriate implementation of scheduling algorithms, utilising proper bandwidth through perfect reservations and meeting the requirements of traffic sources. Equation 6 denote the queuing delay:

$$d_q = \frac{T_{serv}}{N} \tag{6}$$

Where:

Tserv = The service time
N = The number of packets transmitted

$$N = \frac{tot\_data}{s-h} \tag{7}$$

Equation 8 id derived from Little's Law :

$$T_{serv} = \frac{q}{r} \tag{8}$$

Where:

q = The capacity of the queue to hold packets and
r = the rate at which packets arrive

Substituting Eq. 7 and 8 in Eq. 6:

$$d_q = \frac{q*(s-h)}{r*(f+nf)} \tag{9}$$

The delay components contribution can differ considerably. For example, d_p is considered to be negligible and d_{pp} is fixed value. The correlation between the total queuing delay and the load incrementing was considered. Queuing delay is one of the most effectual control parameter that shapes the behavior of total delay.

Substituting the values of various delay components the total delay is computed as:

$$D = \frac{(s+(1*X))*\begin{pmatrix} r*(f+n*f)+ \\ (x*(q*(s-h))) \end{pmatrix}}{x*(r*(f+n*f))} \tag{10}$$

Techniques that reduces both computation time and materials /resource consumption to achieve optimum solution for a given problem is Optimization techniques .It finds optimum solution to several problems than other analytical methods in a reasonably less computation time. Several optimization techniques use mathematical and Meta heuristic algorithm. The latter emerged as effective tool for attaining optimization.

The proposed system uses optimization technique to derive the optimal packet size. The best possible packet size 's' which maximizes the throughput parameter T and minimizes the delay parameter D is obtained using optimization techniques Improved Cuckoo Search (Valian *et al*., 2011), Modified Particle swarm Optimization (Zhu, 2009) and Improved Chaotic Bat Algorithm (Raouf *et al*., 2014) which considers the delay parameter for its quick convergence. This parameter s ' is used in packet dropping algorithm which is described below.

Priority based RED Algorithm: Packet dropping is done by Priority based RED algorithm which differs from SDP and Modified RED (Rani and Suganthi, 2014) where dropping of the packets are done depending upon not only the deadline and optimal size but also considers the parameter priority.

The threshold values minth and maxth are initialised.
The average queue size q_ avg is calculated as General RED algothrim.
If min = q_avg then none of the packets are dropped.
Else if minth =q_avg > maxth calculate dropping probability and mark it for the arriving packet.
Dropping probability Pa is given by

$$Pa = \frac{pb}{1 - count * pb} \tag{11}$$

Where:

$$pb = \frac{\max p * \dfrac{q_{-avg} - minth}{\max th - minth} \times}{\dfrac{pkt\_optsize}{pkt_{size} \times Priority \times Deadline}} \tag{12}$$

Also if q_avg < minth then pb=0 and if q_avg>maxth pb=1;
q_avg is the average size of the queue
count is the number of packets from the packet that has been lastly marked.
pkt_optsize is the size of the packet calculated as optimal for multimedia data.
pkt_{size} is the size of the packet that has arrived.
minth is the queues minimum value for threshold
maxth is the queues maximum value for threshold
maxp - Maximum pb value
pa – probability of marking the current packet
If max is less than q_avg then all the arriving packets are dropped.

Continue from step 3-6 and stop if all the packets got transmitted. Multimedia fragments with minimum deadline are given higher priority. As Multimedia packets are assigned higher priority dropping probability becomes less and dropping of multimedia packets gets reduced.

RESULTS AND DISCUSSION

Simulation was carried out using NS2 to prove that our fragmentation algorithm reduces the delay in multimedia transmission and yields maximum throughput (Wiegand *et al*., 2003) bit streams are used for simulation.

Throughput and delay are calculated for different encoding rates with channel rate fixed to 2Mbps. It was observed the best possible size obtained for each encoding rate produced maximum throughput with minimal delay. Packet drop is simulated using Priority based RED algorithm and the results are compared with two existing methods SDP and Modified RED.

The probability of dropping packets to assess the packet loss rate is computed. Let probability of dropping packets be pdp (RED) and pdp (PRED). Consider the case if the arriving packet size is not same as optimal packet size:

$$\text{pdp}(\text{RED}) = \text{red}_{\text{drop}} \tag{13}$$

$$\text{pdp}(\text{PRED}) = \text{red}_{\text{drop}} * \frac{\text{pkt}_{\text{optsize}}}{\text{pkt}_{\text{size}}} * \frac{1}{\text{priority}} \tag{14}$$

$$\text{Impact} = \text{pdp}(\text{RED}) \sim \text{pdp}(\text{PRED}) \tag{15}$$

Substituting Eq. 13 and 14 in Eq. 15:

$$\text{Im pact} = \text{red}_{\text{drop}} * \frac{\text{pkt}_{\text{optsize}}}{\text{pkt}_{\text{size}}} * \frac{1}{\text{priority}} \tag{16}$$

Consider the case if the arriving packet size is same as optimal packet size. Then, Eq. 16 becomes:

$$\text{Impact} = \text{red}_{\text{drop}}\left(1 - \frac{1}{\text{prioirty}}\right) \tag{17}$$

Equation 16, shows that the packet loss rate of packets not optimal sized is a function of arriving packet size, optimal packet size and priority. Smaller size values of such packet compared with optimal size with lower priority suffers packet drops but with higher value for priority signify less packet drops comparatively to RED.

Equation 17 on the other hands shows that the loss rate of optimal sized packets is either decreases or equals but doesn’t increase more than RED. The loss rate decreases for optimal sized higher priority packets and same as RED for optimal sized lower priority packets. It is also observed that the delay is substantially reduced in Priority based RED when compared with other two methods (Table 1 and Fig. 1- 4).

Table 1: Optimal packet size by TDF computed using ICS, MPSO and IBACH algorithm

Bit rate (in Kbps)	Estimation of ideal packet size(in bytes)
384	290
512	304
640	340
768	382
896	346
1024	296
1152	268
1330	270
1408	182
1536	156
1664	116

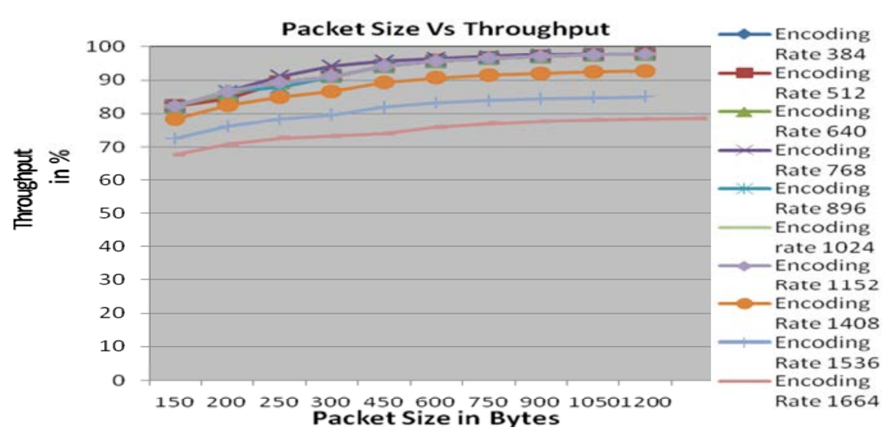

Fig. 1: Effect of packet size on throughput

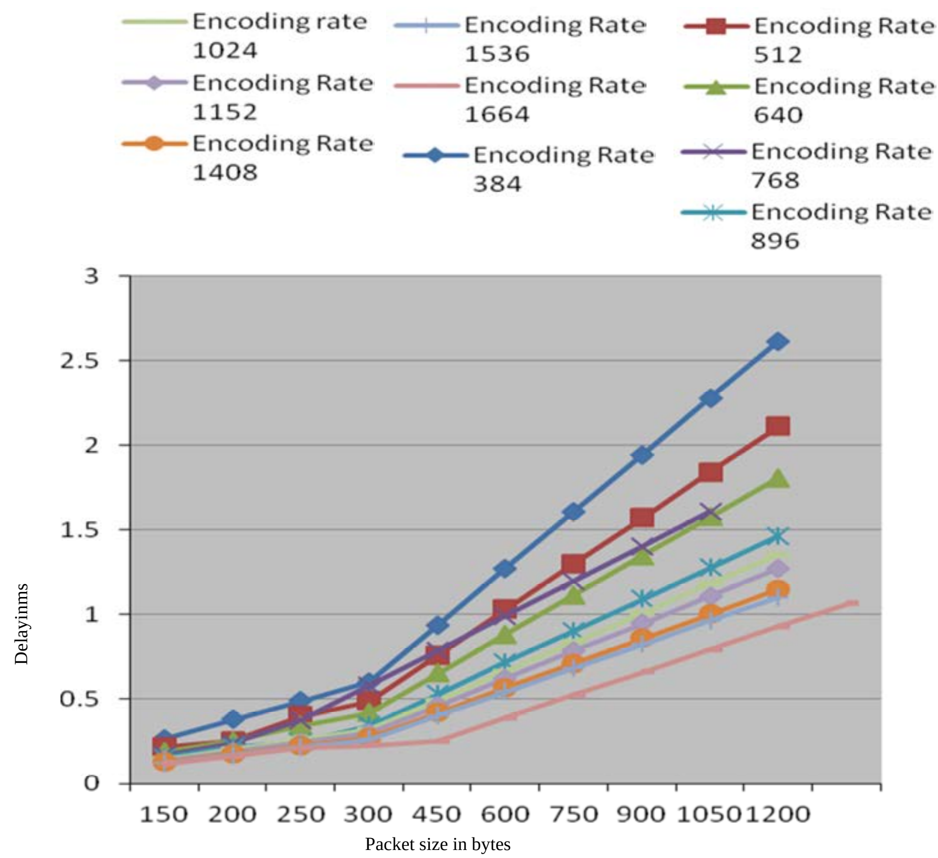

Fig. 2: Effect of packet size on delay

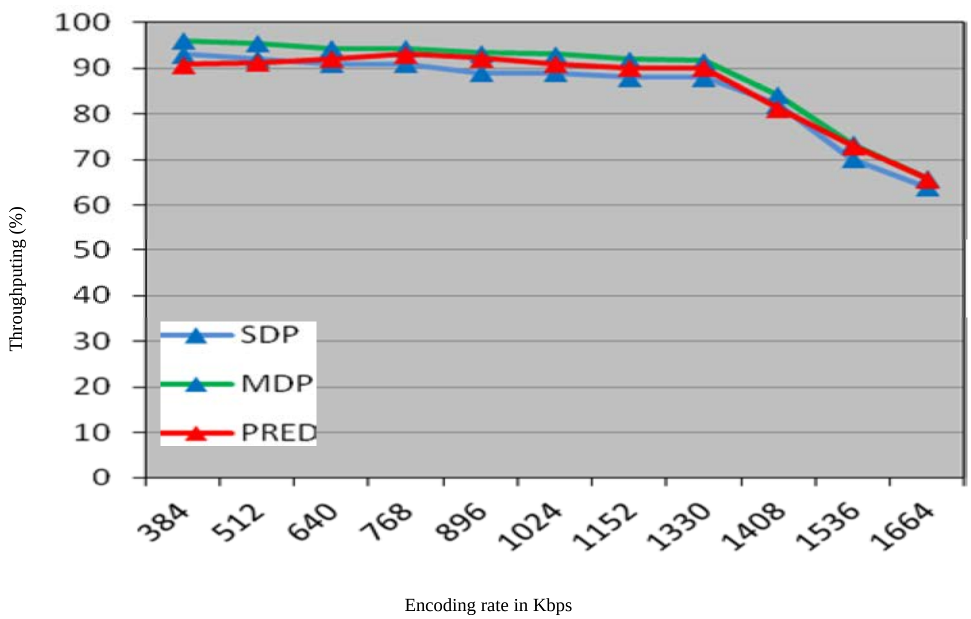

Fig. 3: Throughput comparison of SDP, Modified RED and priority based RED

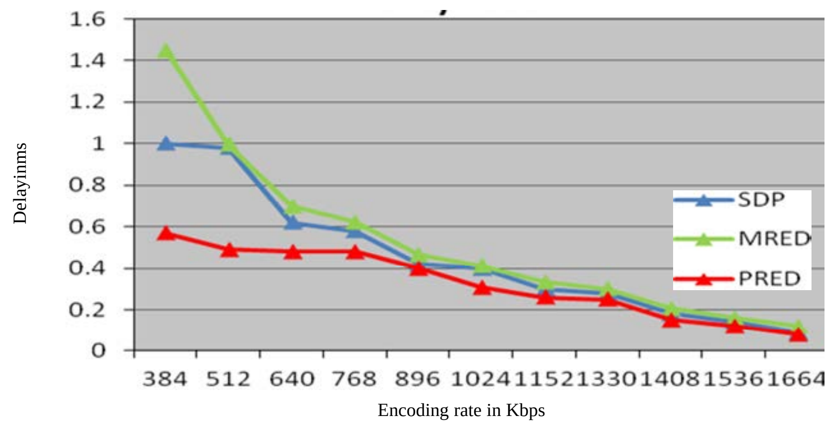

Fig. 4: Delay comparison of SDP, modified RED and priority based RED

CONCLUSION

The system concentrates on throughput and delay based fragmentation to obtain optimal fragment size for different encoding rates .It is done using Improved Cuckoo search, Modified Particle Swarm Optimization and Improved Chaotic Bat algorithm. Using priority based RED the packets are dropped considering deadline, priority and size of the packet. Simulation results show that integration of well suited service differentiation and proper dropping policy improves throughput and also minimizes the delay compared with SDP and Modified RED. It is proved that if the bit rate does not go beyond the bandwidth that is available the packets fragmented into computed ideal packet sizes for different bit rates provides maximum throughput and minimum delay thus enhancing the quality of the received video. Future work is to incorporate suitable scheduling mechanism and multiple queuing principles instead of dropping mechanism to minimise packet drops.

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