

IMPROVING QUALITY OF SERVICE IN IP NETWORKS FOR MULTIMEDIA APPLICATIONS WITH OPTIMAL FRAGMENTATION

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ABSTRACT

The tremendous increase of Internet users and the wide ranging need to support both multimedia traffic and other traffic leads to service differentiation schemes. Multimedia applications have uncompromising and strict deadlines and the packets are worthless if they arrive late than their associated deadlines. The aim of the proposed system is to improve the quality of service by increasing the number of multimedia packets that reaches the destination before their associated deadline without affecting the fairness of other data. The proposed algorithm estimates the optimal packet size for transmission and puts forward a Service Differentiation scheme that drops the packets based on the size and deadline of the packets. Packet loss percentage is observed to be reduced in the proposed system.

Keywords: Optimal Packet Size, Packet Loss, Deadline, SDP, Modified RED

1. INTRODUCTION

The magnificent increase of Internet users has flooded the Internet with diverse applications. The applications contribute heterogeneous data which comprise of real time data like voice, video and traditional data like e-mail, file transfer to the internet. The internet traffic has different QoS requirements. QoS parameters include data rate, delay, jitter, ordered delivery, loss, error rates (Steinmetz and Nahrstedt, 2004). Service principle called Less Impact Better Service (LIBS) (Mamatas and Tsaoussidis, 2010) leads to a novel service differentiation method, namely Sizeoriented Dropping Policy (SDP), which uses packet size to categorize time-sensitive from delay-tolerant flows and prioritize packet dropping probability, accordingly. A significant increase on the perceived quality of real-time applications was perceived.

For example the real time data flows are intolerant to delay, packet loss and jitter. Traditional data flows are intolerant to packet loss but tolerant to delay and jitter. The flows are aggregated in diffserv model where packets are classified as belonging to a flow if they have the same

marking in Type of Service (ToS) byte in the IP header. Scheduling disciplines give priority to QoS sensitive flows in routers but also maintains the fairness of non-priority queue (Bourenene *et al.*, 2008).

More packet loss in the case of multimedia application degrades the quality of the video and voice at the receiving end.

The proposed system focuses on improving the perceived quality of picture and sound by increasing the number of multimedia packets that reaches the destination before expiry of their deadline. Multimedia data are differentiated by the size. Small packets usually correspond to the real time applications like video and voice over internet protocol and bulk data applications use big packets (Dimitriou *et al.*, 2010). The proposed system optimizes Sizeoriented Dropping Policy (SDP) by incorporating the following:

- Calculation of optimal packet size of multimedia data thereby increasing the number of packets that reaches the destination on time
- Dropping packets based on size as well as remaining deadline

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The study is organized as follows:

- Section 2 discusses the related work
- Section 3 explains the proposed system
- Section 4 compares the packet loss percentage of SDP and proposed system

2. RELATED WORK

Multimedia applications such as video streaming, which are delay sensitive and bandwidth sensitive are growing rapidly over internet. Video packets can tolerate some packet loss. Lost video packets induce different levels of quality degradation.

Service differentiation has been developed on the basis of resource allocation with the corresponding service requirements of diverse applications. Applications with strict delay, jitter and packet loss constraints can be satisfied by prioritizing real-time data over bulk data transfers (Georgiou *et al.*, 2011).

Throughput of multimedia applications can be improved by fragmenting multimedia packets into optimal fragments. Fragmentation of packet calls a trade off between reducing the number of overhead bits per packet by using large fragment size and reducing transmission error rate by using small packets (Kambhatla *et al.*, 2012). Hence multimedia data has to be differentiated from other data that can be done using packet size. The diffserv approach enables prioritization by marking with corresponding service identifiers. (Bourenene *et al.*, 2008; Dimitriou *et al.*, 2010)

Each packet has a deadline associated with it. If packets belonging to real time applications are delayed beyond deadline, those packets are considered as lost since those packets will be discarded at their destination (Li, 2009). Non-Congestive Queuing Plus (NCQ+) promotes applications that require comparatively small service times (multimedia applications), as long as their total service times cause insignificant delays to other packets in the queue. The algorithm provides delay guarantees to applications that do not contribute to congestion because of their tiny packet sizes and low transmission rates. The packets are prioritized and in turn corresponding flows, according to their impact on total delay. (Georgiou *et al.*, 2011) introduced the mechanism based on the identification of high bandwidth from the drop history of Random Early Detect (RED). Bodin *et al.* (2000) proposed Weighted RED that calculates a separate average length for the high priority packets preventing starvation of the low priority packets. These methods demand costly memory structures.

(Dag, 2007), introduced static priority with deadline consideration, where the packets are given priorities based on the data type with multimedia data having high priority. If a network node becomes congested 'k' packets of the queue are sorted according to their priorities with the highest priority packet at the head of the queue. A packet with no remaining deadline is dropped. But sorting creates more overhead and time consuming.

Dimitriou *et al.* (2010), introduced Sizeoriented Dropping Policy (SDP) which derives the dropping policy from RED. RED drops the packets on calculated probability. Maximum threshold defines the region where forced dropping occurs. Minimum threshold defines the region where forced dropping occurs (Floyd and Jacobson, 1993). SDP records the size of each incoming packet to calculate the average packet size which serves as a guideline to differentiate small and big packets. If the arriving packet exceeds the average size (Big packets) it is dropped with RED probability, otherwise (for multimedia data) new probability will be calculated based on the size of the packet. In SDP gateways small packets experience less dropping probability than big packets compared to RED.

3. PROPOSED SYSTEM

The pseudocode of the proposed system is as follows:

1. Calculate the optimal packet size(opt_pkt_size) for the given encoding rate. The multimedia packets are split into the size calculated as opt_pkt_size.
2. In each node
Calculate the travel time(travel_time) from current node to destination from the routing table.
for each packet arrival
 if remaining_deadline < travel_time then
 drop the packet
 else
 if pkt_size ≤ opt_pkt_size
 drop_prob = RED_drop
 else
 drop_prob = OPT_drop
 end if
 end if
end for
3. Drop the packets using modified RED.

3.1 Estimation of Optimal Packet Size

3.1.1. Input Parameters

Number of intermediate nodes

Buffer size
 Bandwidth
 Encoding rate (Size of the frames/second in Kb).
 Percentage of other data
 Encoding rate

3.1.2. Fixing the Threshold Values

Minimum threshold- $\frac{1}{4}$ * buffer size.
 Maximum threshold- 3* Minimum threshold

3.1.3. Optimal Packet Size

The packet size for which maximum amount of input data reaches the destination in minimal time is taken as optimal packet size(opt_pkt_size). The optimal packet size is estimated using exhaustive search.

1. The optimal packet size (opt_pkt_size) is calculated by considering the following parameters:
 - a. Available channel rate
 - b. Percentage of other data
 - c. Encoding rate

Throughput(Th) of multimedia data is estimated as follows Equation (1):

$$Th = \frac{\left(\frac{ch_rate - other_data}{p-h}\right)(p) + other_data}{N} \quad (1)$$

p = Optimal fragment size in bytes
 ch_rate = Channel rate in kbps
 oth_data = Other data rate in kbps
 enc_rate = Encoding rate in kbps
 N = enc_rate + other_data(Total data)

The optimal size is determined as the size for which maximum data reach the destination in minimum time. In Equation 1 channel rate available after occupied by other data is divided into equal size fragments of multimedia data. It is added with other data. Throughput is calculated as the ratio of total data sent at the source side to total data received at the destination side.

Number of fragments that will be discarded can be estimated initially as follows (i.e.,) the multimedia packets that can be accommodated in the available channel rate will be discarded. The number of fragments discarded is given in the following Equation (2):

$$Frag - discard = \frac{enc - rate - other - date}{p} \quad (2)$$

3.2. Packet Drop Based on Deadline

The proposed algorithm calculates the travel time to the destination with the routing table information. If the estimated travel time of a packet is less than the remaining deadline the packet is dropped. This is implemented in 1st node as well as in all nodes. This gives way for more useful packets (Packets with remaining deadline).

3.3. Packet Drop Using Modified RED

Packet size is the most typical and easy to extract indicator of the type of application that created the packet (Dimitriou *et al.*, 2010). Transmission delay of a packet is proportional to the size of the packet. Small packets are utilized by the applications that require faster delivery, limited packet losses. Each packet has a deadline associated with it and the multimedia packets have to reach the destination before the expiry of the deadline. In case of congestion the packets are dropped based on the packet size and deadline. Optimal size packets have less dropping probability thereby reducing the packet drops of multimedia data.

Minimum threshold- $\frac{1}{4}$ * buffer size.
 Maximum threshold-3* Minimum threshold

When the average queue size is less than the minimum threshold, no packets are dropped. When the average queue size is greater than the maximum threshold, every arriving packet is marked. If the queue size is between the minimum and maximum threshold the packets are dropped with modified RED probability.

3.3.1. Modified RED Algorithm

3.3.1.1. Initialization

Avg = 0
 Count = -1
 for each packet arrival
 calculate the average queue size(avg)
 if $min_{th} \leq avg < max_{th}$
 calculate probability p_a Equation 3 and 4:

$$p_a = \frac{P_b}{1 - count * p_b} \quad (3)$$

Where:

$$p_b = \max_p \left(\frac{avg - min_{th}}{max_{th} - min_{th}} \right) \quad (4)$$

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    Mark the arriving packet with probability pa
end if
count = 0
end for

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3.3.2. Drop Function of RED

$$p_b = \begin{cases} \frac{\text{avg} - \text{min}_{th}}{\text{max}_{th} - \text{min}_{th}} \cdot \text{max}_p & \text{if } \text{min}_{th} < \text{avg} < \text{max}_{th} \\ 0 & \text{if } \text{avg} < \text{min}_{th} \\ 1 & \text{if } \text{avg} > \text{max}_{th} \end{cases}$$

3.3.3. Drop Function of Modified RED

$$p_b = \begin{cases} \frac{\text{avg} - \text{min}_{th}}{\text{max}_{th} - \text{min}_{th}} \cdot \text{max}_p & \text{if } \text{min}_{th} < \text{avg} < \text{max}_{th} \\ 0 & \text{if } \text{avg} < \text{min}_{th} \\ 1 & \text{if } \text{avg} > \text{max}_{th} \end{cases}$$

3.3.4. Notations

3.3.4.1. Saved Variables

Avg = Average queue size
 Count = Packets since last marked packet
 opt_pkt_size = Calculated optimal packet size for multimedia data.
 pkt_size = Size of the arriving packet.

3.3.5. Fixed Parameters

Minth = Minimum threshold for queue
 Maxth = Maximum threshold for queue
 Maxp = Maximum value for pb

3.3.6. Other

pa: Current packet-marking probability

4. EXPERIMENTAL RESULTS

Videos pre-encoded using H.264/AVC is used to adopt the coding structure of Group of Pictures (GOP). GOP structure specifies the order in which intra- and inter-frames are arranged. A GOP contains the following picture types: I frame, P frame and B frame. Every GOP starts with an I-frame followed by a series of P-frame and B-frame. The typical Group of Pictures

(GOP) structure is IBBPBBP... I-picture or I-frame (intra coded picture)-reference picture, which represents a fixed image and which is independent of other picture types. 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 (Peng and Zhang, 2010).

According to the inter frame dependency relationships it can be derived that B-frame could be dropped at any time. When a P-frame is dropped all other P- and B-frames following this frame should be dropped. When an I-frame is dropped that all other P- and B-frames following within the same GOP should be dropped. Hence I-frame has the highest priority followed by P-frame and B-frame.

The simulation is performed using NS2. Distance vector routing algorithm is used where the vector contains distances to all nodes in the network so that the packet's travel time could be calculated tentatively.

The video is assumed to be encoded in H.264/AVC and the frame size is taken as 16 KB. Each frame has I, B and P type frames.

4.1. Steps

1. Search the 2 consecutive I frames and estimate the play window by subtracting the time stamp values of the I-frames of two consecutive GOPs for given frame rate.
2. Fragment the packets into optimal size where each types of frame (I,B,P) is grouped as different set of fragments
3. Assign weight to each fragment based on priority. I-frame has the priority 3, P-frame has the priority-2 and B-frame has the priority-Weight of each fragment is priority*(Travel_time/deadline) which will change dynamically during travel
4. Mix with 50% of other data
5. Sort the fragments according to weight within each GOP with other data at the end of each GOP. The structure of GOP is IPBBPBBPBBPBB. A low motion clip is assumed where I frame constitutes 50%, P frames 25% and B frames occupy 25% of the total frame space
6. In each intermediate node check
 - if entire GOP could be sent before deadline
 - send the GOP without dropping any fragment
 - else if I fragments and P-fragments could be sent
 - drop B frames
 - else if I-fragments could alone be sent

- drop P frames and all B and P frames following it in the GOP
- else
 - drop the entire GOP
 - (Travel time is calculated from the distance vector routing table).
- 7. In all the cases of step 6 send other data following each GOP without dropping.
- 8. Continue step 2-7 for different fragment sizes for the given frame rate.
- 9. Continue steps 1-8 for different frame rates given in **Table 1**.
- 10. Calculate the weighted throughput multiplying weight of fragment type (I,B,P) with the number of fragments of each type received.

Table 1 shows the determined optimal packet size for different encoding rate with channel rate 2 Mbps and other data constituting 50% of the original data.

The simulation result for throughput for different encoding rates is given as graph in **Fig.1**. Channel rate is assumed to be 2 Mbps for different encoding rates. **Figure 2** compares the throughput of SDP and modified RED for some encoding rates.

It is observed from **Fig. 1** that maximum throughput is gained with optimal fragment size for each encoding rate. For example for encoding rate 640 kbps maximum

throughput is achieved with fragment sizes ranging from 250 to 480 bytes and for 896 kbps 200 to 400 bytes. But the maximum fragment size is used to avoid overhead. But for higher encoding rates the throughput is observed to be low because the data rate exceeds the channel rate leading to more fragment drops. When fragment size becomes high the throughput decreases proportionally because of the excess padding of data in each fragment. For example if the data to be fragmented is of size 1700 bytes. If the fragment size is 300 bytes the data may be split into 6 fragments with the 6th fragment padded with 100 bytes. But if the fragment size is 1500 bytes the data is split into 2 fragments with 1300 bytes padded in the 2nd fragment. With fragment size 300 bytes only 100 bytes extra has to be transmitted but in 1500 bytes fragment size 1300 bytes of extra data has to be sent.

The simulation results for packet drop using modified RED is given graphically in **Fig. 2**.

It is observed from **Fig. 2** that packet drop was less in modified RED than in SDP.

The throughput is affected by the proportion of other data added with video data. The simulation result is shown as graph in **Fig. 3**.

From **Fig. 3** it is observed that when more other data is added the data rate reaches the channel rate leading to more fragment drops due to deadline violations.

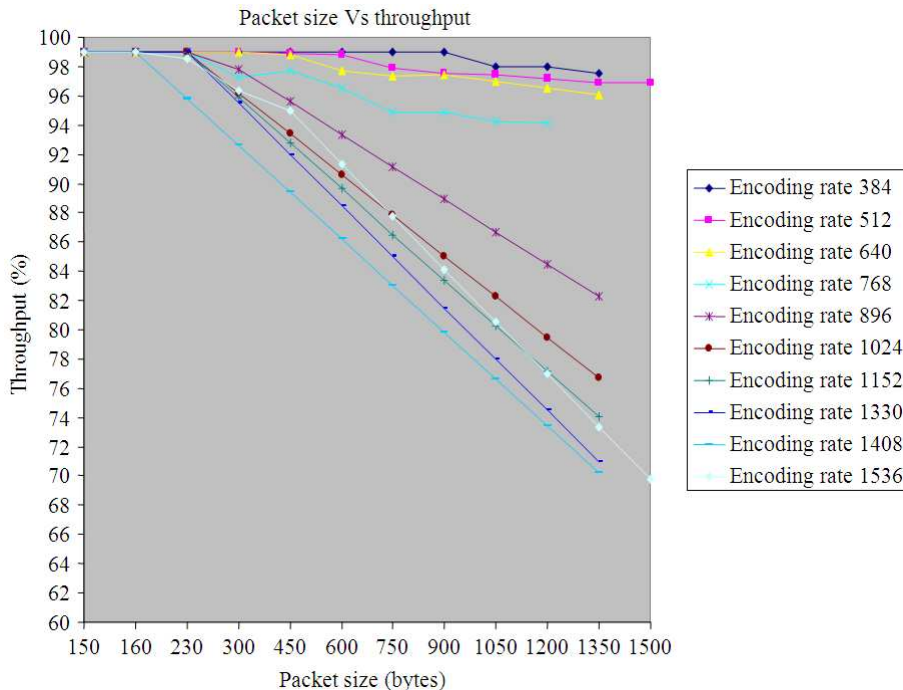


Fig. 1. Packet size vs throughput for different encoding rates

Table. 1. Optimal packet sizes for different encoding

S. No	Frames/second	Encoding rate (in kbps)	Optimal packet size (in bytes)
1	3	384	680
2	4	512	582
3	5	640	480
4	6	768	482
5	7	896	400
6	8	1024	382
7	9	1152	332
8	10	1330	320
9	11	1408	232
10	12	1536	160
11	13	1664	132

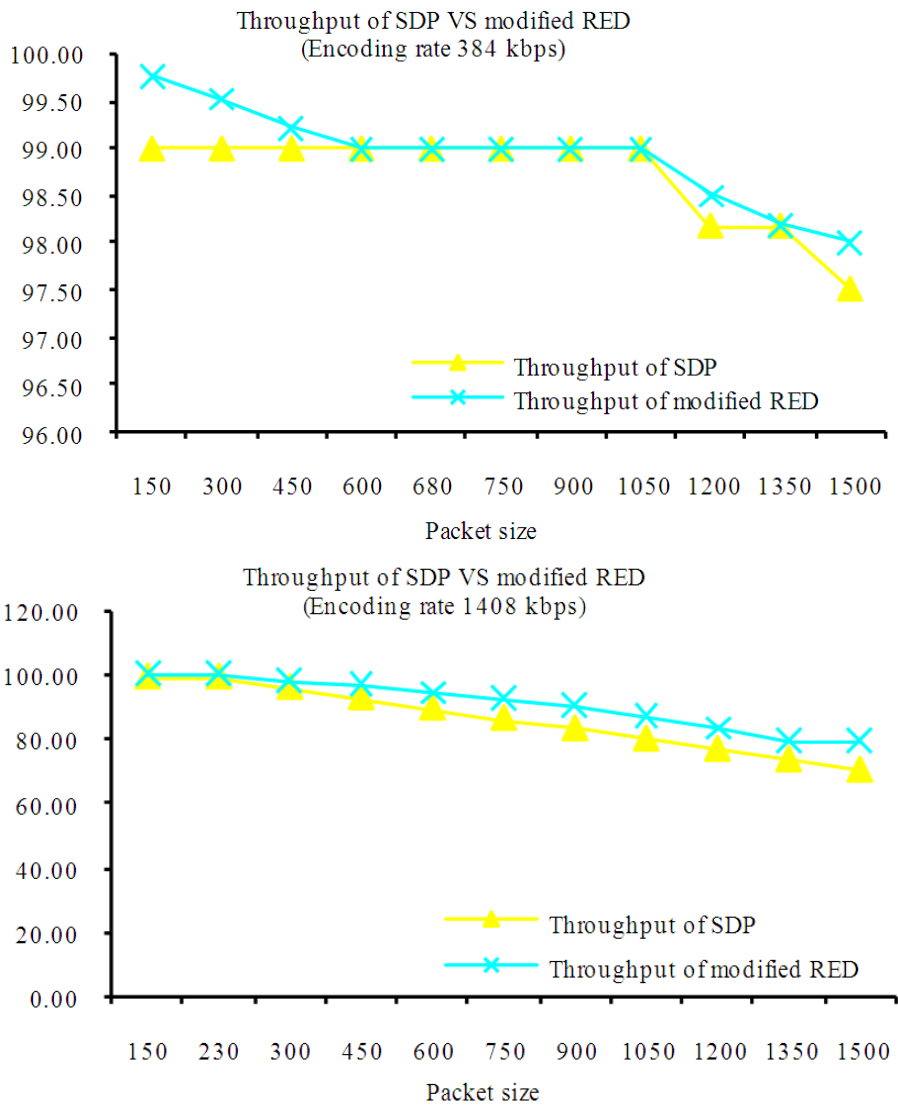


Fig. 2. Comparison of throughput of SDP and modified RED

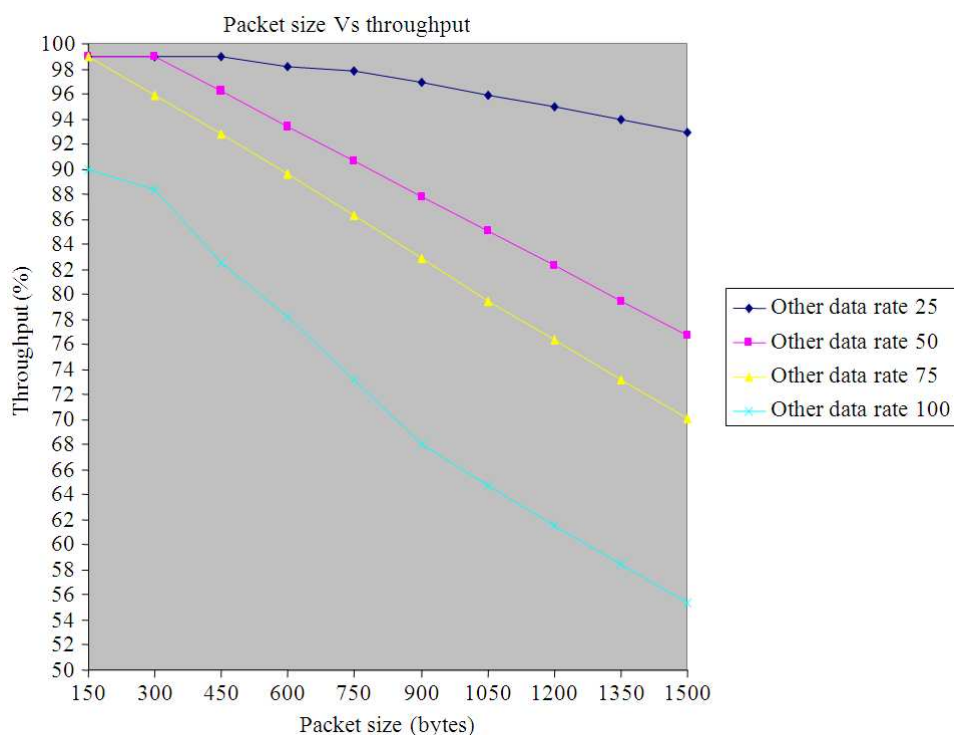


Fig. 3. Packet size vs throughput for different other data rates

5. SUMMARY AND CONCLUDING REMARKS

The proposed optimal fragmentation and modified RED algorithms are used to improve the QoS service of multimedia applications by improving the throughput. The packets are dropped based on the remaining deadline of the packet and packet size. The experimental evaluation shows

- i. Efficient link utilization
- ii. Increase in the arrival of useful packets to the destination thereby increasing the perceived quality of multimedia data.
- iii. The fairness of other packets (non multimedia) are not affected..

The effect of delay and jitter on the perceived quality can also be studied.

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