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Packet Reordering Using Congestion Control Algorithm Based on Data frame Weightage

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ABSTRACT
In network application the data frame is transferred as sub-divided packets which are interdependent. Because of this inter-packet dependencies problems like buffer overflow, degradation of packets may occur in existing application. In this paper packet reordering has been done through Round Trip Delay Time(RTT) technology based on congestion control algorithm & weight based priority algorithm. It has been observed this technology has achieved considerable success to reduce the time delay and also to avoid degradation of packets. It also addressed the problem of buffer management of multiple data streams traffic.
Key Words: Interpacket dependencies, RTT, reordering.

I. INTRODUCTION

In majority network application the data frames are splitted into a small sized packets erfore transmission. The data is useful only after receiving of all packets. Degradation of some packets leads to wastage of whole data. From this is understood that the multistreams has interdependent packet structure. This dependency structure depends on encoding used. Normally re-transmission of losed data packets found in higher level protocol. But dependency structure in MPEG video is so strong that delay of retransmitted packets affects real time traffic.Because of these inter-packet dependency problems, like buffer overflow, loss and degradation of packet usually occurs in existing applications. An usual approach to deal the packet loss problem is to empty encoding schemes which provide considerable improvement in the performance. However this common approach has its own limitations in same network application.

The prime causes for the packet loss is due to congestion.we consider the problem of buffer management of multiple data streams traffic. In this paper packet re-ordering has been done through Round Trip Delay Time (RTT. This technology based on congestion control algorithm and weightage based priority algorithm. These algorithms are guaranteed to give good output, in any traffic pattern. In this paper we studied packet discard policies in depth. Our model is suitable to various traffic encoding schemes that has more inter dependency structure.

BASE STATION:

A base station is what links mobile phones to a wireless carrier’s network. Base station provides local coverage for a wireless network.

1. WSN Info: A wireless sensor network(WSN) consists of spatially distributed autonomous sensors to monitor physical or environmental conditions such as temperature, pressure, sounds, etc. And to cooperatively pass their data through the network to a main location.
2. Packet Node: It is a important part in a packet switching network. The main function of packet node are transmitting and receiving of data packets. Data switches and equipment's are integral part of packet node.
3. Queue Buffer perform
4. Individual Queue Perform
5. Transmission Comparison

MOBILE NODE:

Mobile node is any device having internet connection and having the character of changing location frequently. Usually Mobile phones, handheld and laptop computer are belongs to this kind. Router also comes in this category. As mobile node moves from one place to another place, it changes from one network to another network. Maintaining continuous internet connection needs special support mechanism. In traditional routing procedures each mobile node havesame IP Address. But in standard routing procedure, each and every time a mobile user have to change the IP Address whenever the node changes from one network to another. Mobile Node has three important parts.

1. Mobile Channel
2. Mobile Signal
3. transcoding

Among these, transcoding plays important role in conversion of one encoding to other encoding. It convert analog to analog and digital to digital directly. Transcoding is useful to convert the unsupported file format to supported one.

II. EXISTING SYSTEM

In this work most related to the problems of buffer management in video traffic has been studied.Like MPEG video streaming, most of these data streams has high dependency structure and specific encoding schemes. Due to congestion in data streaming buffer overflow occurs. Because of this overflow existing encoding scheme dropped some data packets in the receiving end. So most of the previous works analyzed on which data packets should be dropped to receive less damage on the output.

Because of loss of some packets, whole data set become unusefull. In the buffer management policy of Ramanathanetalthe frame is dropped when many packets are dropped. In that they suggest latest frame first rule for discard the packets.In many other works priority based discard policies have been studied. In that the packets are discriminated based on importance. Normally less important packets are dropped which would not affect the good put. The model proposed by kesselmanetal has no stream structure which underlies the arrival traffic. In this work they considered restricted arrival pattern. Buffer management in FIFO scheduling regime follows Queuing discipline. Time delay and re-ordering constrained environment happened due to congestion in this method.

III. III.PROBLEM DEFINITION

In a network application, the problem of ensuring the arrival of entire frame at receiving end is crucial. Because of congestion, buffer overflow and degradation of packets occurred. An usual approach to handle the packet loss problems in to empty encoding schemes. In the existing encoding schemes even though it achieved some performance improvement, it has some limitations. It is more costly due to increased traffic. In the regime of FIFO, also more time delay and re-ordering constrained environment have occurred.

The main problem of existing buffer management is low level of good put to the throughput. Throughput is defined as amount of data transmitted as data packets. Good put is defined as amount of data received as data packets in receiving end. In a good buffer management practices good put is almost equal to the throughput. But due to the over congestion loss of data packets reduces the effective good put.In account of all issues discussed the proposed system considered packet re-ordering based on congestion control algorithm and weightage priority algorithm.

In network application the data frame is transferred as sub-divided packets which are interdependent. Because of this inter-packet dependencies problems like buffer overflow, degradation of packets may occur in existing application. In this paper packet reordering has been done through Round Trip Delay Time(RTT) technology based on congestion control algorithm & weight based priority algorithm . It has been observed that technology has achieved success to reduce the time delay and also to avoid degradation of packets. Addressed the problem of managing buffer overflows for traffic consisting of multiple streams, with interpacket dependencies. And provided guidelines for thedesign of such algorithms, and analyzed. Provided guarantees on its performance under any traffic conditions by proving it has a bounded competitive ratio.

IV. PRPOSED SYSTEM

In the existing methodology, only one packet has been sent at a time. Once the packet got received next packet starts to retransmit. But this proposed system randomly initial frame from every packet started too sent at a time.This will reduce the overall delay.The main objective of this paper is to reduce the overflow in the buffer.The input video has been splitted into framesOnce the frames has been delivered to receiver it can be reordered.

Congestion control:

Congestion control controls the traffic by regulating the entry into the telecommunication network. To avoid congestion collapse congestion control algorithm attempts to prevent the over usage of intermediates node and networks process and link capabilities. It also avoids to take resource reducing steps like reducing the packet sending rate.In this framework Lagrange multiplies 'P^{l} loss of probability or delay in the link ^{l} because of queuing.A prime limitation theory is that it assumes.

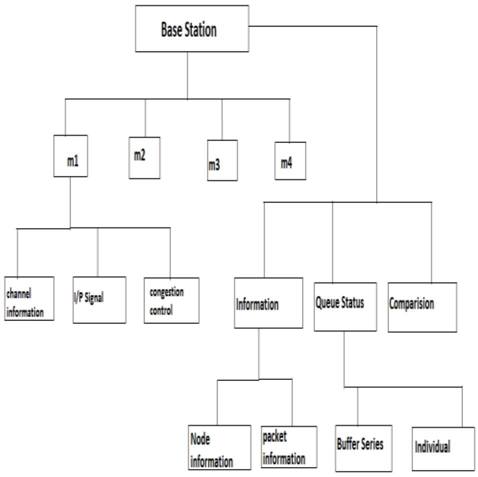

Fig 1: Architecture

PROPOSED ALGORITHM

Theory of congestion control

Theory of congestion control was proposed by Frank Kelly in which be applied two theories namely Microeconomic theory and convex optimization theory. In which to explains how individual packets control their own rates and also concludes that these individual packets interact with each other and achieve the optimum rate in network wide allocation. The mathematical expression for optimal rate allocation is as follows.

Let $x_i$be the rate of flow i, $C_l$be the capacity of link l, and r_{li}be 1 if flow $i$uses link $l$and 0 otherwise. Let x, $c$and Rbe the corresponding vectors and matrix. Let $U(x)$be an increasing, strictly concave function, called the utility, which measures how much benefit a user obtains by transmitting at rate x. The optimal rate allocation then satisfies

$$\max_x \sum_i U(x_i)$$

such that $Rx \leq c$

In this problem the Lagrange dual unlinks. The unlinked flow behaves in different rate based on price of the network in its own way. After unlinks of Lagrange dual each link attract a constraint because of this a Lagrange multiplies has raised.

$$y_i = \sum_l p_l r_{li},$$

Congestion control turned into distributed optimization algorithm to solve the above constraint. The framework of this model is followed in many present congestion control algorithm.In this framework Lagrange multiplies 'P ^{l} loss of probability or delay in the link ^{l} because of queuing. Even though many optimal allocation rates are possible. Max-min rate allocation and Kelly's proportional fair allocation are the dominant examples. A prime limitation theory is that it assumes. Burs tines caused by the sliding of window flow control also causes different flow rate. So it observes different loss or time delay in a given link.

Kinds of congestion control algorithms

Algorithms are classified by different ways. By the type and amount of feedback received from the network: Loss; delay; single-bit or multi-bit explicit signals.By incremental deploy ability on the current Internet: Only sender needs modification; sender and receiver need modification; only router needs modification; receiver, sender, and routers need modification.By the aspect of performance it aims to improve: high bandwidth-delay product networks;lossy links; fairness; advantage to short flows; variable-rate links

Achieving RTT

A first-come-first-served policy (drop tail) can yield TCP connections with large RTT to starvation since packet belonging to connections with small RTT can monopolize the buffer space. To promote RTT fairness among TCP connections, the RDQM divides the buffer space among the flows proportionally to their RTT. Initially, the RTT is taken as that experienced by the SYN TCP segment. After that, the RTT value is measured during connection lifetime. For every packet the BS receives from the wired network, it verifies whether or not the number of enqueued packets of the same flow, *curi*, exceeds the maximum number of allowed packets in queue for that flow, *maxi*.

In case it exceeds, the packet is dropped, otherwise it is enqueued. Such scheme corresponds to a partial sharing scheme with thresholds defined dynamically. RDQM tries to allocate buffer space to accommodate the bandwidth-delay product of that flow estimated as the product of the measured RTT by the wireless link available bandwidth. Such policy is based on the well-known principle for TCP New-Reno that states that for a connection to fully utilize the path between a sender and a receiver the buffer space at a bottleneck link should be at least equal to the bandwidth delay product between the sender and the receiver.

Algorithms 1 to 3 present the details of this approach. Algorithm 1 describes the procedures implemented at the mobile node which essentially inserts the measured RTT values into outgoing LL-ACK packets. Algorithm 2 corresponds to implemented at the BS which either accepts or discards incoming packets. Algorithm 3 updates the queue thresholds, *i.e.*, *curi*, and*maxi*. Algorithm 2 is triggered by the arrival of TCP data packet

Algorithm 1: On LL-ACK packet transmission at MN

1. **If** TCP data packet received
2. Encapsulate feedback info into outgoing LL-ACK
3. Send LL-ACK
4. **Endif**

Algorithm 2: On TCP DATA packet arrival at BS

1. **Get** Flow ID
2. **If** packet belongs to a new flow **Then**
3. Create new entry for incoming flow
4. Set Flow Packet Limit equal to MAXdef
5. Set Flow Packet Count equal to zero
6. Set Flow RTT equal to RTTdef
7. **Else**
8. **If** Flow Packet Count is greater than Flow Packet Limit **Then**
9. Drop incoming packet
10. **Else**
11. Increase Flow Packet Count by one
12. Accept packet
13. **Endif**
14. **Endif**

Algorithm 3: On LL-ACK packet arrival at BS

1.Get Flow ID
2. **If** Flow RTT is greater that zero **Then**
3. Set RTT Sum equal to zero
4. **For**Each Flow **Do**
5. Increase RTT Sum by Flow RTT
6. **Endfor**
7. **For**Each Flow **Do**
8. Set Flow Weight to Flow RTT / RTT Sum
9. Set Flow Packet Limit to max between Flow Weight andbuffersize
10. **Endfor**
11. **Endif**

V. PERFORMANCE EVALUVATIONS

In this section, it is concentrated on describing about MRTT values. wegenerated maximum number of RTT valuesfor every transmissions. It reports about the packet transmission time for every RTT values.

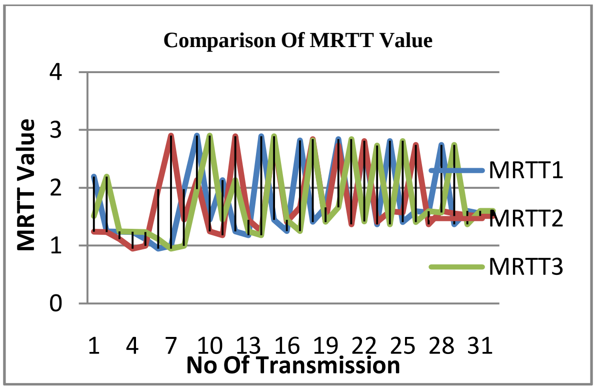

Fig 2:Comparision of MRTT Value

The packets containing video is transmitted from source to destination.Fore every packet RTT values and MRTT values are compared for every number of packet transmissions.

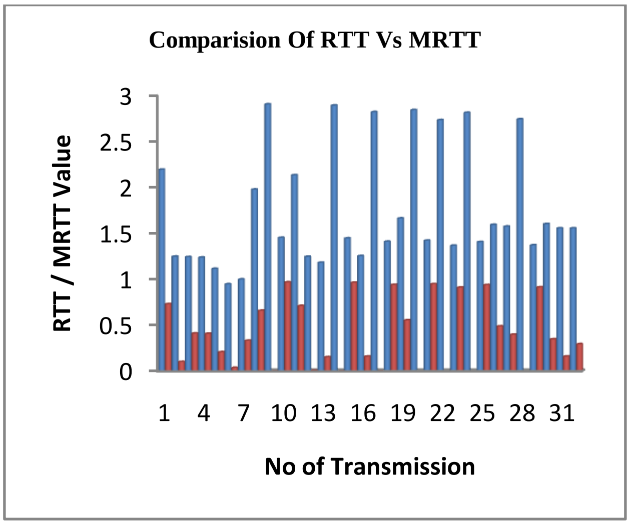

Fig 3: Comparison of RTT Vs MRTT

VI. CONCLUSION AND FUTURE WORK:

The main function of the network layer is routing packets from source to destination. The traffic traversing the network has inter-packet dependencies due to application-level encoding schemes. The proposed scheme has addressed the problem of managing buffer overflows for traffic consisting of multiple streams, with inter-packet dependencies. The algorithms proposed in this paper derives the guidelines to increase the performance of Buffer management.

To increase the merits of the work, it is planned to investigate the following issues in our future research:

In future work encryption and compression method will be applied this will lead to reduce the packet size. Main intention of this method is to avoid the congestion and flooding.

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