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Quality of Service in Real Time Services in Wireless Systems

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Abstract—In Real time message transmission there is no time delay between a message sending and reception. Real time messaging defines standard packet format and data delivery for transmission of audio and video data over IP networks. Video enable applications are mostly used in our life without any delay, which also improve the quality of video. The needs for a central buffer management to achieves better memory utilization by enabling video stream sharing across components and to all network condition. This buffer management avoids congestion in networks. Our work is focused on a queue management scheme to manage the buffer at destination for video enable services which carries huge amount of data through network channel. Video data is generated at source which it reached to destination through various nodes and links. So, there may be delay, packet loss and jitter. To provide the better service at destination, we require a less delay, less amount of packet loss and less jitter. So in this paper we are working on a buffer management mechanism which cares about packet loss and jitter and try to resolve and will find out better scheduling in existing schemes.

Keywords: Buffer management, Scheduling techniques, Congestion, Queue.

1 Introduction

Now a day we need a network with high network capacity and Computer network is a network which is connected with a multiple nodes through a link of different bandwidth. In which sender send the packet through the link to the receiver. But due to congestion in the network or due to less bandwidth available in the network link packet loss in the network and there is also delay of the packet, so to maintain this we need buffer in the network [1].For transmitting or receiving high bandwidth data such as video data high bandwidth. High speed network is made up of cables such as fibre optics that support transmission of high speed data. Continuous media applications such as video and audio are sensitive not only to the packet loss probability but also to the correlation of packet losses. So for transmission of packet we need the buffer which maintains the packet loss in the computer network [14].We widely used video enable applications mostly in our life without any delay, which also

improve the quality of video. The needs for a central buffer management to achieves better memory utilization by enabling video stream sharing across at components and to all network condition. At the same time there can be more than one services required by the any user. So a queue management scheme is required to manage the packet arrival at the buffer at destination for video enables services which carries huge amount of data through network channel. Data is generated at source which is reached to destination through various nodes and links. So, there may be delay, packet loss and jitter. To provide the better service at destination, we require a less delay, less amount of packet loss and less jitter. So we need a buffer management mechanism which care about packet loss and jitter.

1.1 Buffer Management Mechanism

It is a technique which is used to improve the delivery of the packet in the network .We can improve these with the help of implementing different queue in the network. Congestion in a network may occur if the load on the network is more than the carrying capacity of the network. Congestion in a network or internetwork occurs because routers and switches have queues-buffers that hold the packets before and after processing .Hence it suffers from poor quality of service and also can lead to increase delays, lost of data. Congestion can be brought on by several factors [2].

If all of a sudden, streams of packets begin arriving on three or four input lines and all need the same output line, a queue will be created. If there is insufficient memory available to store all of them, then the packet will be lost. This problem cannot be overcome by increasing the memory, because Nagle discovered that if routers have an infinite memory, congestion gets worse, not better. Slow processor can also cause congestion. If routers or processors are slow at performing the tasks required, queues can be formed up, even though there is very high line capacity. Similarly, if the bandwidth of a line is very low it also causes congestion.

1.2 Queue Management

Traffic phase effects occur when different flows of packet from different source and we can see different performances of the network. So we can solve this problem by implementing queue and solved by simply increasing the buffer size in the router. It seems that these effects would not occur or could at least be significantly diminished by increasing the maximum queue length. Since a queue is only meant to compensate for sudden traffic bursts, one may wonder what would happen if the queue length was endless.

Need for Buffer: Congestion occurs when resource demands exceed the capacity [2]. As user come and go, Internet performance is therefore largely governed by these inevitable natural fluctuations. In this case, the gateway would see occasional traffic spikes that go beyond the capacity limit as a certain number of customers use their maximum rate at the same time. Since these excess packets cannot be transferred across the link, there are only two things that this device can do- buffer the packets or drop them.

Network Congestion Control: Managing Internet Traffic limited in time, standard server routers usually place very packets in a buffer, which roughly works like a basic FIFO ('First In, First Out') queue and only drop packets if the queue is full. The usual assumption of this design is that a continuous traffic reduction would eventually decrease or drain the queue, thus making it an enough to compensate for short traffic bursts. Also, it would seem that reserving enough buffers for a long queue is a good choice because it increases the chance of accommodating traffic spikes.

There are mainly two basic problems occurs with this:

1. Storing packets in a queue adds significant delay, depending on the length of the queue.

2. The consequence of the first problem is that packet loss can occur no matter how long the maximum queue, because of the second problem, queues should generally be kept short, which makes it clear that not even defining the upper limit is a trivial task.

2. Literature Survey

Literature review or survey is the study of various research papers published already which we study to gain knowledge about a particular topic. So in this sequence we have studied various research papers and on the basis of that we can give detail our topic in the way as follows:

FIFO: Packets are transmitted on the basis of first come first serve basis. [9].

RED: Red algorithm provides **t**he mechanism which mange the transfer of packet from source to destination through multiple nodes and the mechanism called buffering of the packet into the queue[7].

Proportionally Fair Scheduling: The Proportionally Fair Scheduling (PFS) algorithm was proposed after studying the unfairness exhibited when increasing the capacity of CDMA by means of differentiating between different users[8]. The PFS algorithm seeks to increase the fairness among the users at the same time as keeping some of the high system throughput characteristics [20].

Drop tail: Tail Drop or Drop Tail, is a simple queue management algorithm used by Internet routers to decide when to drop packets. In contrast to the more complex algorithms like RED and WRED, in Tail Drop all the traffic is not differentiated. Each packet is treated identically. With tail drop, when the queue is filled to its maximum capacity, the newly arriving packets are dropped until the queue has enough room to accept incoming traffic [14].

Packet Scheduling in Queue: Queues represent locations where packets may be held (or dropped). Packet scheduling refers to the decision process used to choose which packets should be serviced or dropped [22]. Buffer management refers to any particular discipline used to regulate the occupancy of a particular queue. At present, support is included for drop-tail (FIFO) queuing, RED buffer management [6], CBQ (including a priority and round-robin scheduler), and variants of Fair Queuing including, Fair Queuing (FQ), Stochastic Fair Queuing (SFQ), and Deficit Round-Robin (DRR) [3]. In the common case where a delay element is downstream from a queue, the queue may be blocked until it is re-enabled by its downstream neighbour. This is the mechanism by which transmission delay is simulated [16]. In addition, queues may be forcibly blocked or unblocked at arbitrary times by their neighbours (which is used to implement multi-queue aggregate queues with inter-queue flow control). Packet drops are implemented in such a way that queues contain a "drop destination"; that is, an object that receives all packets dropped by a queue.

Factors Responsible for Occurrence of Congestion: We need buffer limited memory space, channel bandwidth, router capacity load of network, link failure, heterogeneous channel bandwidths the same key forcing the network to stick into congestion.

Effect of Buffer Space*:* The amount of buffer space given at a node is limited, the amount of information that can be stored at the node. For the case sufficient buffer is available more and more packets received at the node may accumulate and get transmitted later on[10]. Here, the packets may suffer very large delay and subsequently leads to congestion. However for lower buffer space the packet may drop very frequently when load is increased and have a lower throughput. This is because packets have less room to wait for their chance. Thus neither using very large amount of buffer at node nor very less amount buffer is able to reduce congestion for all the case for the application have random load higher buffer size may be beneficial whereas medium range buffer is preferred for constant load the effect of channel bandwidth.

Effect of Channel Bandwidths: Traffic originates from a sender; this is where the first decisions are made (when to send how many packets). For simplicity, we assume that there is only a single sender at this point.

• Depending on the septic network scenario, each packet usually traverses a certain number of intermediate nodes. These nodes typically have a queue that grows in the presence of congestion; packets are dropped when it exceeds a limit [12].

• Eventually, traffic reaches a receiver. This is where the final (and most relevant) Performance is seen – the ultimate goal of almost any network communication code is to maximize the satisfaction of a user at this network node. Once again, we assume that there is only one receiver at this point, in order to keep things simple [4]. Traffic can be controlled at the sender and at the intermediate nodes; performance measurements can be taken by intermediate nodes and by the receiver.

Assumption of queue length: Choosing the right queue length is essential. There are two reasons for this. First**,** the source behaviour relies on packet loss as a congestion indicator – thus, the rate of sources will keep increasing until the queue length grows beyond its limit, no matter how high that limit is. Second**,** a queue can always overflow because of the very nature of network traffic, which usually shows at least some degree of self-similarity [6]. Queuing delay is a significant portion in the overall end- to-end delay, which should be as small as possible. Let us look at a single flows and a single link for a moment. In order to perfectly saturate the link, it must have c × d bits in transit, where c is the capacity (in bits per second) and d is the delay of the link (in seconds). Thus, from an end-system performance perspective, links are best characterized by their bandwidth × delay product.

On the basis of this fact and the nature of congestion control algorithms deployed in the Internet, a common rule of thumb says that the queue limit of a router should be set to the bandwidth × delay product, where 'bandwidth' is the link capacity and 'delay' is the average RTT of flows that transverse[2].

3. Implementation

Scheduling is a technique for scheduling data packets in a packet-switched based best-effort communications networks. To maximize the total throughput of the network, or the system spectral efficiency in a wireless network. This is achieved by giving scheduling priority to the least "expensive" data flows in terms of consumed network resources per transferred amount of information.

Drop Tail: The working of drop tail algorithm with the help of flow chart we can see step by step:

If { No Of Packet Incoming at a node >Channel Bandwidth

Then we will go for Check Buffer Size}

else if{ size of buffer >= packet size

Then form a queue of the packets

Else drop the packet}

If {no. of incoming input packets at node < channel bandwidth

Then packet enter the channel

}

RED queue: The algorithms of RED queue in flow chart given in step by step.

1. Average length of queue is calculated

2. If {average length >=threshold} or

Probability to drop the packet is high

3. Else

4. If {Average length of queue <=threshold} or Probability value is lower than enquirer packet s

5. Packet sent to network

}

3.1 Simulation Topology

Implementation is done in ns2, simulation script generally begins by creating an instance of this class and calling various methods to create nodes, topologies, and configure other aspects of the simulation [5]. This network topology is used for the experiment of "Buffer Management Mechanism" and its evaluation; in this topology we used the seven nodes and the link between them. Node 0 connected with node 3, node1-node 3, and node 2-node 3 respectively.

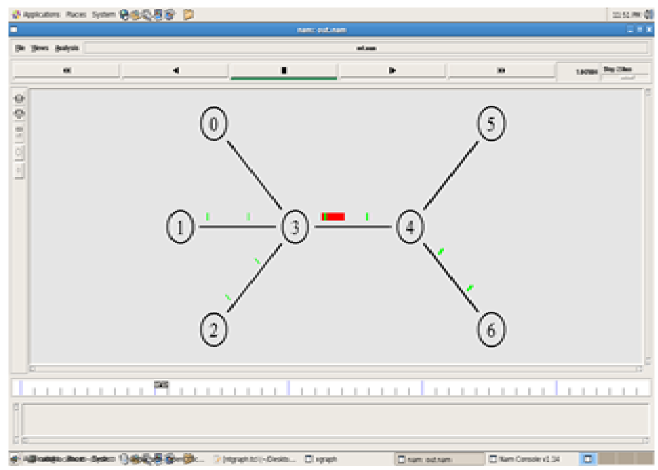

Figure No: 1 Simulation Topology

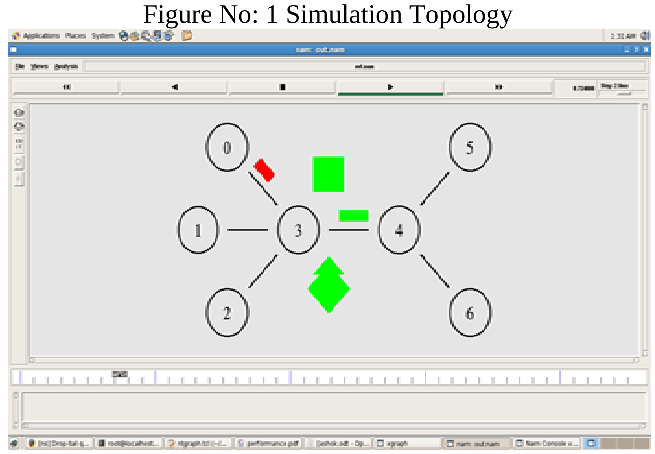

Figure No: 2 Topology After instance

The channel link is node 3 and node 4 through which all the packet passes from source to destination. And node 4 connected to node5 and node6 respectively.

3.2 Parameters Used

The initial parameters are used given below:

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Table 1: Parameters Used

Sr. No.	**Parameters**	**Value**
1	Bandwidth(Mb)	5mb
2	Queue size	20
3	Link Delay	30
4	Packet size	5000
5	Source Rate	1.5Mb
6	Simulation Time	5 minute

So, all the initial parameters shown in the above table which taken for our experiment. The bandwidth between link node 0 to node 3,node 1 to node 3 and node 2 to node 3 is 20Mb, node 3 to node 4 is10Mb and node 4 to node 5 ,n 4 to node 6 is 20 Mb. And delay in all links is 30ms. The initial queue size assumed is 4 packets. But the bandwidth, queue size and transmission delay varies from scenario to scenario.

4. RESULT AND ANALYSIS

We studied and analyzed various types of scheduling algorithm. Here we observe that by using scheduling techniques and algorithms we can eliminate various problems occurring in a WLAN environment such as packet loss, delay, Congestion, flood, node failure and hence improve QoS in Real Time Services in Wireless Systems.

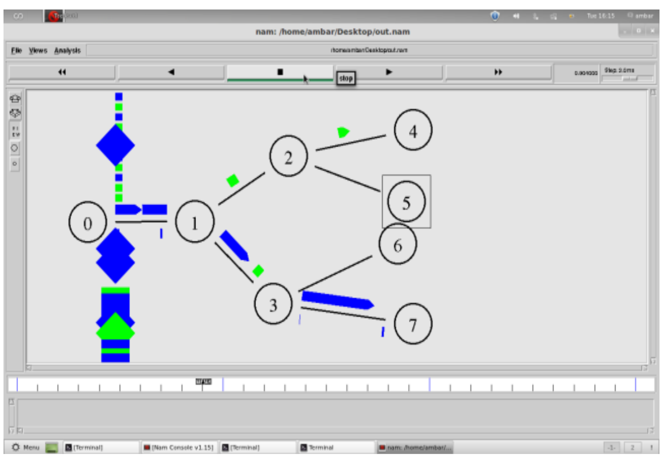

Figure No: 3

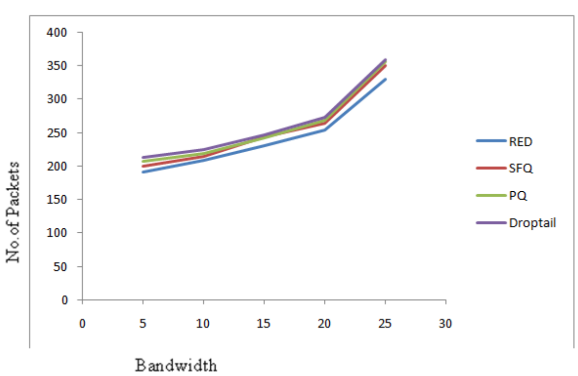

Figure No: 5 Packets vs. Bandwidth Comparison

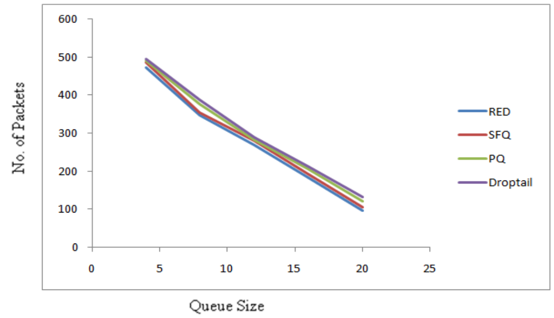

Figure No: 6 Packets Vs. Queue Size Comparison

So the graphs showed above gives the performance comparison of different scheduling algorithms available in wireless system.

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