Router Buffer Sizing in Mixed Traffic

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Abstract— In today’s electronic world, it is observed that 90-95% of Internet traffic is transmitted over TCP, but there’s an increasing demand for real time traffic with the boom of VoIP, online gaming and IPTV applications etc. When the interaction between TCP and UDP traffic happens, they balance well for a specific range of buffer size and later as the buffer size increases, with significantly no change in the TCP traffic’s throughput. So the mixture of TCP and UDP traffic doesn’t go well with large sized buffer. Therefore, it is necessary to decide an appropriate buffer size not only for Internet, but also for optical packet switched networks as they have very low buffering capacity.

Index Terms— Congestion Window Size, Packet Size, NS-2, TCP, UDP, Packet Loss, Router Buffers

1 INTRODUCTION

Internet routers are packet switches, in which packets get buffered during the time of congestion. Hence sizing these buffers is a crucial issue. Conventionally, it was practiced that buffered during the time of congestion. Hence sizing these Internet routers are packet switches, in which packets get buffered during the time of congestion. Hence sizing these buffers is a crucial issue. Conventionally, it was practiced that buffered during the time of congestion. Hence sizing these Internet routers are packet switches, in which packets get buffered during the time of congestion. Hence sizing these buffers is a crucial issue. Conventionally, it was practiced that buffered during the time of congestion. Hence sizing these Internet routers are packet switches, in which packets get buffered during the time of congestion. Hence sizing these buffers is a crucial issue. Conventionally, it was practiced that buffered during the time of congestion. Hence sizing these Internet routers are packet switches, in which packets get buffered during the time of congestion. Hence sizing these buffers is a crucial issue. Conventionally, it was practiced that buffered during the time of congestion. Hence sizing these Internet routers are packet switches, in which packets get buffered during the time of congestion. Hence sizing these buffers is a crucial issue. Conventionally, it was practiced that buffered during the time of congestion. Hence sizing these Internet routers are packet switches, in which packets get buffered during the time of congestion. Hence sizing these buffers is a crucial issue. Conventionally, it was practiced that...
more than 50 flows are active is about 0.005. Therefore, the assumption that a large number of users exist in the system does not hold. Thus, large reductions in buffer sizes due to statistical multiplexing effects the performance of the router. In fact, reducing buffer-sizes in these networks would result in dramatic degradation in the overall performance. If the core router is heavily loaded up to 98%, it can still operate with very small buffers if the core to access speed ratio is small. Here when a particular access to core router ratio is assigned, there exists a threshold operating load below which small buffers seem to be adequate and above this threshold, a buffer of size \( O(C^*RTT) \) is required.

The internal architecture of commercial routers and the precise set of queues a packet goes through are not easily available hence it becomes difficult to analyze the effect of reducing buffer size. Further, there some routers that tend to have hidden buffers that are not accessible to the user, and hence buffer size cannot be controlled directly. The main result using the drop-tail scheme is that while link utilization is largely independent of router architecture, buffer size and offered load, other metrics such as loss and delay are much more sensitive.

Hence it is prudent to size router buffers at a value that balances the performance of both TCP and UDP tracer route appropriately. Operating the router buffers at a very small value can adversely impact the performance of both TCP and UDP tracer. Furthermore, operating it in the "anomalous region" can result in increased UDP packet loss, with only a marginal improvement in end-to-end TCP throughput. Moreover, building an all-optical packet router and buffering of packets in the optical domain is a rather complex and expensive operation [5].

Understanding the nature of network traffic is critical in order to properly design and implement computer networks and network services like the WWW, recent examinations of local-area network traffic and wide-area network traffic have challenged the commonly assumed models for network traffic, for example, the poisson process. If the traffic were to follow a poisson or markovian arrival process, it would have a typical burst length which would tend to be smoothed by averaging over a long enough time scale. The measurements of real traffic indicate that significant traffic variance i.e. burstiness is present on a wide range of time scales. Traffic that is bursty on many or all time scales can be described statistically using the notion of self-similarity. Self-similarity is the property that is observable at a wide range of time scales, it can exhibit long-range dependence [6].

In the case of stochastic objects like time series, self-similarity is used in the distributional sense: when viewed at varying scales, the object's correlation structure remains unchanged. As a result, such a time series exhibits bursts—extended periods above the mean—at a wide range of time scales. Since a self-similar process has observable bursts at a wide range of time scales, it can exhibit long-range dependence [6].

The Internet was designed as a best-effort network, which means delivering information without any delay guarantees. Many years later, it has grown up and is used to provide new services, which sometimes have real-time requirements. One of the first real-time services deployed was VoIP, which is nowadays widely used, and is replacing traditional telephony systems. The problem of using a best-effort network for deploying a real-time interactive service has been largely discussed, as the users of traditional telephony would not like to change to a new technology unless the offered quality was similar to the one they are used to. Thus, many studies were carried out, trying to identify the different network impairments, and to quantify their effect on the quality perceived by users. A number of hosts are connected via the same access router to a server. If gaming service is considered, this is also a scenario in an Internet café. Gaming has been reported as one of the main activities deployed by the users of these businesses, which are still a very popular way for connecting to the Internet.

Fig. 1 Scenario of Internet Café

In Fig 1, the connection will be shared by many services, some of them real-time. The total traffic offered to the router may vary and in some moments it can even be over the link capacity. So the router will add delays and discard packets, and the buffer size and policy will have an influence on the distribution of these impairments. In particular, some policies are packet-size aware, as they penalize different sizes in different manners [7].

In the field of communication safe and reliable transmission of data is an important issue. Emergence of computer networks and internet brings tremendous changes in the field of communication. A computer network forms, when two or more computer systems interconnect over a same link for the sharing of resources. Wired computer network is vastly recommended because they are flexible and multi adaptive in nature. In wired network various queues has been used out for the safe and reliable transmission of data. Thus, the implementation and proper selection of the queues is one of the most crucial issues. The choice of the particular queue from the various queues is totally depended upon the need of transmission of data. The transmission of packet over a medium at any instance of time requires a packet processing routine. Thus, to maintain a proper processing of the packets over a source an interface must be deployed. This interface object must be able to accept the request from source objects to transmit a packet, even when the medium is busy in transmitting a previous packet. The various queues, which were implemented, can be discussed into the following categories: Drop Tail Queue, Random Early Detection (RED Queue), Stochastic Fair Queuing, Deficit Round Robin (DRR), Fair Queue
(FQ). The simulator is a tool for demonstrating the various protocols, algorithms and to serve as an aid in the better understanding of the protocols. The RED shows maximum number of packet loss in case of CBR traffic while SFQ shows maximum number of packet loss in case of FTP. The FQ shows minimum number of packet loss in CBR and FTP traffic [8].

High quality multimedia applications present in Internet Protocol networks have rigid requirements in terms of packet loss, delay, and delay jitter. In IP networks, loss characteristics affect the quality of the established video and/or voice connections, and influence the throughput of bulk data transfers. These characteristics cannot be captured by specifying only the loss probability but requires a detailed study of loss patterns. Gilbert model which are widely used for measuring TCP/UDP traffic flows was originally derived from UDP loss traces obtained from genuine Internet measurements. This is used to validate the loss process generated by simulations [9].

A large congestion window size in the wireless multi-hop networks worsens TCP performance by increasing the probability of packet contention and packet losses from excessive collisions. There have been some previous works on TCP bandwidth measurement methods. In TCP-Westwood, the sender estimates the available bandwidth dynamically by measuring and averaging the rate of returning ACKs. In TCP-Peach, the sender probes the available network bandwidth in only one RTT with the help of low-priority dummy packets. A similar method is used in TCP-Jersey, which was derived from the time-sliding window estimator. Hence in wireless multi-hop networks, it is necessary to limit the TCP congestion window size so as not to increase the probability of congestion loss, thereby ensuring a good TCP performance [10].

3 OUR CONTRIBUTIONS TO THE STUDY OF ROUTER’S BUFFER SIZING

Our work consisted of 6 phases.

Phase I: NS2 simulations are performed to study the UDP packets loss by varying the number of UDP flows. Here the number of nodes created was 52. Of which 25 pairs of Source – Destination nodes in which TCP traffic flowed. And 1 pair of Source-Destination node in which UDP traffic flowed. For every Source-Destination pair different start times were allotted to get the real world traffic results. Both TCP and UDP traffic was made go through 2 routers named r0 and r1. All the Source nodes were connected to r0 and all the Destination nodes were connected to r1. Buffer sizes between these two routers were varied from 5-50kB.

Phase II: In this phase different topologies like Ring, Bottle Neck, and Derived with similar network configurations were simulated.

Phase III: NS2 simulations are performed to study the UDP packets loss by varying the Queue types like Class Based Queuing, Fair Queuing, and Stochastic Fair Queuing, Drop Tail.

Phase IV: In the above 2 phases, the total number of nodes used were 54. In order to understand the real world scenario...
the number of nodes was increased to 200. 100 nodes are Source-Destination pairs through which TCP traffic were passed. The number Source-Destination pairs were varied according to the percentage of UDP traffic. When 5% of UDP traffic was simulated, 95 source nodes of TCP and 5 source nodes of UDP and as the percentage of UDP traffic was increased, the number of TCP nodes were reduced. In these simulations, congestion window size and time to live field was also set for every traffic flow. These two parameters have an immense effect on the traffic flow. The queue type used was Drop Tail as it said to be one of efficient queues for Bottleneck network topology. The Buffer Size was varied from 5kB to 50kB.

Phase V: Here AWK scripts were coded to measure TCP Throughput, UDP Throughput and UDP Loss Probability.

Phase VI: Here graphs are plotted on the values obtained from the trace files and AWK script. Graphs help in analysing the data better. So graphs were plotted for different topologies, different queue types, and varying number of TCP and UDP nodes. On the x axis, Buffer Size was plotted and on y1 axis TCP throughput and y2 axis UDP throughput were plotted.

4 Results and Discussions

The graphs shown below are obtained by running the simulations and measuring the parameters like TCP Throughput, UDP Throughput and UDP Loss Probability.

Fig. 3. TCP & UDP Throughput with varying buffer sizes with 25 traffic flows of TCP and 1 traffic flow of UDP

Here Figure 3, shows the Throughput values when 25 traffic flows of TCP are sent and 1 traffic flow of UDP is sent. It can be observed that throughput of TCP traffic keeps on increasing with the increasing Buffer Size. However, the throughput of UDP traffic keeps increasing till the buffer size is 20 and then throughput reduces and it reaches the peak, and then remains stable. Hence it seems that keeping a buffer size of 40kB is favourable for both UDP and TCP traffic.

Fig. 4. TCP & UDP Throughput with varying buffer sizes with 25 traffic flows of TCP and 2 traffic flows of UDP

As it seen that the throughput of TCP traffic is linearly increasing whereas the throughput of UDP is at peak when the buffer size is fixed as 45-50kB.

Fig. 5. TCP & UDP Throughput with varying buffer sizes with 25 traffic flows of TCP and 3 traffic flows of UDP

Here Figure 5 shows the throughput values of 25 traffic flows of TCP and 3 traffic flows of UDP. The throughput values of TCP traffic is steadily increasing, whereas the throughput of UDP traffic is the highest when the buffer size is 40-45kB. Hence a buffer size of 40-45 kB is beneficial when mixed traffic is flowing in the network.

Fig. 6. TCP, UDP Throughput and UDP Loss Probability with varying buffer sizes with 25 traffic flows of TCP and 4 traffic flows of UDP

Here Figure 6 shows the Throughput values for 25 traffic flows of TCP and for 4 traffic flows of UDP. The throughput values of TCP traffic is steadily increasing, whereas the throughput of UDP traffic is the highest when the buffer size is 40-45kB. Hence, a buffer size of 40-45 kB would be very much favourable for both types of traffic.
Fig. 7. Varying Buffer Sizes on Derived Topology Drop Tail
For Figure 7, a network of 25 pairs of Source-Destination nodes was designed in Derived topology with the queue type as Drop Tail. Here there’s only 1 flow of UDP traffic and 25 flows of TCP traffic. The loss probability of UDP traffic is extremely good because there’s only 1 flow, whereas for TCP it was steadily increasing. Hence it can be said that with the buffer size of 35-40kB would suffice the needs of both the traffic.

Fig 8: Varying Buffer Sizes on Bottle Neck Topology with FQ
For Figure 8, a network of 25 pairs of Source-Destination nodes was designed in Bottle-Neck topology with the queue type as Fair Queuing. Here there’s only 1 flow of UDP traffic and 25 flows of TCP traffic. The loss probability of UDP traffic is constant because there’s only 1 flow, whereas for TCP it is constant. Hence it can be said that with the buffer size of 35-40kB would be enough for the needs of both the traffic.

Fig 9: Varying Buffer Sizes on Bottle Neck Topology with RED
For Figure 9, a network of 25 pairs of Source-Destination nodes was designed in Bottle-Neck topology with the queue type as Random Early Detection. Here there’s only 1 flow of UDP traffic and 25 flows of TCP traffic. The throughput of UDP traffic is increasing but there’s a slight reduction when the buffer size is between 35-45kB. Then it steadily increases and reaches the peak. Whereas for TCP traffic, the throughput values steadily increases except when the buffer size is 55kB.

Hence, it can be stated that by keeping the buffer size to 45-50kB would be beneficial for both the traffic.

Fig 10: Varying Buffer Sizes on Bottle Neck Topology with SFQ
Here in Figure 10, a network of 25 Source-Destination pairs were simulated with 25 traffic flows of TCP traffic and 1 traffic flow of UDP. In this network, topology used was Bottle Neck and the queue type used was Stochastic Fair Queuing. Due to the type of topology used and the type of the queue used, the throughput values remain constant for both the traffic.

Fig 11. Varying Buffer Sizes on Ring Topology with FQ
Here in Figure 11, a network of 25 Source-Destination pairs was simulated with 25 traffic flows of TCP and 1 Source-Destination pair with traffic flow of UDP. Here the topology used was Ring and queue type used was Fair Queuing. The throughput value of both TCP and UDP traffic remains constant.

Fig 12: Varying Buffer Sizes on Ring Topology with RED
Here in Figure 12, a network of 25 Source-Destination pairs was simulated with 25 traffic flows of TCP and 1 Source-Destination pair with traffic flow of UDP. Here the topology used was Ring and queue type used was Random Early Detection. The throughput value of both TCP and UDP traffic remains constant.
Here in Figure 13, a network of 25 Source-Destination pairs was simulated with 25 traffic flows of TCP and 1 Source-Destination pair with traffic flow of UDP. Here the topology used was Ring and queue type used was Stochastic Fair Queuing. The throughput value of both TCP and UDP traffic remains constant.

Fig. 13. Varying Buffer Sizes on Ring Topology with SFQ

Fig. 14. Varying Buffer Sizes on Bottle Neck Topology with parameters like congestion window size and time to live with 5% of UDP Traffic

Here in Figure 14, a network of 95 Source-Destination pairs was simulated with 95 traffic flows of TCP and 5 Source-Destination pairs with traffic flow of UDP. Here the topology used was Bottle Neck and queue type used was Drop Tail. The throughput value of both TCP and UDP traffic kept on increasing as the Buffer Size was increased. But there’s a drop in the UDP loss probability when the buffer size is kept as 45kB. Hence it seems favourable for both UDP and TCP traffic to maintain the buffer size between 35-45kB.

Fig. 14. Varying Buffer Sizes on Bottle Neck Topology with parameters like congestion window size and time to live with 5% of UDP Traffic

Fig. 15: Varying Buffer Sizes on Bottle Neck Topology with parameters like congestion window size and time to live with 10% of UDP Traffic

Here in Figure 15, a network of 90 Source-Destination pairs was simulated with 90 traffic flows of TCP and 10 Source-Destination pairs with traffic flow of UDP. Here the topology used was Bottle Neck and queue type used was Drop Tail. The throughput value of both TCP and UDP traffic kept on increasing as the Buffer Size was increased. But there’s a drop in the UDP throughput when the buffer size is kept as 30kB. Hence it seems favourable for both UDP and TCP traffic to maintain the buffer size between 35-45kB.

Fig. 15: Varying Buffer Sizes on Bottle Neck Topology with parameters like congestion window size and time to live with 10% of UDP Traffic

Fig. 16. Varying Buffer Sizes on Bottle Neck Topology with parameters like congestion window size and time to live with 15% of UDP Traffic

Here in Figure 16, a network of 85 Source-Destination pairs was simulated with 85 traffic flows of TCP and 15 Source-Destination pairs with traffic flow of UDP. Here the topology used was Bottle Neck and queue type used was Drop Tail. The throughput value of both TCP and UDP traffic kept on increasing as the Buffer Size was increased. But there’s a drop in the UDP loss probability when the buffer size is kept as 30kB. Hence it seems favourable for both UDP and TCP traffic to maintain the buffer size between 35-45kB.

Fig. 16. Varying Buffer Sizes on Bottle Neck Topology with parameters like congestion window size and time to live with 15% of UDP Traffic

Fig. 17: Varying Buffer Sizes on Bottle Neck Topology with parameters like congestion window size and time to live with 20% of UDP Traffic

Here in Figure 17, a network of 80 Source-Destination pairs was simulated with 80 traffic flows of TCP and 20 Source-Destination pairs with traffic flow of UDP. Here the topology used was Bottle Neck and queue type used was Drop Tail. The throughput value of both TCP and UDP traffic kept on increasing as the Buffer Size was increased. But there’s a drop in the UDP throughput when the buffer size is kept as 15kB. Hence it seems favourable for both UDP and TCP traffic to maintain the buffer size between 35-45kB.

Fig. 17: Varying Buffer Sizes on Bottle Neck Topology with parameters like congestion window size and time to live with 20% of UDP Traffic
Fig 18: Varying Buffer Sizes on Bottle Neck Topology with parameters like congestion window size and time to live with 25% of UDP Traffic

For Figure 18, a network of 75 Source-Destination pairs was simulated with 75 traffic flows of TCP and 25 Source-Destination pairs with traffic flow of UDP. Here the topology used was Bottle Neck and queue type used was Drop Tail. The throughput value of both TCP and UDP traffic kept on increasing as the Buffer Size was increased. But there’s a drop in the UDP throughput when the buffer size is kept as 15kB. Hence it seems favourable for both UDP and TCP traffic to maintain the buffer size between 35-45kB.

Fig 19: Varying Buffer Sizes on Bottle Neck Topology with parameters like congestion window size and time to live with 30% of UDP Traffic

For Figure 19, a network of 70 Source-Destination pairs was simulated with 70 traffic flows of TCP and 30 Source-Destination pairs with traffic flow of UDP. Here the topology used was Bottle Neck and queue type used was Drop Tail. The throughput value of both TCP and UDP traffic kept on increasing as the Buffer Size was increased. But there’s a drop in the UDP throughput when the buffer size is kept as 25kB. Hence it seems favourable for both UDP and TCP traffic to maintain the buffer size between 30-45kB.

Fig 20: Varying Buffer Sizes Bottle Neck Topology with parameters like congestion window size and time to live with 35% of UDP Traffic

For Figure 20, a network of 65 Source-Destination pairs was simulated with 65 traffic flows of TCP and 35 Source-Destination pairs with traffic flow of UDP. Here the topology used was Bottle Neck and queue type used was Drop Tail. The throughput value of both TCP and UDP traffic kept on increasing as the Buffer Size was increased. But there’s a drop in the UDP throughput when the buffer size is kept as 30kB. Hence it seems favourable for both UDP and TCP traffic to maintain the buffer size between 40-45kB.

5 CONCLUSION

The Router’s Buffer Size is a very important parameter which has to be carefully set for lowering the congestion rates and thus lower the packet loss rates too. NS-2 is very useful tool for studying the packet losses by the trace files generated by the same.

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