Survey on Router's Buffer Sizing in Mixed Traffic

Sherrin Benjamin Jacob
Assistant Professor
D J Sanghavi College of Engineering
Mumbai

Shilpa Verma
Assistant Professor
Thadomal Shahani Engineering College
Mumbai

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 the loss of UDP packet also 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.

General Terms
Congestion Window Size, Packet Size, NS-2

Keywords
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 buffer size of a router B should be RTT*W, where RTT=average Round trip time and W=capacity of bottleneck link [1]. Arguably, router buffers are the single biggest contributor to uncertainty in the Internet. Buffers cause queuing delay and delay-variance; when they overflow they cause packet loss, and when they under- flow they can degrade throughput [3]

Buffer bloat is a very common phenomenon found in routers, switches, gateways and broadband gear. This occurs due to the presence of excessively large buffers in systems, particularly in communication systems. Buffers with bloated sizes have latency measured in seconds, rather than microseconds or milliseconds. According to the telephone standards, the maximum desirable latencies are in the 150–200 ms range, and human perception for some latency is as low as 10 ms. Although some buffering is required to smooth bursts in communications systems, but packet loss is the only way to signal the presence of congestion in the network, and congestion-avoiding protocols such as TCP rely on timely congestion notification to regulate their transmission speeds. When an oversized buffer is put into the system, there are traces of higher packet loss.

According to the networking fundamentals, datagram network with infinite storage, first-in-first-out queuing, and a finite packet lifetime will under overload or drop all packets. However some of the buffers in the Internet are effectively infinite in size. Not all packet loss is evil: some packet loss can be essential for correct operation. However once bloated buffers fill, there's no timely congestion notification by packet loss or explicit congestion notification. Only the buffers on either side of the bottleneck link fill, and if those buffers are not well managed, they can fill up completely, causing much higher packet loss. Other buffers in the path, remaining nearly empty, remain dark and undetectable. Any application that saturates a link with bloated buffers can induce buffer bloat pain, for instance uploading videos to YouTube, emailing messages with large images attached, backing up large files or file systems, downloading large files, such as ISO images, a Linux distribution image, or a movie via Bit torrent, watching Netflix, and even visiting certain kinds of web pages can all fill these buffers [2].

There has been vital discussions regarding the size of buffers required at core Internet routers from past few years. Recently researchers have made arguments with the help of theory and experimentation that proves

2. REVIEW OF LITERATURE
Operating the router buffers in the "anomalous region" can result in increased UDP packet loss, with only a marginal improvement in end-to-end TCP throughput, which is undesirable from a network operator's point of view. Extensive simulations in [1] show the presence of anomalous loss performance for real time traffic. These simulations consider the real video traces, short range Poisson models and long range fractional Brownian motion models along with several other TCP flows. These two quantitative models are used to explain the anomaly of packet loss. The 1st model captures the buffer sharing dynamics between real time and TCP traffic and also shows the impact of greedy nature of TCP traffic on the effective buffers available due to the real time traffic. The 2nd model is Markov chain based approach which gives a numerical evaluation of packet loss. In [1], it is suggested that several other TCP parameters such as round trip times, number of flows, and relative mix of long lived and short lived TCP flows also impact on the severity of anomaly. The sizes of UDP packets have an effect on the anomaly. When the sizes of the UDP packets were larger, the losses were also large. [1]

According to [4], impact of flow arrivals and departures in networks are the main motivation for the extensive research in Buffer sizing. Networks are characterized by a vast disparity in the operating speeds of access routers and core routers. When there are flow arrivals and departures, [4] show that the core routers are rarely congested even at high loads of 98%. Since there is no congestion on the core router, the flows are largely limited by their access speeds. The impact of small buffers on a single congested link where the access limitations are absent is very trivial. It is rather well known that TCP approximates processor sharing, when the file-sizes are large. Therefore, at any time, very few active flows are present in the network even at significantly high loads. For example, under processor sharing, even at 90% loading, the probability 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 trace appropriately. Operating the router buffers at a very small value can adversely impact the performance of both TCP and UDP trace. 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 associated with one type of Fractal—an object whose appearance is unchanged regardless of the scale at which it is viewed. 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.

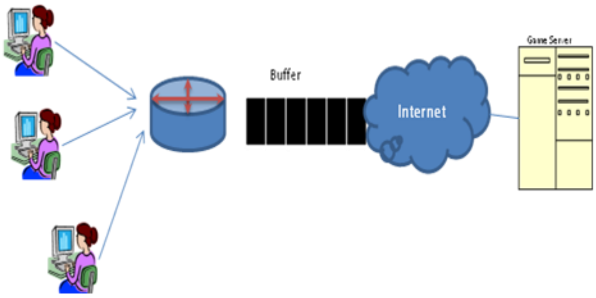

Fig 1 Scenario of Internet Café

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].

NS-2 (Network Simulation Version 2) is an object-oriented simulator developed as part of the VINT project at the University of California in Berkeley. This project was funded by DARPA in collaboration with XEROX Palo Alto Research Centre (PARC) and Lawrence Berkeley National Laboratory (LBNL). NS-2 is extensively used by the community of networking research. It also provides support for simulation of multicast, routing, TCP protocols over wired and wireless networks, etc. This simulator is event-driven and runs in a non-real time fashion. It consists of C++ core methods and uses TCL and Object TCL shell as an interface allowing the input file (simulation script) to describe the model to simulate. Users can define arbitrary network topologies composed of nodes, routers, links and shared media. A rich set of protocol objects can then be attached to nodes, which are usually called as agents. In NS-2 network physical activities are translated to events, events are queued and processed in the order of their scheduled occurrences. And the simulation time progresses with the events processed. And the simulation "time" may not be the real life time as "inputted". NS-2, can model essential network components, traffic models and applications. Typically, it can configure transport layer protocols, routing protocols, interface queues, and also link layer mechanisms.

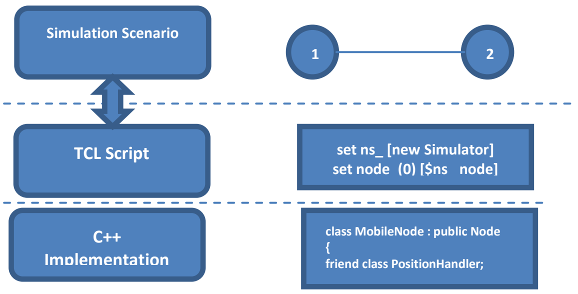

Fig. 2 Layered Structure of NS-2

From the NS-2 developer view, Figure 2 shows the layered architecture of NS. The event schedulers and most of the network components are implemented in C++ and available to TCL Script, thus the lowest level of NS-2 is implemented by C++, and the TCL script level is on the top of it to make simulations much easier to be conducted. Then, upon the TCL level, there's overview of the network, i.e. the simulation scenario. These elements are combined as so called NS-2 software. In order to successfully carry out one simulation, the data must be provided to NS-2:

1) Appearance of the network: The whole topology view of sensor network or mobile network, this includes the position of nodes with (x, y, z) coordinate, the node movement parameters, the movement starting time, the movement is to which direction, and the node movement speed with pausing time between two supposed movement.

2) Internal of the network: Since the simulation is on the network traffic, so it is important to inform the NS-2 about which are source nodes, about the connections to be made, what kind of connection to be used (simplex/duplex).

3) Configuration of the layered structure of each node in the network: This includes the detailed configuration of network components on sensor node, and a need to drive the simulation. It is necessary to direct the simulation results to a particular trace file, which is required to be mentioned in the TCL script, and also about how to organize a simulation process [11].

3. 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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