# Bufferbloat Problem

**CSC458 Computer Networks Programming Assignment 2**

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**Abstract** Bufferbloat refers to the existence of excessively large and frequently full buffers inside a network. [Sivov and Sokolov, 2012]

Most TCP congestion control algorithms rely on packet drops to determine the available bandwidth between two ends of a connection.[Allman and Paxson, 2009, El-Sayed et al., 2011] In general, TCP congestion control algorithms speed up the data transfer (via increasing congestion window size) until packets start to drop, then slow down the transmission rate. Under ideal conditions, we expect an equilibrium speed to be reached after a period of time of adjustments.

In a fast to slow transition hop, bufferbloat can easily occur. Let's consider the internet topology illustrated in the assignment handout (Assignment Topology). TCP will continue to increase the cwnd size since packets sent out are being buffered inside the intermediate router $s_0$ and with no packet being dropped. It will only decrease the cwnd size when buffer of $s_0$ is saturated, but that is already too late. In other words, the buffer in the intermediate router has turned the packet drops into an **un-timely** indication of congestion, which is bad since TCP rely on timely communication of congestion via packet drops.

Bufferbloat causes increase in queueing delay, and thus causes end users to experience increase in latency, which is the sum of transmission delay, processing delay, and queueing delay. [Sivov and Sokolov, 2012] Bufferbloat also causes jitters and decreases the overall throughput of the network.

## Methods

We emulate our Assignment Topology using mininet. Then, we simultaneously perform the following three tasks

- start a long lived TCP flow sending data from $h_1$ to $h_2$, and
- send 1 ping per 0.1 second from $h_1$ to $h_2$, and
- spawn an web server on $h_1$, and download the webpage from $h_1$ once every two seconds.

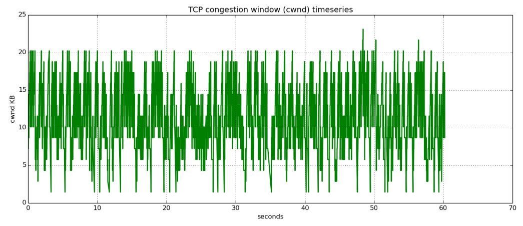
This simulation is repeated for three different queue sizes, $Q = 5/20/100$ pkts. Then, for each max queue size, we plot the time-series of the long-lived TCP flow's cwnd, the RTT reported by ping, the webpage download time, and the queue size at the router $s_0$.

## Results

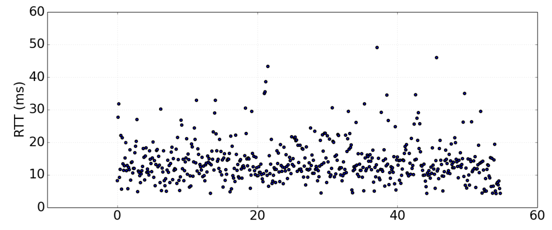
Table 1 provides an summary of mean and standard deviation of the fetch time in all three experiments with queue size 5, 20, and 100 pkts respectively, while Figures 1, 2 and 3 are plotted time-series for the long lived TCP flow's cwnd, RTT reported by ping, webpage fetch time, and queue size at the router.

| Queue Size | 5 | 20 | 100 |
|---|---|---|---|
| Mean(ms) | 0.72167 | 0.48278 | 1.40365 |
| Stddev(ms) | 0.47233 | 0.14989 | 0.71954 |

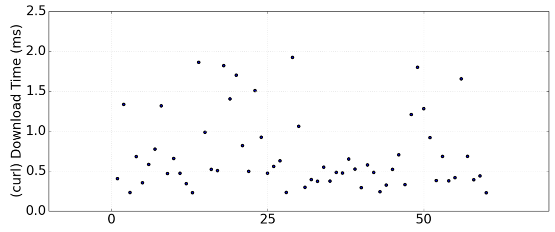
Table 1: Mean and standard deviation in fetch time for all three experiments.



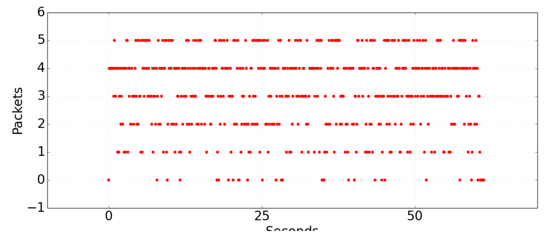
(a) Long-lived TCP flow's cwnd



(b) RTT reported by ping

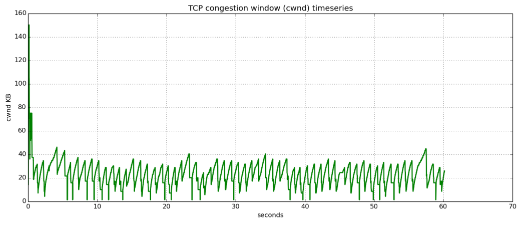


(c) Webpage download time

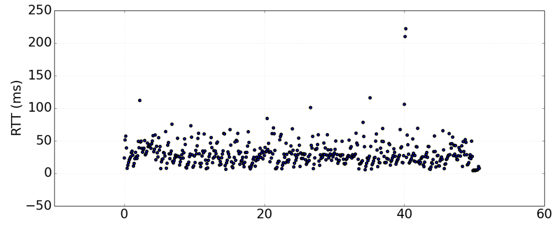


(d) Queue size at the router

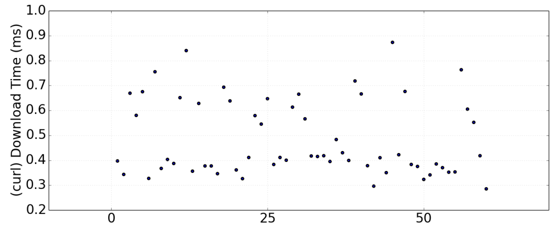
Figure 1: Long-lived TCP flow's cwnd, RTT reported by ping, webpage download time, and queue size at the router with max buffer size of 5.



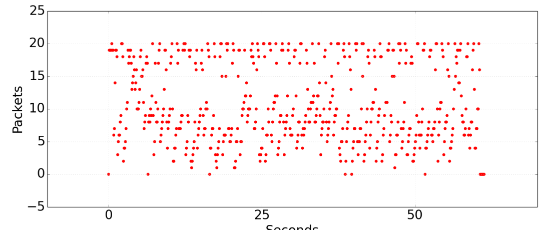
(a) Long-lived TCP flow's cwnd



(b) RTT reported by ping

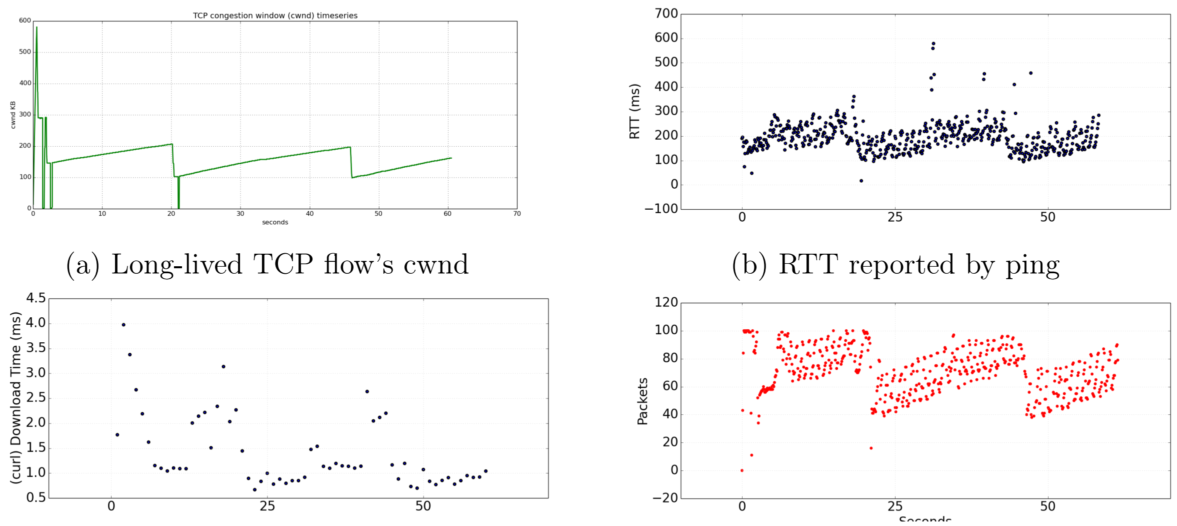


(c) Webpage download time



(d) Queue size at the router

Figure 2: Long-lived TCP flow's cwnd, RTT reported by ping, webpage download time, and queue size at the router with max buffer size of 20.



(a) Long-lived TCP flow's cwnd

(b) RTT reported by ping

(c) Webpage download time

(d) Queue size at the router

Figure 3: Long-lived TCP flow's cwnd, RTT reported by ping, webpage download time, and queue size at the router with max buffer size of 100.

## Discussion

Although the buffer is designed to smooth bursts out, it can hurt to have a buffer that is too large just the same as having a too small buffer. We illustrate this with the results we described in the "Results" section. When the queue size is 5, since the buffer is too small, packets get dropped frequently, and this is causing both a high delay and a high variance in this case. On the contrary when the max queue size is 100, the buffer size is too big, and this is causing bufferbloat to happen. Figure 3d illustrates that this big buffer is frequently (unnecessarily) full and seldomly empty, indicating the bufferbloat problem is occurring. When the max buffer size is 20, however, the buffer is neither too big nor too small; we see the lowest mean fetch time and standard deviation in all three experiments in this case.

Ipconfig reports a maximum queue length of `txqueuelen:1000` packets, and a MTU of 1500 bytes. Then, if we have a saturated buffer, we need

$$(1000 \times 1500 \times 8)/(100 \times 10^6) = 0.12\,\text{second} \tag{1}$$

Figure 4 shows[1] the three data points that we have, which are pairs of queue size and the average RTT time taken reported by ping. On the plot, the coordinates are marked as text next to the data points. We than fit a linear regression model to our three data points, and then the fitted line is plotted on the same plot. We have reason to believe, with some allowance for error, that the relationship is linear and can be described as

$$\text{RTT(ms)} = 2 \times \text{queueSize} \tag{2}$$

[1]This plot is generated with my script `plot_relationship.py` available in the same directory as all the other plot python files. It is also automatically invoked in my modified version of `run.sh` script. You will need NumPy and Scipy.stats to run this script, I used them for the fitting of linear regression model.

For the sake of mitigating the bufferbloat problem, we can try

- The probably simplest approach is to decrease the buffer sizes at each hop. This way, when congestion happen, buffers fill up quickly, and then rely on packet drops as a timely indication of congestion. This approach is, however, not optimal. We originally introduced buffers to deal with bursts of packets, and make networking more smooth in general. Reducing buffer size will cost us the ability to deal with bursts in the network.
- Use a delay based congestion avoidance algorithm rather than delay based. In this way, excessive buffering, even when there is no drop in packets, will signal that a congestion is occurring. We can then control the cwnd size based on this information. [Sivov and Sokolov, 2012]

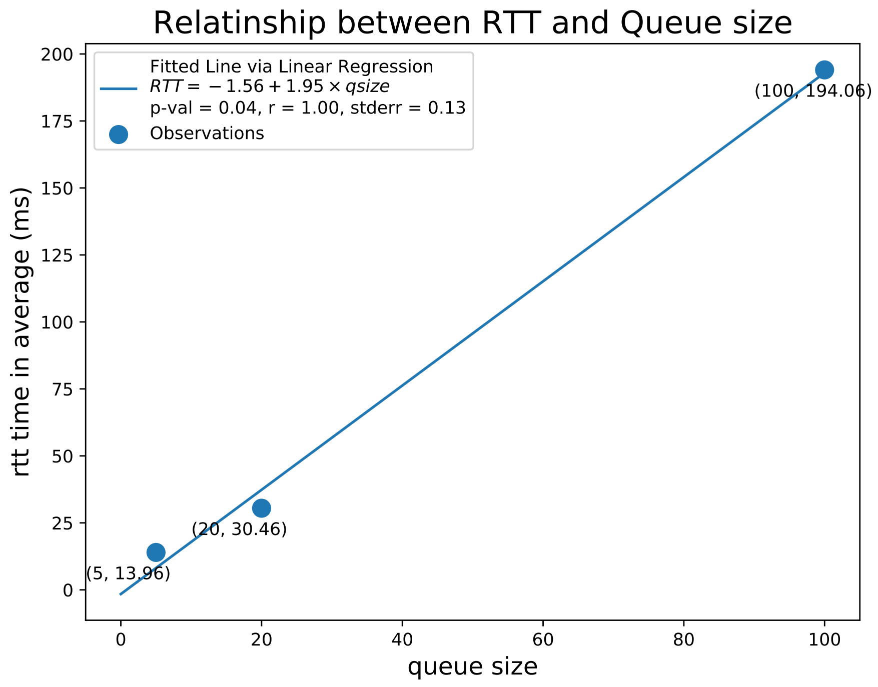


Figure 4: Scatter plot of RTT (in ms) and the max queue size used respectively. Plotted with the fitted line via linear regression. This plot is generated with my script `plot_relationship.py` available in the same directory as all the other plot python files. It is also automatically invoked in my modified version of `run.sh` script. You will need NumPy and Scipy.stats to run this script, I used them for the fitting of linear regression model.

## References

[Allman and Paxson, 2009] Allman, M. and Paxson, V. (2009). Tcp congestion control. Rfc, RFC Editor.

[El-Sayed et al., 2011] El-Sayed, A., HAGGAG, S., and EL-FESHAWY, N. (2011). A survey for mechanisms for tcp congestion control. *International Journal of Research and Reviews in Computer Science (IJRRCS)*, 02:676–682.

[Sivov and Sokolov, 2012] Sivov, A. and Sokolov, V. (2012). The bufferbloat problem and tcp: Fighting with congestion and latency.