COL724: Advanced Computer Networks

Assignment#:1

Buffer Bloat

Submitted by: Arihant Jammar [2022JCS2669]

Q1. Why do you see a difference in webpage fetch times with short and large router buffers?

Answer.

When a larger buffer size is employed, TCP adopts an approach where it doubles the sender's window size. This action allows more packets to be sent before waiting for acknowledgements. In situations of high traffic, this might initially appear advantageous, as more packets can promptly arrive in the buffer. However, it's important to consider the propagation delay, which in this experiment is measured at 6ms.

As the buffer accumulates an increasing number of packets due to the larger window size, the round-trip time (RTT) for these packets experiences an exponential increase. This phenomenon occurs because a substantial number of packets await processing in the buffer before being transmitted. Consequently, the packets eventually start to be dropped as the buffer overflows. This marks the initiation of congestion control mechanisms, as packet loss is perceived as a sign of network congestion. Consequently, TCP reduces the sender's window size in response to the detected congestion.

This reduction in the sender's window size, though necessary for congestion control, results in another increase in the RTT. The process unfolds in a cycle, where a larger buffer leads to a more pronounced RTT increase due to the prolonged queueing of packets, and this eventually prompts congestion control mechanisms that cause the sender's window size to contract, thereby inducing further RTT increments.

Conversely, when dealing with a smaller buffer size, a different dynamic emerges. The limited buffer capacity causes the sender's window size to be reduced each time the buffer limit is nearly reached. This swift response to buffer congestion allows the system to circumvent the substantial RTT increases experienced with larger buffers. This rapid adjustment is possible due to the congestion avoidance and control mechanisms, which take effect sooner in scenarios with smaller buffers.

Ultimately, the presence of congestion avoidance and control in a low buffer size context facilitates quicker adjustments to network conditions, ensuring that the web page request is transmitted more expeditiously. This dynamic contrasts with the situation in larger buffer sizes, where the prolonged RTT increments due to excessive packet accumulation in the buffer can lead to slower webpage fetch times. Therefore, the size of the router buffer can significantly impact network performance and the efficiency of congestion control mechanisms.

Q2. Bufferbloat can occur in other places such as your network interface card (NIC). Check the output of ifconfig eth0 on your VirtualBox VM. What is the (maximum) transmit queue length on the network interface reported by ifconfig? For this queue size, if you assume the queue drains at 100Mb/s, what is the maximum time a packet might wait in the queue before it leaves the NIC?

Answer: In the context of the **ifconfig** command, the **txqueuelen** parameter refers to the transmit queue length associated with a network interface. The **txqueuelen** value specifies the maximum number of packets that can be queued for transmission on that interface. This parameter is used to manage the flow of outgoing network traffic from the host.When a network interface sends out data, the data is usually queued in a transmit queue before being transmitted over the network.

Let's consider that the **txqueuelen** reported by my **ifconfig** for **eth0** is 1000 packets, and we know that the queue drains at a rate of 100 Mb/s.

Given:

- Transmit Queue Length (**txqueuelen**): 1000 packets
- Queue Drain Rate: 100 Mb/s = 100,000,000 bits/s
- Packet Size: 1500 bytes = 12000 bits

Calculation:

Time = (Queue size * Packet size) / (Queue drain rate)
Time = (1000 packets * 12000 bits) / (100,000,000 bits/s)

Time = 0.12 seconds

Therefore, with a txqueuelen of 1000, a queue drain rate of 100 Mb/s, and a packet size of 1500 bytes, a packet might wait a maximum of 0.12 seconds (or 120 milliseconds) in the queue before it leaves the NIC.

Q3. How does the RTT reported by ping vary with the queue size? Describe the relation between the two.

Answer: The Round-Trip Time (RTT) reported by the ping command is influenced by multiple factors in the network path. It can be broken down into different components, each contributing to the total RTT. Specifically, we can express the RTT as follows:

RTT on Ping = Propagation Delay on the Path to the Host Pinged + Queuing Delay on the Path to the Host + Propagation Delay on the Path from the Host to the Server/Host Who Pinged + Queuing Delay on the Path from Host to Server/Host

:*Queueing Delay* = Queue Size * Packet Delay

:*Packet Delay* = Time spent for the all the packets in the Queue to be serviced.

In our experiments, where the propagation delay remains constant (e.g., at 6ms), it's evident that RTT is also dependent on queuing delay. The queuing delay occurs when packets need to wait in network queues due to congestion or other factors. As the queue size increases, the waiting time for packets in the queue also increases, leading to a longer queuing delay. Consequently, the queuing delay contributes to the overall RTT, and a larger queue size results in an extended RTT.

Q4.Identify and describe two ways to mitigate the bufferbloat problem.

Answer: Bufferbloat is a phenomenon that occurs when large buffers in network devices lead to excessive delays and latency in packet transmission. To mitigate bufferbloat and improve network performance, there are several strategies and techniques that can be employed.

Here are two ways to mitigate the bufferbloat problem:

1>**Active Queue Management with ECN (Explicit Congestion Notification):**

ECN allows routers to notify endpoints about network congestion before bufferbloat becomes a severe problem.

ECN works by marking packets with congestion indications instead of dropping them by changing some bits in it, providing a more subtle way to signal congestion. By combining Active Queue Management (AQM) with ECN, bufferbloat can be effectively controlled.

The endpoints (receivers) monitor the ECN markings and respond by reducing their transmission rates, acknowledging the network's congestion status. This leads to proactive congestion control, preventing queues from growing excessively.

2> Bufferbloat can be mitigated through effective queue length management (prioritizing smaller length queues) and the implementation of the **RED (Random Early Discard)** algorithm. By actively monitoring queue lengths and randomly dropping packets before queues become overly congested, RED prevents excessive buffering. This ensures that network traffic remains smooth and responsive, reducing the likelihood of bufferbloat-related latency spikes.

Q5. Describe how and why your results change when you re-run the emulation.

Answer:

When re-running network emulations, we observe changes in the results due to the dynamic nature of traffic patterns.

Bursty Traffic: The nature of the traffic being generated can lead to variability. Bursty traffic, where packets arrive in quick succession, can impact queue occupancy and RTT differently compared to more evenly distributed traffic.

Packet Sizes: Variations in packet sizes can affect queuing behavior. Smaller packets might be processed faster, leading to lower queuing delays, while larger packets can contribute to buffer congestion and higher delays.

Background Activities: Other applications and devices on the network can introduce varying levels of load and congestion, impacting overall network performance.

Data Rate Variability: Applications with fluctuating data rates can lead to varying levels of congestion, affecting buffer occupancy and RTT.

The accuracy of measurement tools can introduce variability in reported results.

Note:

Queue Size :20 packets

Average Download time: 39.54833 units

Standard Deviation: 0.00216

Queue Size :100 packets

Average Download time:39.54933 units

Standard Deviation: 0.00141