Receive Buffer Dynamics and OS Scheduling

Thesis by

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To my mom, whose advanced degree set a wonderful example.

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Abstract

We perform an empirical study of the effects receive-side resource sharing has on a TCP transaction. Our variable of focus is the buffer used to deliver packets to the application. Specifically, our interest lies with how its size affects throughput amidst background load.

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Chapter 1

Introduction

TCP provides robust and reliable stream-based communication between pairs of clients. TCP not only maintains ordering amongst packets, but has the ability to dynamically control the rate at which those packets are exchanged. One of the ways in which flow is controlled is by grouping sequences of packets into windows. Within TCP, a window is the maximum amount of data a sender is allowed to transmit before receiving an acknowledgment for that data. The maximum size of the window is negotiated between endpoints during the connection establishment phase. Once a connection is established, a sender will transmit no more than the agreed upon amount of data during one transaction. It follows that during the lifetime of a connection, the amount of data outstanding, or unacknowledged, will never be more than the size of this window.

Senders do not necessarily have to wait for all data to be acknowledged before sending more. When a receiver sends acknowledgment of data received, it includes the amount of remaining buffer space it has available—this is commonly referred to as the advertised window. Upon receiving the acknowledgment, if the sender has more data to send, it will send no more than this advertised amount. Thus, one of the limiting factors in data transmission is the amount of data the receiver can handle: the receive buffer.

The proper size of the receive buffer is not necessarily a straightforward calculation. As a general rule, the bandwidth-delay product (BDP) is commonly used to determine its minimum size. BDP is the product of bandwidth and round-trip time (RTT) between two end points. It can vary dramatically depending on network topologies and traffic dynamics. In our test network BDP was an underestimate of proper receive buffer size.

In the past, the interaction between TCP operation and process scheduling put a physical upper limit on the size of the data structure [2]. In modern operating systems, however, this is not the case: the maximum size of the receive buffer is limited only by system resources. As such, it is not uncommon for network operators to use an arbitrarily large receive buffer size in their network configuration. In our testing, large receive buffer sizes did not hinder transmission, but they did put an unnecessary strain on system resources. Thus, the importance of the receive buffer, not just as a network tool, but as a system resource, motivates our work.

1.1 Motivation

Packet buffering is an essential part of TCP network transmission. It can be found at the sender, receiver, and all points in between. For intermediary nodes, buffering facilitates packet transmission amidst bursty traffic and parallel flows. At the endpoints, buffering enforces the reliable, in-order, delivery guarantees that TCP specifies.

It is our hypothesis that buffering at the endpoints also shelters a connection from shared system resources. In much the same way that router buffers can minimize the negative effects of cross traffic, receive side buffering should be able to reduce the impact of resources, such as memory and CPU, that are shared amongst other applications. To test our hypothesis, we look at receive buffer dynamics from the vantage point of the scheduler. We study the manner in which the buffer operates while sharing resources and draw conclusions as to how its size not only affects throughput, but the receiving system as a whole.

Chapter 2

Methodology

Performing a study of this nature required operating system modifications to monitor the networking subsystem and a very low-latency network. We monitored OS networking on a per-process, per-cycle basis to identify how the buffer affected bandwidth. Often, when monitoring system behavior, samples are taken at periodic time intervals; however, this method was insufficient for the type of analysis we were trying to perform. For this, fine granularity and an emphasis on process scheduling were paramount. To properly test our hypothesis we needed to find out whether or not the work being performed to receive data on behalf of the application was sufficient for the amount of data TCP was trying to transmit. Looking at processing from the standpoint of the scheduler was the best way to achieve that.

Aside from OS customizations, we also required a specialized, low-latency network. Our goal was to isolate the operating system as the only bottleneck within the connection, requiring a network fast enough that packet transmission was not a concern. The specifics of this, as well as our OS changes, are described here.

2.1 Implementation

Socket monitoring works on a per-socket basis. When a socket is created, monitoring of that socket is turned off by default, allowing existing network applications to operate normally. Monitoring is turned on and off by sending an `SO_MONITOR` option to the `setsockopt` system call. Once enabled, that socket had its snapshot taken prior to the owning process beginning execution, and again once execution was finished; that is, on process activation and deactivation. We often refer to this period as a "timeslice" or a "cycle."[1] To acquire very specific TCP related statistics, some metrics were tracked while an owning process was not executing. Often the case in Linux, data can enter or leave

[1] These terms are meant to describe the time that a process is executing, not necessarily when it is running. In many operating systems, including Linux, a process may be marked as running and even on a run queue, but may not have the CPU. We are primarily interested in the time that a user process is given CPU time—our use of the terms timeslice, active time, and cycle refer to such.

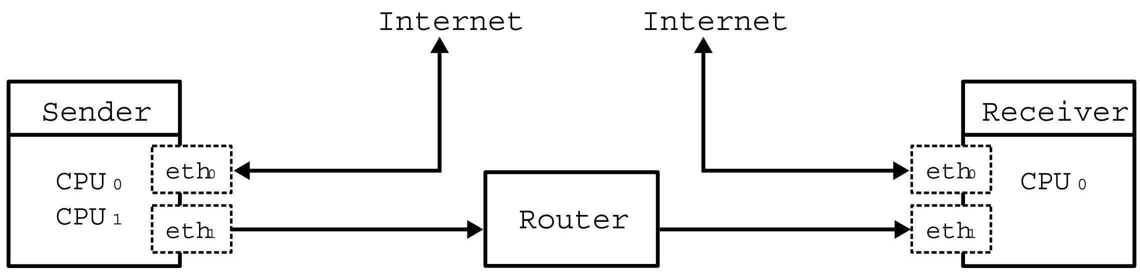


Figure 2.1: Our experimental network setup, consisting of a sender and receiver connected via an intermediary router. Link speed across our network was 10 Gigabits per second with an average round trip time of 0.057 milliseconds. Separate NICs with public IPs connecting each machine to the Internet.

a socket even while the owning process is not active. Out of context monitoring was helpful, for example, to gain TCP window statistics. We implemented these socket and scheduler changes within the Linux 2.6.19.2 kernel. When set, socket monitoring decreased bandwidth by approximately 3%. To maintain accuracy, the option was set immediately prior to any calls that transmitted data, and always by the same thread that handled I/O on the socket. It was disabled once the TCP connection was terminated.

2.2 Experimental Setup

2.2.1 Hardware setup

Our test bed consisted of a sender and receiver connected via a Cisco 7609 Router (Figure 2.1). Our sender and receiver had equivalent 2.4GHz 64-bit AMD processors, however, our sender had two cores enabled, while our receiver had only one. Each machine had two Ethernet interfaces: one connecting them to the Internet, the other connecting them to the private, experimental, network. Their private network interface cards were Neterion Xframe II's operating at 10Gb per second on a 64-bit 133MHz PCI-X I/O bus. When the connection was otherwise idle, we achieved an average ping-based round trip time of approximately 0.053 milliseconds. We used maximum transfer unit of 9000 bytes for our testing. Each machine also had 1 MB of L2 cache, and 4GB of main memory.

2.2.2 Traffic Generation

Traffic for our tests was generated using iperf.[2] We altered the standard 2.0.2 version as described in Section 2.1 so that socket measurements could be made. An experiment consisted of several data transmission sessions. For a given session, data was transferred from the sender to the receiver for

[2] *iperf* is a tool specifically designed "for measuring TCP and UDP bandwidth performance." It is distributed and maintained by the Distributed Applications Support Team at the National Laboratory for Applied Network Research.

60 seconds. Socket receive buffer size varied between transmission sessions, varying between 6KB and 1.5MB. We found this range to be small enough to observe peaks in bandwidth, yet large enough to see trends that remained consistent through much larger (≈100MB) buffer sizes. On both sender and receiver, the application buffer remained constant at 8KB.[3]

2.2.3 Background Load

To generate background load on our receiver, we used a custom application that stressed a tunable combination of CPU and memory. In a tight loop, the application read from and wrote to random places in a given memory area, sleeping for a given amount of time (at microsecond granularity) between iterations. Both memory and sleep time were controlled via command line parameters. Our application was able to allocate anywhere from 0 bytes to 200% of the system RAM amount (going beyond 100% invoked the system pager). The sleep parameter served as a sort of CPU usage throttle.

To simulate load within the receiving application itself, we forced the receiver to sleep between network read calls. As with our CPU/memory application, this was controlled at microsecond granularity.

2.3 TCP Parameters

Buffer size was controlled via the `/proc` file system. Among other things, the `/proc` file system allows users to access kernel-level variables, including several pertaining to TCP operation. Many of the variables within `/proc` allow not only read access, but write access as well without interrupting OS operation. Only a small subset of `/proc` variables were altered for our testing:

`tcp_*mem` There are three TCP memory variables that affect memory usage: tcp_rmem, tcp_wmem, and tcp_mem. Respectively, they control memory management of TCP receive socket buffers, send socket buffers, and the overall TCP stack size. Each variable consists of three values the kernel interprets as minimum, maximum, and default memory levels (referred to herein as v_{min}, v_{def}, and v_{max}, respectively). These values act as memory bounds for dynamic allocation of TCP related objects.

`tcp_rmem` During each test $v_{min} = v_{def} = v_{max}$ on the receiver. This ensured internal dynamic memory management was disabled. On the sender, v_{def} was larger than the maximum receive value, and that v_{max} was twice the value of v_{def}.

[3] The application buffer should not be confused with the socket buffer. The application buffer is a concept that exists in user space. It is the memory region that is accessed by the system calls `read` and `write`, for example. The socket buffer, on the other hand, is a kernel space data structure controlled by the internal networking stack.

`tcp_wmem` On both the sender and the receiver, we set v_{def} to be larger than the maximum receiver buffer size and $v_{max} = 2 * v_{def}$. This ensured that we were not congestion window limited.

`tcp_mem` v_{max} of this variable was more than the v_{max} variables of tcp_rmem and tcp_wmem combined.

`tcp_moderate_rcvbuf` This Boolean value is used by the kernel to turn TCP socket buffer autotuning on or off. On the receiver, this was always set to "false" in order to ensure that our requested buffer sizes were respected.

`tcp_no_metrics_save` A Boolean value that controls TCP caching within the kernel. To get consistent results between tests, this value was set to false on both the sender and receiver.

All other networking related variables within `/proc` remained at their system defaults.

Chapter 3

Results

Using the methods described in the previous chapter, our goal was to determine the effects the receive buffer has on throughput. Specifically, whether or not increased buffering at the receiver can minimize the effects of actions adverse to receiving. We considered two cases:

1. The receiver itself was the bottleneck. In this case the receiver emulated a typical server, doing performing work over the incoming packets.

2. The system as a whole was the bottleneck. Although the receiver was able to operate unhindered, the system was not entirely devoted to receiving. Instead, other applications were running in parallel that consumed shared resources.

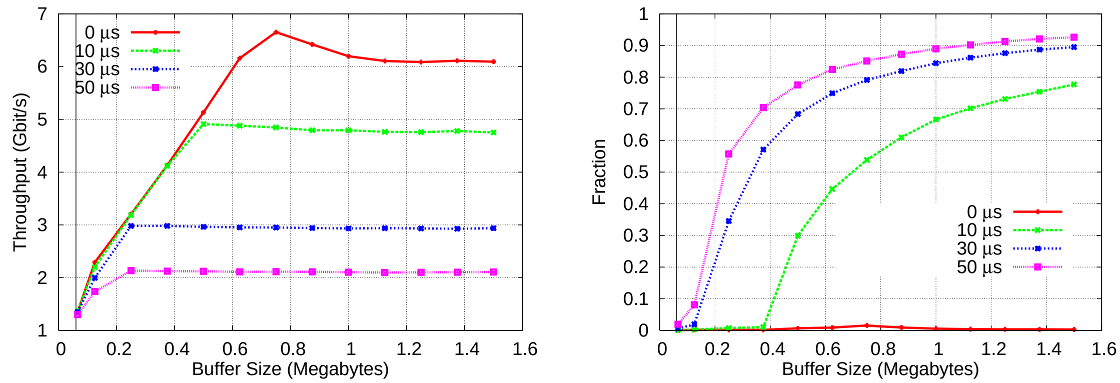
Our primary metric of comparison was the amount of bandwidth the TCP connection was able to achieve at various receive buffer sizes. We also, however, look at various scheduling-based metrics to help explain our bandwidth curves.

3.1 Working Receiver

In our working receiver tests, the receiving application was working while it received packets. To simulate work, the receiver delayed consecutive calls to `read` by a given number of microseconds.[1] We studied delay values ranging from 0 to 50 microseconds. The zero microsecond case represented a non-working, uninterrupted, receiver. While a somewhat unrealistic scenario, it provided a good baseline for comparison. Under these ideal conditions, we were able to achieve a maximum throughput of approximately 6.66Gbps.[2] From Figure 3.1(a), throughput increased with increased buffer size. However, after a certain point (≈1 MB) this one-to-one correspondence stopped—no matter how much larger the receive buffer size became, the number of bits transferred per second remained constant.

[1] This was implemented using a tight loop; the application did not sleep.

[2] Although we had 10Gb of total bandwidth available, our I/O bus limited us to 7.93Gbps. Thus, our maximum actual transmission rate was about 84% of the theoretic limit.



(a) Throughput with receiver delay. Curves represent delay values between consecutive `read` calls at the receiver, in microseconds.

(b) Receive buffer utilization. Curves represent the percentage of buffer used with respect to the amount of buffer available.

Figure 3.1: Bandwidth and buffer utilization at increasing receive buffer sizes. The vertical line near the y-axis represents the BDP based buffer size.

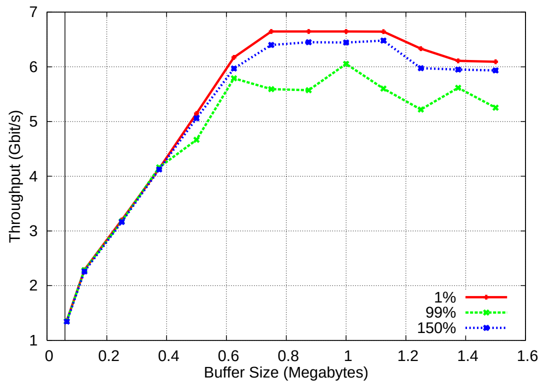
The amount of time spent between `read` calls had a dramatic affect on throughput, as seen in Figure 3.1(a): increasing receiver delay decreased overall bandwidth. Figure 3.1(b) shows the percentage of buffer space used at the end of the receivers active time. As the amount of time spent by the receiver increases, so to does the amount of buffer space used. In line with our hypothesis, TCP is trying to shield the connection from a slow receiver via buffering.

3.2 Resource Sharing

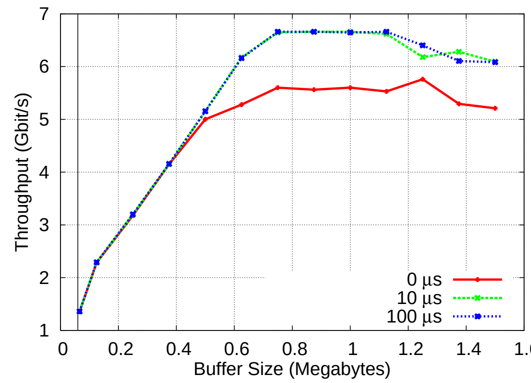
We sought to ascertain the effect of receive buffer size in the midst of background CPU and memory load. One of the benefits of the receive buffer is that it prevents a TCP connection from collapsing during periods of bursty traffic; we want to know whether the same can be said for periods of high system load. As mentioned in Subsection 2.2.3, we used a custom application to generate CPU and memory load. In these tests, the receiver did no processing of incoming packets.

3.2.1 Memory

First, we considered the affect memory allocation had on bandwidth. For our experiments, we considered between 1 and 150 percent system memory allocations; roughly 41MB and 6GB, respectively. From Figure 3.2(a), while memory allocation did have an affect on perceived throughput, receive buffer size did not help. If our initial hypothesis was correct, the correspondence between buffer allocation and maximum throughput should have been different depending on the amount of system



(a) Throughput at various memory utilization. Lines represents the percentage of system memory (4GB) used.



(b) Throughput at various background CPU utilization. Each line is the amount of time our background load application slept between successive operations.

Figure 3.2: Bandwidth observed by our receiving application as it shared different system memory and CPU. In neither case did the size of the receive buffer play a role in shielding the receiver from the effects on the background load.

memory allocated. That is, if r_i represents the optimal buffer size[3] with i% of system memory consumed, then the following relation should have held: $r_1 < r_{99} < r_{150}$. In reality, when 99% of the system memory was utilized, a receive buffer of approximately 768KB was optimal—the same buffer that was optimal in the 1 and 150% memory utilization cases.

As an aside, we consistently observed that the relationship between memory usage and throughput was not completely linear: bandwidth at 150% memory utilization is better than bandwidth experienced when 99% of the memory was used—a counter intuitive observation. Our memory application was independently verified, so we assume the effect to be an artifact of the Linux paging system.

3.2.2 CPU

Our hypothesis was also tested against CPU sharing. To conduct the experiments, our memory application slept between successive memory accesses. Sleeping for zero seconds caused the application to become not only memory bound, but CPU bound as well, as it had to calculate the array offsets. This effect was minimized by sleeping for longer periods of time between loop iterations. We studied sleep values between 0 and 100 microseconds. Results from these tests can be seen in Figure 3.2(b). As was the case in our memory testing (Subsection 3.2.1), the receive buffer was unable to shield the receiver from the effects of processor sharing. In this case, the optimal buffer size was the same, irrespective of background load.

[3] The smallest buffer size that allows our receiver to achieve the highest noticeable bandwidth.

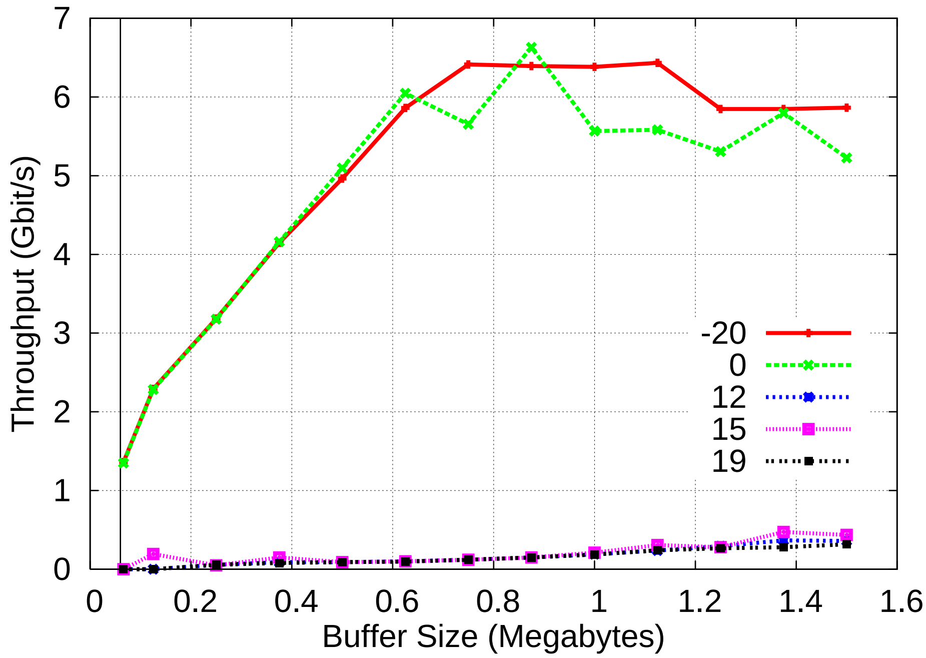


Figure 3.3: Bandwidth of priority sharing.

3.2.3 Priority

Finally we studied how a processes priority affected bandwidth and resource sharing. Tests were run in the same fashion as described in Subsection 3.2.2, however, the priority of the receiving application was varied. More specifically, we ran our receiving application in conjunction with the background load generator, which used 99% of the system memory and did not sleep between memory accesses. Linux assigns priorities to processes with what is known as a *nice* value—the "nicer" a process, the more willing it is to share the CPU. Thus, high priority processes have negative nice values, while low priority processes have higher nice values. By default, processes have a nice value of 0. In the case of our experiments, the priority of our background application remained constant at its default value. The priority of our receiving application varied, however, between minimum and maximum Linux nice values of -20 and +19.

Overall, and in line with theory [3], scheduling priority had the most notable effect on throughput in all of our background load tests; results can be seen in Figure 3.3. The default priority allowed our receiving application to be hampered somewhat. At the highest priority however, maximum bandwidth was achieved irrespective of the background load.

In line with our hypothesis, buffering did have an effect on throughput, albeit slight. When the receiving application was assigned a low priority there was a linear relation between buffer size and throughput.

Chapter 4

Conclusion

Linux does a very good job, overall, of getting data from the network card to the receiver. Under ideal conditions, data was transferred from internal buffers to the application within the application context. We found that usage of the receive buffer only became noticeable when the receiving application was internally hindered; when it had to both receive and do work. External factors, such as resource sharing amongst other applications, had an adverse affect on throughput; contrary to our hypothesis, however, receive buffer size did nothing to improve this perceived throughput.

4.1 Future Work

4.1.1 Networking disconnect

As mentioned in Section 3.1 if the receiver had more than a few microseconds worth of work on the received data, the relationship between buffer allocation and buffer used became linear. That is, the more space available to TCP, the more space it used. Unfortunately, this relation did not carry over to throughput—TCP was able to get the same amount of throughput using much less buffer space. We think of this as a disconnect between TCP and the receive buffer. The operating system should not use all available resources just because it can.

This problem could probably be solved by increasing the amount of feedback between TCP and the application. The current Linux kernel is able to do receive aware auto-tuning, which is a step in this direction. Currently, the kernel keeps track of the amount of time a packet is in the TCP receive state—the time difference between a packet being put in the receive queue and that same packet being delivered to the application. The shortcoming we notice, however, is more so in the length of time between packet delivery and subsequent packet transmission. We are not out to improve windowing, but to improve system resource utilization.

4.1.2 Logarithmic throughput

During the course of testing, several interesting, but somewhat tangential observations were made. One such observation that warrants further investigation is the effect of the maximum transfer unit (MTU) on bandwidth at varying receive buffer sizes. It is well known, and confirmed with our own testing, that MTU plays a large role in the maximum achievable throughput between two points. We also observe that the buffer size required to achieve that maximum throughput is logarithmically dependent on the MTU. Some have speculated that this phenomenon is a result of Linux memory management [1]; however, that has not been definitively proven, nor formally modeled. Having an understanding of this relationship would greatly improve the accuracy of receive buffer auto-tuning.

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