EE542-Lab 03-Project Report

Network Measurement

Presenter: Boyang Xiao

USC id: 3326-7302-74

Email: boyangxi@usc.edu

Index

- Exploring Bandwidth and Throughput
- Bandwidth, Delay, and Loss
- Secure copy

Exploring Bandwidth and Throughput

- 1. Measuring TCP & UDP throughput using iPerf3
    - UDP testing results:

```
ubuntu@ip-10-0-2-86:~$ iperf3 -u -c 10.0.2.163 -p 5201 -b 2000M
Connecting to host 10.0.2.163, port 5201
[  4] local 10.0.2.86 port 48455 connected to 10.0.2.163 port 5201
[ ID] Interval           Transfer     Bandwidth       Total Datagrams
[  4]   0.00-1.00   sec   179 MBytes  1.50 Gbits/sec  22915
[  4]   1.00-2.00   sec   177 MBytes  1.49 Gbits/sec  22714
[  4]   2.00-3.00   sec   171 MBytes  1.44 Gbits/sec  21927
[  4]   3.00-4.00   sec   182 MBytes  1.53 Gbits/sec  23345
[  4]   4.00-5.00   sec   184 MBytes  1.54 Gbits/sec  23501
[  4]   5.00-6.00   sec   181 MBytes  1.51 Gbits/sec  23113
[  4]   6.00-7.00   sec   180 MBytes  1.51 Gbits/sec  23082
[  4]   7.00-8.00   sec   184 MBytes  1.54 Gbits/sec  23535
[  4]   8.00-9.00   sec   185 MBytes  1.55 Gbits/sec  23695
[  4]   9.00-10.00  sec   186 MBytes  1.56 Gbits/sec  23845
- - - - - - - - - - - - - - - - - - - - - - - - -
[ ID] Interval           Transfer     Bandwidth       Jitter    Lost/Total Datagrams
[  4]   0.00-10.00  sec  1.77 GBytes  1.52 Gbits/sec  0.038 ms  452/231670 (0.2%)
[  4] Sent 231670 datagrams
```

The bandwidth should be **1.42 Gbits/sec** for UDP.

Exploring Bandwidth and Throughput

- 1. Measuring TCP & UDP throughput using iPerf3
  - TCP testing results:

```
ubuntu@ip-10-0-2-86:~$ iperf3 -c 10.0.2.163 -p 5201
Connecting to host 10.0.2.163, port 5201
[  4] local 10.0.2.86 port 45728 connected to 10.0.2.163 port 5201
[ ID] Interval           Transfer     Bandwidth       Retr  Cwnd
[  4]   0.00-1.00   sec   451 MBytes  3.78 Gbits/sec   21   1.60 MBytes
[  4]   1.00-2.00   sec   495 MBytes  4.15 Gbits/sec   12   1.43 MBytes
[  4]   2.00-3.00   sec   502 MBytes  4.21 Gbits/sec    4   2.07 MBytes
[  4]   3.00-4.00   sec   500 MBytes  4.19 Gbits/sec    7   2.37 MBytes
[  4]   4.00-5.00   sec   504 MBytes  4.23 Gbits/sec  122   1.89 MBytes
[  4]   5.00-6.00   sec   504 MBytes  4.23 Gbits/sec   24   1.75 MBytes
[  4]   6.00-7.00   sec   488 MBytes  4.10 Gbits/sec    8   1.48 MBytes
[  4]   7.00-8.00   sec   512 MBytes  4.30 Gbits/sec    7   2.07 MBytes
[  4]   8.00-9.00   sec   490 MBytes  4.11 Gbits/sec   20   1.72 MBytes
[  4]   9.00-10.00  sec   510 MBytes  4.28 Gbits/sec    7   1.74 MBytes
- - - - - - - - - - - - - - - - - - - - - - - - -
[ ID] Interval           Transfer     Bandwidth       Retr
[  4]   0.00-10.00  sec  4.84 GBytes  4.16 Gbits/sec  232             sender
[  4]   0.00-10.00  sec  4.83 GBytes  4.15 Gbits/sec                  receiver
```

The bandwidth should be **4.16 Gbits/sec** for TCP.

Exploring Bandwidth and Throughput

- 2. Measuring TCP & UDP throughput with Latency on Router VM

Protocol Type	Latency(ms)	Bandwidth	Jitter	Datagram Loss
UDP	10	1.36 Gbits/sec	0.032 ms	0.1%
	50	1.25 Gbits/sec	0.030 ms	5.5%
	200	1.46 Gbits/sec	0.029 ms	81%
	500	1.32 Gbits/sec	0.032 ms	91%

- Conclusions:
  - Latency will not affect UDP's bandwidth or jitter
  - Latency will make UDP's datagram loss increase more and more faster

Exploring Bandwidth and Throughput

- 2. Measuring TCP & UDP throughput with Latency on Router VM

Protocol Type	Latency(ms)	Throughput-sender	Throughput-receiver
TCP	10	999 Mbits/sec	999 Mbits/sec
	50	101 Mbits/sec	101 Mbits/sec
	200	9.71 Mbits/sec	7.77 Mbits/sec
	500	1.80 Mbits/sec	1.02 Mbits/sec

- Conclusions:
  - Higher latency will make TCP's throughput grow lower
  - When latency is higher than some point, the throughputs as a sender/as a receiver will become different from one single node.

Exploring Bandwidth and Throughput

- 3. Measuring TCP & UDP throughput with Ethernet link limit on Router VM

Protocol Type	Ethernet link limit	Throughput-sender	Throughput-receiver	Datagram Loss(UDP)
UDP	10Mbps	1.37 Gbits/sec	NULL	99%
	100Mbps	1.40 Gbits/sec	NULL	93%
	1000Mbps	1.51 Gbits/sec	NULL	35%
TCP	10Mbps	10.3 Mbits/sec	9.97 Mbits/sec	NULL
	100Mbps	99.4 Mbits/sec	99.0 Mbits/sec	NULL
	1000Mbps	966 Mbits/sec	964 Mbits/sec	NULL

- Conclusions:
  - Ethernet link limit can affect both UDP and TCP transferring.
  - For TCP, it can directly make TCP throughput the same as the link limit, because TCP will make sure every datagram is ACK-ed and the throughput can be only the link limit
  - For UDP, the whole throughput does not change too much with various link limit, but the limit affect the datagram loss, because not all datagrams can be delivered with a link limit on the router.

Exploring Bandwidth and Throughput

- 4. Measuring TCP & UDP throughput when changing the window sizes on Client and Server
- Experiment results:
  - For TCP:
    - Changing the window sizes **does not** affect the results when there are latency on the Router
    - Changing the window sizes **does not** affect the results when there are link rate limits on the Router
  - For UDP:
    - Changing the window sizes **does not** affect the results when there are latency on the Router
    - Changing the window sizes **does not** affect the results when there are link rate limits on the Router

Exploring Bandwidth and Throughput

- **5. Measuring TCP throughput with different window sizes**
  - **Router link rate limit scenarios:**

Router Link Latency	Window size	Throughput-sender	Throughput-receiver
10 ms	64 Kbytes	1.12 Gbits/sec	1.12 Gbits/sec
	128 Kbytes	1.12 Gbits/sec	1.12 Gbits/sec
	256 Kbytes	1.12 Gbits/sec	1.12 Gbits/sec
100ms	64 Kbytes	105 Mbits/sec	104 Mbits/sec
	128 Kbytes	107 Mbits/sec	107 Mbits/sec
	256 Kbytes	105 Mbits/sec	104 Mbits/sec
500ms	64 Kbytes	7.30 Mbits/sec	6.41 Mbits/sec
	128 Kbytes	2.01 Mbits/sec	0.92 Mbits/sec
	256 Kbytes	4.85 Mbits/sec	3.22 Mbits/sec

Exploring Bandwidth and Throughput

- **5. Measuring TCP throughput with different window sizes**
  - **Router link latency scenarios:**

Link rate limit	Window size	Throughput-sender	Throughput-receiver
10Mbps	64 Kbytes	10.4 Mbits/sec	9.98 Mbits/sec
	128 Kbytes	10.7 Mbits/sec	9.96 Mbits/sec
	256 Kbytes	10.7 Mbits/sec	9.97 Mbits/sec
100Mbps	64 Kbytes	99.4 Mbits/sec	99.0 Mbits/sec
	128 Kbytes	99.2 Mbits/sec	99.8 Mbits/sec
	256 Kbytes	99.4 Mbits/sec	99.0 Mbits/sec
1000Mbps	64 Kbytes	965 Mbits/sec	963 Mbits/sec
	128 Kbytes	966 Mbits/sec	964 Mbits/sec
	256 Kbytes	957 Mbits/sec	956 Mbits/sec

Exploring Bandwidth and Throughput

- 5. Measuring TCP throughput with different window sizes
  - Conclusions:
    - In the internet latency and rate limit scenarios, changing the window sizes seems do help to the TCP throughput
    - When the latency is very large, the throughput can be unstable under different window sizes
    - If the link is at 100Mbit/s and 25ms of delay, the window size should be 128 Kbytes to maintain a highest throughput. But my experiments tell me that it cannot get a full 100Mbit/s throughput (actually only 99.2Mbit/s), because there is latency on the link and the ethernet connections are not stable.

Bandwidth, Delay, and Loss

- 1. Using ping to test RTT between machines

Router latency	RTT between Server and Client
0 ms	0.774 ms
25 ms	50.972 ms

- 2. Measuring the bandwidth-delay product for this link

Protocol Type	Router latency	Bandwidth	RTT	bandwidth-delay product
UDP	25 ms	1.42 Gbits/sec	50.972 ms	74.11 Mbits/sec
TCP	25 ms	4.16 Gbits/sec	50.972 ms	217.13 Mbits/sec

Bandwidth, Delay, and Loss

- **3. If you left the send/receive windows at their default values, would this link perform well?**
  - From the experiments above, the send/receive window sizes do not affect the link performance that much, so this link will not perform well.

- **4. Can we scale the send/receive windows to see the effect of a bandwidth-delay product/TCP window size mismatch?**
  - This does not make sense, because we should decrease the window sizes to see the difference but not scale it.

Bandwidth, Delay, and Loss

- **5&6. Set the send/receive window values to something too small and measure the TCP throughput**

Router link latency	Window size	Throughput-sender	Throughput-receiver
1 ms	10 Kbytes	958 Mbits/sec	956 Mbits/sec
5 ms	10 Kbytes	962 Mbits/sec	960 Mbits/sec
25 ms	10 Kbytes	970 Mbits/sec	970 Mbits/sec

  - If we use a very small window size, with larger link latency, the TCP throughput can even grown higher. From the iPerf3 log printed on the terminal, we can find that the TCP will adjust the window size to maintain a highest speed even if the network kernel buffer sizes have been set to a very small value.

Bandwidth, Delay, and Loss

- **7. Measuring networks with link losses**

Router link Loss	iPerf3 testing results – UDP datagram loss
0%	0.081%
0.1%	0.42%
5%	5.2%

  - From the experiments above, we can see that when we set a certain value of link loss on the Router side, the iPerf3 testing can get the same results from the Client and Server VM sides. Therefore we can validate the link loss setting is valid.

Secure Copy

- **Preparation works**
  - Set the Router link bandwidth to be 100Mbps
  - Test the RTT between Client and Server: **0.798 ms**
  - Test the bandwidth delay product between Client and Server: **81.22 Kbits**
- **Explain what we have done in Step 3 & 4**
  - We create a folder called test/
  - We copy the file /dev/urandom to the file ./filesystem.bin
  - We make file system on ./filesystem.bin
  - We use loop mode to mount on the file ./filesystem.bin
  - We make all the files in the test/ folder have all permissions type

Secure Copy

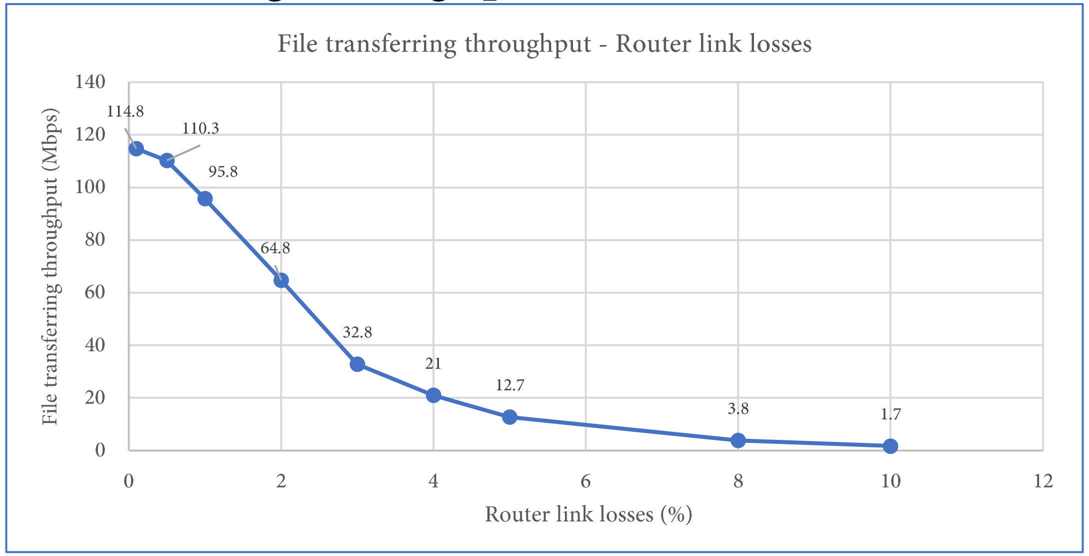
- **Step 7&8:** Throughput of the file transferring: **50.3MB/s**
- **Step 9:** Increase the link delay up to 25ms and measure the file transferring throughput:
  - RTT (tested using ping) : **50.901 ms**
  - File transferring throughput: **28.1MB/s**
  - **Improvement:** If we use smaller window sizes, throughput can be improved:
    - **Window size:** 10KBytes; **Throughput:** 31.9MB/s
    - **Window size:** 10KBytes; **Throughput:** 34.1MB/s

Secure Copy

- **Step 10:** Does scp seem to have some sort of built-in limitation? Can you guess what it is? Hint: scp uses the SSH protocol to transfer data.
    - SCP can only be operated on one thread but not on multi-thread simultaneously. This is because scp is built on ssh, and ssh can only use one port-22 and it can handle only one connection each time.

Secure Copy

- **Step 11** Measuring file transferring throughput under link losses



  - With increasing link losses(from 0.1% to 10%), file transferring throughput grows lower.
  - The descending trend of throughput is not linear, but it experiences a rapid decreasing when the router link loss is between 1% to 5%. This might be because TCP connections are very sensitive during this period due to its ACK mechanisms and sliding windows mechanisms. Before this period, data loss can be ignored, and after this period, the throughput has already been low. So this period it drops very fast.

Secure Copy

- **Step 12** Estimate (or measure) the delay and loss from Los Angeles to Switzerland. If you had a physicist for a colleague, and he wanted to download some data from the new atom smasher at CERN, would you expect him to come to you for help? Can you help him? If so, how?
  - From the ping results from this website, the delay between LA and Zurich can be 156.87ms ,and data loss datum cannot be found
  - If this physicist asked me for help, I would configure his computer and set the TCP buffer size to be little as 10-50 Kbytes. This should help him to obtain a higher throughput and higher network rate.

Thanks for listening!

Presenter: Boyang Xiao

USC id: 3326-7302-74

Email: boyangxi@usc.edu