EE 542 – Laboratory Assignment #3: Network Measurement (individual project)
Instructor: Young H. Cho
Due date: September 9 at 11:59pm

Due to the difficult of the next Lab, there will be no video submission for this lab. Remember to start the Lab 4 ASAP with your team !

1. Exploring Bandwidth and Throughput

This exercise will let you explore measuring bandwidth and throughput on your testbed on AWS. Your report should contain tables or graphs that show your results in a meaningful manner, and more importantly your thoughts and conclusions as to what the results mean. Remember, never perform an experiment just once, make sure to collect several runs and average the results. Please include any files you create as an addendum to your report. Have fun!

You'll need to familiarize yourself with the iperf tool. Use Linux OS to do all of the tests. Be complete in tests and explanations.

Go through some/all of the following tutorials before you begin (depends on your level of familiarity):

http://www.ee.surrey.ac.uk/Teaching/Unix/
https://www.garron.me/en/linux/visudo-command-sudoers-file-sudo-default-editor.html
https://openmaniak.com/iperf.php
https://netbeez.net/blog/how-to-use-the-linux-traffic-control/
https://www.poftut.com/linux-ethtool-tutorial-usage-examples/

1.1. Bandwidth-Delay Product.

1) Use a network of Linux nodes (two – one client, one server) connected via an IP router (vyos) from Lab 2 in AWS.
   Use the ping tool to measure the round-trip delay between two machines.
   Use iperf or iperf3 (look at iPerf hint section 1.3) in UDP mode to measure the maximum bandwidth (explore -b <bw> option).
   Now use iperf or iperf3 in TCP mode to measure the TCP throughput.
   You can use either iperf or iperf3 for all your tests.

2) Now use ‘netem’ on router to artificially inject a delay to the link in one direction. For example, if your ethernet link is named ‘eth0’

   ```
   # sudo tc qdisc add dev eth0 root netem delay 100ms
   ```

   should add an approximate delay of 100ms to the network link. You only need to do this in interface of vyos connected to client.
   Verify the added network delay using ping and iperf.

   Perform 4 additional emulated network latency experiments using netem. (i.e. 10ms, 50ms, 200ms, and 500ms)

Test the correct functioning of the tool with ping and iperf

3) Now use 'tbf' in tc to change characteristic of Ethernet link speed.
```
# sudo tc qdisc add dev eth0 root tbf rate 10mbit latency 1ms
burst 90155
```

This should change the speed of your Ethernet link to 10Mbps. In order for the entire network to have the same speed, this command needs to be entered on both interfaces of router VM connected to client, server side respectively. You can tune burst parameter to achieve higher throughput (burst should be greater than 9015, else you will get an error logged in dmesg). Your throughput might also be impacted due to VM scheduling done by AWS scheduler for vyos router instance. Also tune burst to lower and higher value and observe impact on throughput. Can you comment what this parameter does?

Verify the changed speed using iperf.
Change the network speed to 10Mbps, 100Mbps, and 1000Mbps and do the step (2).
Comment on the results at this point. Were you surprised at all?

Find out the receive and send network buffer sizes on server, client VM.

```
# sudo cat /proc/sys/net/core/rmem_max
# sudo cat /proc/sys/net/core/wmem_max
```

4) Change the maximum size for the send/receive windows to 2Mbyte on server, client VM:

```
# sudo sysctl –w net.core.rmem_max=2097152
# sudo sysctl –w net.core.wmem_max=2097152
```

Perform step 2 and 3 again and comment on the result. What do you think is going on?
Experiment with the window sizes and comment on the results.

5) Now use the -w option for iperf to change the send and receive windows to 64, 128 and 256 kilobytes while again measuring the TCP throughput.

6) Comment on your new results. You might want to put all of these results in a table.
At 100Mbit/s and 25ms of delay, what is the theoretically optimal send/receive window size?
If you run your experiment at this size, do you get 100Mbit/s of throughput?
If yes, why? If not, what could cause your link to run at less than full bandwidth?

1.2.Bandwidth, Delay, and Loss.

Delete the set configuration on router by executing:
```
# sudo tc qdisc del dev eth0 root
# sudo tc qdisc del dev eth1 root
```
This would cause the network to operate at its max speed. Then configure a delay of 25ms.

1) Use ping to determine the average Round-Trip-Time (RTT) between the computers.

2) What is the bandwidth-delay product for this link?
   Use iperf in UDP mode to measure the bandwidth.
   Use iperf in TCP mode to measure the TCP throughput.

3) If you left the send/receive windows at their default values, would this link perform well?

4) Can we scale the send/receive windows to see the effect of a bandwidth-delay product/TCP window size mismatch?
   Does it make sense to do so?

5) Set the send/receive window values to something too small (i.e. about the value calculated in (2). Now measure the TCP throughput. What happened?

6) Comment on these results. Specifically, what happens if we have a gigabit speed link (deleting tbf and netem cause link to operate at gigabit speed) with 1ms of delay? 5ms?

   Would you be happy with a 1GB DSL connection knowing you can only control the receive window size (hint: the RTT to the east coast of the US from Los Angeles can be 50ms or greater)?

7) Use 'netem' on router VM to Emulate packet loss. For example, if your ethernet link is named 'eth0'

```
# sudo tc qdisc change dev eth0 root netem loss 0.1%
```

This causes 1/10th of a percent (i.e 1 out of 1000) packets to be randomly dropped.
Verify the packet loss using iperf.
You cannot have netem and tbf operating at same time in tc. You must delete one of them to add other.

1.3. iPerf hints

1) To test the available bandwidth using iperf in UDP mode, use the following on the server machine:

```
# iperf –s –u –p <port num>
```

Where `<port num>` is any port number you choose > 1024

2) Then on the client:

```
# iperf –c <server name> –u –p <port num> –b <BW>
```

   Where `<server name>` is the name of the server (defined in your experiment setup), and <port num> is the same as chosen above. `<BW>` is the attempted bandwidth. Keep increasing this number towards the link speed until you see packet loss. Report the highest bandwidth possible with no packet loss.

3) To test TCP throughput using iperf use the following server line:

```
# iperf -s -p <port num> -w <window size>
```

4) And the following client:

```
# iperf -c <server name> -p <port num> -w <window size>
```

Where `<server name>` and `<port num>`are defined as above. `<window size>` is the window size you would like to use for send (on the client) and receive (on the server).

2. Secure Copy

The `secure copy program' scp is a standard tool on modern UNIX-like machines. It is used to copy files between machines, securely and reliably. However, as we will see, it does not always provide good throughput.

1) Set the network to operate at 100Mb/s using tc.

2) What is the RTT delay between the nodes? What is the bandwidth delay product?

3) On each node (client, server) create and mount a temporary local file system:

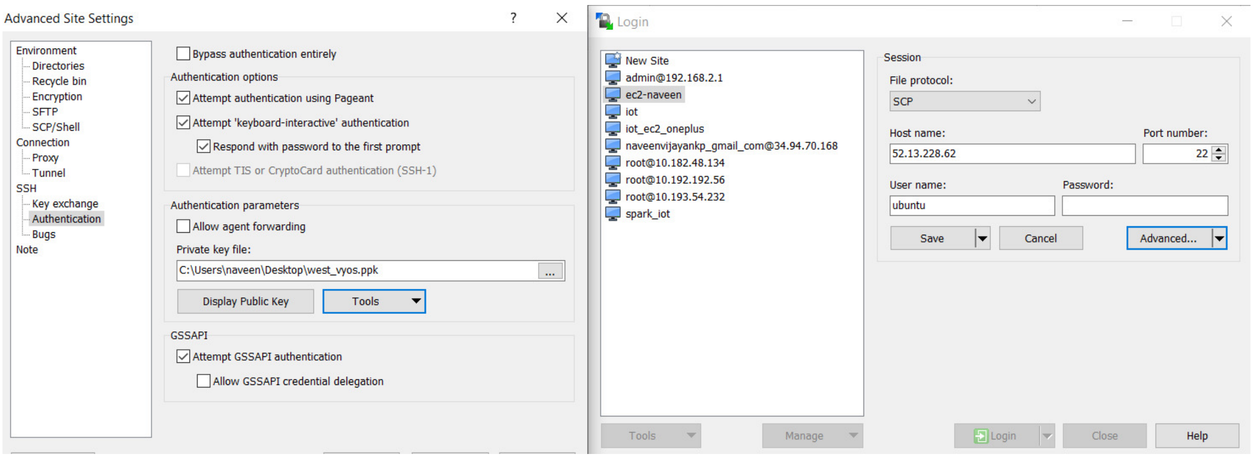
```
# mkdir test/
# dd if=/dev/urandom of=filesystem.bin bs=1M count=400
# sudo mkfs filesystem.bin
# sudo mount -o loop filesystem.bin test/
# sudo chmod -R 777 test/
```

4) On client node, execute below command inside folder test/:

```
# cd test/
# dd if=/dev/urandom of=data.bin bs=1M count=200
```

5) Explain briefly what the above commands do.

6) Now copy private key of the EC2 instance to which file is being copied to the VM where scp command is being executed. (you can use scp with private key from your windows/mac for copying it to the client VM)



7) Copy the file (data.bin) to the other node:

```
#scp -i <privatekey> data.bin USER@othernodeIP:/home/ubuntu/test/
```

8) What transfer throughput do you get?

9) Explore the problem: increase the delay on the link. What happens to the throughput? Can you improve the performance using the kernel parameters (i.e. default and maximum TCP window sizes)? Make sure to explore RTT delays at least up to 50ms.

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.

11) Now set your delay back to zero. At the same time add a small amount of packet-loss to the link. Start at 0.1% (i.e. 1 packet dropped in 1000) and test at various loss levels up to 10%. Graph throughput vs. loss. Discuss these results. Does this seem extreme? Can you explain why this happens?

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?

Create a report slide deck with results and answers to all of the above questions.