2014cs50296 Udayin Biswas

NETWORKS HW1

-Udayin Biswas, 2014CS50296

1. **testmyspeed->** Download speed 127 mb/s
   Upload speed 0.56 Mb/s

   speedtest.net-> Download speed 56.91 Mb/s
   Upload speed 43.86 Mb/s

   Tested at General Computing Lab.

No the results differ by a great extent.
The possible reasons for this variation are:
(1) How much bandwidth is available to us depends on whether others using the same connection are putting load on it. So if one person downloads a youtube video, it will decrease the bandwidth available to us, hence reducing the download and upload speed.
(2) Both the sites use different algorithms and use servers to fulfil our task. So if a server node is located closer to us, it will help eliminate Internet congestion and give us an accurate speed than a server node located farther. (There might be latency issues with a farther server as it has to travel through many nodes in a network graph.)
(3) Multithreading:the server manage can allow more than one user at a time and to even manage multiple requests by the same user without having to have multiple copies of the programming running in the computer.

2.
(a)Ping is a computer program that helps to verify the reachability of a host computer on an Internet protocol network. Ping uses the ICMP protocol which stands for Internet Control Message Protocol.Ping is used to check connectivity in the case of troubleshooting and measures the RTT(Round Trip Time) for a message which is sent from a source to destination and echoed back.

It uses the ICMP Echo function to send small packets to a particular IP address. This is done by sending ICMP Echo Request packets to host and then waiting for an ICMP Echo reply.The request packet has 64 bytes- 56 data bytes and 8 bytes of protocol reader information. The source then waits for a return packet after sending message. If the destination computer is up and there is proper connectivity, then a return packet is sent back.

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(The echo request ICMP has a type field of 8 plus a Code field of 0 whereas the echo replies have a Type field of 0 plus a Code field of 0.) (source-www.wikipedia.org)

RTT	minimum(ms)	average(ms)	maximum(ms)	std deviation(ms)
Google 74.125.200.147	93.514	96.627	99.771	1.181
Rice.edu 128.42.204.11	344.596	421.711	838.398	95.381
iitd 103.27.9.20	16.755	23.362	52.308	7.572

The IITD server is closest and the rice.edu is farthest in terms of Round Trip Time.
The reason for this is that the IIT Delhi server is the closest possible host computer to me. While google has multiple servers all over the world and uses Anycast to answer to our request through a nearby server. Rice University meanwhile has its server located far far away.(Probably in the US itself). The RTT gives us a measure of the distance between us and the closest server(rtt* speed of light) and hence we can determine approximately how far a server is.

3. ifconfig run on Mac OS X Yosemite.

(a)IPv4 addresses

```
lo0: 127.0.0.1
gif0: -
stf0: -
en0: -
fw0: -
en1: 10.192.49.85
en2: -
p2p0: -
awdl0: -
bridge0: -
```

(b) Interfaces having ethernet address

```
en0: ether 38:c9:86:0f:e8:d7
     mtu 1500

en1: ether 38:c9:86:e8:dc:e4
     mtu 1500

en2: ether d2:00:1d:77:24:80
     mtu 1500

p2p0: ether 0a:c9:86:e8:dc:e4
      mtu 2304

awld0: ether 02:65:07:89:96:4a
       mtu 1452
```

```
   bridge0: ether 3a:c9:86:f0:59:00
            mtu 1500
```

MTU stands for Maximum Transmission Unit. An MTU is the largest size packet that can be transmitted in a packet/frame based network like Internet. It is specified in octets which means 8 bit bytes. TCP uses MTU to find the maximum size of a packet in any transmission.

(c) Interfaces with IPv6 address
    lo0: IPv6 ::1 prefixlen 128
    en1: IPv6 fe80::3ac9:86ff:fee8:dce4
    awdl0: IPv6 fe80::65:7ff:fe89:964a

    An IPv6 address is composed of 128 bits.

4. Run trace route from Mac OS X at GCL using IITD Wi-fi.

```
   Udayins-MacBook-Pro:~ udayin$ traceroute www.cse.iitd.ac.in
traceroute to bahar.cse.iitd.ac.in (10.208.20.4), 64 hops max, 52 byte
packets
 1  10.192.32.14 (10.192.32.14)  1.252 ms  0.842 ms  0.840 ms
 2  10.254.238.1 (10.254.238.1)  3.155 ms  1.875 ms  1.620 ms
 3  10.254.208.2 (10.254.208.2)  1.292 ms  1.212 ms  1.165 ms
 4  bahar.cse.iitd.ernet.in (10.208.20.4)  0.926 ms  0.941 ms  0.927 ms
Udayins-MacBook-Pro:~ udayin$ traceroute www.iitd.ac.in
traceroute to www.iitd.ac.in (10.7.174.111), 64 hops max, 52 byte
packets
 1  10.192.32.14 (10.192.32.14)  5.013 ms  1.045 ms  0.881 ms
 2  10.254.238.1 (10.254.238.1)  1.200 ms  1.364 ms  1.158 ms
 3  10.254.236.18 (10.254.236.18)  1.374 ms  1.285 ms  1.234 ms
 4  www.iitd.ac.in (10.7.174.111)  1.199 ms  0.962 ms  0.927 ms
```

The routers 10.192.32.14 and 10.254.208.2 are common so there are 2 common routers and 2 different routers for each.

No, it is not true in real life that RTT to routers farther along the path should be larger than those closer to the source.

Possible suggestion for this might be that the farther route has an alternative closer path to the source than along the path. So overall it might have a shorter path than the router we initially assumed to be closer to source along the path. Other factors might be latency issues.

5. (a) x-axis denotes the packet size(in bytes) and y axis is cumulative RTT in seconds. The slope of the graph is quite steep and variable. Acc to (https://www.coursehero.com/file/p3vbc44/Estimate-the-RTT-multiple-times-for-various-size-packets-The-minimum-RTT-of/) the slope S denotes the reciprocal of first link capacity.

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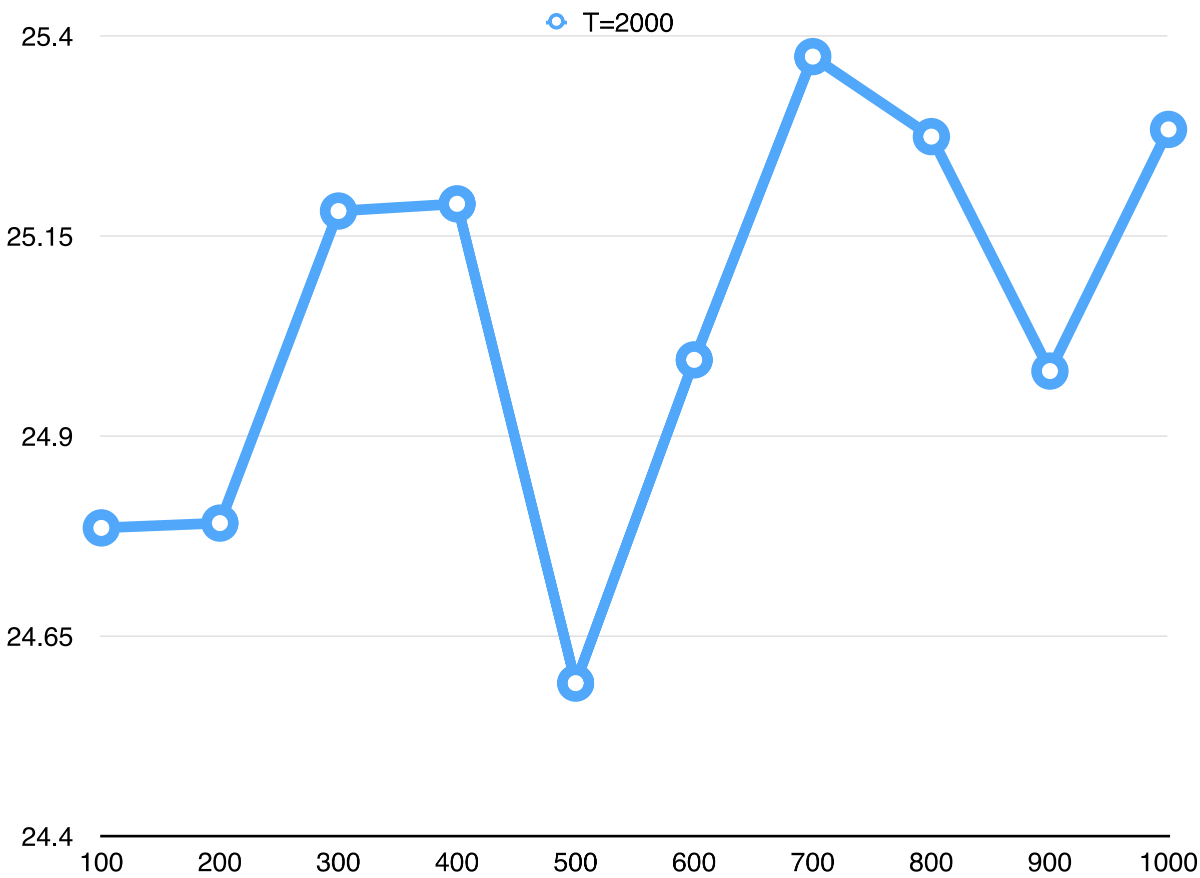

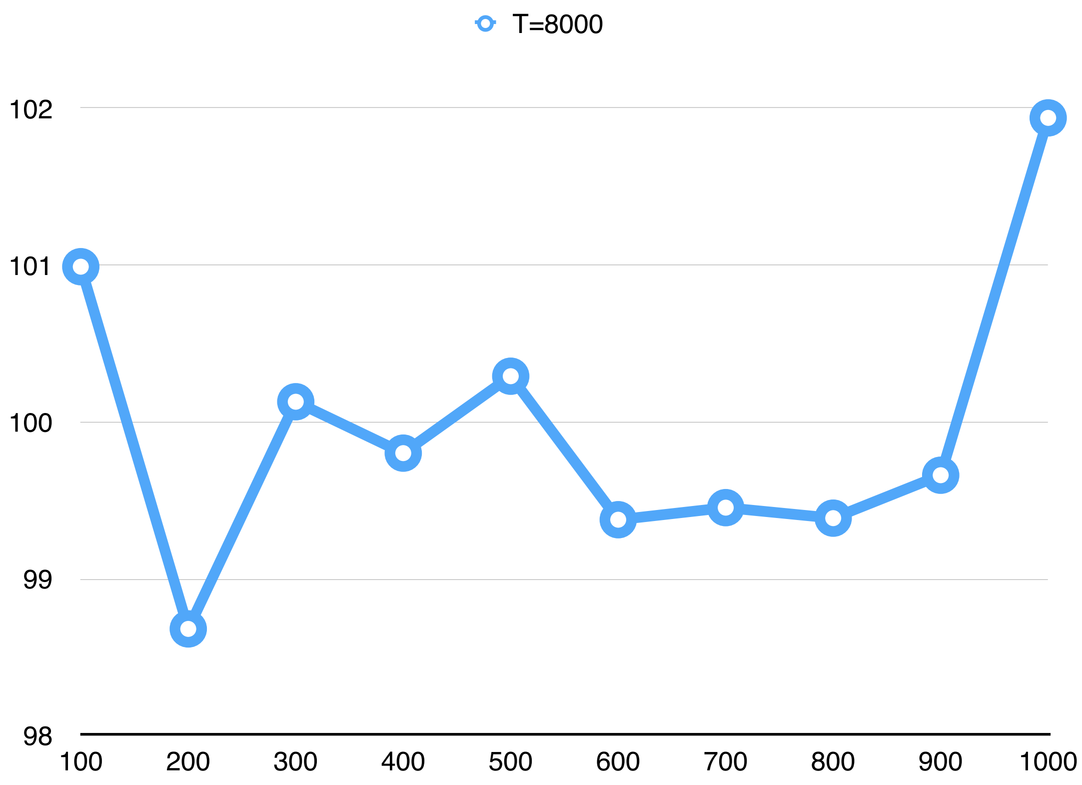

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(b)Netem delay, mean 5ms, standard deviation 2ms, random packet loss 0.01%
From the scatterplot, we can deduce that a packet will take more cumulative RTT in the case of netem. This is because a delay is being added as well a packet loss enhances the probability of a packet being dropped and using extra time in timeout scenario.