Homework1

P3. Consider an application that transmits data at a steady rate (for example, the sender generates an N-bit unit of data every k time units, where k is small and fixed). Also, when such an application starts, it will continue running for a relatively long period of time. Answer the following questions, briefly justifying your answer:

a. Would a packet-switched network or a circuit-switched network be more appropriate for this application? Why?

b. Suppose that a packet-switched network is used and the only traffic in this network comes from such applications as described above. Furthermore, assume that the sum of the application data rates is less than the capacities of each and every link. Is some form of congestion control needed? Why?

a. For the application it is mentioned in the question, that data is being transmitted at a steady rate and it is fix that there will be N bits of data in every K seconds (also K is small is fix-which means the data will be almost continuous also.Looking into all the characteristics and the reasonable amount of data that is to be sent a circuit switched network will be more appropriate. In the circuit switched network a particular link will be dedicated with a specific bandwidth that will work with FDM or TDM allowing continuous flow of data without any extra overheads. It is also mentioned that the application will be for a long time so we can bear the cost of setting up a dedicated connection and tearing it down.
b. In case a packet switched network is used with traffic coming to this network only from such kind of applications, there should not be any congestion control needed. Since it is mentioned that the sum of application data is less than the capacity of every link. Since there is enough capacity and the incoming data rate is fix there should be no congestion in the links. However, there might be some queuing delay if in case a lot of packets come together on a particular router.

P4. Consider the circuit-switched network in Figure 1.13. Recall that there are 4 circuits on each link. Label the four switches A, B, C and D, going in the clockwise direction.

a. What is the maximum number of simultaneous connections that can be in progress at any one time in this network?

b. Suppose that all connections are between switches A and C. What is the maximum number of simultaneous connections that can be in progress?

c. Suppose we want to make four connections between switches A and C, and another four connections between switches B and D. Can we route these calls through the four links to accommodate all eight connections?

A.
Maximum Simultaneous connections will be when we consider the connection to be between 2 switches.
So in this case, it is like 4 between A and B
4 between B and C
4 between C and D
4 between D and A

This makes a total of 16 connections simultaneously.

B.
Suppose all the connections are between A and C there will be a total of 8 connections.
4 between A-B-C
4 between A-D-C

C.
We want to make 4 connections between A and C, and 4 between B and D. Yes we can make them together.

2 between A and C will be A-B-C
2 will be A-B-D
So a total of 4

2 Between B and D will be B-C-D
2 between B and D will be B-A-D
So a total of 4

P6. This elementary problem begins to explore propagation delay and transmission delay, two central concepts in data networking. Consider two hosts, A and B, connected by a single link of rate R bps. Suppose that the two hosts are separated by m meters, and suppose the propagation speed along the link is s meters/sec. Host A is to send a packet of size L bits to Host B.
a. Express the propagation delay, dprop, in terms of m and s.
b. Determine the transmission time of the packet, dtrans, in terms of L and R.
c. Ignoring processing and queuing delays, obtain an expression for the endto-end delay.
d. Suppose Host A begins to transmit the packet at time t = 0. At time t = dtrans, where is the last bit of the packet?
e. Suppose dprop is greater than dtrans. At time t = dtrans, where is the first bit of the packet?
f. Suppose dprop is less than dtrans. At time t = dtrans, where is the first bit of

the packet?
g. Suppose s = 2.5 · 108, L = 120 bits, and R = 56 kbps. Find the distance m so that dprop equals dtrans.

a. dprop=m/s seconds
b. Transmission time = L/R seconds
c. End to End delay = (m/s + L/R) seconds
d. The last bit of the packet will be at Host A only or going to leave Host A as dtrans is the time taken for the entire packet to be put on the link to be propagated.
e. If dprop is greater than dtrans then the dtrans at dtrans the first bit will be on the link.
f. If dprop is less than dtrans then at dtrans the first byte will be at Host B as the time will be sufficient for it to reach at B.
g. Distance m such that dprop and dtrans are equal
   L/R=m/s
   120/56000=m/2.5*10^8
   m= 535714.286 mts

P7. In this problem, we consider sending real-time voice from Host A to Host B over a packet-switched network (VoIP). Host A converts analog voice to a digital 64 kbps bit stream on the fly. Host A then groups the bits into 56-byte packets. There is one link between Hosts A and B; its transmission rate is 2 Mbps and its propagation delay is 10 msec. As soon as Host A gathers a packet, it sends it to Host B. As soon as Host B receives an entire packet, it converts the packet’s bits to an analog signal. How much time elapses from the time a bit is created (from the original analog signal at Host A) until the bit is decoded (as part of the analog signal at Host B)?

Solution.
Host A converts analog voice to a digital 64 kbps bit stream on the fly
Then -> 56 bytes packets
Transmission rate -> 2Mbps
Propagation delay=10msec

The time taken for bits to be generated in the packet=
56*8/64*1000=7 ms

Transmission time =56*8/2*10^6seconds=0.000224 seconds

Propagation time = 10 ms

Total = 17.224 ms

P10. Consider a packet of length L which begins at end system A and travel over three links to a destination end system. These three links are connected by two packet switches. Let di, si, and Ri denote the length, propagation speed, and the transmission rate of link i, for i = 1, 2, 3. The packet switch delays each packet by dproc. Assuming no queuing delays, in terms of di, si, Ri, (i = 1,2,3), and L, what is the total end-to-end delay for the packet? Suppose now the packet is 1,500 bytes, the propagation speed on all three links is 2.5 · 10^8 m/s, the transmission rates of all three links are 2 Mbps, the packet switch processing delay is 3 msec, the length of the first link is 5,000 km, the length of the second link is 4,000 km, and the length of the last link is 1,000 km. For these values, what is the end-to-end delay?

Solution.
The end-to-end delay for 3 links will be =
L/R1+L/R2+L/R3+d1/s1+d2/s2+d3/s3+2*processing delay

3*(1500*8/2*10^6)+5000000/2.5*10^8+4000000/2.5*10^8+1000000/2.5*10^8+2*3
3*6+20+16+4+6
64ms

P13.
(a) Suppose N packets arrive simultaneously to a link at which no packets are currently being transmitted or queued. Each packet is of length L and the link has transmission rate R. What is the average queuing delay for the N packets?
(b) Now suppose that N such packets arrive to the link every LN/R seconds. What is the average queuing delay of a packet?

Solution.
Delay for packet 1 = 0
Delay for packet 2 = L/R
Delay for packet 3 = 2L/R
Delay for packet N = (N-1)L/R

A. Average is:
(0+1+2-----N-1)L/(R*N)
(N-1*N*L)/(R*N*2)
(N-1)L/2R

B. If there are N packets coming to the link after every LN/R seconds
The link will be empty when each set of N packets will come.
As the time LN/R > (N-1)L/R
Now every batch of N will have a queuing delay of
(N-1)L/(2.R)

P19. (a) Visit the site www.traceroute.org and perform traceroutes from two different cities in France to the same destination host in the United States. How many links are the same in the two traceroutes? Is the transatlantic link the same?

(b) Repeat (a) but this time choose one city in France and another city in Germany.

(c) Pick a city in the United States, and perform traceroutes to two hosts, each in a different city in China. How many links are common in the two

Ans.
In this question I have done traceroute from my local computer to 2 IP addresses in France and determined the results:

```
C:\Users\rajsh>tracert 78.31.44.118

Tracing route to 78.31.44.118 over a maximum of 30 hops

  1     3 ms     3 ms     4 ms  10.153.0.2
  2     3 ms     9 ms     2 ms  smdf-csdis-aruba-a1k-1--cmdf-cscore-c6k-1.ncstate.net
[10.250.16.113]
  3    16 ms     3 ms     3 ms  hmdf-bbcore-c6k-1--smdf-cscore-c6k-1.ncstate.net [10.250.1.61]
  4    26 ms     6 ms     3 ms  ncsugw-gi2-1.ncstate.net [152.1.6.70]
  5     *        *        *     Request timed out.
  6     *        *        *     Request timed out.
  7    88 ms    90 ms    88 ms  ae-2-3204.edge3.Paris1.Level3.net [4.69.161.114]
  8    94 ms    90 ms   103 ms  CELESTE.edge3.Paris1.Level3.net [213.242.111.50]
  9    89 ms    89 ms    88 ms  78.31.44.118

Trace complete.
```

```
C:\Users\rajsh>tracert 88.176.5.1

Tracing route to cse35-2-88-176-5-192.fbx.proxad.net [88.176.5.192]
over a maximum of 30 hops:
```

```
 1     5 ms     5 ms     7 ms  10.153.0.2
 2     *        *        *     Request timed out.
 3     9 ms     4 ms     4 ms  cmdf-bbcore-c6k-1--smdf-cscore-c6k-1.ncstate.net [10.250.1.65]
 4     *        *        *     Request timed out.
 5     *        *        *     Request timed out.
 6    12 ms    10 ms    12 ms  9-1-3.ear3.Washington1.Level3.net [4.31.77.189]
 7     *        *        *     Request timed out.
 8    17 ms    23 ms    17 ms  Cogent-level3-200G.WashingtonDC111.Level3.net [4.68.73.198]
 9    33 ms    24 ms    20 ms  be3084.ccr42.dca01.atlas.cogentco.com [154.54.30.65]
10    23 ms    19 ms    20 ms  be2807.ccr42.jfk02.atlas.cogentco.com [154.54.40.109]
11    90 ms    90 ms    93 ms  be3628.ccr42.par01.atlas.cogentco.com [154.54.27.170]
12    94 ms    92 ms    92 ms  Iliad.demarc.cogentco.com [149.6.114.10]
13   100 ms    93 ms    97 ms  194.149.166.61
14    99 ms   104 ms    99 ms  rennes-9k-1-be2000.intf.routers.proxad.net [194.149.162.98]
15   105 ms    99 ms   102 ms  213.228.11.157
16   227 ms   176 ms   174 ms  cse35-2-88-176-5-192.fbx.proxad.net [88.176.5.192]

Trace complete.
```

The above 2 traceroute outputs we cannot find any links in common.

Also we can see that the reply from the second router got timed out.

The transatlantic link in the first looks to be 7 as the above are time outs.

The transatlantic link in the first looks to be 11 as the above are time outs.

P25. Suppose two hosts, A and B, are separated by 20,000 kilometers and are connected by a direct link of R = 2 Mbps. Suppose the propagation speed over the link is 2.5 108 meters/sec.

a. Calculate the bandwidth-delay product, R dprop.

b. Consider sending a file of 800,000 bits from Host A to Host B. Suppose the file is sent continuously as one large message. What is the maximum number of bits that will be in the link at any given time?

c. Provide an interpretation of the bandwidth-delay product.

d. What is the width (in meters) of a bit in the link? Is it longer than a football field?

e. Derive a general expression for the width of a bit in terms of the propagation speed s, the transmission rate R, and the length of the link m.

Solution:
A.
dprop=20000*1000/2.5*10^8 seconds=0.08 seconds
R=2*10^6 bits per second

Bandwidth delay product =dprop*R=160000 bits

B.
The number of bits in the link at a given time cannot be more than the bandwidth delay product. So the bits in the link will be 160000 bits.

C.
The number of bits in the link at a given time cannot be more than the bandwidth delay product.

D.
So the total length of the link is 20000km
And at one time number of bits in the link is 160000 bits
So the width is
20000*1000/160000=125 mts
It can be more or less than the field depending on the football field.

E.
general expression
m/(R*dprop)
m/(R*(m/s))
s/R

P27. Consider problem P25 but now with a link of R = 1 Gbps.
a. Calculate the bandwidth-delay product, R dprop.
b. Consider sending a file of 800,000 bits from Host A to Host B. Suppose the file is sent continuously as one big message. What is the maximum number of bits that will be in the link at any given time?
c. What is the width (in meters) of a bit in the link?

Solution..

A.
Bandwidth delay product = R*dprop

dprop=0.08 seconds (from previous question)
Bandwidth delay product = 1*10^9*0.08=80000000 bits

B.

The maximum number of bits in the entire link at a given time are :
800000 bits which is the size of the message that can be put in the link at once

C.

width of bit = 20000000/80000000 = 0.25 metre

P31. In modern packet-switched networks, including the Internet, the source host segments long, application-layer messages (for example, an image or a music file) into smaller packets and sends the packets into the network. The receiver then reassembles the packets back into the original message. We refer to this process as message segmentation. Figure 1.27 illustrates the end-to-end transport of a message with and without message segmentation. Consider a message that is 8 · 106 bits long that is to be sent from source to destination in Figure 1.27. Suppose each link in the figure is 2 Mbps. Ignore propagation, queuing, and processing delays.

a. Consider sending the message from source to destination without message segmentation. How long does it take to move the message from the source host to the first packet switch? Keeping in mind that each switch uses store-and-forward packet switching, what is the total time to move the message from source host to destination host?

b. Now suppose that the message is segmented into 800 packets, with each packet being 10,000 bits long. How long does it take to move the first packet from source host to the first switch? When the first packet is being sent from the first switch to the second switch, the second packet is being sent from the source host to the first switch. At what time will the second packet be fully received at the first switch?

c. How long does it take to move the file from source host to destination host when message segmentation is used? Compare this result with your answer in part (a) and comment.

d. In addition to reducing delay, what are reasons to use message segmentation?

e. Discuss the drawbacks of message segmentation.

A.

The time taken for the message to go from Source Host to the first packet switch=
8*10^6/2*10^6=4 seconds.

Further, the time taken for the for the message to go from Source Host to the Destination using Store and Forward Delay is= 4*3=12 seconds.

B.

Time taken by the first packet to move to the first switch: 10^4/2*10^6=0.005 seconds

The time when first packet reaches the second switch= 2* 0.005 seconds = time taken by the second packet to reach the first switch = 0.010 seconds

C.

Time taken for the entire packet to reach the destination in case of segmentation=

Time taken by the first packet to reach the destination host = 3*0.005 = 15 ms

Time taken by 2nd packet to reach the destination= 20 ms

Time by 3rd packet to reach the destination = 25 ms

Time by 800th packet to reach the destination = 15 ms + 5*799ms

That is 4.01 seconds

This result is lesser by approximately 3 times as compared to the result in case 1 when the packet was sent all at once.

D.

Below can be the reasons to prefer message segmentation:

In case there is an error in the bits of a particular packet then the error can be corrected by retransmission of a single packet, otherwise we will need the retransmission of the entire packet.

Also, if the entire packet is sent at once and the packet sits on the router due to queuing delay. If there are smaller packets at the back of this huge packet these smaller packets might face a terrible delay and might also get dropped.

Also, sending one packet having all the data will lead to terrible delay.

E.

When we implement message segmentation there can be an overhead of segmenting the packets and then joining them back at the destination routers. Also, all the packets have to have a seperate header that might lead to sending extra data over the link.

Question.

Read the paper: D.D. Clark, “The Design Philosophy of the DARPA Internet Protocols.”

Proceedings of ACM SIGCOMM 1988.
1. What are the top four goals of the Internet architecture that are discussed at length in the paper?
2. What goals led to the success of the Internet?
3. Why was the “datagram” model selected?
4. If you were to design the Internet today, are there any other design goals that you would include?

Answer 1.
The top four goals of the Internet architecture are :

1. The top level goal for building a network architecture was to be able to develop an effective solution or technique for multiplexed utilization of the existing interconnected networks. For example, connecting the ARPANET with the ARPA packet radio networks so that the packet radio network has access to machines on the ARPANET .Hence, the main objective was to incorporate the existing network architectures and connect the several separately administered entities to build a common entity.
2. Another, important goal was Internet communication must continue despite loss of networks or gateways.
   Several ways to achieve this as well have been discussed in the paper which include maintaining state information like number of packets acknowledged, number of outstanding flow control permissions.
   Another method was by saving the state at the intermediate packet switching nodes of the network, but this became complicated due to a number of robust replication algorithms that were required would have been difficult to build.
   Yet another alternative to the above replication method was to put all the information on an endpoint of the Internet. This is known as Fate sharing.
3. The Internet Architecture must be able to accommodate a variety of Networks.
4. The Architecture must permit distributed management of its resources.

Answer 2.
The Goals that led to the success of Internet were:
The goal to have all time Internet communication despite loss of networks or gateways was a major goal that lead to the success of the internet. The use of datagram eliminated the need for connection state within intermediate switching nodes. This enabled the internet to be reconstituted after a failure without any concerns about the state. Another goal that has been suggested in the paper that lead to the success of the Internet was flexibility of the service offered. This goal was also eased and achieved by the usage of Datagrams that led to building multiple services on top. The architecture also tried to not restrict the range of services that the Internet could be engineered to provide. The term Realization has been used in the paper to describe a set of networks, gateways and hosts connected together in the context of Internet Architecture. There are a wide variety of realizations like one having 1200 bits per second phone lines, networks with speed greater than 1 megabit per second and realizations having

throughput varying by orders of magnitudes. The Internet has been built and engineered in such a way that it tolerated a variety of realizations that have been interconnected and work together.

Answer 3.

There are a lot of reasons why the datagram model is selected:

1. They remove the requirement for connection state within the intermediate switching nodes. This ensures that the internet can be reconstituted after some kind of failure without being concerned about the failure.
2. The datagram gives a basic building block out of which a variety of services can be implemented.
   The datagram provides a more elemental and basic kind of service that various endpoints can combine to build a particular type of service that is needed.
3. The datagrams represent a minimum network service assumption which has allowed a lot of networks to be incorporated into various network realizations based on the need requirement and use case.

Answer 4.

If I was to design the Internet today, some of the goals that I would include are as follows:

1. Try to build or incorporate a model other than using the Datagram model, as it includes a few factors that in condition of problem make the issues more difficult. Let us say there is a lot of congestion in the network and all the packets are very small (ie. they have very few number of bytes). In case there are numerous retransmissions in the network due to congestion. The small packets will have huge headers and will carry the data repeatedly without any success. This will lead to the problem getting worse instead of ideally correcting in the first place. Hence, there should be mechanisms that should work on self improvement of the network issues rather than a fixed static set of protocols.
   We can look ahead to implementing more dynamic and robust protocols that will determine the problems and inform the hosts and switches about the same.
2. Further, another goal can be to implement a model with a less redundant architecture. Lets say, there are protocols on different layers achieving the same task but not completely. For example, error correction is done in the TCP layer and also in the lower layers like the Data link layer.

Question:

Read the paper: J.H. Saltzer, D.P. Reed, D.D. Clark, "End-to-End Arguments in System Design." ACM Transactions on Computer Systems, vol. 2, no. 4, November 1984.

1. What alternative solutions are possible for reliable file transfer?

2. A "moderate" interpretation of the E2E argument is that, if hosts can implement certain functionality correctly, then implement this functionality at lower layers only as a performance enhancement, if doing so does not impose a burden on applications that do

not require it. Based on this interpretation, argue whether or not these functionalities belong in the network:
– multicast
– routing
– quality of service (QoS)
– name resolution (DNS)
– web caches

Answer1.

The alternate solutions that are possible for reliable file transfer are:

1. To again and again implement the steps along the way that has duplicate copies, timeout and retries, crash recovery, carefully located error detection mechanisms. This will reduce the probability of all the threats to a relatively lesser value. However, in this case the program will have to be written for retry and error mechanism will be a bit complicated. But since the probability of the issue occurring regularly is less the repeated checking of all the problems is uneconomical.
2. Another, alternate solution that we can suggest is “end to end check and retry”. So if we are trying to detect the error in file after transfer. There will be a checksum stored with each file. The file will be transferred from A to B and at B the file transfer application will copy the file from the disk storage system to its own memory. The sum will be recalculated and the value will be sent back to A. A will compare it with the original checksum. If the checksums match the file transfer will be successful, otherwise there will be a retry.

Answer2.
-Multicast:
Multicast is basically group communication where the communication can be 1 to many or many to many. Multicast messages are basically sent in the ethernet multicast or the IP multicast by just specifying the particular IP address in the destination. This is how it can be implemented in the network However, there can be applications that might also require Multicast messages. However, it might be based on different criteria. Let us say an application wants to send multicast messages on the basis of the text it receives, or after analysing some kind of data. Then the application will generally build its own multicast application. This multicast will be an application layer multicasting. Also, all the applications do not even require multicast. Hence, this is a purely optional functionality and can be implemented end to end, instead of building it in the network.

-Routing
Routing belongs to the network as it helps in sending the packets to the particular destinations, by determining the specific paths. Implementing routing on the application end to end will be a hassle as to make it compatible to the network requirements would require a lot of code and this

all should be parallel to the already implemented network protocols. Further this functionality is used with all kinds of applications hence it is better in the network instead of end to end.

-(Quality of Service)QOS- It is the overall performance of the service Cloud Computing, Telephony or Computer Network. All the applications do not use performance evaluation methods to build better applications. Hence this tool should not be in the network and should be end to end as it can always be implemented on the applications depending on what is to be measured.

-Domain Name Resolution- This should be implemented by the network as a separate protocol and is not preferred to be done end to end as almost all the applications and web usages require DNS resolution.

-Web Caches - Web caching should be end to end and not on the network. As there are some applications that might require the data to be cached like the web browsers and some applications that would not like file transfer applications. Hence web caching is preferred to be implemented on the application rather to make it a part of the core network as it would add over head.