CS-358
Computer Networks
Mid Semester Assignment

Tarusi Mittal 1901CS65

Que 1:-

(a) Packet Switching is said to employ Statistical Multiplexing because in packet - switching network when the packets flow from different source into a link, they do not follow any static fixed route or pattern.

Packet switching is advantageous over circuit switching because:-

1) Efficiency: Packets can find their own path (data) to their destination address, without the need for a dedicated channel.

2) Reduces lost data Packets:- With packet switching data packets can be resent if they don't reach their destinations. So, it is more realiable.

3) Cost Effective: Packet switching is more cost effective as there is no need for a dedicated channel for voice or data traffic as opposed to the circuit switching.

(b) For such application, a circuit-switched network would be more suitable, because this application involves long sessions with predictable smooth bandwidth requirements.
Now, as the ~~bandwidth of~~ transmission rate is not bursty and is known, the bandwidth for each application session can be reserved ~~by~~ without any significant waste. Moreover, the overhead costs of setting up and tearing down connections are low compared to the time of a typical application session.

(c) Distance of the links from Router = D
Bandwidth = R bps
No. of packets sent by A = N
Length of each packet = L
Time taken to sent N packets = $\frac{2LN}{R}$ sec.
Propagation speed = C m/s.

Various forms of delay that the packet would encounter assuming negligible node processing delays at hosts and routers are:

(1) Transmission Delay :- The time taken to transmit a packet from the host to the transmission medium.

$$T_t = \frac{\text{length of packet}}{\text{Transmission Rate}}$$

$$= \frac{L}{R} \text{ sec}$$

Now our process is

$A \longrightarrow \text{Router} \longrightarrow B$

So total transmission delay

$$= 2 \times \frac{L}{R} \text{ sec}$$

$$= \boxed{\frac{2L}{R} \text{ sec.}}$$

(2) Propagation Delay: when the packet is transmitted to the transmission medium, it goes through the medium to reach the destination. So the time taken by the last bit of packet to reach the destination is called the propagation delay

$$T_p = \frac{\text{Distance}}{\text{Velocity}}$$

$$= \boxed{\frac{2D}{C} \text{ sec}}$$

(3) Queueing Delay: After reaching the destination, the amount of time the packet waits in the queue before getting processed is known as queueing delay.

For our example. (we will consider for 1 batch)

The Queueing delay for 1^{st} Packet $= 0$

2^{nd} Packet $= L/R$

3^{rd} Pc $= \frac{L}{R} + \frac{L}{R}$

$\vdots$

n^{th} Packet $= \frac{(n-1)L}{R}$

↳ because queue will be empty befor 2nd batch arrivs.

Total queueing delay $= \frac{L}{R}\left(1 + 2 + \cdots + (n-1)\right)$

$$= \frac{L}{R}\left(\frac{n(n-1)}{2}\right)$$

$$= \boxed{\frac{N(N-1)L}{2R}}$$

Average Packet Delay $= \frac{\text{Total of All the 3 Delays}}{\text{Total no of Packets}}$

$$= \frac{\frac{2LN}{R} + \frac{2DN}{C} + \frac{N(N-1)L}{2R}}{N}$$

$$\boxed{\text{Avg. Packet Delay} = \frac{2L}{R} + \frac{2D}{C} + \frac{L(N-1)}{2R}}$$

Que 2.

(a) Loss Rate $(L) = \dfrac{\text{No of Packets Lost}}{\text{No of Packets Sent}}$

Given: 1 Packet Lost in 1 cycle.

window size varies from $\frac{W}{2RTT}$ to $\frac{W}{RTT}$

The number of packets sent in a cycle is

$$= \frac{W}{2} + \left(\frac{W}{2}+1\right) + \cdots + \left(\frac{W}{2}+\frac{W}{2}\right)$$

$$= \sum_{k=0}^{W/2}\left(\frac{W}{2}+k\right)$$

$$= \sum_{k=0}^{W/2}\frac{W}{2} + \sum_{k=0}^{W/2}k = \left(\frac{W}{2}\right)\left(\frac{W}{2}+1\right) + \frac{\left(\frac{W}{2}\right)\left(\frac{W}{2}+1\right)}{2}$$

$$= \frac{1}{4}W(W+2) + \frac{1}{8}W(W+2)$$

$$= \frac{W^2}{4} + \frac{2W}{4} + \frac{W^2}{8} + \frac{2W}{8} = \frac{3W^2}{8} + \frac{3}{4}W$$

So,

$$\text{Loss Rate } (L) = \frac{1}{\frac{3}{8}W^2 + \frac{3}{4}W}$$

For large W; $\frac{3}{8}W^2 >> \frac{3}{4}W$

$$\Rightarrow L \approx \frac{8}{3W^2} \Rightarrow \boxed{W \approx \sqrt{\frac{8}{3L}}}$$

Now;

$$\text{Average Bandwidth} = \frac{3}{4} \times W \times \frac{MSS}{RTT}$$

$$= \frac{3}{4}\sqrt{\frac{8}{3L}} \times \frac{MSS}{RTT}$$

$$= \sqrt{\frac{3 \times 3 \times 8}{4 \times 4 \times 3}} \times \frac{MSS}{RTT\sqrt{L}}$$

$$= \boxed{\frac{1.22\ MSS}{RTT\sqrt{L}}}.$$

Average Bandwidth for the given model

$$= \boxed{\frac{1.22 MSS}{RTT\sqrt{L}}}.$$

(b) Segment Size (MSS) $= 1500$ bytes $= 1500 \times 8$ bits

RTT $= 100$ ms $= 100 \times 10^{-3}$ sec.

Avg. Throughput $= 100$ Gb/s $= 100 \times 10^{9}$ b/s.

We know, $W = \frac{\text{Avg. Throughput} \times RTT}{MSS}$

$$W = \frac{\overset{10}{\cancel{100}} \times 10^{9} \times \cancel{100} \times 10^{-3}}{\underset{3}{\cancel{1500}} \times \cancel{8}4}$$

$$W = \frac{10^{7}}{12}$$

$$\boxed{W = 833,333}$$

⇒ To calculate loss rate.
Here considering ideal model

$$\text{Avg Throughput} = \frac{1.22 \times MSS}{RTT\sqrt{L}}$$ (from previous pg)

$$100 \times 10^{9} = \frac{1.22 \times 1500 \times 8}{100 \times 10^{-3} \times \sqrt{L}}$$

By calculating

$$\sqrt{L} = \frac{1.464}{10^{6}} \Rightarrow \boxed{2.143 \times 10^{-12} = L}$$

Hence.

$$\boxed{\begin{aligned} L &= 2.143 \times 10^{-12} \\ W &= 833,333 \end{aligned}}$$

(c) As we saw in the previous parts, the

$$\text{Average Throughput} \propto \frac{1}{RTT}$$

i.e the Average throughput is inversely proportional to RTT which implies that the throughput of TCP will be increased when the RTT is decreased. This alone tells us that the performance of TCP will be biased against long RTT connections. This RTT mechanism which our TCP congestion control uses results in the RTT unfairness as all flows sharing a level transmission medium operate at the scale of their RTT, flow's with shorter RTT's have faster sending rate.

CUBIC is a congestion control protocol for TCP and the current default TCP algorithm in Linux. This protocol modifies the linear window growth function to be a cubic function in order to improve the scability of a TCP over fast and long distance networks.

It achieves more equitable bandwidth allocation among flows with different RTT by making the window growth to be independent of RTT and thus those flow grow their congestion window at the same rate.

During the steady state, the TCP CUBIC increases the window size aggressively when the window is far from the saturation point, and slowly when it is closer. This feature makes the model highly stable, scalable and also fair to the standard TCP flows.

(d) For all parts we assume that TCP model can buffer out of order packets also.

Now,
given; the current expected sequence no which is 2001.

(i) Packet with sequence no = 2001
Length = 1000 bytes

Current sequence no. = 2001

Now according to RFC 1122, RFC 2581 if the event at receiver is arrival of in order segment with expected sequence no. All data upto expected seq no. already Acked, ~~so~~, In that case the TCP Receiver Action is a delayed ACK, wait upto 500 ms for next segment and in case of no next segment, then ACK.

So, now as next segment arrives at 1000 ms as given, ACK will be sent at 500 ms according to the above rule.

→ ACK of 3000+1 = 3001 sent at time = 0.5 sec

and.

the current expected sequence no = 3001

(ii) Client received packet with no = 3001

Length = 1000 bytes (assumed)

Time = 1 sec.

Current sequence no = 3001.

According to the rule in previous part:

A delayed ACK after 500 ms will be sent:

Now.

ACK of 4000 + 1 = 4001 sent at time = 1.5 sec.

and

the current expected sequence no = 4001

(iii) Packet with sequence no = 6001
Length = 1500 bytes
Arrival Time = 2s.
Expected sequence no = 4001
current sequence no = 6001
Now, expected and current sequence no does not match.

Now, according to RFC(1122), RFC 2581) rule when the event at receiver is arrival of out of order segment higher than expected sequence number A gap is detected basically then the TCP Receiver Action is that it immediately send duplicate ACK indicating sequence of next expected byte.

So, duplicate of 4001 → sent at 2s.
The TCP model buffers current packet & waits for ACK.

(iv) Packet with sequence no = 5001
Length = 1000 bytes.
Arrival Time = ~~sec~~ 2.2 sec.
Expected sequence no = 4001
Again expected sequence no and current does not match.

By the same rule used in previous part.
Duplicate Ack of 4001 ⟶ sent at 2.2 sec
The TCP model buffers current packet & waits for ACK.

(vi) Packet with sequence no = 4001
Length = 1000 bytes.
Arrival Time = 3.2 sec.
Expected sequence no = 4001

Now according to (RFC 1122), RF(2581) rule when event at receiver is arrival of in order segment with expected seq. one other segment has Ack pending then the TCP Receiver Action is that it immediately send single cumulative Ack, Acking both in order segments.

⇒ ACK 7501 ⟶ sent at 3.2 sec

- 4001 ⟶ 5000
- 5001 ⟶ 6000
- 6001 ⟶ 7501

—x —x —x —END —x —x —x —