CS358

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Q1 (a)

Ans Packet switched Networks do not reserve bandwidth for flows but instead sends packets whenever there are packets in the queue (which do not follow any fixed route). Thus packet switching method is said to employ statistical multiplexing.

Some of the advantages of packet switching over circuit switching are :-

1. More no. of active users :- Since circuit switching has to allocate certain bandwidth to any user, only that grp of people can use it. While packet switching method can support more users

2. Cost effective and more efficient :- Maintaining dedicated bandwidth costs more in general. Also if a link has sending rate of 1 MbPS with 10 users, the effective speed for circuit sw. becomes 100 kbps but packet switching user will enjoy greater speed in general

3. Easy retransmission of data :- The packet which got corrupted can be detected and only that packet needs to be retransmitted.

Q1 (b)

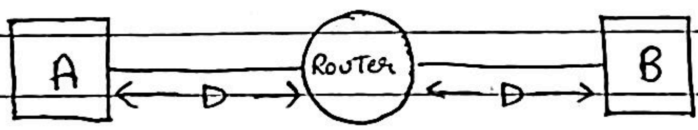
Ans Given :- N-bit units of data at every 'k' time for a very long duration of time

A circuit switched network would be well justified for given application. Since the period of transmission is quite long with predictable bandwidth requirement throughout the session, we can reserve a bandwidth without any significant waste. Also, the overhead cause of setting up the connection initially gets balanced out by the ~~t~~ time the application is running

Thus, circuit switching network should be chosen.

Q1 (C)

Ans



Bandwidth speed = "R" bps

Packet size = L bits

Propogation speed = c m/s

Distance between A & router = "D" m

Distance between router & B = "D" m

N packets are sent every $\frac{2LN}{R}$ seconds

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Assumption :- Node processing delays are negligible

Different type of delays that packet will face are :-

1.) Transmission delay :- The time taken to transmit a packet from host to link.

$$d_t = \frac{\text{Packet length}}{\text{Transmission rate}} = \frac{L}{R} s$$

For a given packet, total transmission delay to travel from A to router and from router to B

$$= \frac{L}{R} + \frac{L}{R} = \left(\frac{2L}{R}\right) s$$

2.) Propogation delay :- Time required to travel from source to destination

$$d_{prop} = \frac{\text{Distance}}{\text{speed}} = \frac{D + D}{C} = \left(\frac{2D}{C}\right) s$$

3. Queuing delay :- Amount of time, a packet waits in queue

The queuing delay is 0 for first transmitted packet,

(L/R) for 2^{nd} packet and so on...

Thus i^{th} packet has to wait $\frac{(i-1)L}{R}$

Total queuing delay for all packets $= \sum_{i=1}^{N} \frac{(i-1)L}{R}$

$\therefore$ Total queuing delay $= \frac{L}{R} \sum_{i=1}^{N} (i-1)$

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

Since it takes $\frac{LN}{R}$ seconds to transmit a given batch of N packets. Thus, the queue is empty each time a new batch of N packets arrive after $\frac{2LN}{R}$ seconds

$$\frac{2LN}{R} > \frac{LN}{R}$$

Thus average delay with one batch is average delay across all batches.

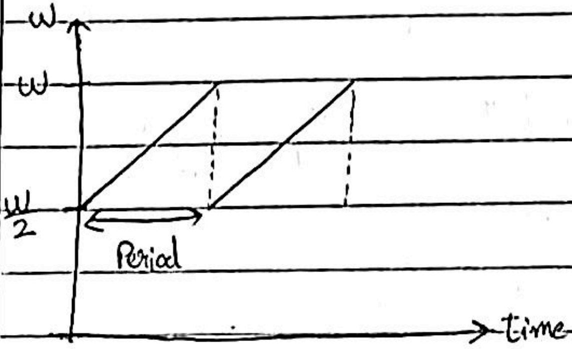
$\therefore$ Average delay $= \frac{(\text{Total transmission delay} + \text{Total propogation delay} + \text{Total queuing delay})}{\text{Total packets}}$

$$= \frac{\left(\frac{2L}{R}\right)\times n + \left(\frac{2D}{C}\right)\times n + \frac{N(N-1)}{2}\left(\frac{L}{R}\right)}{n}$$

Ans Avg. delay $= \frac{2L}{R} + \frac{2D}{C} + \frac{(N-1)L}{2R}$

Q2 (a)

Ans In a steady state, dynamics of TCP can be viewed as:-



* Let loss rate be "L"

$$L = \frac{\text{Number of packets Lost}}{\text{Number of packets sent}}$$

* In each cycle 1 packet is lost.

* Number of packets sent in each cycle

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

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

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

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

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

$$\therefore L = \frac{1}{\left(\frac{3W^2}{8}\right) + \left(\frac{3W}{4}\right)}$$

for large W; $\frac{3W^2}{8} \gg \frac{3W}{4}$

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$\therefore \; L \approx \dfrac{8}{3W^2}$

$$W = \sqrt{\frac{8}{3L}}$$

Average Bandwidth $= \dfrac{3}{4} \times W \times \dfrac{\text{max. packet size}}{RTT}$

$$= \frac{3}{4}\sqrt{\frac{8}{3L}} \times \frac{mss}{RTT} = \sqrt{\frac{3}{2L}}\;\frac{mss}{RTT}$$

$$= \frac{1.22\ mss}{RTT \times \sqrt{L}}$$

Thus for an ideal TCP model in its steady state

avg. bandwidth $= \dfrac{1.22\ MSS}{RTT \times \sqrt{L}}$

Hence, proved.

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Q2 (b)

Ans Given ⇒ Max. segment size = 1500 bytes = 1500×8 bits
RTT = 100 ms = 100×10^{-3} s = 0.1 s
Average throughput = 100 Gbps = 100×10^{9} bps

To find ⇒ Window size = W &
Allowable losse rate = L

In an ideal model for the steady state dynamics of TCP

$$\text{average throughput} = \frac{1.22 \times MSS}{RTT \times \sqrt{L}}$$

$$100 \times 10^{9} = \frac{1.22 \times (1500 \times 8)}{0.1 \times \sqrt{L}}$$

$$\sqrt{L} = \frac{1.22 \times (1500 \times 8)}{100 \times 10^{9} \times 0.1} = \frac{1.22 \times 1500 \times 8}{10^{10}}$$

$$= \frac{1.22 \times 15 \times 8}{10^{8}} = \frac{146.4}{10^{8}} = \frac{1.464}{10^{6}}$$

$$\sqrt{L} = \frac{1.464}{10^{6}}$$

$$L = \frac{2.143}{10^{12}}$$

We know that,

$$W = \frac{\text{bandwidth} \times RTT}{MSS}$$

$$= \frac{100 \times 10^9 \times 0.1}{1500 \times 8} = \frac{10^8}{120}$$

$$\boxed{W = 8{,}33{,}333}$$

$\therefore$ $L \approx 2 \times 10^{-12}$

$W = 833333$

Q2 (C)

Ans TCP congestion control uses retransmission timeouts as one of the technique. However, this RTT based mechanism results in RTT-unfairness issue. Since, all flows sharing a bottleneck link operate at the scale of their respective RTT, ∴ flows with shorter RTTs grows their sending rate at faster pace. This is also quite evident from the fact that average throughput at steady state is inversely proportional to RTT

∴ Avg. throughput/bandwidth $\propto \frac{1}{RTT}$

∴ TCP is thus RTT unfair and flows with different RTT's grow their congestion window different.

TCP CUBIC, the next version of BIC-TCP, mitigates the RTT unfair problem. The key feature of CUBIC is that its window growth depends only on the real time between 2 consecutive congesation events.

One congesation event is the time when TCP undergoes fast recovery. we call this real time a congestion epoch. Thus, the window growth is independent of RTTs, TCP CUBIC, during steady state, increases the window size more aggressively but reduces it pace when it comes ~~close~~ to saturation point wich make this model quite stable and scalable. as well.

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Q2 (d)

Ans As per the RFC 1122 & RFC 2581

	Event at Receiver	TCP Receiver action
(i)	Arrival of in order segment with expected sequence no. All data upto expected seq no already ACKed	Delayed ACK, wait upto 500 ms for next segment. If no next segment, then ACK
(ii)	Arrival of in order segment with expected seq. one other segment has ACK pending	Immediately send single cumulative ACK, ACKing both in order segments
(iii)	Arrival of out of order segment higher than expected sequence number. !! Gap detected !!	Immediately send duplicate ACK indicating sequence of next expected byte.
(iv)	Arrival of segment that partially or completely fills the gap.	Immediate send ACK provided that segment starts at lower end of gap.

Now given that All previous segments before 2001 have been acknowledged.

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Assumption ⇒ TCP model can buffer out-of-order packets too.

Therefore,

· current expected sequence number = 2001 (let it be "ESN")

let Time = 0

(i) Packet with sequence number = 2001 ; length = 1000 bytes

∴ ESN = current packet number = 2001

As per RFC 1122, RFC 2581 rule (i)

⇒ A delayed ACK after waiting upto 500 ms will be sent if no next segment arrives within this time we, know that next segment will arrive at 1000 ms (given) thus ACK will be sent at 500 ms.

\# ACK of (3000+1 = 3001) sent at time = 0.5 s

∴ Now, ESN becomes 3001

(ii) The client receives packet with seq. no. = 3001

· Time = 1 sec

ESN = current packet no. = 3001 ; length = 1000 bytes (Assumed, as not mentioned in question)

by rule (i) of RFC 1122, RFC 2581

A delayed ACK after 500 ms will be sent

ACK of (4000+1 = 4001) sent at time = 1.5 s

ESN becomes 4001

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iii Packet with seq no. = 6001 length = 1500 bytes arrived at time = 2 s

Expected seq no. = 4001 ≠ current seq no = 6001

→ Rule (iii) RFC(1122) RFC 2581

Immediately sends duplicate ACK indicating seq. no. of expected byte

duplicate ACK of 4001 sent at 2 s

TCP model buffers current packet and waits for ACK

(iv) Packet with seq. no. = 5001 & length = 1000 bytes arrived at time 2.2 s

Expected seq. no. = 4001 ≠ 5001 = current seq. no.

→ Rule (iii) RFC 1122, RFC 2581

Immediately sends duplicate ACK indicating seq. no. of expected byte.

duplicate ACK of 4001 sent at 2.2 s

TCP model buffers current packet & waits for ACK

(V) Packet with seq no. = 4001 & length = 1000 bytes
arrived at time 3 2 sec

Expected seq. no. = 4001 = 4001 = current seq no.

rule (ii) single cumulative ack. to ACK pending
in order segments

ACK 7501 sent at 3.2 sec

4001 - 5000	5001 - 6000	6001 - 7501