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Answering to the question no →02

Leaky bucket algorithm control the flow of irregular data and make the data regular flow. This is helps the network to remain congestion free. How it works described ~~before~~ below. Let us assume a bucket with a leak underneath. So the water will come out through the bucket constantly no matter how

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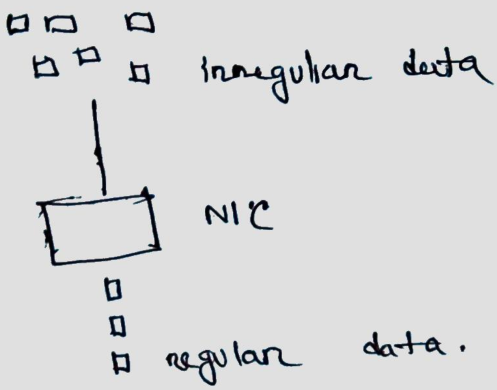
~~wu~~ much water the bucket contains.

Similarly this idea applies to control the flow of irregular data in the ~~m~~ network interface card.



This is how this algorithm helps congestion free network.

Given,

Data = 496 Mbs

Time = 950 ms

$$\therefore \text{Data} = \frac{496}{1000} \times 950$$

$$= 471.2 \text{ mb}$$

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$\therefore$ duration of output $= \dfrac{471.2}{24}$ s

$= 19.63$ sec

Answering to the question no → 03

Given scenario,

scenario 1,

$P_1 \longrightarrow P_1$

Delay in arrivals 89.5, 91.6, 89.4, 90.7, 90.8, 89.9

scenario 2;

Delay in arrivals 6, 23, 12, 59, 22, 90

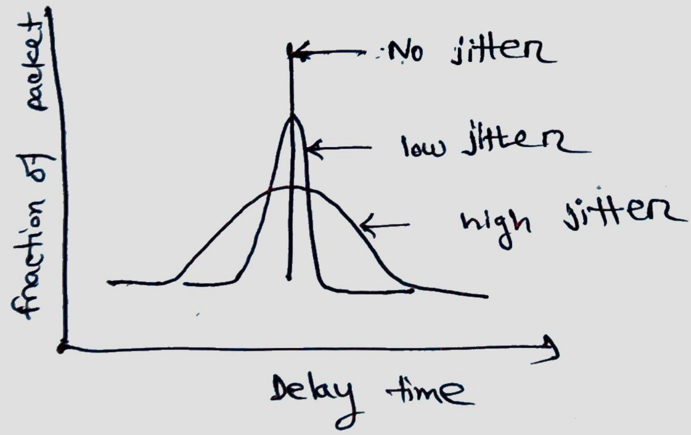
If we observe the above scenario we can see that, in senario 1, the variation or differrence betwen arrival time are very low. On the other hand the variation of arrival time in scenario 2 is very high.

Let us see how this variation affect jitter. If the variation is low the jitter is low. so, the delay between arrival packet is low. If the variatio is high the jitter is also high.

So, from the above scenario, scenario 1 has the low jitter.

Low jitter is very important in multimedia communication. If the jitter is low the packet will process faster and there will be no problem in the audio or video quality of multimedia communication.

If the jitter is high there will be poor connection in the multimedia communication. So, low jitter is very importan in multimedia communication.



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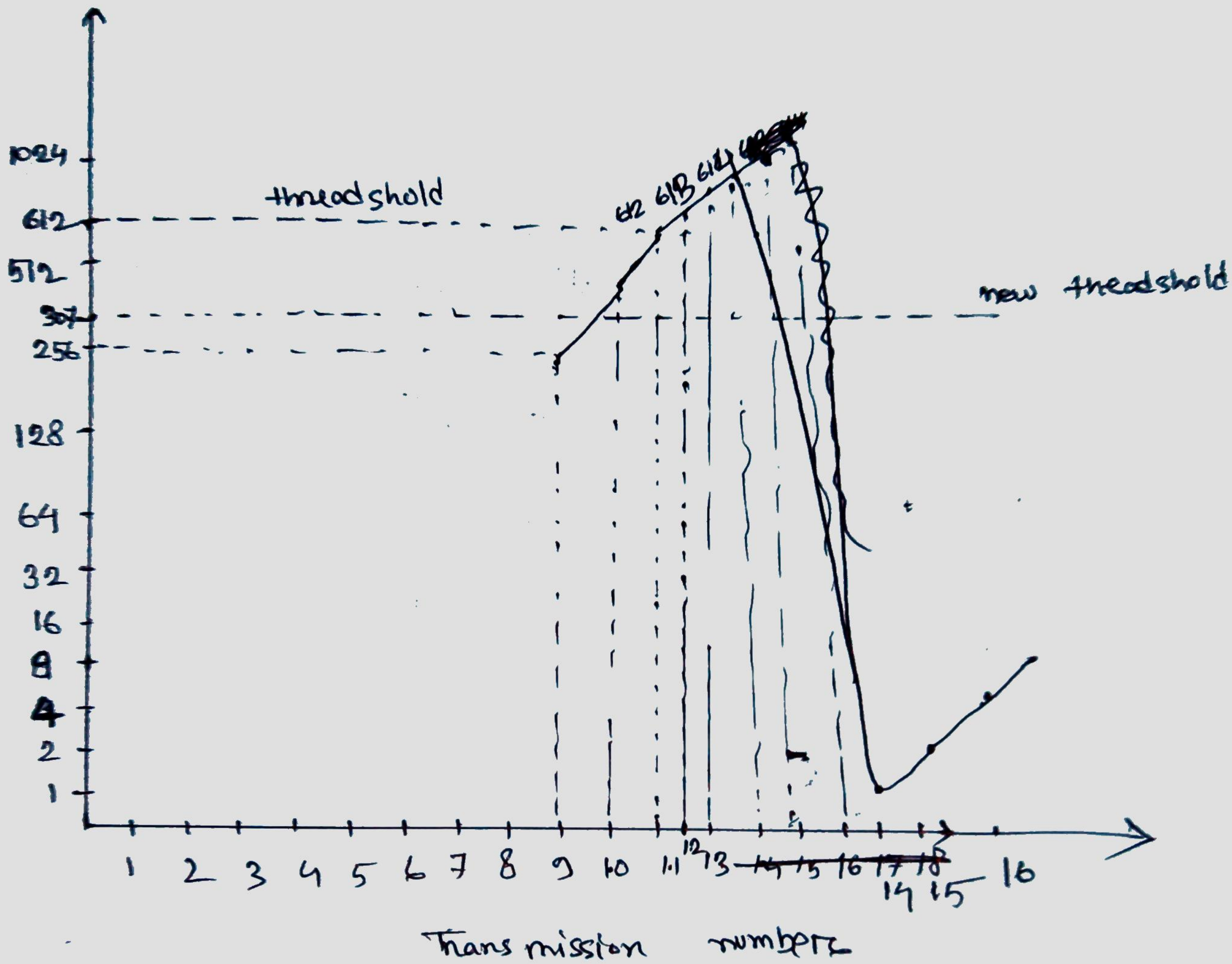
Answering to the question no → 04



New threadshold = $\frac{1}{2} \times$ time out = $\frac{1}{2} \times 614$

= 307

Congestion window:

10 → 512

11 → 612

12 → 613

13 → 614

14 → 1

15 → 2

16 → 4

17 → 8

Answering to the question no→05

From the scenario, we can see that the server farm does not have shared memory because each processing node has its own cache memory. So that the processing node can direcly access the cache ~~m~~ memor they don't need to seek for shared memory. If there is a shared cache memory then it would have been in a separate network like SAN. The node will require the access of that shared memory from the network. Which is time comsuming.

But, Every node has its own cache. so, it can access the cache memory faster. So the performance of individua direct access memory is more efficient than shared cache memory. This is how the performance of the server farm will increase by using own cache memory of each processing node.

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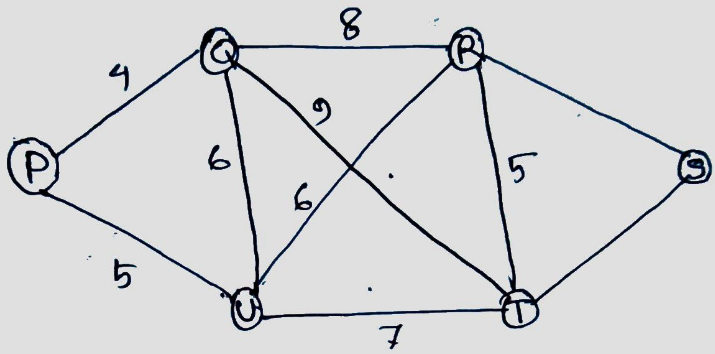
Answering to the question no-1

Given subnet,



So, the link-state packet for router 'T' is

'T'	
Sequence #	
Age	
'S'	2
'R'	5
'Q'	9
'U'	7

Here, A link state packet contains the ip address of router. A sequence number which maintain the updated information about state of another router. Sequence number is updated when a higher sequence number is read.

Age in link state packet is the time limit of a sequence number it in

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the buffer.

Age is a time limit where a packet will be removed from the buffer after that age time. So, if there is any read error of higher or lower sequence numbers, we can solve the errors by using this 'age' in the link state packet. But there is a problem with the duration of age.

Let, 32 bits of sequence nuber is used in a link state packet and packet is updated in every 10 seconds.

if the duration of age is short, assuming 15 sec then

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Seq# 1 ✓

10

seq# 2 ✓

15 sec seq# 2 will ~~be~~ acknowledged

20 Seq# 3 ✗

30 Seg# 4 ✓ at 30 sec seq#4 will upadted, seq#3 lost

So, due to short duration of age seq# 2 will die and new seq# 4 will be updated in buffer. Here the loss of packet occurs which will be problemetic establishing connection. This cycle will go on and we will loss so many packets. So age can't help to solve the sequence number wrap up problem.

if the duration of age is high, assuming 100 sec

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Seq#1

10 Seq#2

20 Seq#3 (read error 333)

30 Seq#4 ×

so many packet will be lost

130 Seq#14 ✓

So, in long duratation age there will be backdated information in the buffer for a long time and many packet will be lost.

This is how age plays a vital role in the link state packet. We ~~usuall~~ ideally use 50 - 60 see for age so that we maintain the above scenarios.