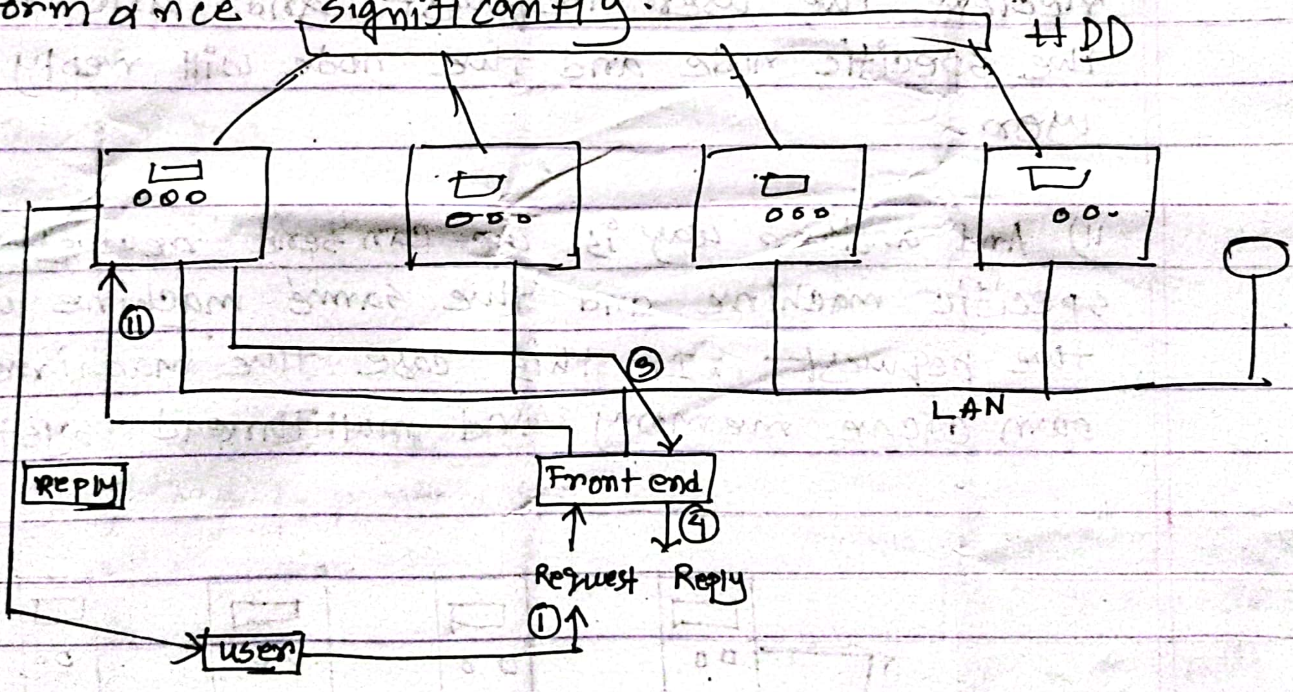
Ami's sir Ami's CSE405 final

Here front end is busy to handle receiving and replying request. To improved performance we can send the reply to the user directly which is called "TCP hand off" and another way is we can send the request to the specific machine and that same specific machine will reply the request. In this case ~~each~~ mechine has used its own ~~ch~~ cache memory and its multithread system. so, it easily and fast search in its cache memory and reply the request It can improve the performance significantly.



Hossain Mim

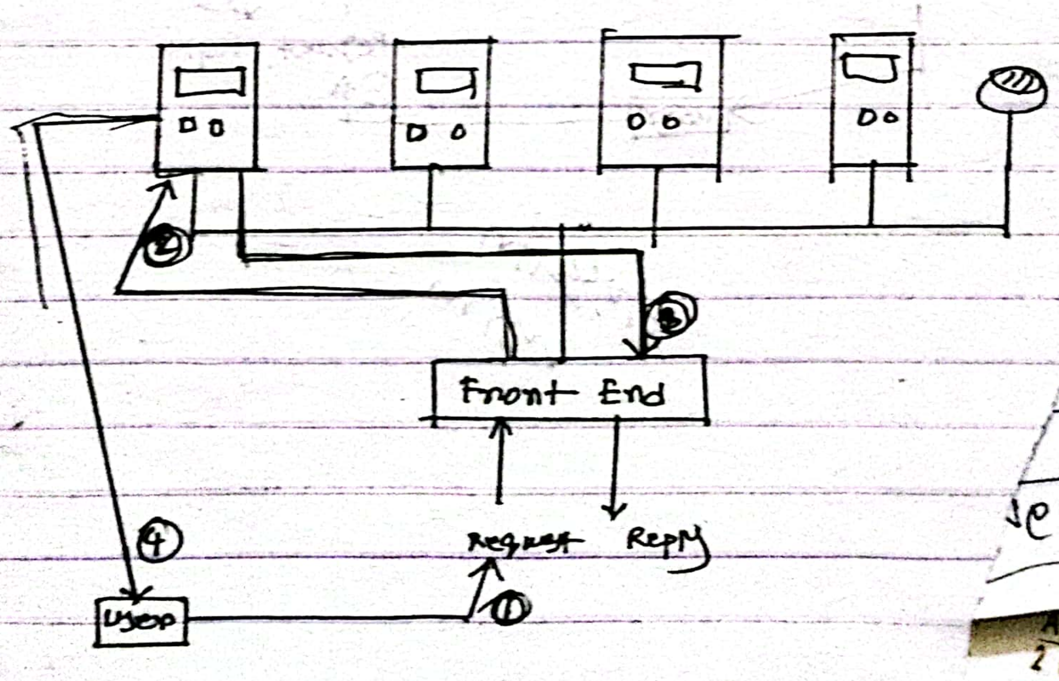
S
par
In
fragm

And another way is we send the ~~&~~ req to the specific ~~node~~ machine and the same mach will be reply to the rq. In

In the given server farm each processing nodes has its own cache memory. It is already optimized server fram because each processing nodes has its own cache memory. when a user request for something then the processing nodes can sever this fram thier own cache memory.
For inceaseing performance of the server fra

I) The TCP handoff that is front end will ~~respons~~ reciere the user request and hand it over to the specific node and the node will reply directly user.

II) And another way is we can send request to the specific machine and the same machine will reply the request. In this case the machine is used own cache memory and multithread system.



Ans to the question no: 3

given,

present threshold 612 KB
present conjestion window 256KB
time out point 614 KB

Next conjestions:—

1st: $256 \times 2 = 512$ KB
2nd: 612 → Threshold
3rd: 613 KB
4th: 614 → time out
5th: 1 KB → start again
6th: 2 KB
7th: 4 KB
8th: 8 KB Ans

given,

present threshold is 612 KB
present congestion window is 256 KB
time out point 614 KB

Next congestion :—

1st: $256 \times 2 = 512$ KB
2nd: 612 KB
3rd: 613 KB
4th: 614 KB
5th: 1 KB KB
6th: 2 KB
7th: 4 KB
8th: 8 KB

Sc
pag
In
fragh

Ans to the question no: 2 (2nd part)

(calculated the leak of the Bucket)

- Data transmission rate from the pc = 512 MB/s
- Time for transmission
$= 950\ \mu s$
$= 0.0095$ ms
$= 0.000095$ s
- Network data transmission capacity 64 MB/s
- Time interval transmission
25 ms
$= 0.025$ s

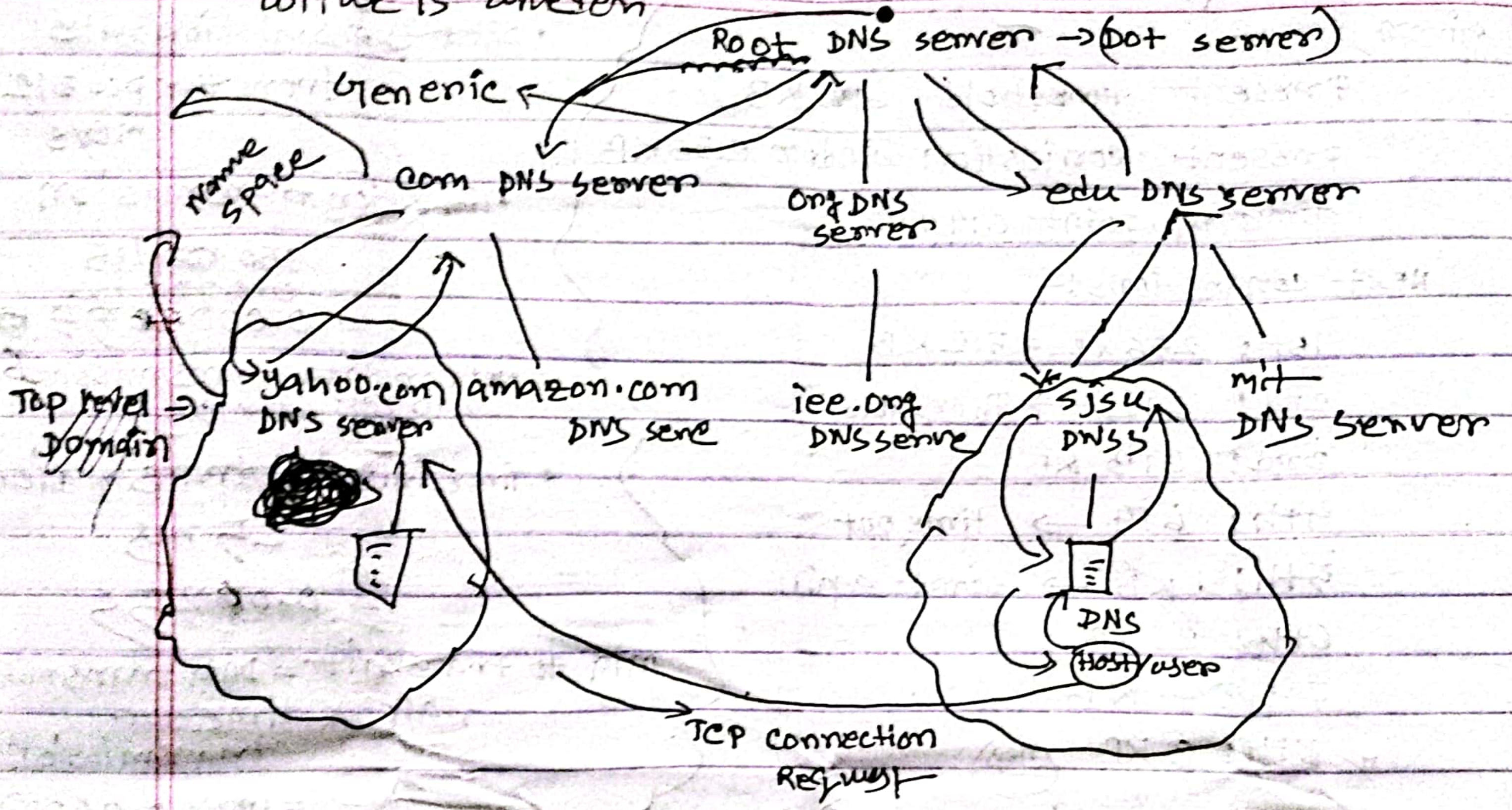
Data to transmit = Data transmission rate × time for transmission
$= 512 \times 0.000095$
$= 0.4864$ MB

Network capacity $= 64 \times 0.025$
$= 1.6$ MB

Leak Rate $= \frac{\text{Data transmit}}{\text{time interval}}$
$= \frac{0.4864}{0.025}$
$= 19.456$ MB/sec

of sjsu.edu for stuco os DNS

state the steps of a TCP connection with web server of yahoo from a host withe is under



Step of the host →:

- Browser ~~determines~~ find the yahoo ~~.com~~ URL.
- Browser asked the DNS server for the IP address of yahoo URL.
- DNS Replies with th IP address
- Browery makes a TCP connection with machin
- Get the request file.
- TCP Connection ~~succes~~ successful.
- Dispry the reply file.

Now for the server side: →

- Accepet the TCP connection
- Gets the file from the Disk
- Return to the client
- TCP connection Re'lease.

Release

Hossain Mim

jitter is negative factor. jitter refers to the variation in the packet arrival times in a network

From the given scenario 1 :—

Highest Delay : ~~...~~ 91.6 ms

release

Lowest Delay : 89.~~..~~4 ms

so, the jitter variation is = (91.6 − 89.4) ms

= 2.2 ms

Hossain

Form the given scenario 2:

Highest Delay: 90 ms

degrade

Lowest Delay: 6 ms

so, the jitter variation is (90−6) ms

= 84 ms

~~...~~ clearly ~~...~~ scenario 1 has low jitter because the variation in the packet ~~..~~ arrival time is very much smaller scenario 2.

low jitter is for high quality multimedia communication.

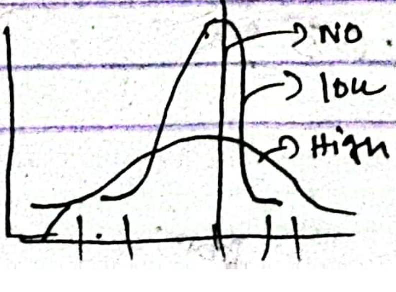
In the multimedia communication, if all data packet ~~...~~ ~~...~~ delay a long time but arrival time at the same or very closer time than the quality remains good.

But if jitter is high ~~...~~ and all data packet arriva time on closer time in different that the quity of the media degrades.

So, the multimedia comumincation ~~...~~ do not used high jitter. ~~...~~

Pack. jitter is good scenario. so, it is very necessary the jitter to be low in multimedia communication

In the fragm



1ms = 0.001s

7600 ms

7.6 ms 7.6×10^{-3}

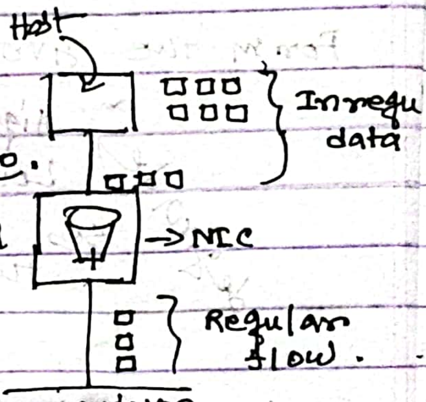
=> 0.0076 s

$\times 10^{-6}$ 0.000001

The LBA is a network traffic shaping mechanism used to control rate at which data transmitted.

It purpose is to ensure a consisten and controlled output rate from a network.

- If all data packet tries to flow at the same time then conjestion occurs.
- Incomming data packets are added to the bucket at a certain rate.
- If the bucket overflow, excess data packets are dropped or later transmission.
- packets are removed from the bucket at a consisten output rate and ~~controlled~~ continuous flow of the data.



Input data rate 496 MB/S

Time 950 ms

Output data rate 24 MB/S

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

$$= 471.2 \text{ MB}$$

$$\text{Duration output time} = \frac{\text{Data}}{\text{output rate}}$$

$$= \frac{471.2}{24}$$

$$= 19.63 \text{ S}$$

The duration of output for a Leak would be 19.63S considering network data rate 24 MB/S.

Input data rate 51

time: 950 ...

output data

64

Data: $\frac{5}{10}$

= 6

owner Source Age Seq# link state pack sent A B C D Ack sent

In short length of a sequence number. The sequence length is only 4 (four) Bit long. we know. is link state algorithm sequence number is update by its next number which is greater.

when we used 4 Bit sequence number, after "1111" which is 15, there is no member which greater then 15.

So, it is not possible to update the data. That's way 32 bit sequence number comes to solve this proble.

Distribution table for Q: —

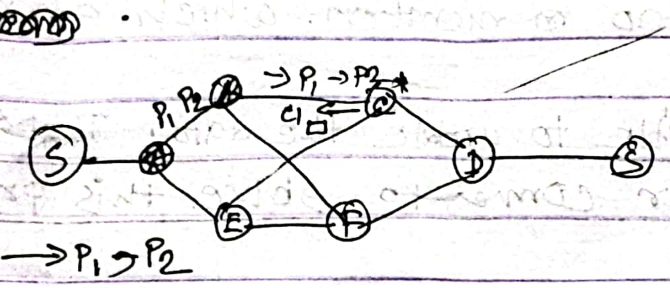
owner	Source	Age	Seq#	link state packet sent P	U	T	R	ACK sent P	U	T	R
P	T	60	118	0	1	0	1	1	0	0	0
T	R	60	98	1	1	0	0	0	0	1	1
P(x) P	U	60	116								

P link state packet from U sequence 116 but Q already receives P to T with sequence 118. So, it will not include in the distribution table.

In short length of a sequence number. The sequence number length only four (4) bit long. we know that,
So, link SA Sq No is update by its next number which is greater than.
packe when we used to the 4 bit squence number after
In the which is 15, there is no greater than
fragme 15.

tis not possible to update the data after 17 which 15. That's way 32 bit comes to solve this

The "Hop-by-Hop ckp" conjection control Algriothum is an improvement over the basic 'CKP' Algriothm because it allows for more fine-grained control of congestion at each hop or intermediate node in the network.



when source generating packets routers forward these packets from one to another Assuming that router C is suffering conjection It send chock packet to source. But when C pases the ckp to the B router B will understand that it suffer from congeation To reduce it B router store some data it

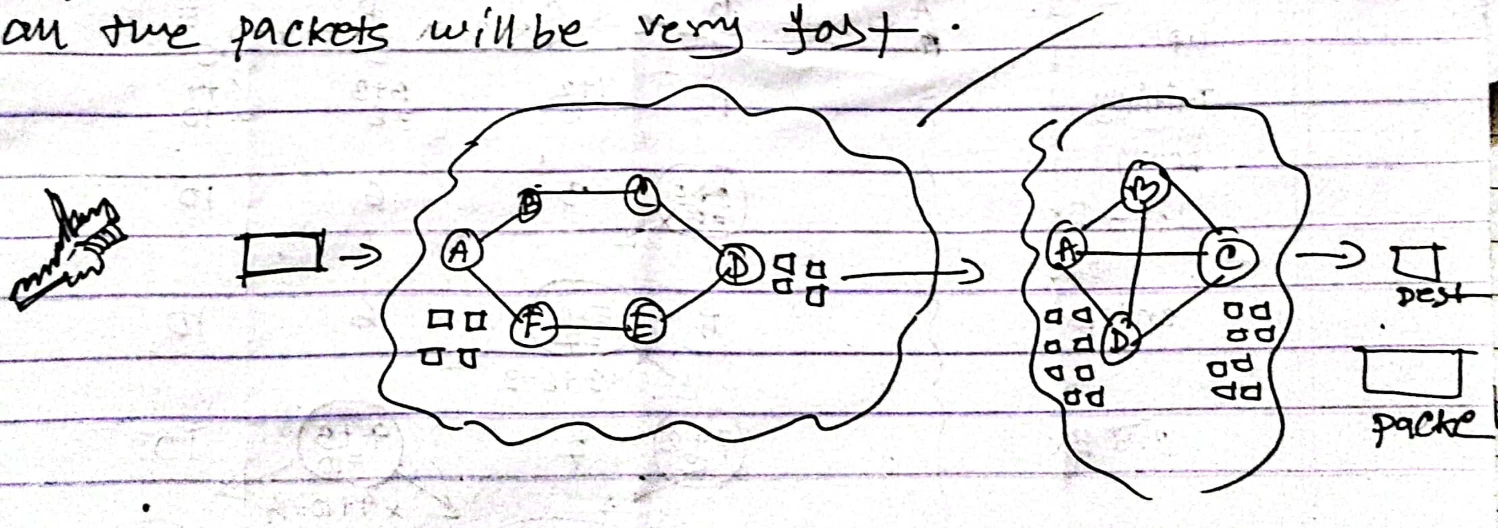
This way follow that congestion can be detected and address more quickly and improving overall network performance.

The "Hop by Hop cp" congestion control algorithm is improvement over the basic 'chock packet' algorithm because it allow fine-grained control of congestion at each hop in the network when the source generating packet frouter forward these packets from one to another Assuming that router C is suff conjetion. It send the choke packet to source But when C

It intermediate networks are connection oriented then the "Non-transparent Fragmentation" would be better.
we know that, two types of fragmentation :-

i) Transparent
ii) Non transparent

In Non-transparent fragmentation, the packet size is larger than it transporting capacity and if the router has fragmentation permission then the packet will be fragmented and send to the next routers. And a fragmented packet can be frament again.
the network has good connection then non-transparent fragmentation will be good because there will be less chance to ~~lose~~ lost the packet
So, connection oriented network to reassmble all the packets will be very fast.



So, the connection oriented network to reassemble all the packets will be fast.

In the network has good connection then non-transparent fragmentation will be good because there will be less chance lost the packe.

2nd ⇒ C থেকে B আসতে লাগবে 2 s
2 আর 5 = 7 s

E থেকে option নাই
D শুধু release করবে
E থেকে D যেতে 4s
D থেকে A 7 10

up to Down

(1st) B থেকে C যেতে লাগে 2 sec আর C থেকে A তে যেতে লাগে 3
immediately করে

A ←1→ B ←2→ C ←3→ D ←4→ E

	A	B	C	D	E
(up) Initially		1	1+2 = 3	3+3 = 6	6+4 = 10 ⇒ A তে release করার value
(Down) 1st		2+3 = 5	3	6	10
2nd		5	5+2=7	6	10
			×3+6=9		
3rd		7+2 = 9	7	7+3 = 10	
				×4+10 = 14	
4th		9	9+2=11	10.	4+10 = 14

4th: D থেকে E যেতে লাগে 4s
D থেকে A যেতে 10

3rd:- C পাঠাবে 7 s
B এর 2 s পাঠাতে
D কাছে যাবে, D কাছে দুইটি value আছে A থেকে 1টি E হতে, অন্যটা C
D থেকে E তে পাঠাবে যখন 3 s
C 7s

A ←1→ B ←2→ C ←3→ D ←4→ E

	A	B	C	D	E
up Initially		1	1+2 = 3	3+3 = 6	6+4 = 10
Down 1st EX		2+3 = 5	3	6	10
2nd. EX		5	5+2 = 7	6	10
			×3+6=9		
3rd EX		7+2 = 9	7	7+3 = 10	10
				×4+10=14	
4th EX		9	9+2=11 ×10+3=13	10	10+4 = 14

Ans

Hossain mim