MNC PA2 Report

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I have read and understood the course academic integrity policy.

TimeOut Scheme.

1. For ABT Implementation the timeout used is 30 units. As the average RTT is 10 units timeout has to be more than 10units. It is found from observation that for the lossy and corrupted scenarios the throughput is higher if timeout of 30 units is used.
2. For GBN throughput is found to be maximum for timeout of 15 units. Even though there are some retransmissions in ideal case of without loss or corruption the usage of 15 time units is giving good throughput results because if the timeout is increased then the window size won't move faster as we need to wait for more time for the lost or corrupted packets.
3. Similarly for sr protocol the timeout used is 30 units. By running the experiment tests by varying loss and corruption probability throughput is decreasing by increasing the timeout. But for much smaller timeouts there are lots of retransmissions to balance both results timeout of 30 units is selected.

Multiple Timers Implementation.

Data Structures Used:

```
struct pkttimer
{
  float timeout;
  int seqno;
};
```

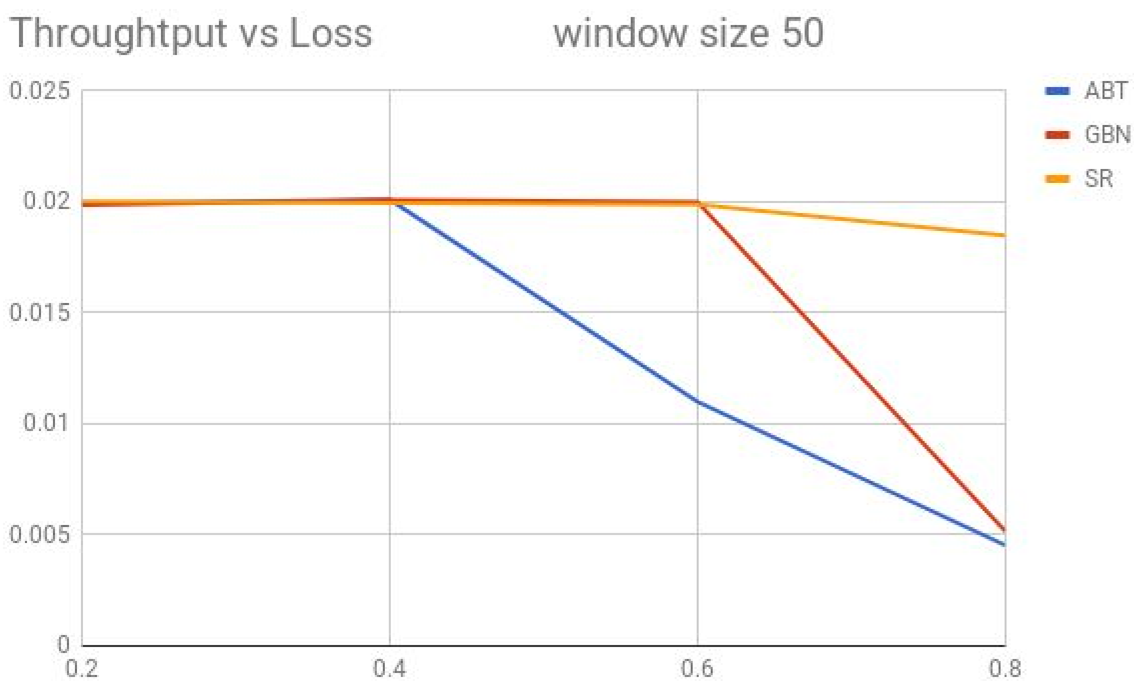
```
std::list<pkttimer> timerlist;
```

Multiple Software timers is SR is implemented by using lists std::list<pkttimer> timerlist where pkttimer is a structure with two parameters float and int. Pkttimer is used to populate the current simulation time when the packet is sent from A and its sequence number. When a packet is sent from a the corresponding Pkttimer is populated and added in to the list at back. Timer is started when first packet is transmitted. And when the timeout occurs we can check whether the Ack for the packet is received and retransmit accordingly. Each time the timeout occurs timeout value in next pkttimer is subtracted with the previous timeout value and the Timer is Restarted with this value.

Performance Comparison

Experiment1:

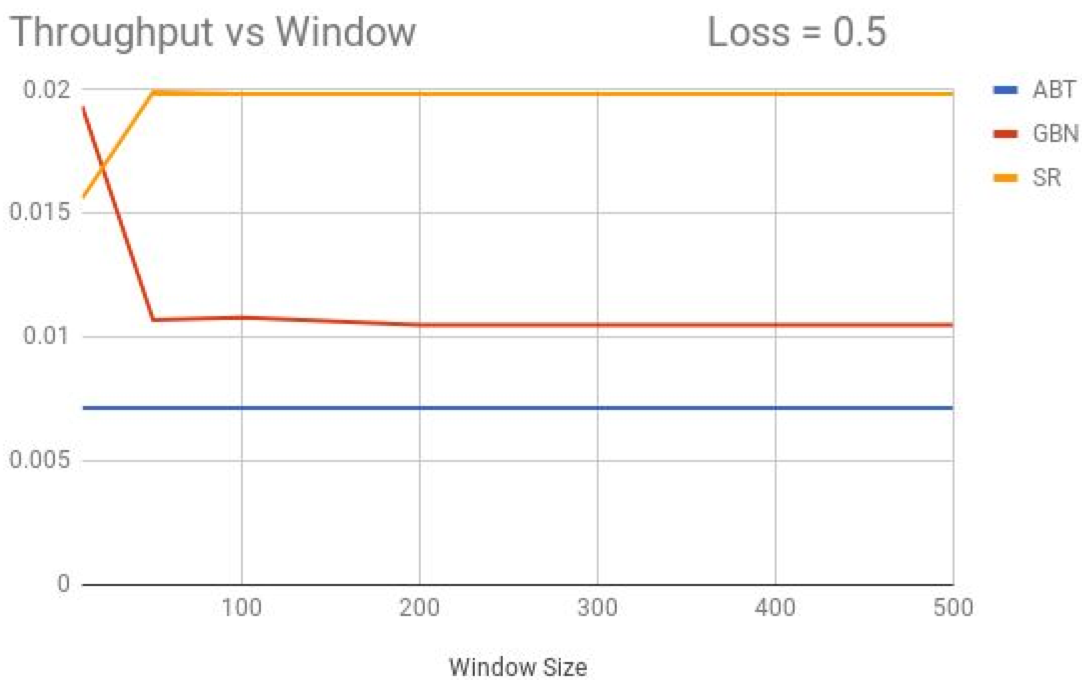
In the lossy environment generally packets gets dropped and receiver has to retransmit the lost packets after the timeout this reduces the number of packets delivered per unit time as a result throughput decreases with increase of loss. This behaviour is common for all the protocols and it is depicted in the graph.

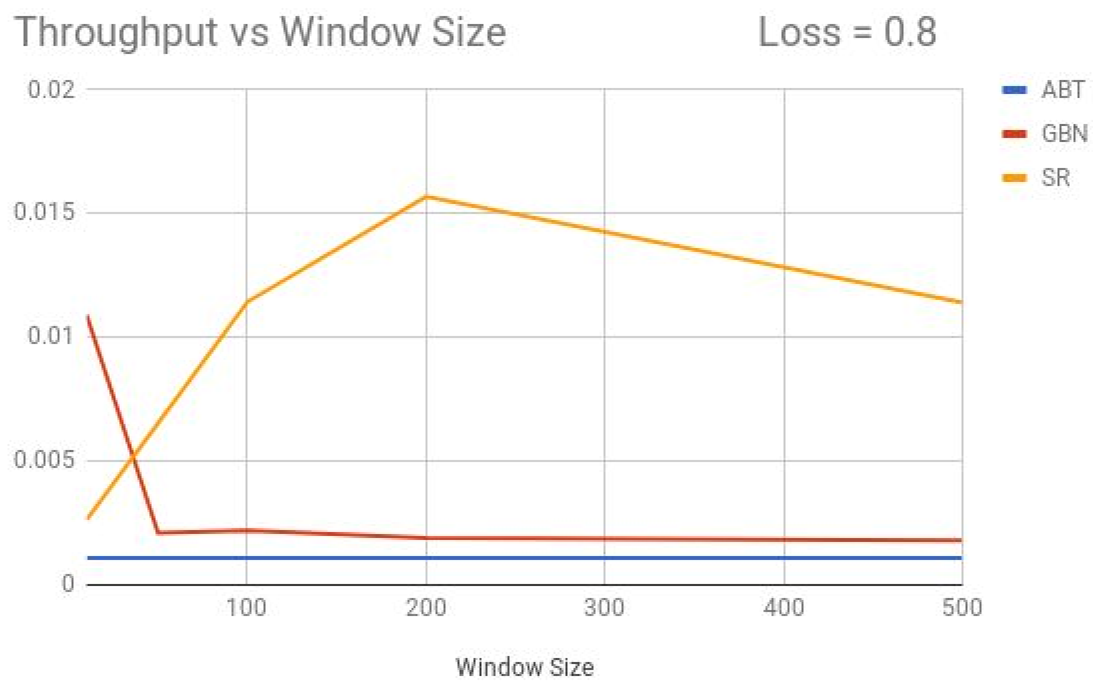


Similarly for the window size 50 the throughput decreases with increase in loss probability. But the rate of decrease of throughput changes with the window size. For higher window size initial increase in loss probability won't have much impact on the throughput. And from the graphs it is evident that SR is the preferred protocol if the channel is very lossy.

Experiment 2:

For the small or no loss in the channel the throughput is not much dependent on the window size.because if there is no packet loss then there will be steady increase in sender and receiver windows of both GBN and SR protocols.


Throughput vs Window
Loss = 0.5
0.02
0.015
0.01
0.005
0
100
200
300
400
500
Window Size
ABT
GBN
SR



From the above graphs we can say that for no/small loss and small window size GBN is the preferred protocol and in the high loss and large window size availability the SR is the preferred protocol. And ABT is independent of window size and is least preferred protocol in lossy channel.