A Report on NS3 Project

CSE 322

A Comparative Study on Performance of Different Congestion Control Algorithms

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Contents

1 Introduction

This report briefly discusses the varied network parameters and performance metrics of two built-in and one implemented congestion control algorithms, with three different network topologies simulated.

2 Network Topologies Under Simulation

Two different topologies were simulated for task-A, one wired and one wireless low rate static. For task-B, a suitable topology for data centers was used.

2.1 Task A1 - Wired topology

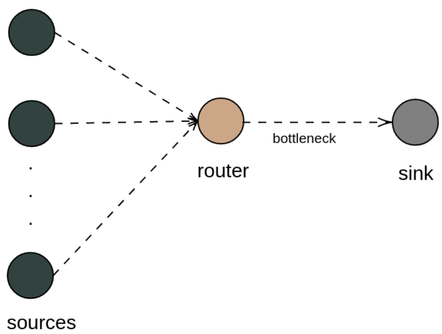


Figure 1: Wired Topology

In this topology, we have n number of sources, one gateway and one sink. All nodes have point to point wired connections

between them. After creating point to point channels between each pair of nodes, the data rate and delay value of the channel were set to 2Mbps and 0.01ms. A rate-error model was installed on the devices to simulate error on the network and the error rate was set to be 0.01%. Then FIFO-queues were installed on the devices. Finally individual ipv4 networks were created for each wired connection.
Two congestion control algorithms - TcpNewReno and Tcp-Westwood were tested on this topology.

2.2 Task A2 - Wireless low-rate (802.15.4, static)

This topology was created using low rate PAN devices (wifi

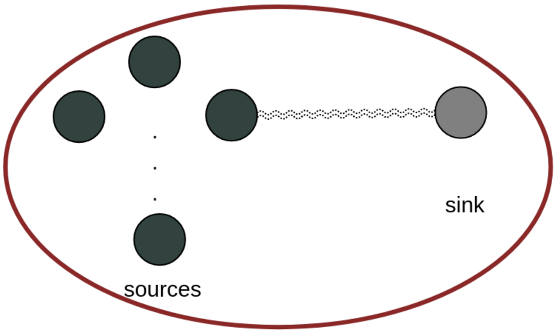


Figure 2: Wireless Topology (Low Rate Static)

standard 802.15.4) in a static wireless IPv6 network. There are n number of sources and one sink.
After creating the nodes, the mobility model was used to set the position of the nodes in a square shaped grid. The coverage area of the grid was varied. For creating static nodes, constant position mobility model was used. The maximum transmission range of the nodes was set to be 80 meters using range propagation loss model.
The low rate PAN devices were created using LrWpanHelper. Finally, an ipv6 network was installed on all the nodes.
Both congestion control algorithms for task A1 were also applied and tested on this topology.

2.3 Task B - Topology for Data Centers

This topology was inspired from the topology provided in [1] (Alizadeh et al., 2010) for data centers. This is a fully wired topology simulated with a lot of flows. The steps to create this topology is similar to Task-A1 topology. Two congestion control algorithms were tested on this topology - TcpDctcp, a popular congestion control algorithm for data centers, and TcpSwift, an algorithm implemented in NS3 in this study. The algorithm was inspired from the algorithm presented in [2] (Kumar et al., 2020).

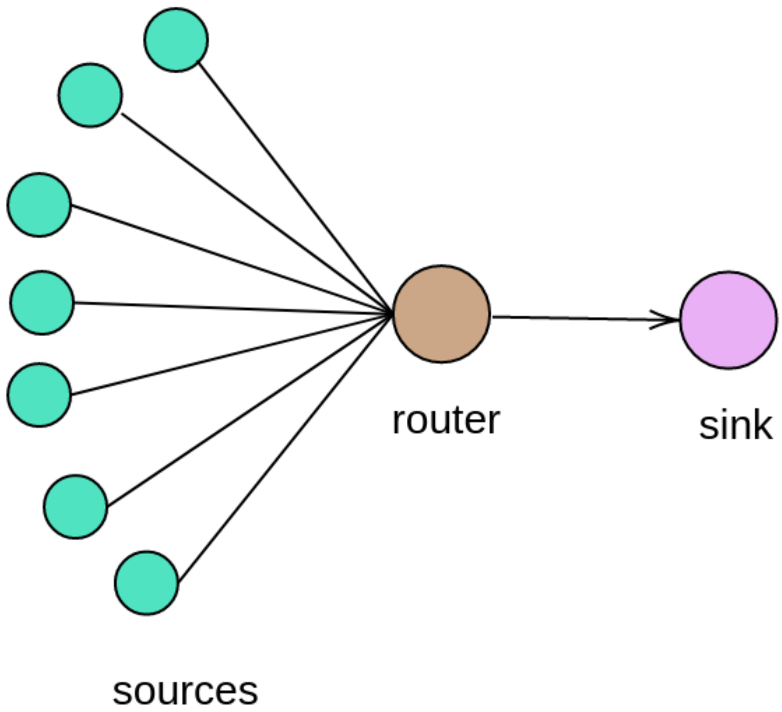


Figure 3: Wired Topology Suitable for Data Centers

3 Parameters Under Variation

The following parameters were varied to analyze performance metrics. Here Txrange = 20 and Maxrange = 80.

#nodes	#flows	#packets per second	coverage area (wireless)
20	10	100	1*Tx_range
40	20	200	2*Tx_range
60	30	300	3*Tx_range
80	40	400	4*Tx_range
100	60	500	5*Tx_range

4 Overview of the Proposed Algorithm

The algorithm, Swift, proposed in this study was inspired from [2] (Kumar et al., 2020). Swift uses delay as the primary congestion signal. The accurate measurement of delay can result in a high performing congestion control algorithm. Swift was designed for data centers which are prone to extreme congestion. The DCTCP algorithm [1] is standard for congestion control in data centers. Swift was tested in Google datacenters and it performed much better than DCTCP, with near-zero packet drops, while sustaining 100Gbps throughput per server. Following is the pseudocode of the algorithm :

Algorithm 1: SWIFT REACTION TO CONGESTION

```
Parameters: ai: additive increment, β: multiplicative decrease
  constant, max_mdf: maximum multiplicative decrease factor
cwnd_prev ← cwnd
bool can_decrease ←                          ▷ Enforces MD once every RTT
  (now − t_last_decrease ≥ rtt)

On Receiving ACK
  retransmit_cnt ← 0
  target_delay ← TargetDelay()               ▷ See S3.5
  if delay < target_delay then               ▷ Additive Increase (AI)
    if cwnd ≥ 1 then
      cwnd ← cwnd + (ai / cwnd) · num_acked
    else
      cwnd ← cwnd + ai · num_acked
  else                                       ▷ Multiplicative Decrease (MD)
    if can_decrease then
      cwnd ← max(1 − β · ((delay−target_delay) / delay),
        1 − max_mdf) · cwnd

On Retransmit Timeout
  retransmit_cnt ← retransmit_cnt + 1
  if retransmit_cnt ≥ RETX_RESET_THRESHOLD then
    cwnd ← min_cwnd
  else
    if can_decrease then
      cwnd ← (1 − max_mdf) · cwnd

On Fast Recovery
  retransmit_cnt ← 0
  if can_decrease then
    cwnd ← (1 − max_mdf) · cwnd

cwnd ←                                       ▷ Enforce lower/upper bounds
  clamp(min_cwnd, cwnd, max_cwnd)
if cwnd ≤ cwnd_prev then
  t_last_decrease ← now
```

Figure 4: Pseudocode of Swift

5 Modifications Made in the Simulator

- **New Files** : The following files/classes were added to incorporate Swift :
  - tcp-swift.cc
  - tcp-swift.h
  - class TcpSwiftRecovery in tcp-recovery.ops.cc

  All of these additions were done to the internet module. The wscript file of the module had to be modified to introduce the additions.

- **Modified Files** : The following files/classes were modified to incorporate Swift :
  - tcp-socket-state.cc, tcp-socket-state.h
  - tcp-socket-base.cc

5.1 Variables

The following variables were needed to implement Swift. They were added to TcpSocketState class in the tcp-socket-state files.

- **bool m_canDecrease** - for deciding when the congestion window can decrease
- **uint32_t retransmit_count** - to trigger recovery algorithm
- **uint32_t m_gamma** - multiplicative decrease parameter (max_mdf), initial value was set to be 1
- **uint32_t max_cwnd** - A parameter to control maximum congestion window, initial value was set to be nearest segment size of 20000

- **uint32_t min_cwnd** - A parameter to control minimum congestion window, initial value was set to be nearest segment size of 500

The following variables were added to TcpSwift class in tcp-swift files.

- **uint32_t m_alpha** - additive increment (ai), initial value was set to be 2
- **uint32_t m_beta** - multiplicative decrease constant, initial value was set to be 4

5.2 Functions

- **TcpSwift** :
  - *PktsAcked*

    This function is triggered when a packet is acknowledged. In this function the retransmit count was set to be zero. Also the minimum and base round trip time were updated.
  - *IncreaseWindow*

    The congestion window was adjusted based on different conditions, following the Swift algorithm. When congestion window is less than slow start threshold, we follow the slow start method of New Reno algorithm. When it reaches the threshold, we move on to congestion avoidance phase by adjusting the congestion window. Here we use delay as a threshold.

    If rtt is less than target delay, we get the congestion window in segments and do the following :
    * if congestion window is greater than one segment size,

      $$cwnd = cwnd + \frac{\alpha}{cwnd} * segmentsAcked$$

* else,

$$cwnd = cwnd + \alpha * segmentsAcked$$

If rtt is not less than target delay, we check if the can decrease boolean value has been set to true. If it is, we do the following :

$$value1 = 1 - \beta * (\frac{delay - targetDelay}{delay})$$

$$value2 = 1 - gamma$$

$$cwnd = max(value1, value2) * cwnd$$

– *TargetDelay*

Since the values of the parameters used in the target delay function of the paper [2] were not defined clearly, a placeholder value was used in this function to be fine tuned later. From the following plot of [2](Kumar et al., 2020), we set the target delay to be 25 microseconds.

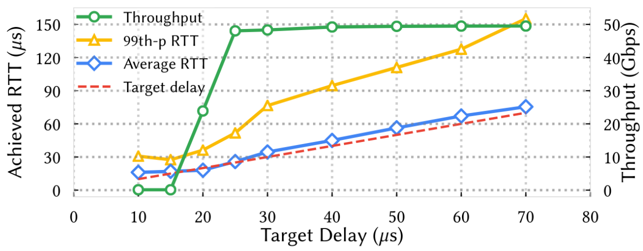


Figure 17: T_1: Achieved RTT and throughput vs. target delay, 100-flow incast.

Figure 5: Target Delay vs Throughput

- **TcpSocketBase** :

– *ReTxTimeout*

This function is triggered when a retransmit timeout event occurs. We increment the retransmit count here.

If it is greater than retransmit threshold (retxThreshold), the congestion window is set to be the minimum value defined by us. Else, if the boolean canDecrease is set, we do multiplicative decrease :

$$cwnd = (1 - gamma) * cwnd$$

- **TcpSwiftRecovery** :
  - *Enter Recovery*

    In this function we set the retrasmit count to be zero. Then, if the boolean canDecrease is set, we do multiplicative decrease :

    $$cwnd = (1 - gamma) * cwnd$$

    Then we apply the clamp function on the congestion window and if it decreases from the previous congestion window, we set the time of last decrease to the current time.

- **TcpSocketState** :
  - Clamp

    This function sets and upper and lower bound on the congestion window.

6 Results with Graphs

For each task, the following metrics were calculated using the help of flow monitor.

- **Network throughput (Kbps)**
  The network throughput was calculated for each flow with the following formula -

  $$\frac{receivedBytes \times 8}{(TimeofLastReceivedPacket - TimeofFirstSentPacket) \times 1024}$$

  The throughput was summed for each flow then finally it was divided with the number of flows to get the average throughput.

- **End-to-end delay** The end to end delay for each received packet of the network was summed and then divided with total received packets. The unit is nanoseconds.

  $$\frac{EndtoEndDelaySum}{ReceivedPackets}$$

- **Packet delivery ratio**

  $$\frac{ReceivedPackets \times 100}{SentPackets}$$

- **Packet drop ratio**

  $$\frac{LostPackets \times 100}{SentPackets}$$

6.1 Task A1 - Wired Topology

Figure 6 describes the performance metrics of two congestion control algorithms - TcpWestwood and TcpNewReno against varied number of flows in each simulation. Here number of nodes = 40 and packets per second = 400.

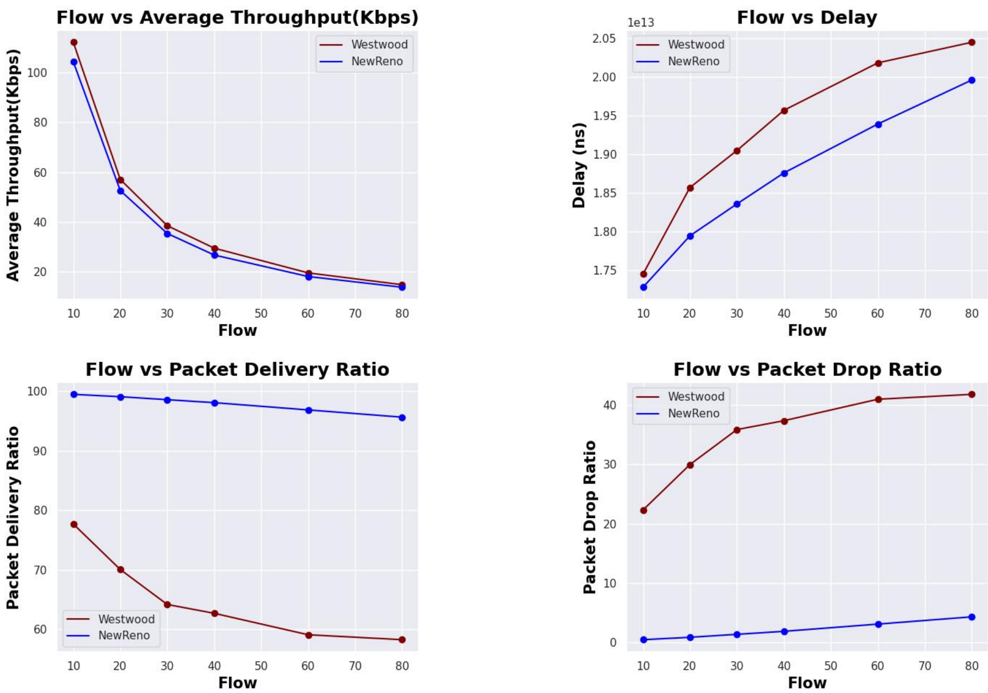


Figure 6: Flow vs Metrics

Figure 7 describes the performance metrics against varied number of nodes in each simulation. Here number of flows = 80 and packets per second = 400.

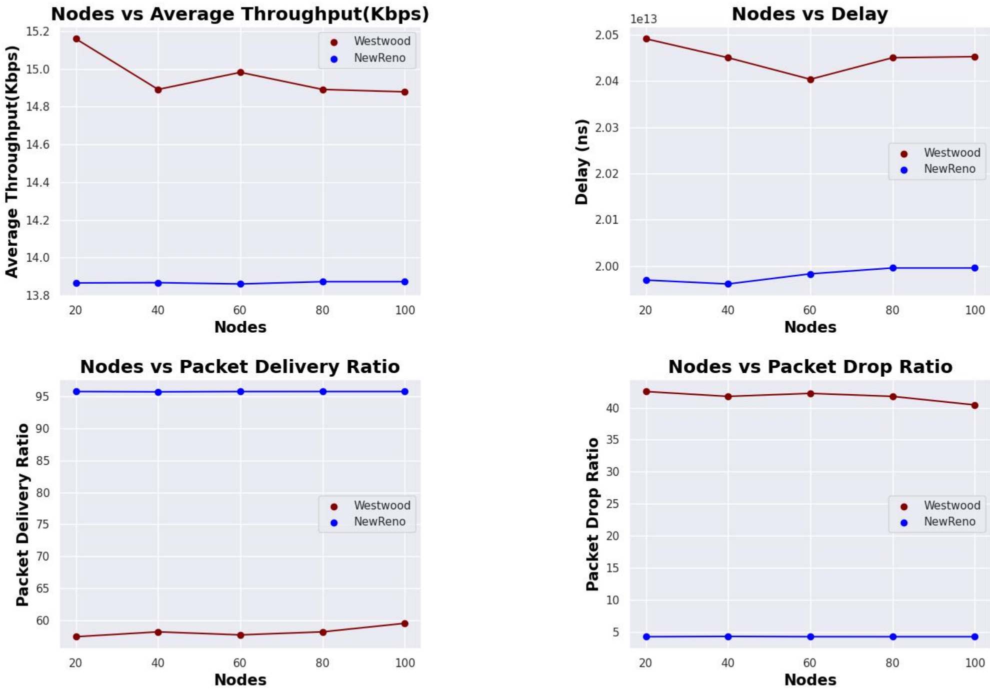


Figure 7: Nodes vs Metrics

Figure 8 describes the performance metrics against varied number of packets per second in each simulation. Here number of flows = 80 and nodes = 40.

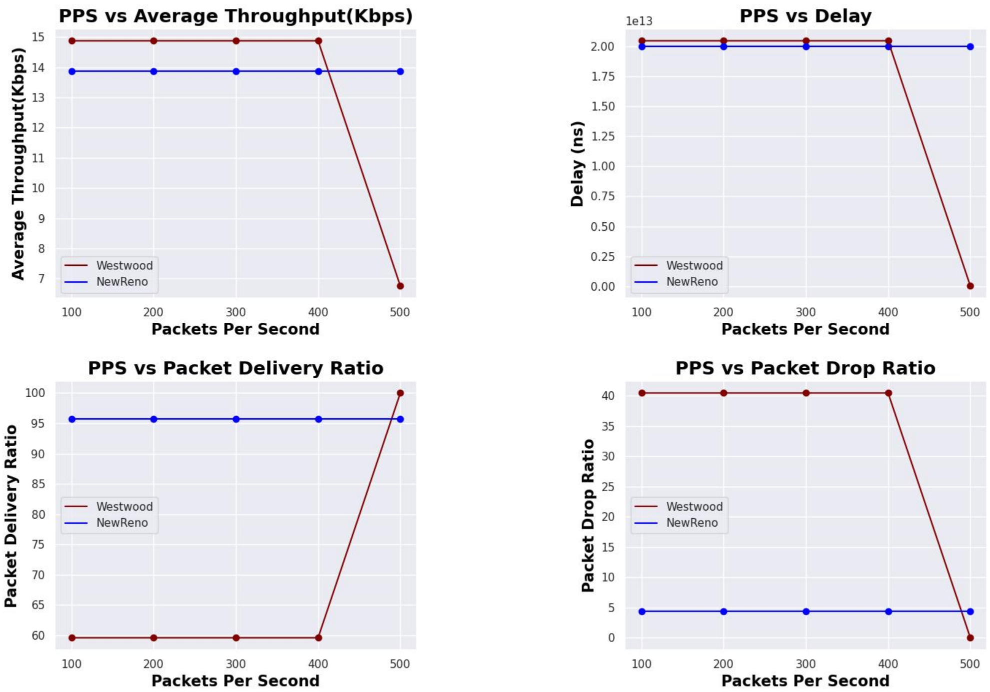


Figure 8: Packets Per Second vs Metrics

6.2 Task A2 - Wireless Low Rate Static Topology

Figure 9 describes the performance metrics of two congestion control algorithms - TcpWestwood and TcpNewReno against varied number of flows in each simulation. Here number of nodes = 40 and packets per second = 400.

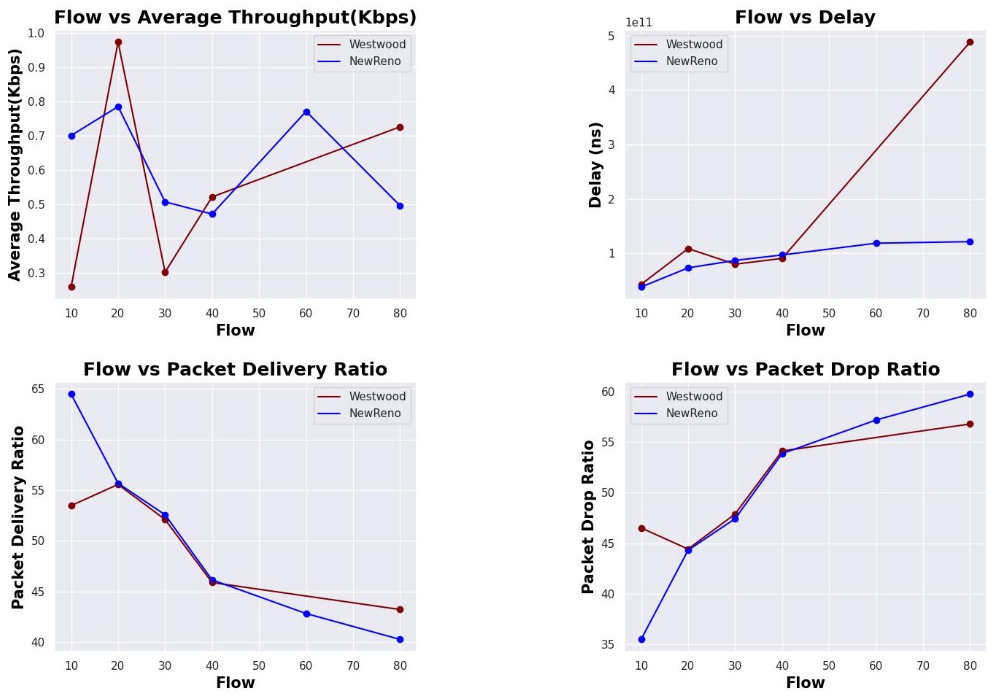


Figure 9: Flow vs Metrics

Figure 10 describes the performance metrics against varied number of nodes in each simulation. Here number of flows = 80 and packets per second = 400.

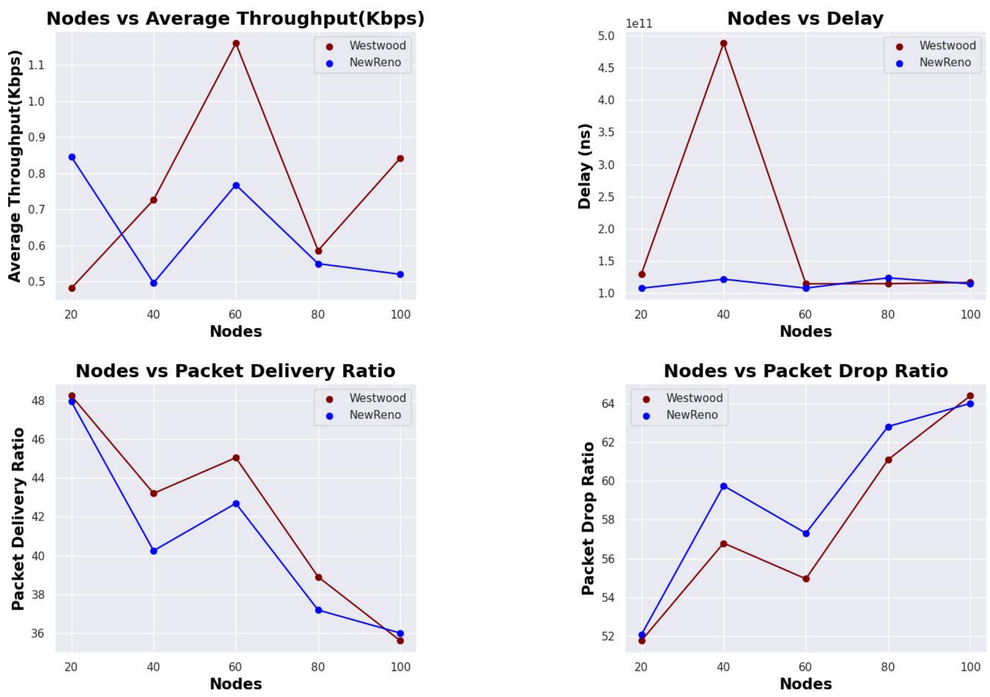


Figure 10: Nodes vs Metrics

Figure 11 describes the performance metrics against varied number of packets per second in each simulation. Here number of flows = 80 and nodes = 40.

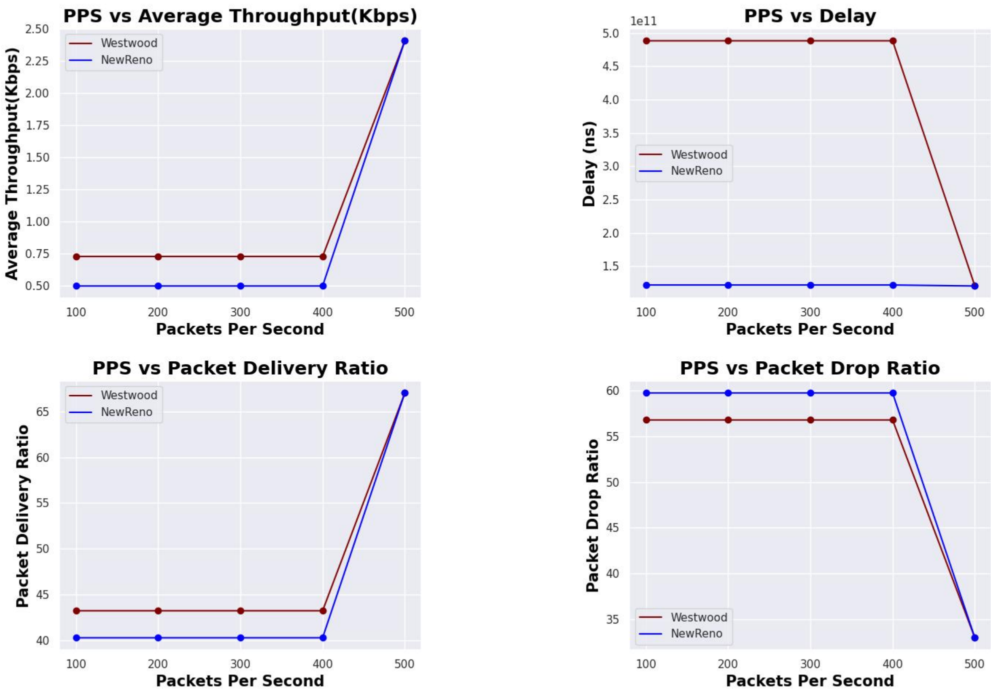


Figure 11: Packets Per Second vs Metrics

Figure 12 describes the performance metrics against varied coverage area in each simulation. Here number of flows = 80, nodes = 40, packets per second = 400 and Max range = 80 meters.

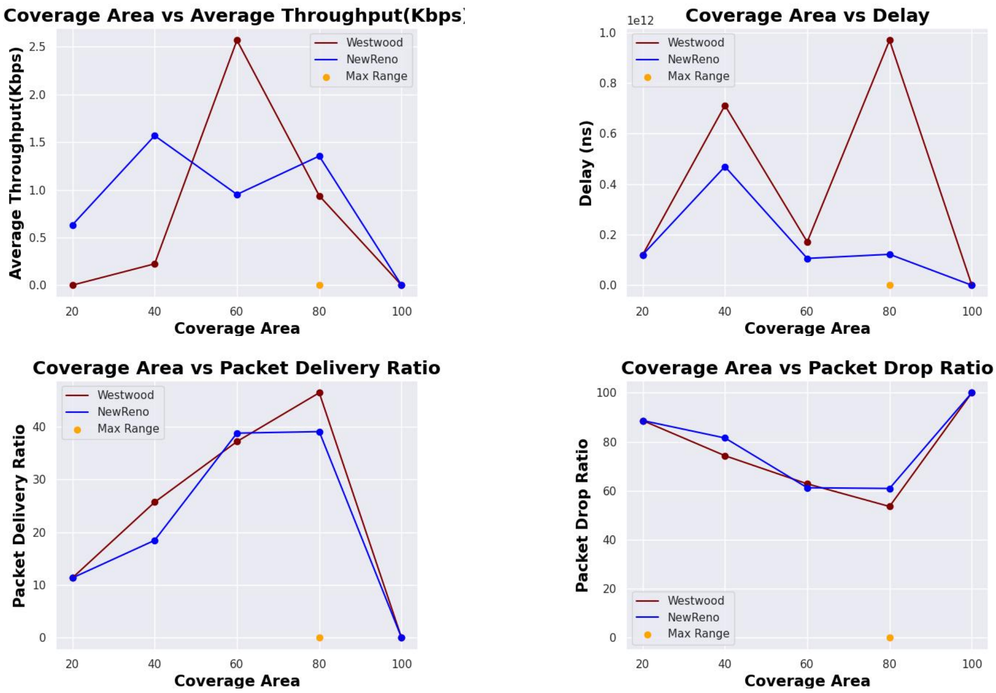


Figure 12: Coverage Area vs Metrics

6.3 Task B - Modified Algorithm

6.3.1 Required Two Metrics

Figure 13 and 14 describes the performance metrics implemented algorithm TcpSwift and standard algorithm DCTCP. In [2] it was claimed that Swift achieves loss rates that are atleast 10x less than that of DCTCP while maintaining equal throughput. So we look at the metrics packet drop ratio and average throughput by varying different parameters.

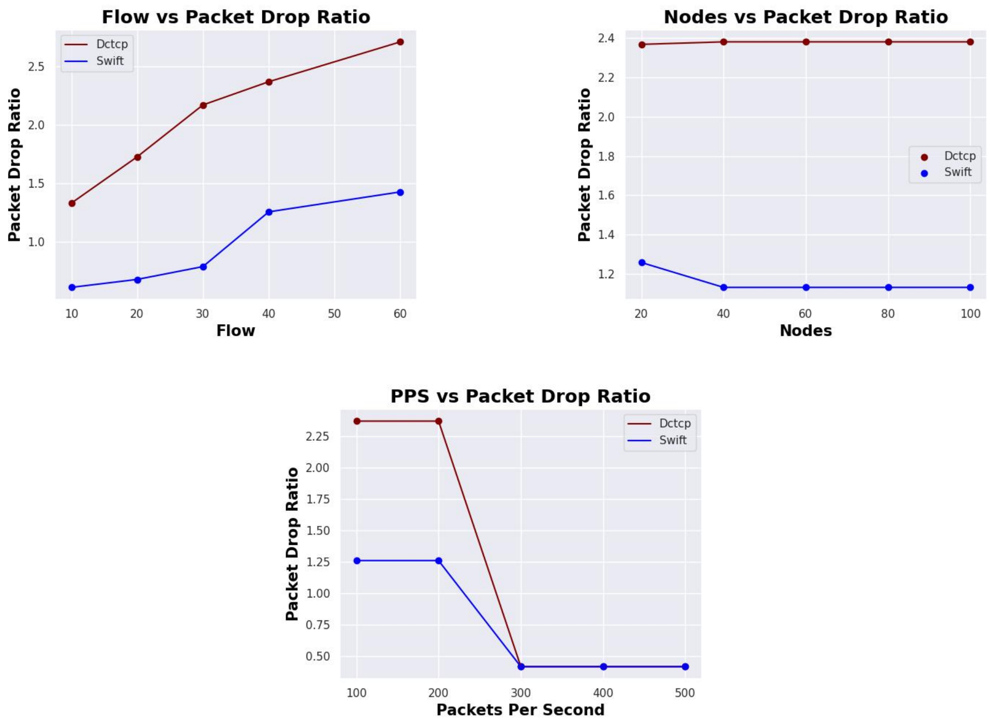


Figure 13: Parameters vs Packet Drop Ratio

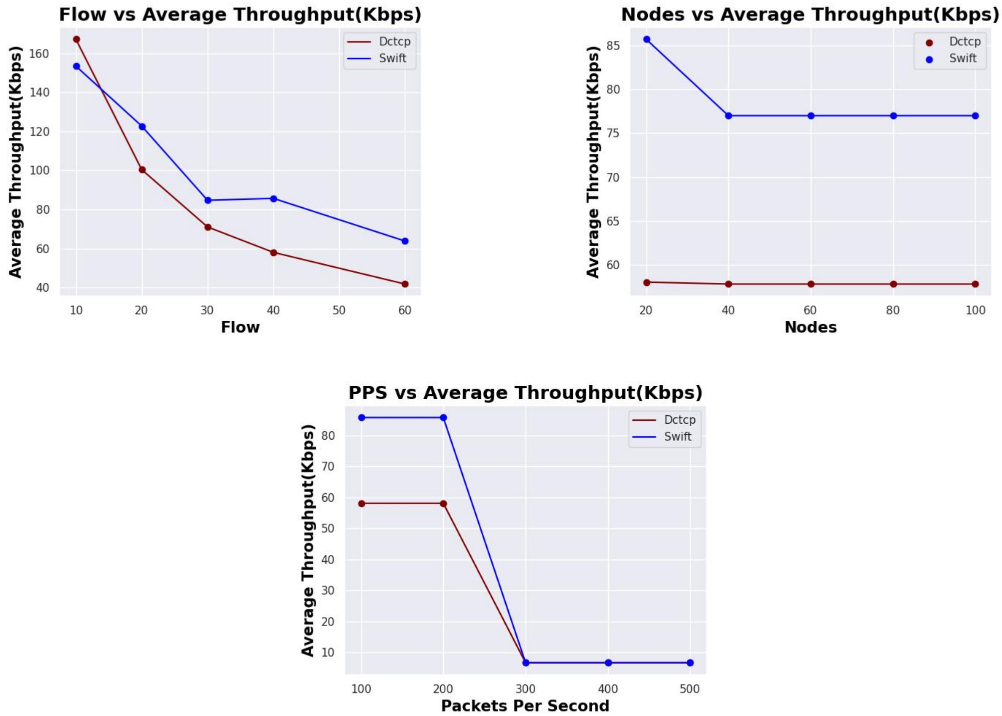


Figure 14: Parameters vs Packet Drop Ratio

6.3.2 Additional Study

The following plot shows the congestion window of Swift and Dctcp. This plot implies the correctness in implementation of Swift algorithm.

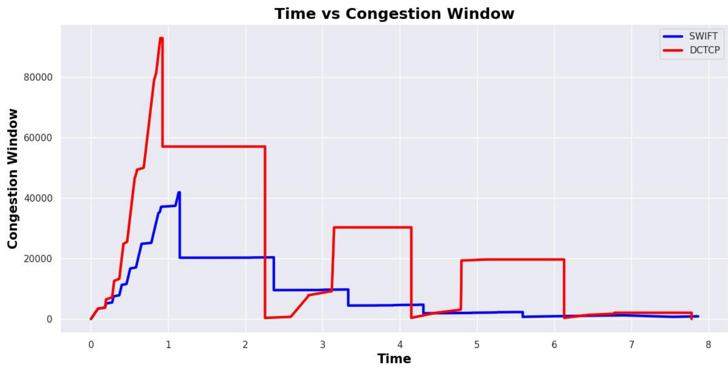


Figure 15: Comparison of congestion window of TcpDctcp and TcpSwift

Figure 16 shows fairness achieved by Swift. Here the figure in the left has been taken from (Kumar et al., 2020) [2]. Since simulating Gbps level throughput is unachievable by point to point link of NS3, the throughput has been shown in Kbps range.

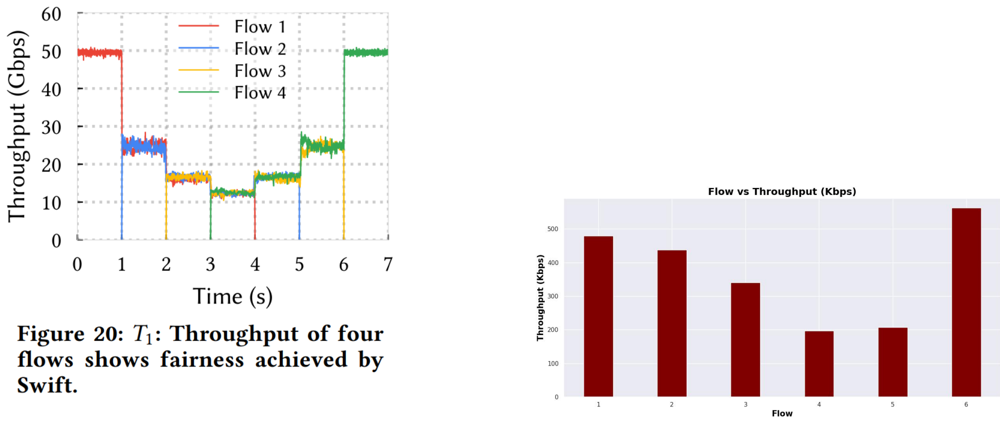
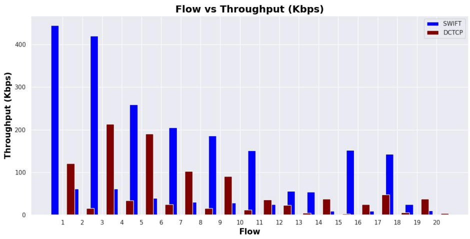


Figure 16: Throughput of individual flows of Swift

Following figure shows a comparison between throughput and lost packets of TcpDctcp and TcpSwift.



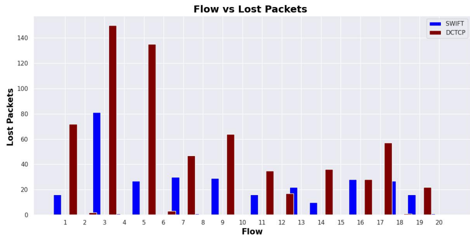


Figure 17: Comparison of performance of TcpDctcp and TcpSwift by individual flows

7 Summary Findings

7.1 Task A1 - Wired

	Throughput	End to End Delay	Packet Delivery Ratio	Packet Drop Ratio
Flows	If we increase number of flows while keeping number of nodes the same, the throughput of the network decreases. Here the performances of both algorithms are quite similar.	End to end delay increases as we increase flows. The delay of Westwood is greater than NewReno.	The packet delivery ratio is much higher in NewReno than Westwood. So, NewReno is more reliable for wired networks. Both values decrease as we increase flow.	Packet Drop Ratio is also much higher in TcpWestwood. Drop ratio also increases as we increase flow.
Nodes	Increasing number of nodes does not affect the performance of the network very much, as there is a bottleneck and we are not increasing number of flows. Here we can see that the throughput of Westwood is actually higher than New Reno.			
Packets Per Second	**As we increase packets per second (PPS),the performance does not change as the congestion control algorithms are adjusting the congestion window according to the data rate. Here, we can consider the value 500 PPS to be an outlier for TcpWestwood. Further simulations showed that if packets per second is increased beyond 500 (e.g 600), the number of sent packets decreases significantly for Westwood. Very few packets are actually sent so the packet loss ratio is very low, so is the throughput.**			

7.2 Task A2 - Wireless

Analysis on wireless topology revealed that Westwood performs better in wireless networks that wired networks. Further analysis is described on table 1

	Throughput	End to End Delay	Packet Delivery Ratio	Packet Drop Ratio
Flows	Average throughput is low for low-rate devices. The values are very similar even if flows are increased. They fluctuate in a range between 0.3-1.0.	End to end delay increases as we increase flows. The delay of Westwood is greater than NewReno.	The packet delivery ratio is similar in NewReno and Westwood. Both values decrease as we increase flow.	Drop ratio also increases as we increase flow.
Nodes	Average throughput is low for low-rate devices. The values are very similar even if nodes are increased. They fluctuate in a range between 0.5-1.2.	Delay is similar in both Westwood and NewReno. Nodes=40 is an outlier.	The packet delivery ratio is a little higher in Westwood. Both values decrease as we increase nodes.	The packet drop ratio is a little higher in NewReno. Both values decrease as we increase nodes.
Packets Per Second	As we increase packets per second (PPS),the performance does not change as the congestion control algorithms are adjusting the congestion window according to the data rate. Here, we can consider the value 500 PPS to be an outlier. Further simulations showed that if packets per second is increased beyond 500 (e.g 600), the number of sent packets decreases significantly. Very few packets are actually sent so the packet loss ratio is very low, so is the throughput.			
Coverage Area	Throughput and delay is similar.Packet delivery ratio increases and drop ratio decreases as we increase the coverage area. Since max transmission range is 80 m, increasing area beyond that ceases all transmission.			

Table 1: Analysis on results of Task A2

7.3 Task B - Modified Algorithm

The plots generated clearly shows that TcpSwift achieves much lower packet loss rates while maintaining similar throughput to TcpDctcp, validating the claim made in the paper. The exact results described in the paper could not be reproduced because of missing data of various parameters of the paper. More time and fine tuning may result in much better performance of the algorithm. Another issue was the limitation of point to point connections of NS3 in simulating Gbps level data rate. Because of that the exact scenario of a data center could not be depicted in the simulation.

References

[1] Alizadeh, M., Greenberg, A., Maltz, D., Padhye, J., Patel, P., Prabhakar, B., Sengupta, S. and Sridharan, M., 2010. *Data center TCP (DCTCP). ACM SIGCOMM Computer Communication Review, 40(4), pp.63-74.*

[2] Kumar, G., Dukkipati, N., Jang, K., Wassel, H., Wu, X., Montazeri, B., Wang, Y., Springborn, K., Alfeld, C., Ryan, M., Wetherall, D. and Vahdat, A., 2020. *Swift. Proceedings of the Annual conference of the ACM Special Interest Group on Data Communication on the applications, technologies, architectures, and protocols for computer communication,.*