Dynamic Receiver Buffer Management Scheduler for Multipath TCP in mmWave Networks

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1 Motivation and Abstract

In this independent study we tried to study how MPTCP (Multipath Transmission Control Protocol) is going to impact the throughput and latency in mmWave networks. This is continuation of our effort from ECE/CSC 773 research paper where we evaluated performance of variants of TCP in mmWave networks. We were tasked to study the issues and challenges posed by current variants of Transport layer Protocol and if possible develop new Transport layer protocol for upcoming 5G (mmwave networks). Next generation applications will not only require high throughput but also low latency.

TCP and its variants were found to perform poorly over cellular networks due to high capacity variability, self-inflicted queuing delays, stochastic packet losses unrelated to congestion, and large bandwidth-delay products.

Our study also found that one of the key features of 5G [3GPP.23.501] is dual connectivity (DC). The Internet Engineering Task Force has proposed the Multipath TCP by utilizing the feature of dual connectivity in 5G, where a 5G device can be served by two different base stations as shown in Figure 1. DC may play an essential role in leveraging the benefit of 5G new radio, especially in the evolving architecture with the coexistence of 4G and 5G radios.

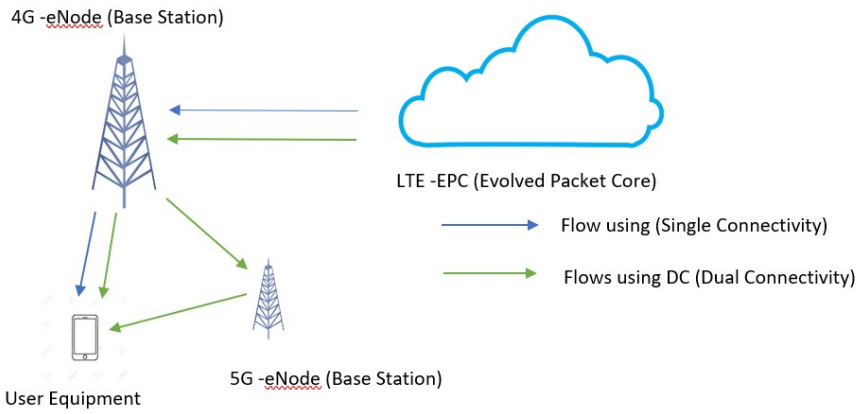

Figure 1: Use Case of Dual Connectivity in 5G Networks

On a 5G device with DC, an application can send data to the destination (e.g., a single-home server) through multiple radio links, over one or more PDU sessions. This all could be achieved by MPTCP. MPTCP's main rationale is that using multiple sub-flows will increase the throughput and reliability; the protocol is now implemented in most major operating systems and used in data centers and wireless networks.

However, it has it's own share of problems. More specifically, the packet with the lower sequence number arrives at the receiver later than the packet with the higher sequence number. The receiver has to buffer the packets with the lower sequence number. Until it receives all the packets whose sequence number is lower than the higher sequence number, the data can be submitted to the upper layer. The complex interactions that exist between the reliability of the different paths and the different congestion control algorithms require a deep understanding of MP-TCP performance before deploying it on mmWave links. Also [9] demonstrated the impact of mmWave

links on the performance of the transport layer and pointed out that reducing latency while maintaining a high throughput is a challenging issue for 5G networks. Current MPTCP standard falls short of this objective. Lately there have been many works which tries to address this problem. Some has proposed new window control algorithms (congestion control) like LIA (Linked Increase Algorithm)[10] , OLIA(Opportunistic Linked-Increases Algorithm) [3], BALIA (Balanced Linked Adaptation) [8]while others have tried to propose new scheduling algorithm so that packet reordering time is reduced to minimum, thus thereby reducing overall latency. Most relevant scheduling algorithms are covered in Related Work section.

Reducing packet reordering also solves problem of buffer bloat whereby large buffers are employed to store out of order packets received from other sub-flows which has better channel characteristics and low RTT (Round Trip Time). Having studied recently proposed algorithms and improvements we find that concept of dividing the receiver buffer logically into number of sub flows in proportion to their capacity.the problem of latency due to reordering of packets in receive buffer can be addressed. We have tested this approach on NS-3 and results are promising. We also feel that this approach could be implemented on top other recently proposed techniques to have additive effect in further optimizing the MPTCP.

2 MPTCP Overview

Before we move further this section will briefly explain the working of MPTCP. It is important to have basic understanding of MPTCP so as to figure out one important shortcoming in the proposed MPTCP RFC and our solution to it in proposed algorithm in Secion 4 . As an extension of the TCP protocol, MPTCP supports simultaneous transmission of data through multiple paths, i.e. multiple TCP flows by using multiple network interfaces. As shown in Fig the MPTCP is located between the application layer and the network layer, and can be further divided into an MPTCP layer and a TCP sub-flow layer.

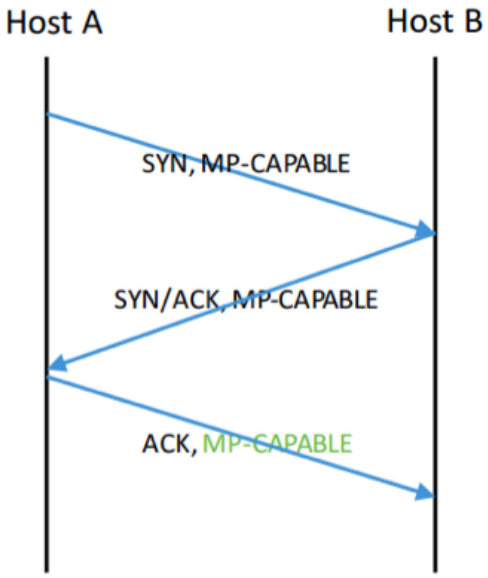

Figure 2: MPTCP Connection Set Up

The life cycle of an MPTCP connection includes three phases, namely initial connection, data transfer, and closing connection. In the initial connection phase, the difference between MPTCP and TCP is that MPTCP has a four-way handshake before the multipath is enabled. MPTCP and TCP have similar three-way handshake. But the SYN, SYN/ACK and ACK packets have the MPTCP_CAPABLE option. After the TCP connection has been established, the client can advertise other addresses by sending a TCP segment with the ADD_ADDR option.

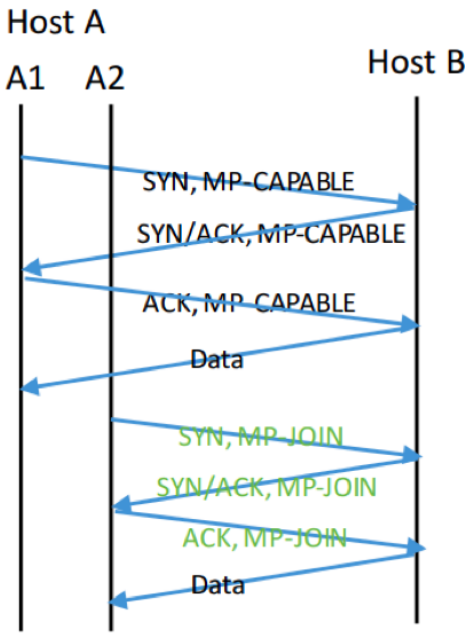

Figure 3: MPTCP Subflow

During the MPTCP data transfer phase, the subfows are linked together by a single multipath TCP connection, and both can transmit data. To ensure the reliability of the orderly transfer of data on subfow, MPTCP uses two principles. First, each subfow is equivalent to a regular TCP connection, and with its own 32-bit sequence number space. Second, MPTCP maintains a 64-bit data sequence number space. The DSN_MAP and DSN_ACK options use these data sequence numbers.

MPTCP's FIN semantics also allow sub-flows to be closed cleanly while allowing the connection to continue on other sub flows. Finally, MPTCP provides a REMOVE ADDR message, allowing one sub-flow to indicate that other sub-flows using the specified address are closed. This is necessary to cleanly cope with mobility when a host loses the ability to send from an address and so cannot send a sub-flow FIN.

It is also important to understand how acknowledgement works in case of MPTCP especially when data is being sent through heterogeneous subflows. Since it is difficult to visualize by reading the RFC, refer following for better understanding.

First we will quote the from the MPTCP RFC verbatim and then explain it further as this concept is very fundamental to operation of MPTCP and is heavily used in all the modifications

Standard says **The "Data Sequence Signal" carries the "Data Sequence Mapping". The data sequence mapping consists of the subflow sequence number, data sequence number, and length for which this mapping is valid. This option can also carry a connection-level acknowledgment (the "Data ACK") for the received DSN. With MPTCP, all subflows share the same receive buffer and advertise the same receive window. There are two levels of acknowledgment in MPTCP. Regular TCP acknowledgments are used on each subflow to acknowledge the reception of the segments sent over the subflow independently of their DSN. In addition, there are connection-level acknowledgments for the data sequence space. These acknowledgments track the advancement of the bytestream and slide the receiving window.**

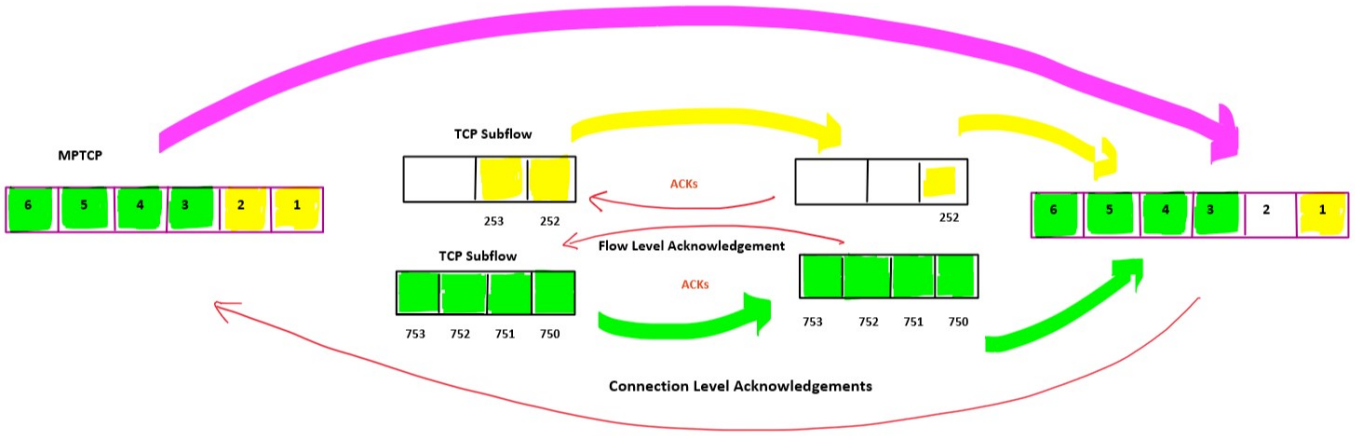

Figure 4: MPTCP Subflow

In the above figure we can clearly two types of ACKs. One is flow level acknowledgments which pertains to individual subflow and is just like normal TCP ACKs. On top of it there are connection level ACKs as well.

Connection level ACKs are used in determining number of unordered data at the connection level for $subflow_i$. For example in figure 4, corresponding to yellow flow since Packet with Data Sequence Number 2 is not acknowledged at the connection level.

Flow level acknowledgements are as specified in standard are normal TCP ACKs at each subflow level and just like in any TCP flow control used to determine the amount of send data once the Send Window is calculated for each subflow.

3 Introduction to proposed work

For the sake of brevity, complete MPTCP operation is not mentioned here as of now.

In the original MPTCP, receive buffer is shared, and receiver notifies the overall buffer ($rwnd$), which controls the data flow of the total MPTCP connection. Following lines are directly quoted from the latest MPTCP standard [1]

"With MPTCP, all subflows share the same receive buffer and advertise the same receive window"

Reasoning given behind it is that maintaining per subflow receive windows, could end up stalling some subflows while others would not use up their $rwnd$ window. However, if we divide the receiver buffer window in accordance with some parameter such that a subflow which has better channel characteristics and low RTT, thereby having low chance of packet loss

In proposed scheduler the sender first tries to estimate the occupancy of total receiver buffer by each subflow which will be dependent on the subflow's congestion window and RTT So let us assume that $acwnd_i$ is the short time average congestion window of $subflow_i$ Round Trip Time (RTT)of $subflow_i$ RTT_i

Estimated Average Buffer Occupancy of flow Buf_i

Allocation of receiver Buffer $RecvB_i$

The total receiver buffer will be allocated to each sub-flow in proportion to their maximum Buffer Occupancy.

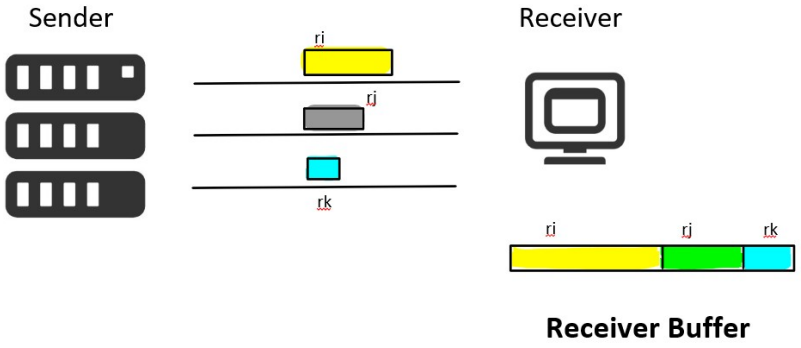

Figure 5: Receiver Buffer Division Approach

Each subflow will obtain individual receive window $rwnd_i$ In earlier approaches $rwnd_i$ is the receiver window advertised to all the flows. The send window of $subflow_i$ will be now restricted by congestion window and receive window calculated as per each flow rather than total receive window

For the sake mathematical calculation let us assume two $subflows_0$ and $subflow_1$. RTT_0 is less than RTT_1 refer Figure 3. If packets from $subflows_0$ are transmitted earlier than $subflows_1$, then out of order packets in receive buffer $= \left\lceil \frac{RTT_0}{RTT_1} \right\rceil \cdot acwnd_0$

otherwise Out of Order packets $= (\left\lceil \frac{RTT_0}{RTT_1} \right\rceil - 1) \cdot acwnd_0$

If we assume both the scenarios are equally probable, then Out of order packets $= (\left\lceil \frac{RTT_0}{RTT_1} \right\rceil - 1/2) \cdot acwnd_0$

Above reasoning can be easily extended to multiple flows for MPTCP. Here average buffer occupancy of $subflow_i$ becomes

$$Buf_i = (\left\lceil max_j \neq i \frac{RTT_j}{RTT_i} \right\rceil - 1/2) \cdot acwnd_i$$

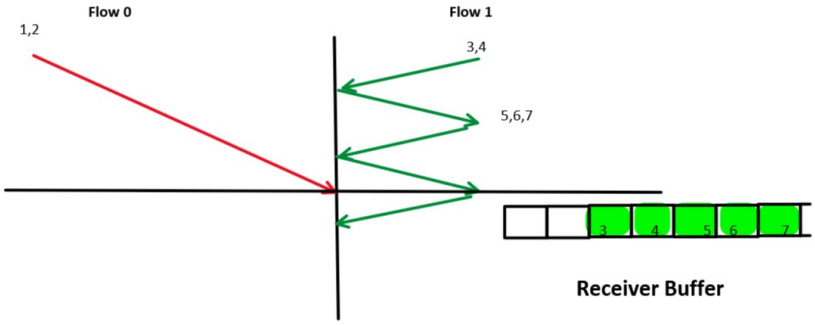

Figure 6: Out of Order packet

The receiver buffer will be distributed among each subflow in proportion to the

$$RecvB_i = recvBuffer \cdot \frac{Buf_i}{\sum_{i=0}^{N-1} Buf_i}$$

$recvBuffer$ is the size of total receiver buffer

Receiver window $rwnd$ will be now distributed to each subflow as $rwnd_i$

$$rwnd_i = rwnd \cdot \frac{RecvB_i - unordered_i}{\sum_{B_i > unordered_i}(RecvB_i - unordered_i)}$$

Here $unordered_i$ is the unACKed data at the connection level as described in Section 2

$$rwnd_i = \begin{cases} 0, & RecvB_i < unordered_i \\ rwnd \cdot \frac{RecvB_i - unordered_i}{\sum_{B_i > unordered_i}(RecvB_i - unordered_i)}, & else \end{cases}$$

Once the receiver buffer window is distributed among each subflow The send window of each sublow $SendWindow_i$ slides as per the congestion window $acwnd_i$ and receiver window $acwnd_i$ calculated by algorithm which is typical of TCP flow and congestion control.

$$SendWindow_i = min(cwnd_i, rwnd_i)$$

The final data which is send is send window of subflow minus the pending unACKed data.

$$SendData_i = SendWindow_i - Outstanding_i$$

where $Outstanding_i$ is unACKed data at flow level as decribed in section 2.

As we can see the algorithm does not tempers with the fundamental operation of TCP, hence we believe that it can be easily ported to other schemes which tends to optimize MPTCP

4 Algorithm Pseudo-code

Algorithm 1: Receiver Buffer Division Algorithm

Input:
Receiver window: $rwnd_i$;
Congestion Window of $subflow_i$;
Output:
New Data to be send on $subflow_i$;
for $i \leftarrow 0$ **to** $N-1$ **do**
  $Buf_i = (\left\lceil max_j \neq i \frac{RTT_j}{RTT_i} \right\rceil - 1/2) \cdot acwnd_i$
end
for $i \leftarrow 0$ **to** $N-1$ **do**
  $RecvB_i = recvBuffer \cdot \frac{Buf_i}{\sum_{i=0}^{N-1} Buf_i}$;
end
if $RecvB_i < unordered_i$ **then**
  $rwnd_i = 0$;
else
  $rwnd \cdot \frac{RecvB_i - unordered_i}{\sum_{B_i > unordered_i}(RecvB_i - unordered_i)}$;
end
$SendWindow_i = min(cwnd_i, rwnd_i)$ $SendData_i = SendWindow_i - Outstanding_i$

Above is the pseudo-code of our algorithm.

5 Related Work

In this section we have tabulated the related work in the field.

5.1 DAPS - Intelligent Delay-Aware Packet Scheduling For Multipath Transport [5]

The main idea behind packet/path scheduling is not to transmit packets in monotonically increasing sequences on any available path, but to carefully choose which packet should be sent when and over which path, based on some characteristic of the associated path. The crux of the algorithm is that the sequence number of the packets transmitted on the slow path is determined depending on the ratio between the two RTTS (RTT of slowpath/ RTT of faster path). n ps. DAPS does not explicitly consider how much storage the receiver has available while scheduling the packets on different subflow. Out-of-order Transmission for In-order Arrival Scheduling for Multipath TCP is similar to DAPS. Whenever MPTCP sender wants to send data, it has to make three decisions 1. Which subflows must be used to send data 2. After selecting a subflow how much data should be sent in

5.2 FPDPS - Fine-grained Forward Prediction based Dynamic Packet Scheduling [11]

Fine-grained Forward Prediction based Dynamic Packet Scheduling (FPDPS). It allocates a number of packets to each under-scheduling TCP subflow by estimating the number of packets that will be transmitted on all other TCP subflows simultaneously. The estimation is done by utilizing TCP characteristics of each TCP subflow within a MPTCP connection. It adopts the idea of TCP modelling and takes account of packet loss. FPDPS further enhances above algorithm using Dynamic Adjustment with the Feedback of SACK (DAF) to make an offset to eliminate the previous scheduling deviation for this round of scheduling. The sender gets feedback information over SACK options, which can help the sender distinguish the scheduling deviation in the last round, and then the sender modifies the value in the next round accordingly. By this way, it will further eliminate the accumulated estimation error brought out by the not-entirely-accurate FPDPS.

5.3 DEFT - Delay Equalized Fast Transmission [6]

The basic problem which this paper seeks to solve is current MPTCP CCAs do not consider the additional reordering delay resulting from the path heterogeneity of MPTCP. Different paths have different round-trip time (RTTs), causing packets to receive out of order. DEFT considers the characteristics of 5G networks and aims to guarantee fast responsiveness of each sub-flow by maintaining a certain number of backlogged packets. They have analytically proved that RTTs of all paths should be equalized to minimize the reordering delay. Since delay-based schemes can control the queuing delay of each subflow to their equilibrium point. DEFT aims to find the optimal number of backlogged packets to equalize the RTT values. Hence, DEFT can be summarized as a window control scheme along with a delay-equalizing scheme. The window control scheme updates the *cwnd* of the routes to achieve fast responsiveness and fairness. On the other hand, the delay-equalizing scheme updates the protocol-dependent parameter α_r to minimize the application-level E2E delay.

6 Experimental Setup and Simulations

Having identified one shortcoming in MPTCP and proposed an algorithm to solve it, now we describe the learnings we had while trying to simulate it. There are primarily two approaches and are hereby discussed.

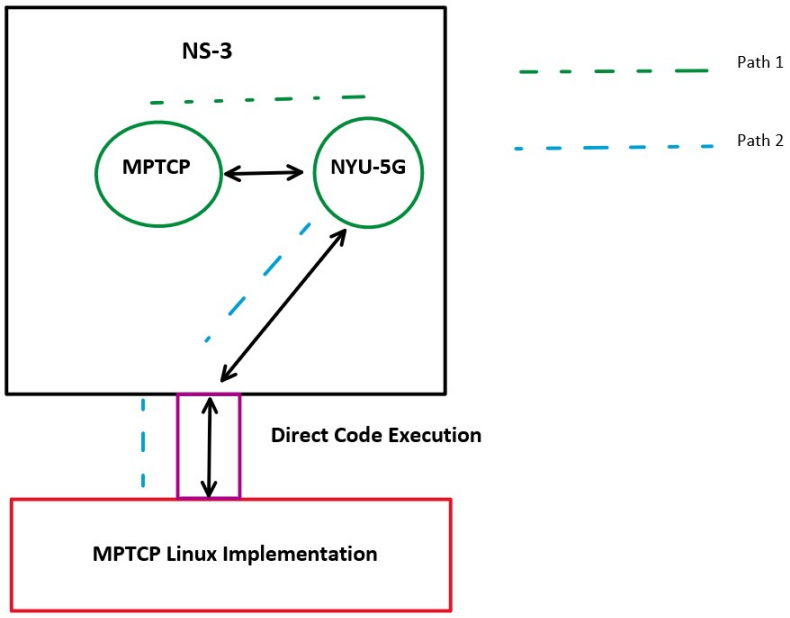

Figure 7: MPTCP Connection Set Up

6.1 Using NS-3 with extension of NYU mmWave module and MPTCP implemented in Linux Kernel

This approached is taken by many research group to simulate MPTCP vis a vis mmWave networks as shown by dotted line Path 2. Other approach is shown by dotted lines of Path 1 which will be discussed next.

The simulations were based on ns-3 with the extension of the NYU mmWave module v1.2 [36], where the Linux kernel implementation including MPTCP v0.92 is plugged in using net-next-nuse and Direct Code Execution (DCE). Specifically, the ns-3 module in dce-linux was replaced with the NYU mmWave module v1.2, and net-next-nuse was added to support the merge with the MPTCP Linux kernel v0.92.The DCE ns-3 module provides facilities to execute within ns-3 existing implementations of userspace and kernelspace network protocols. Hence in our case to implement the proposed algorithm, we have to modify the kernel implementation of MPTCP and using DCE have to integrate with NYU mmWave module

6.2 Creating MPTCP support in NS-3 and then integrating it with NYU mmWave module

Another approach is shown by green dotted line (path 1). Here MPTCP module is to be directly connected with NYU mmWave module. As of now MPTCP is not officially supported by NS-3 but time to time few people or research groups have come out with their own implementation and also published papers about it and made their source code open for further research. Our study found three such implementations and they will be now discussed in chronological order

In first implementation [4] author creates MPTCP functionality by creating MpTcpSocketBase Class which inherits from TcpSocketBase Class. MPTCP socket provides an interface between the application and TCP flows. The MpTcpSocketBase object controls path management, scheduling, packet reordering and congestion control algorithms, etc. Another class MpTcpSubflow defines TCP subflows that communicate with the network layer. This implementation is oldest of all and other two extends its feature and add many new ones. Hence there is little incentive to discuss and work on it.

The second implementation [2] is a major improvement from the previous one. Paper shows a well-defined state diagram (not shown here for sake of brevity) and many things which are part of MPTCP RFC are implemented.

The MPTCP connection starts with a TCP socket and the client sends a SYN + MP CAPABLE option with its key. If SYN is MPTCP capable then server upgrades socket to MPTCP socket and replies with SYN/ACK + MP CAPABLE option along with its server key. The client upon receiving SYN/ACK upgrades to a MPTCP socket and replies with an ACK completing the 3-way MPTCP handshake. This MPTCP socket provides an interface between the application and TCP subflows for packet scheduling, reordering and retransmission, etc. Round robin and fastest RTT schedulers are implemented and divide the application byte stream into segments for transmission on established subflows.

The various feature in MPTCP implementation [2] is described in brief in following figure which taken directly from the paper.

Table 2
List of supported and missing features

SHA1 support	We added an optional SHA1 support in ns3 to generate valid MPTCP tokens and initial DSNs. This allows to communicate with a real stack and also proved necessary for wireshark to be able to analyze the communication.
Scheduling	The fastest RTT and round robin schedulers are available.
Congestion control	Subflows can be configured to run TCP ones such as NewReno or LIA.
Mappings	As in the standard, data is kept in-buffer as long as the full mapping is received. This is necessary when checksums are used, otherwise this can be disabled to forward the data faster.
Subflow handling	It is done directly by the application that can choose to advertise/remove/initiate/close a subflow at anytime if it is permitted by the protocol.
Packet (de)serialization	Packets generated along with MPTCP options can be read/written to a wire, allowing an ns3 MPTCP stack to interact with other MPTCP stacks, such as a linux one.
Fallback	If the server does not answer with an MP_CAPABLE option, the client falls back to legacy TCP. Other failures are not handled, e.g., infinite mapping or MP_FAIL handling as simulating these features is of little interest.
Buffer space	Buffer space is not shared between subflows, data is replicated between the subflow and the meta send/receive buffers rather than moved.
Path management	We drifted away from the specifications in order to be able to identify a subflow specifically, i.e., we associate a subflow id to the combination of the IP and the TCP port. Nevertheless the implementation is modular so it is possible to replace the subflow id allocation with a standard scheme.

Figure 8: Features part of MPTCP Implementation

However there few shortcomings of this implementation and some are worth noteworthy. This implementation does not implement path management which is a core MPTCP function. Although there are three classes for MPTCP congestion control i.e., MpTcpCongestionCoupled, MpTcpCCOlia and MpTcpCCUncoupled, but they are incomplete and nonfunctional. NS-3 is a continuously evolving project wherein each new version potentially introduces new models and classes as well as modifies existing models and classes. NS-3.25 refactored TCP removing, modifying, and appending

some TCP classes, functions, and variables. The TcpSocketBase class has had major changes over the evolution of ns-3. The class no longer handles congestion control and new congestion control classes that are subclasses of the TcpCongestionOps class have been introduced.

All the above changes in TCP classes makes [2] incompatible with the current NS-3 stack. We have tested it and it is not even compiling. The support for this version is not available has been confirmed by author in communication with us.

Hence to test the our algorithm we were left with only last implementation [7]. Authors say they have taken the base code and implemented some missing functionality critical of which is Path Management. They implemented a path management component to initiate subflows for three path managers default, ndiffPorts and fullmesh and made appropriate changes to the TcpL4Protocol, TcpRxBuffer and TcpSocketBase classes to make existing MPTCP code compatible with ns-3-dev.

We are simulating and testing our MPTCP scheduler in source code provided by above implementation of MPTCP in Network Simulator-3. We will be taking base implementation of MPTCP and adding our simulator alongside with it as shown in figure below. Highlighted scheduler is our implementation.

While simulating our scheduler in [7] set up, we found that certain files necessary for end to end simulation were missing. Repeated contact with authors were futile. So in the end we have decided to first refactor the [7] code and then add our module. This unforeseen issue has increased the work and digressed us from our main objective of adding a new scheduler to refactoring whole MPTCP patch. But I feel these added efforts will help us understanding MPTCP even more and once published will be much more appreciated by NS-3 community.

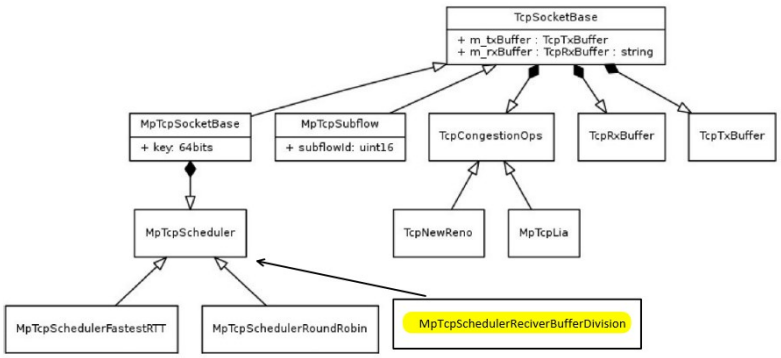

Figure 9: Receiver Buffer Scheduler

7 Conclusion and Future Work

This independent study has helped understand MPTCP in detail along with current trends in optimizing it. As mmwave networks are going to be part of our daily life, we will need connectivity through multiple interfaces. We study what approach various research groups have taken in past and after studying MPTCP standard came out with an approach and algorithm which can further optimizes the operation of MPTCP vis a vis packet reordering delay. Also this algorithm can be applied over existing optimizations thus having synergistic effect. This however needs to be thoroughly tested and can be part of future work.

We thought of testing this algorithm in NS-3 but got stuck as previous implementation MPTCP of NS-3 were no longer supported. Efforts to reach out to some of the scholars for clarification were also not successful although we are in touch of couple of them. Few who replied back said they have stopped supporting their MPTCP patch. Hence, now it seems our task was not confined to implement a new scheduler but whole of MPTCP. Since the independent study has to be completed within a short span of single semester, we feel that rigorous testing based upon simulation can be done in near future. I personally would be working on the project post completion of this semester. Any MPTCP simulator with interface to attach with mmwave module is going to be very helpful for upcoming research in mmwave communications.

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