eMPTCP: Towards High Performance Multipath Data Transmission By Leveraging SDN

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***Abstract*—Motivated by the poor performance of MPTCP when used for bulk transfers in multipathed networks, in this paper we propose an efficient MPTCP protocol variant named eMPTCP aiming to achieve high throughput, low latency and good bottleneck fairness. The novel MPTCP variant eMPTCP can prevent in-network buffer overflow and packet loss using a distributed and reactive approach for bandwidth allocation and adjusting the congestion window of subflows on multiple paths in a coordinated fashion. It employs ECN feedback and latency (in terms of RTT) to modulate the congestion window via a gamma-correction function. This novel congestion- and latency-aware window adjustment mechanism behaves very helpful in handling the traffic bursts, easing the buffer pressure on switches, and greatly mitigates the incast issue. Besides, in this solution SDN is leveraged to compute a set of optimal available routes for subflows and actively adjust the number of subflows of each MPTCP flow according to the instantaneous traffic condition. Moreover, this novel MPTCP protocol variant eMPTCP works well with existing switch hardware and is able to coexist with legacy TCP. It ensures that a multipath flow will not take up more capacity on any shared paths than if it was a single path TCP flow using only one of those paths, which guarantees it will not unduly harm other flows. Simulation results show that eMPTCP achieves smaller MPTCP convergence time, higher aggregate throughput and lower flow completion time than existing legacy competitors, and largely improves the application performance and user experience, making the network more robust and faster.**

I. Introduction

Nowadays the multihomed devices (e.g., smartphones, tablets) are enabled with both cellular (e.g. 3G, 4G) and 802.11 (WiFi) access, and many computers or servers (e.g., in data centers) are equipped with several network interface cards(NICs) connecting to multiple network devices. This has significantly motivated the development of Multi-path TCP (MPTCP), which aims to simultaneously use several access mediums to spread flows on different paths. Currently, MPTCP has been standardized in IETF RFC [1]. Figure 1 depicts the comparison of standard TCP and MPTCP protocol stacks. Admittedly, MPTCP greatly improves the application performance in terms of end-to-end throughput by using multiple paths transparently. However, the current existing MPTCP solutions have various limitations, which cause many critical issues, such as path collision, bottleneck fairness, incast [2], resource wastes, inefficient congestion control, and so on [3]. There is large room for improvement on MPTCP solutions in high speed networks.

All existing MPTCP algorithms adopt TCPs additive-increase and multiplicative-decrease (AIMD) fashion to adjust window sizes, where additive increase is executed when no

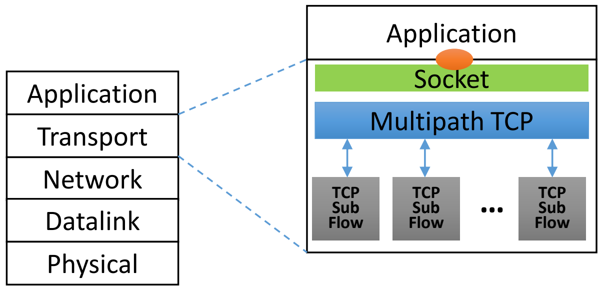

Fig. 1. Comparison of Standard TCP and MPTCP Protocol Stacks.

loss is detected, and multiplicative decrease is taken for packet loss. Take EWTCP [4] for example, the window-sizes of subflows are adjusted independently other than in a coordinated way, which can not make very efficient use of the network [5]. Correspondingly, some other MPTCP algorithms are proposed, such as Coupled MPTCP [6][7], Semicoupled MPTCP [8], Max MPTCP [8], aiming to adjust the window increasement granularity on multiple routes in a coordinated way [8]. However, all these existing algorithms fail to modulate the congestion window decreasement in case of packet loss, behaving latency agnostic. Besides, the revised window increasement mechanisms still have their own deficiencies, incurring low aggregate throughput, slow responsiveness (long convergence time), etc. The impairments of the current existing MPTCP solutions can be summarized as below:

1) Current existing MPTCP solutions usually determine a fixed number of subflows to use during flow connection establishment, and cannot dynamically adjust the number of subflows reacting to the actual network conditions. Admittedly, using more subflows can theoretically achieve higher throughput, however, it is not always the case more is the best [9]. On the contrary, the number of established subflows should adjust to the actual traffic condition. Specifically, to use a fixed number of subflows without adjustment may lead to the following potential deficiencies:
   - At the end host side, each started subflow creates and maintains its own sub-socket, which is associated to the meta-socket. Thus, more subflows mean more sockets to store and maintain, which cost more resources. In addition, each sub-socket maintained in the kernel requires additional system calls to manipulate it, which consumes more CPU capacities. Therefore, more subflows will cause more overhead

to the system resources.
- The subflows on different paths may experience different transmission time, which incurs packets out-of-order at the last hop. Thus, the receiver should maintain a large buffer to assemble them in the right order, which increases memory consumption. The number of subflows should be dymanically adjusted so as to achieve a higher application performance at a lower cost.

2) The multipath routings increase the possibility of incast issue, but the current existing congestion control mechanisms cannot handle this issue and are not congestion aware.
3) Most of existing solutions are latency (RTT) agnostic, and thus cannot efficiently choose best paths.
4) Window adjustment scheme issues:
   - Although the bottleneck fairness problem is considered in some solutions, they fail to take the limited buffer space in switches into consideration when adjusting window sizes; This leads to high application latencies, where bandwidth hungry large flows may build up queues at the switches and thus throttle the latency sensitive small flows.
   - The window decreasement is neither latency-aware nor congestion precautionary. In case of packet drops, the congestion window of each subflow is halved directly, which is not responsive to level of network congestions. This not only incurs an abrupt drop of throughput, but also leads to high convergence time.

Based on these careful investigations and observations, we propose a novel MPTCP variant protocol named as eMPTCP, and leverage SDN to help achieve better traffic engineering. Specifically, the goal is to take advantage of SDN [10] to implement optimal multipath routing calculations, dynamically adjust the number of subflows with computed best paths, and gracefully modulate the subflow windows in a coordinated fashion adapting to current traffic state (e.g., throughput, RTT, congestion level) so as to maximize the application performance that can be achieved, such as low latency, high throughput, faster reaction to congestions, and incast prevention.

The rest of the report is organized as follows. The related work is reviewed in Section II. Section III presents the proposed SDN-based eMPTCP solution, which is evaluated in Section IV. Finally, Section V concludes this paper.

II. Related Work

The current existing solutions can be divided into two categories: SDN-based MPTCP approaches [11], and MPTCP protocol design without SDN [12][13]. The former approaches simply employ SDN controller only to compute the routes for all subflows with the legacy MPTCP protocol which has many potential issues. However, SDN's ability is not fully exploited, and it ought to undertake more efficient roles using its global network knowledge. The later one focuses on designing more efficient MPTCP protocols with no SDN controllers, but there is still large room for improvement on protocol designs. Take EWTCP for example, it uses TCP-NewReno like algorithm to adjust window sizes on each route independently of other routes, as below:

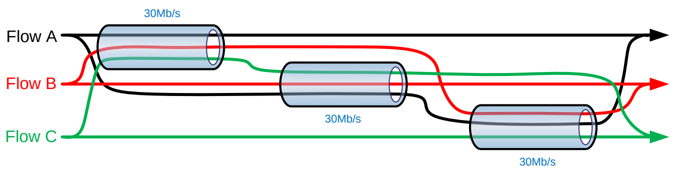

Fig. 2. An example of EWTCP multipath routing.

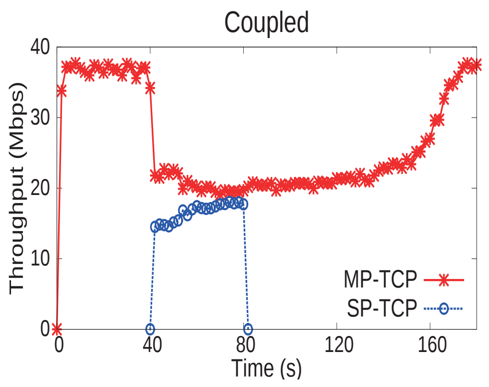

Fig. 3. Responsiveness performance of coupled algorithm.

- For each ACK on route r, $w_r = w_r + \alpha / w_r$, $\alpha = 1/\sqrt{n}$, where n is the number of subflows.
- For each loss on route r, $w_r = w_r/2$

EWTCP cannot efficiently make use of the multiple paths, which results in relatively low throughput. Figure 2 illustrates a scenario to depict this issue. As shown, there are three different flows, all of which evenly split flows into two subflows on two different paths. There will be three flows on each path, and each subflow will get 10 Mbps, and thus each flow will achieve a throughput of 20 Mbps. However, if each flow only uses the one-hop path, each of them will get 30 Mbps. This reveals that EWTCP cannot make very efficient use of the network. In order to deal with this issue, a coupled algorithm is proposed in [4] and it adjusts the window size of multiple subflows in a coordinated way, as below:

- For each ACK on route r, $w_r = w_r + 1/w_{total}$, where w_{total} is the total window size across all subflows.
- For each loss on route r, $w_r = w_r/2$

This solution can mitigate the issues of EWTCP, but the coupled algorithm is not responsive suffering a long convergence time, as shown in Figure 3. Typically, a route with a large throughput can greatly suppress the throughput on other routes even though the other routes are underutilized.

Semi-coupled MPTCP and Max MPTCP [8] are then proposed. Semi-coupled MPTCP adjusts window size as below:

- For each ACK on route r, $w_r = w_r + \alpha / w_{total}$, where α is a constant which controls the aggressiveness.
- For each loss on route r, $w_r = w_r/2$

Max MPCTP adjusts windows as below:

- For each ACK on route r, $w_r = w_r + min\{\alpha/w_{total}, 1/w_r\}$, where α is a constant which controls the aggressiveness.
- For each loss on route r, $w_r = w_r/2$

Max MPTCP still has a relatively high convergence time, though lower than Coupled MPTCP.

Moreover, all of these existing algorithms only modified the way of window increasement without modulating the window in case of packet loss. Furthermore, all of these algorithms do not have congestion precautionary mechanisms and are not latency aware. The limitations are listed as below:

- They cannot prevent the congestion;
- They cannot avoid background large flows dominating the buffer increasing the latency of small flows;
- Switches suffer high buffer pressure, where in shared memory switches the congested ports will exhaust the buffer resources harming flows on other ports;
- The multipath routing could result in serious incast issue but the existing MPTCP protocols cannot resolve it;
- The existing algorithms behave latency-agnostic, where the paths with lower latency cannot be efficiently used.

In order to deal with these limitations, we propose a SDN-based MPTCP solution aiming to achieve high application performance. The key idea of this solution is given as below:

1) SDN is leveraged to compute a set of optimal available routes for subflows and actively adjust the number of subflows according to the instantaneous traffic condition.
   - The SDN controller computes the best routing paths for all flows based on the network condition.
   - SDN application running on end device monitors the real-time achieved overall throughput of all subflows. Whenever it is lower than the theoretical achieveable throughput received from SDN controller, then additional subflow paths are requested to be established and assigned with the computed best routes.
2) Use a distributed and reactive approach for bandwidth allocation and adjusts the congestion windows of subflows on multiple paths in a coordinated fashion.
   - For each ACK on path i, increase window by $\alpha * \frac{1}{r_i(\sum_{k\in S} w_k/r_k)}$, where $\alpha = \frac{1/r_i}{(\sum_{k\in S} 1/r_k)/n}$, w_i indicates the window size of subflow i, r_i denotes the RTT of subflow i.
3) Design a novel congestion avoidance algorithm, which employs ECN (explicit congestion notification) feedback and latency (in terms of RTT) to modulate the congestion window via a gamma-correction function.
   - If packets are marked with ECN, the window size is adjusted according to current congestion level, as $w_i = w_i(1 - \frac{\delta}{2})$, where $\delta = f^{\tau}$ is the penalty function, $\tau = \frac{r_i}{(\sum_{k\in S} r_k)/n}$ is a time (RTT) factor.

The key differences that this proposal differentiates the existing solutions are given as below:

1) The number of subflows is adjusted dynamically according to the real-time achieved throughput and traffic condition, other than using a fixed number of subflows which wastes much network and host system resources.
2) SDN is used in a different way to globally optimize the traffic engineering and guarantee applications throughput, which is very beneficial for load balancing, efficient path choosing and link utilization.
3) The novel congestion avoidance algorithm adjusts the window size according to the current congestion extent. This is different from the existing MPTCP solutions which directly halve the windows when packet drops without considering the level of congestion on different paths.
4) The special new designed congestion control mechanism mitigates the incast issue, reduces the queuing delays, and well handles traffic bursts, all of which are the limitations of existing MPTCP solutions.
5) The coordinated window adjustment mechanism achieves better bottleneck fairness and more efficient path selection. It is prone to use less congested paths (with lower latency) to carry more traffics, and the traffics on congested paths automatically shift to the less congested paths.

III. SDN-based eMPTCP Design

In this section we present the design, analysis of our proposed SDN-based MPTCP variant. The architecture of this solution mainly consists of three key components:

1) Controller control plane, which is responsible for:
   a) Calculate optimal paths for new subflows of each MPTCP, and installing forwarding rules in all involved switches;
   b) Estimation of the maximum achievable sending rate of each MPTCP flow;
   c) Actively adjust the number of subflows of each MPTCP flow to meet achievable throughput.
2) Data plane has two parts:
   a) The SDN application running on end user's device monitors the achieved sending rate and initiates adding path request when the achieved sending rate is lower than a designated threshold;
   b) The switch notifies the controller about new MPTCP flow arrivals and expirations.
3) MPTCP transport protocol design
   a) Coupled algorithm to adjust windows of subflows on multiple paths in a coordinated fashion;
   b) Latency-aware congestion control mechanism

Figure 4 illustrates the architecture of the proposed SDN-based eMPTCP solution. In this solution, each MPTCP-enabled end host / device runs a SDN application monitoring the states of MPTCP flows, e.g. the current sending rates on connected paths. The working procedure is as below:

1) When a new TCP flow starts, MPTCP adds the MP_CAPABLE option to the SYN segment starting the handshake procedure.
2) The SYN packet with MP_CAPABLE option arrives at the switch initiating a PACKET_IN message to be sent to controller notifying the controller of this new flow.

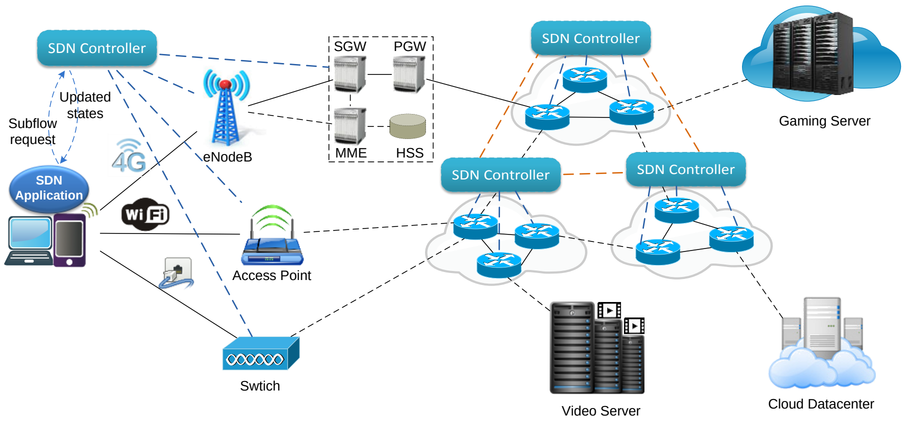

Fig. 4. The architecture of SDN-based eMPTCP solution.

3) SDN controller computes the best route for the new flow, and installs the forwarding rules on all involved switches.
4) If a server supports MPTCP, it replies to the SYN segment with an ACK message containing MP_CAPABLE option. Then, the sender confirms the MPTCP connection by sending the final ACK with MP_CAPABLE option, and the multipath connection is established. From this time, more subflows can be added to this connection by using MP_JOIN option.
5) The controller constantly estimates the maximum theoretical achievable sending rate (denoted as ζ) of each MPTCP flow according to the current network state, and delivers the updated estimated ζ values to the respective source senders of flows.
6) SDN applications running on users device monitors the exact sending rate (ξ) of each MPTCP flow, where $\xi = \sum w_i/r_i$, w_i denotes the window size of subflow i, r_i indicates the round trip time of subflow i, which is computed using a smoothed RTT estimator similar to TCP.
7) When it detects that the achieved sending rate (throughput) is largely lower than theoretical achievable sending rate, i.e. $\xi < \lambda * \zeta$, where λ is a adjustable marginal parameter, then the SDN application sends adding-path-request to notify controller. Then, the controller calculates the number of subflows to add and the best route for each subflow, and notifies the sender so as to achieve the expected overall sending rate.
8) Then, the sender and receiver use the MP_JOIN option to add new subflows.
9) All the flows are transmitted using our new proposed MPTCP transport protocol designed as below.

In order to further improve the throughput and application performance, we design a novel MPTCP transport protocol eMPTCP, which is latency-aware, and can handle traffic bursts. This new transport protocol eMPTCP employs a distributed and reactive approach for bandwidth allocation. The window adjustment mechanism applies ECN feedback and latency (in terms of RTT) to regulate the congestion window taking the bottleneck fairness, latency and congestion into consideration.

The protocol is designed as below:

1) **Switch side:** The switch employs an active queue management, where the queue uses randomly early detection (RED) scheme. The arriving packet will be marked with the CE (congestion encountered) codepoint DiffServ field (right most two bits) in IP header when the instantaneous queue occupancy is greater than K, which is pre-defined. This ensures the quick notification of queue overflow.
2) **Receiver side:** When the receiver receives the packets, it ACKs each packet with ECN-Echo (ECE) flag in the TCP header if the packet has a marked CE codepoint.
3) **Sender side:** The congestion window size is adjusted at the sender side. The sender calculates the fraction of marked packets, denoted as f, which is updated once for every window of data (per RTT) as below:

$$f = (1-p) * f + p * h$$

where h is the fraction of packets that are marked in the previous window, and $0 < p < 1$ is a weighted parameter given to new samples against the past. A larger f implies a higher congestion level. The window size is adjusted as below:

- If $f = 0$, for each ACK on path i, increase window by $\alpha * \dfrac{1}{r_i * (\sum_{k \in S} w_k/r_k)}$, where $\alpha = \dfrac{1/r_i}{(\sum_{k \in S} 1/r_k)/n}$, n is the total number of subflows, and S is the set of paths of all subflows.
- If $f > 0$, the window is decreased to be $w_i(1 - \frac{\delta}{2})$, where $\delta = f^{\tau}$ is the penalty function, $\tau = \dfrac{r_i}{(\sum_{k \in S} r_k)/n}$ is a time (RTT) factor, and r_i is the round trip time on path i.

To summarize, the window is adjusted as below:

$$w_i = \begin{cases} \dfrac{n}{r_i^2(\sum_{k \in S} 1/r_k)(\sum_{k \in S} w_k/r_k)} & f = 0 \\ \\ w_i(1 - \dfrac{\delta}{2}) & f > 0 \end{cases}$$

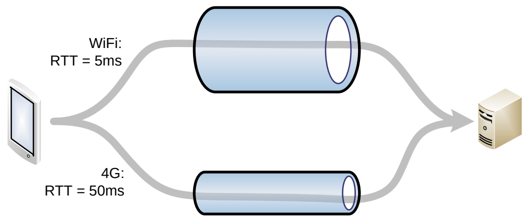

Fig. 5. An example of multipathing via WiFi and 4G.

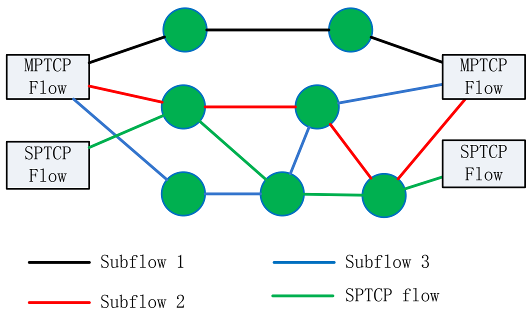

Fig. 6. An example of MPTCP and SPTCP coexsistance.

By doing this, a larger f (higher congestion) leads to a larger δ , and thus causing more window decreasement. If $f = 1$, which means 100% packets are marked, the window will be cut by a factor of 2. The penalty function δ with factor τ ensures that the window size of subflow on congested path (higher RTT/latency) will be decreased more than that of subflows on less congested paths (lower RTT/latency). Meanwhile, the window increasement considers the bottleneck fairness on the shared links, and is prone to shift its traffic onto less-congested paths with lower latency. Figure 5 gives an example. In this scenario, the RTT of WiFi route is $r_1 = 5\ ms$, and RTT of 4G route is $r_1 = 50\ ms$, therefore in case of no congestion ($f = 0$), the window of WiFi subflow will be increased by $\frac{20}{11} * \frac{1}{5 * (\sum_{k \in S} w_k / r_k)}$ for each received ACK, while the window of 4G subflow will be increased by $\frac{2}{11} * \frac{1}{50 * (\sum_{k \in S} w_k / r_k)}$. Likewise, if congestion occurs on 4G path ($f > 0$), the window size will be decreased more than that on WiFi path. As a result, most of traffic will be quickly moved to WiFi route with lower latency and higher sending rate, and the traffic will be dynamically balanced according to the instantaneous network condition on each path.

Consequently, this solution can achieve the following unique benefits.

1) SDN helps compute the best routes for MPTCP subflows using its global knowledge of network conditions. This is better than legacy ECMP-based random hashing approach that may lead to path collision and cause significant throughput degradation.
2) With the help of SDN, the sender actively adjusts the number of subflows of each MPTCP flow, which significantly saves more server resources and network resources compared with traditional scheme which uses a fixed number of subflows.
3) Running with the modified MPTCP protocol eMPTCP the system can dynamically and automatically migrate the flows on the congested paths to the less congested paths by applying a congestion avoidance mechanism. According to the theoretical analysis results, it greatly increases the overall throughput and decreases average flow completion time.
4) The novel window adjustment mechanism plays a great role in handling the traffic bursts, easing the buffer pressure on switches, and greatly mitigates the incast issue which degrades the performance of end-to-end throughput.
5) This solution can largely improve the application performace and user experience, making the network more robust and faster.
6) This novel MPTCP protocol variant works well with existing switch hardware and is able to coexist with legacy TCP. It ensures that a multipath flow will not take up more capacity on any shared paths than if it was a single path TCP flow using only one of those paths, which guarantees it will not unduly harm other flows. Figure 6 depicts an scenario illustrating the coexistence of MPTCP flows and single path TCP (SPTCP) flows.

IV. System Evaluation

In this section, we evaluate the performance of eMPTCP through simulations from various aspects.

A. Simulation Settings

All simulations are conducted based on a 8-*ary* Fat-Tree topology [14], which consists of 8 PODS, 16 core switches, 32 aggregation switches, 32 edge switches and 128 servers connected using 1*Gbps* links capable of bidirectional communications, and contains 16 different paths between any intra-pod node pairs. Each server is configured with two different IP addresses so that it allows up to 4 subflows per MPTCP connection. The propagation delay of a link is set to be 5 μs. The default forwarding time (pipeline processing time at each switch) is 10 μs, excluding the queuing time. The flow distribution (the way choosing source and destination) follows uniformly random mode, and the Weibull distribution is applied to determine the interval time of packet generation. The maximum segment size (MSS) is 1460 bytes (MTU=1500 bytes). The TTL is set to be 128.

B. Evaluation Results

In the simulations, eMPTCP is evaluated using the following various metrics: 1) the network latency, which is measured by the flow completion time; 2) the throughput of eMPTCP flow; 3) the bottleneck fairness.

The network latency is evaluated in terms of flow completion time. As shown in Figure 7, it reveals that eMPTCP significantly reduces the overall network latency, with 35% and 50% on average latency reduction compared with MPTCP and TCP New Reno, respectively. Figure 8 presents the simulation results of goodput, which is evaluated at the receiver side, between a certain pair of servers using eMPTCP, MPTCP and

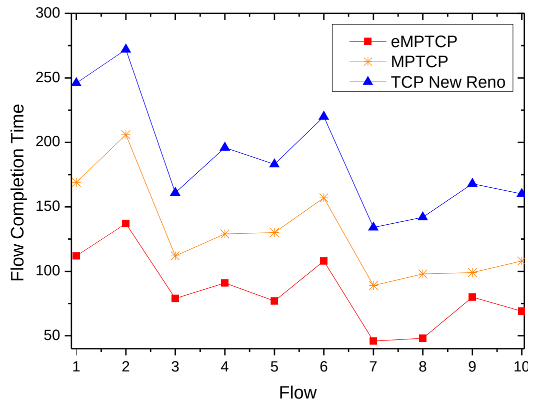

Fig. 7. The flow completion time.

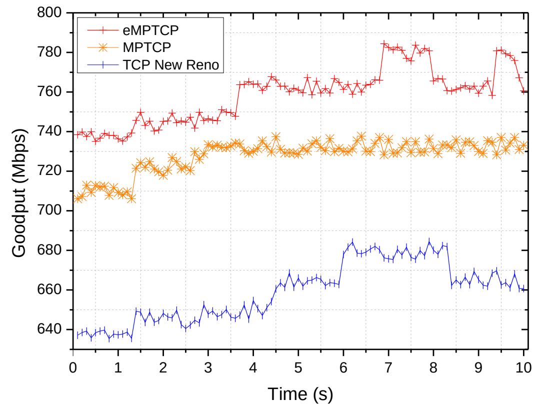

Fig. 8. The performance of goodput.

traditional TCP New Reno. The goodput evaluates the application level throughput which only refers to the really transferred "useful" data per time unit (application data rate), excluding protocol overhead bits. The simulation results exhibit that eMPTCP increases goodput by 10% at most compared with MPTCP, and 18.6% better than TCP. The results shown in Figure 9 convinces a better fairness achieved by eMPTCP, where all the flows achieve their fair share of the bandwidth during the whole simulation period when the number of flows are increased and decreased.

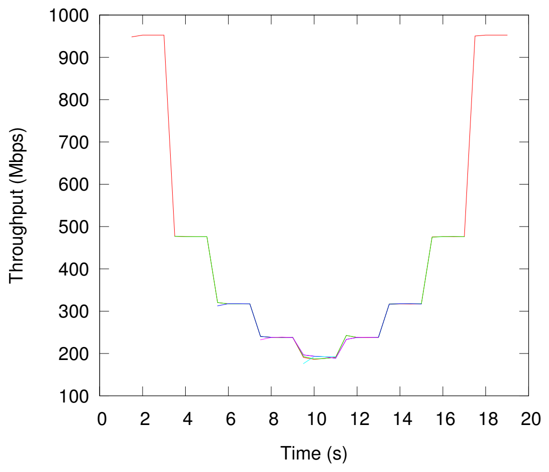

Fig. 9. The fairness between eMPTCP/MPTCP and the traditional TCP New Reno.

V. Conclusion

In response to the poor performance of MPTCP when used for bulk transfer in high speed networks, this research proposed an efficient SDN-based eMPTCP solution. In this solution, SDN is leveraged to optimize path selection rather than using hashing functions which often leads to collisions resulting on congestions, and dynamically adjust the number of subflows according to instantaneous traffic condition. We design a congestion control mechanism by adjusting windows on multiple paths in a coordinated fashion. Besides, it employs ECN feedback and latency (in terms of RTT) to modulate the congestion window via a gamma-correction function, allowing enough headroom of queue buildup. This largely reduces queueing delays on congested switches, and minimizes the impact of long flows on the completion time of short flows. Besides, it also helps mitigate the buffer pressure, and enables faster reactions to congestions. Consequently, it can achieve higher throughput and lower latency than regular TCP and MPTCP.

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