Understanding User Perceived Video Quality using Multipath TCP over Wireless Network

Sohee Kim Park, Arani Bhattacharya, Mallesham Dasari, Samir R. Das
Stony Brook University
{soheekim, arbhattachar, mdasari, samir}@cs.stonybrook.edu

***Abstract*—Modern computing devices such as smartphones and personal computers provide multiple network interfaces such as WiFi and 3G/LTE cellular network. Multipath TCP (MPTCP) is a commonly proposed method of aggregating the bandwidth of these interfaces. However, how MPTCP affects the quality of experience of existing services, especially to variances in bandwidth and latency among the individual paths over the wireless networks is not well-known.**

In this work, we explore the quality of service (QoS) and quality of experience (QoE) of adaptive video streaming using MPTCP over wireless network given its vast popularity and significant bandwidth demand. Unlike prior works, we conduct systematic measurements over three mobile network operators, AT&T, Verizon Wireless (VzW), and T-Mobile, along with WiFi. Based on extensive measurements, we show that MPTCP can improve the QoS and QoE of video streaming only if the network interfaces have the roughly similar bandwidth and latency. Our studies also show that MPTCP can perform worse than TCP in case of extreme differences between the network interfaces.

***Keywords*—Multipath TCP, Adaptive Video Streaming, QoS, QoE, Performance, Wireless Network**

I. Introduction

Multipath TCP (MPTCP) is a TCP extension that establishes one or more TCP connection(s) among two multi-homed hosts and diverts the packet deliveries over multiple paths to achieve higher throughput and improve resilience to the path failure [1]. Industries have started implementing and deploying MPTCP services [2]–[4]. There has been increasing efforts in refining the protocol, applying MPTCP in various areas, and measuring the MPTCP performance [5]. However, the user perceived QoE and the performance of MPTCP on existing Internet services are not well explained.

Video streaming is a crucial Internet application because a significant fraction of mobile data traffic and fixed access network traffic is consumed as video streaming [6], [7]. The recent proliferation of 4K video, Virtual Reality (VR) or 360°video streaming [8], [9] demands more bandwidth than ever before. However, the average Internet connection speed in the US falls short of meeting the bandwidth needs [10]. Given the significant bandwidth demands of video streaming, MPTCP's growing popularity and adoption to aggregate bandwidths, it is essential to know how video streaming performs over MPTCP. Thus, we focus on studying the performance of video streaming over MPTCP under different network conditions.

In this paper, we present a framework for measuring the QoS and QoE of adaptive video streaming over MPTCP. We look at QoS metrics such as downlink throughput and round-trip time while streaming video over MPTCP. We also study the correlation between these QoS metrics and other QoE metrics such as resolution, number of resolution changes, initial start time, rebuffering duration, and total play time using combination of different network interfaces over MPTCP. We run MPTCP over multiple existing wireless network carriers and study the QoE in different combinations of network and compare with TCP's. We measure several quality measurement metrics that impacts the user engagement [11] and showcase both limitations and benefit of MPTCP over the cellular and/or WiFi network. We seek to answer the following questions in our study: 1) How good is the QoS and QoE of video streaming over MPTCP on Wireless Network, compared to TCP?, 2) Are there scenarios where MPTCP offers better QoS and QoE than TCP?, 3) Are there scenarios where MPTCP degrades the QoS and QoE compared to TCP?, and 4) Can we improve MPTCP to achieve better QoS and QoE?

Our measurements show that MPTCP can provide a better QoE than TCP. However, the gains from MPTCP are limited to the cases where the different interfaces offer similar bandwidths and latency. The higher the divergence between the interfaces, the worse the QoE. After a point, the QoE becomes worse than even TCP. Thus, our measurements show when using MPTCP is beneficial as compared to TCP.

We summarize the contributions of this paper as follows:

- We measure the user perceived video streaming QoE over MPTCP, which gives insights on how the network characteristics and MPTCP impact the user video streaming experience.
- We analyze the performance of video streaming over both TCP and MPTCP under various network conditions. We find the conditions where we can achieve higher video streaming QoE using MPTCP.
- Using the insights we obtain from our measurements, we propose alternatives to the current MPTCP path connection and scheduling algorithm to improve video streaming quality.

The rest of the paper is organized as follows. Section 2 explains our experiment methodology and Section 3 explains our observations. We discuss related works in Section 4 and conclude in Section 5.

II. Experiment Methodology

We first describe the metrics for measuring video streaming QoS and QoE. Then, we present measurement methodology under different scenarios.

A. Quality Measurement Metrics

We describe video streaming QoS and QoE metrics in Table I. We measure network quality using iPerf [12] and ping before streaming. We capture streaming packet traces using tcpdump and log them for further analysis.

TABLE I
Video Streaming QoS and QoE Metrics

Parameter	Description
Th_{avg}	Downlink Throughput average and standard deviation
RTT_{avg}	Average Round Trip Time
RTT_{diff}	Maximum RTT differences between the paths
DB_{total}	Total packet size downloaded for the streaming
S_o	Initial playback start time (Initial Delay), i.e. Duration between the first HTTP request for the streaming and the playback time
P_{total}	Total play time. Duration the video is played
R_{total}	Rebuffering duration. The duration that the streaming halts to fill the playback buffer and restart playback
Res_{avg}	Average Video Resolution Played
Res_{ch}	Number of resolution increase and decrease

From the tcpdump traces, we measure downlink throughout streaming and average it over time, Th_{avg}. From the traces, we also calculate round-trip time (RTT) as difference between timestamps of a data packet and the corresponding acknowledgment packet. We calculate average round trip time, RTT_{avg} for each path. Given $RTT_{avg,i}$, average RTT of i^{th} path, we calculate RTT_{diff}, the RTT difference of two path with longest average RTT and shortest average RTT.

$$RTT_{diff} = \max_{i=1}^{n} RTT_{avg,i} - \min_{i=1}^{n} RTT_{avg,i}$$

P_{total} is total play time measured from the first HTTP request until the playback is completed. S_o is initial start time, duration between the first HTTP request and the playback starts. Rebuffering duration R_{total} is difference S_o and video duration from P_{total}. For video resolution of each segment, average video resolution is calculated such that $Res_{avg} = 1/N * \sum_{n=1}^{N} Res_n$, where N is the number of segments and Res_n is the resolution of n^{th} segment.

Th_{avg} and RTT_{avg} are network QoS metrics. User QoE is measured in terms of start time S_o, rebuffering duration R_{total}, average resolution Res_{avg} and the number of resolution changes Res_{ch}. The smaller the values of S_o, R_{total}, and Res_{ch}, the better the user experience and the quality of streaming. Higher Res_{avg} indicates better QoE.

B. Experiment Setup

Fig. 1 illustrates our experiment setup. We use Amazon EC2 servers at different locations. One server, ServerVA (located in Virginia, USA), is a EC2 location close to the vantage

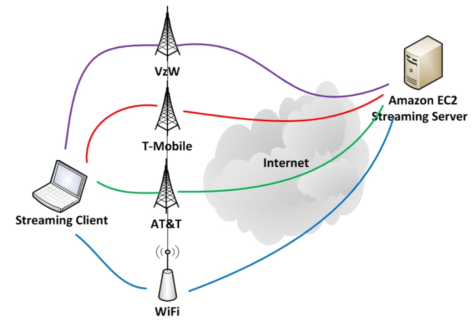


Fig. 1. Experiment Setup

point, and the other server, ServerJP (located in Japan) is chosen to give large RTT from the vantage point. Average delays are about 20 ms $\sim$ 300 ms depending on the network interface types and server locations. The streaming server is configured with Ubuntu 14.04, Apache version 2.4.12, and MPTCP Kernel. Adaptive video streaming client sends HTTP request over port 80, IPv4. Port 8080 is also used to support MPTCP streaming with a specific carrier: it drops MPTCP Options field with the MP_CAPABLE option at HTTP request over port 80, which prevents establishing MPTCP connection and falls back to TCP connection.

The streaming server hosts video files using Apple HLS (HTTP Live Streaming) adaptive video streaming, which enables adaptation of bit-rate and resolution. The video is divided into segments of a fixed duration of $10s$ in our experiment, and client requests the corresponding quality video segment based on the bandwidth available at the point of request. Table II and III list the video quality offered at ServerJP and ServerVA. ServerJP hosts a 10 minute video with maximum resolution 720p and ServerVA offers a 5 minute video with maximum resolution of 1080p.

TABLE II
Video Quality Offered at ServerJP

Category	Resolution	Bitrate	Remark
Audio Only	N/A	65 Kbps	
240p	428 x 240	246 Kbps	
288p	512 x 288	473 Kbps	
360p	640 x 360	1.018 Mbps	
432p	768 x 432	1.399 Mbps	
480p	854 x 480	2.017 Mbps	SD
576p	1024 x 576	2.559 Mbps	
720p	1280 x 720	3.318 Mbps	HD

TABLE III
Video Quality Offered at ServerVA

Category	Resolution	Bitrate	Remark
Audio Only	N/A	64 Kbps	
288p	512x288	544 Kbps	
360p	640x360	1.054 Mbps	
480p	854x480	2.764 Mbps	SD
720p	1280x720	4.564 Mbps	HD
1080p	1920x1080	12.064 Mbps	2K

Streaming client runs on desktop computers: Intel NUC (Next Unit of Computing) with Ubuntu 14.04, VLC player version 2.2.0 [13], This configuration offers computing power for high quality video, space for log data volume and tethering that enables us to connect multiple cellular network simultaneously using MPTCP.

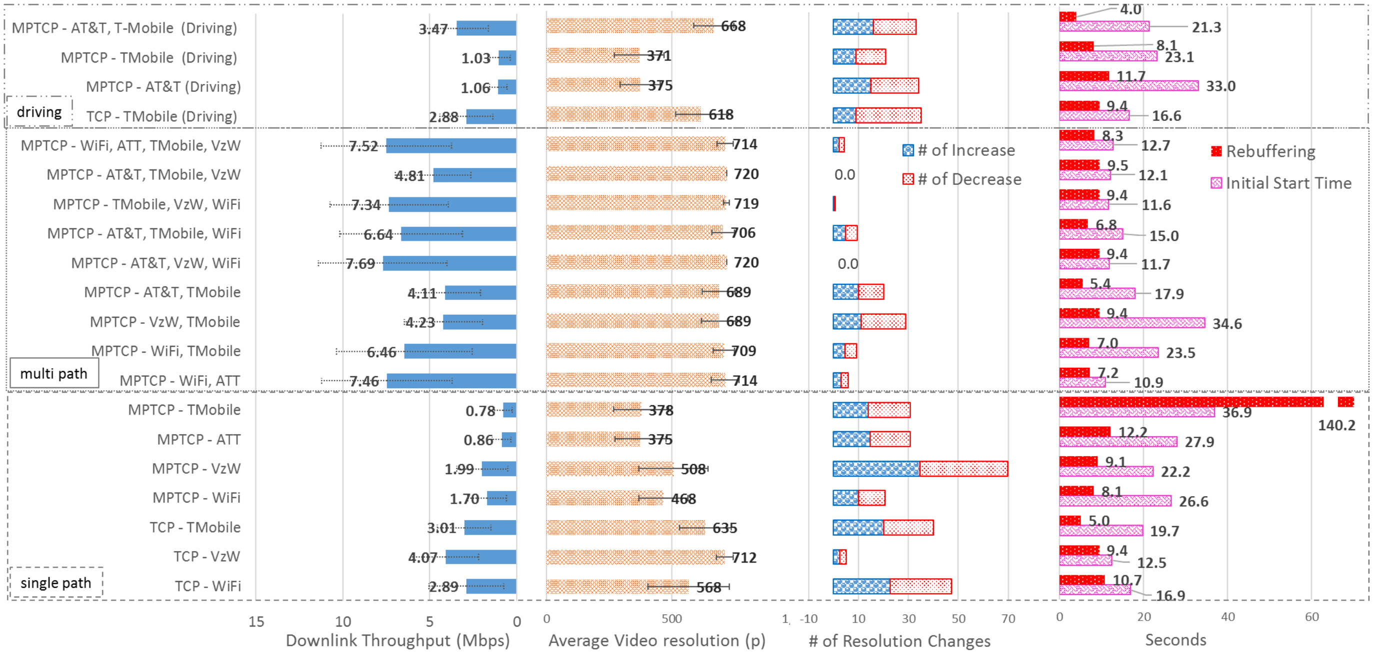


Fig. 2. Video Streaming QoS and QoE over WiFi, T-Mobile, AT&T, and VzW, from ServerJP: TCP vs. MPTCP, Stationary vs. Driving

Using WiFi and major cellular network (VzW, T-Mobile, and AT&T), we measure streaming experience from different vantage points such as university campus indoor and outdoor, office buildings, and residential areas. Since office buildings show similar results as campus or residential areas, we do not discuss it separately among our results.

C. Experiment Scenario

Experimental scenarios include changes in 1) streaming mode, stationary or driving, 2) server location, 3) network interface and 4) transport protocol. The stationary mode is streaming at fixed locations. Mobility testing is conducted on a driving vehicle. The experimental scenario is summarized in Table IV.

TABLE IV
VIDEO STREAMING EXPERIMENT SCENARIO

Mode	Network	Server	Resolution
Stationary	Cellular, WiFi	ServerJP	Max 720p
Driving	Cellular	ServerJP	Max 720p
Stationary	Cellular / WiFi	ServerVA	Max 1080p

III. EXPERIMENT RESULTS

We now present the video streaming quality measurement and the analysis over the wireless network using MPTCP/TCP.

A. Delay, Bandwidth and Streaming Quality

MPTCP is designed to perform at least as good as TCP. However, our evaluation shows that TCP performs better than MPTCP under two scenarios: 1) TCP always outperforms the single-path MPTCP, 2) The quality degradation on MPTCP is larger when network delay is larger: With 155 ms delay, VzW throughput over MPTCP is 49% of TCP and the average

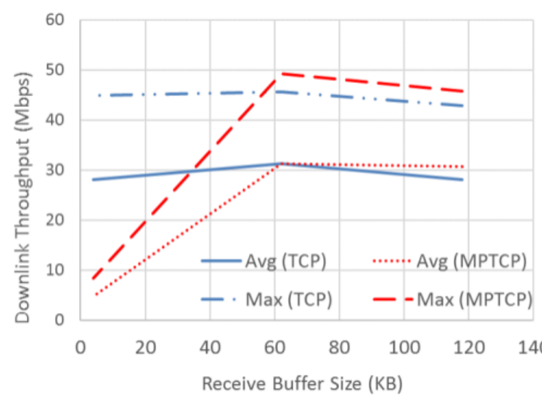


Fig. 3. Receive buffer Size and Streaming Quality

video resolution is 71% of TCP. With 400 ms delay, T-Mobile throughput over MPTCP is 26% of TCP, and the average resolution is 59% of TCP (Fig. 2).

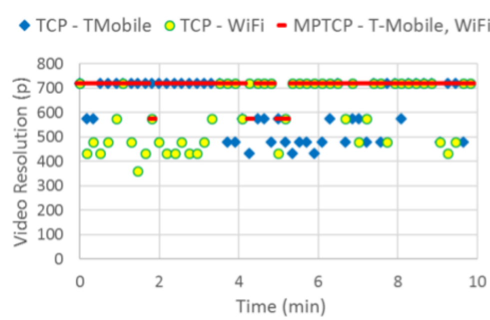
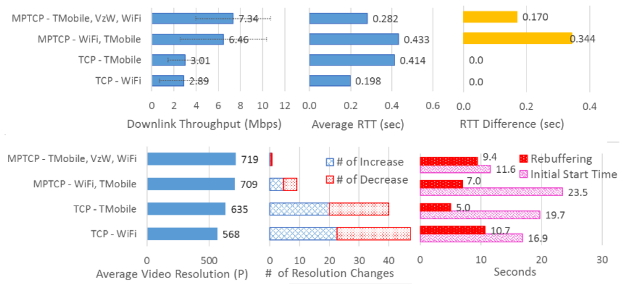
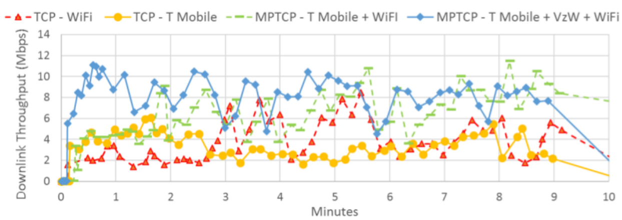


Fig. 4. Video resolutions: TCP vs. 2-path MPTCP

We further experimented with different receive buffer sizes and look at the throughput / resolution correlations. Fig. 3 shows the downlink throughput versus receive buffer size over single-path MPTCP and TCP using WiFi. With default receive



(a) QoS/QoE over similar network: TCP vs. 2, 3-path MPTCP



(b) Downlink throughput over time: TCP, 2, 3-path MPTCP

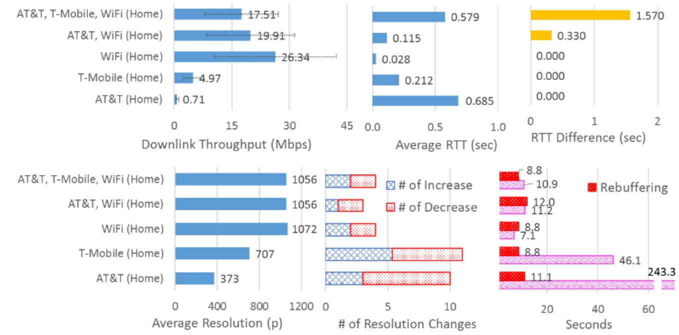
Fig. 5. Higher QoS / QoE over similar 2 paths MPTCP

buffer size, MPTCP throughput is 15% of TCP. Initial start time with MPTCP is twice the TCP and the resolution over MPTCP is 59% of TCP. When the receive buffer size increases to BDP/2, MPTCP throughput increases by more than 7 times over throughput with default receive buffer, close to TCP's. The recommended receive buffer size for single-path TCP is twice that of Bandwidth Delay Product (BDP). For MPTCP, it is sum(BW_i)*RTO_{max}, where BW_i = Bandwidth for each subflow i and RTO_{max} is the largest RTO (Retransmission TimeOut) across all subflows [14]. However, we find TCP shows similar throughput and streaming quality regardless of the receive buffer size.

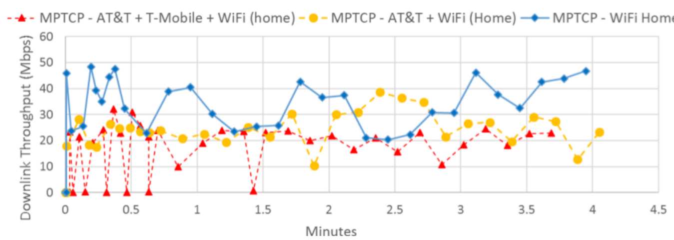
B. Streaming over similar network

We observe that QoE improves over the multiple paths than single path under specific conditions. Fig. 4 shows the video resolutions played over time using TCP and MPTCP. Streaming over two paths on MPTCP gives better video resolution than TCP video streaming. It plays the maximum resolution 93% of the time over MPTCP using WiFi and T-Mobile combined, while 47% and 57% of the time with TCP over WiFi and T-Mobile respectively. MPTCP with two interfaces provides better quality than TCP when streaming on a driving vehicle as is depicted in Fig. 2.

We analyze streaming quality and RTT/throughput correlations at Fig. 5. We observe that quality increases when both downlink throughput differences and RTT differences are low: with RTT_{diff} 344 ms and the DL throughput difference about 0.1 Mbps, two path MPTCP throughput increases 215%, average video resolution increases 117%, rebuffering decreases 35%. number of changes decrease 75% than TCP's. However, multiple paths could vary in network characteristics and we show the QoE over MPTCP on such network paths in the next section.



(a) QoS/QoE over network with high delay differences: 1, 2, and 3-path MPTCP



(b) Downlink throughput over network with high delay differences

Fig. 6. Degraded QoS / QoE over MPTCP with heterogeneous paths

C. Streaming over network with high delay differences

Our experiments show that video streaming over MPTCP using largely heterogeneous paths degrades the streaming quality than single path streaming Fig. 6.

QoS and QoE degrades over MPTCP when multiple paths have largely different capacity and delay. Using AT&T with 255 times higher RTT than WiFi's and bandwidth only 26% of WiFi's, MPTCP (AT&T, WiFi) downlink throughput drops to 76% of the single path throughput and video resolution also drops to 1056p from 1072p of single path. Initial start time increases 160% and rebuffering increases 136% (Fig. 6a). When three interfaces are used, compared to the single interface, throughput drops 34%, resolution drops to 1056p, and initial start time increases 153%. RTT_{diff} is over 1.57 seconds and the throughput difference is over 25 Mbps. This shows that under large variance, MPTCP with two or more interfaces does not offer better QoS and QoE than single interface streaming. The more interfaces it adds to WiFi, the worse the QoS and QoE.

Using the shortest RTT scheduling, data packets are transmitted over WiFi with shortest RTT until it is congested. Then the data is transmitted through cellular network. As soon as the WiFi path is freed, more data is transmitted over the WiFi path. While packets sent through the WiFi are already received, some packets sent earlier over the cellular path have not arrived because the capacity is low and the network delay is high. This is because of the heterogeneity of paths which introduces head of line blocking resulting in out-of-order packets from the slow path, holding the receive buffers.

D. Optimal number of subflows

We now investigate how many MPTCP subflows would be necessary to stream HD, 4K, or 360 videos. We intend to

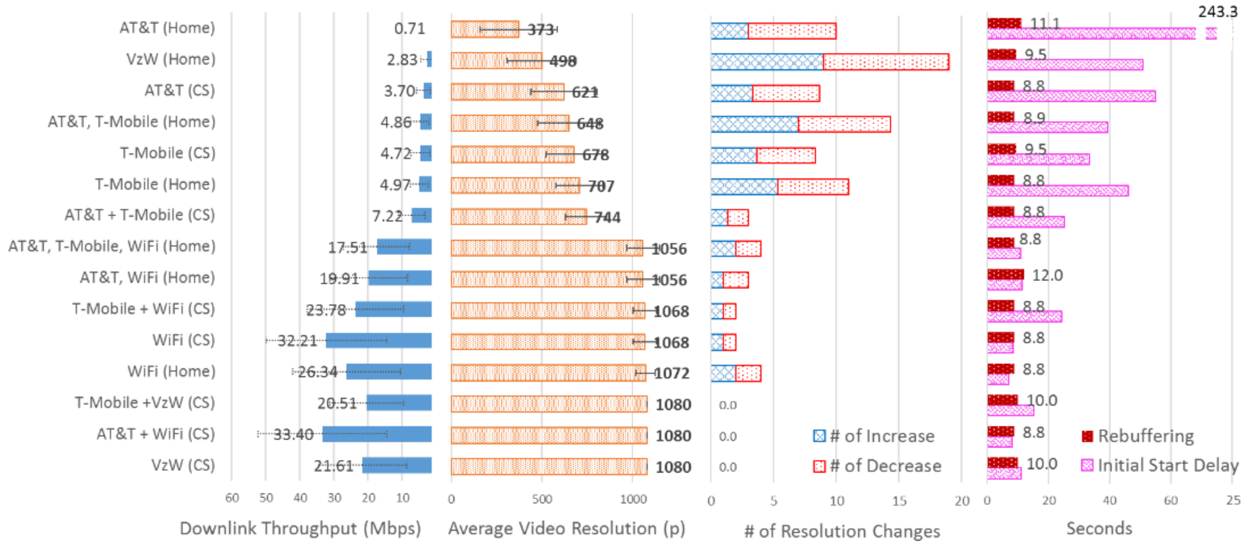


Fig. 7. Video Streaming QoS and QoE over WiFi, T-Mobile, AT&T, and VzW, from ServerVA using 1, 2, and 3-path MPTCP

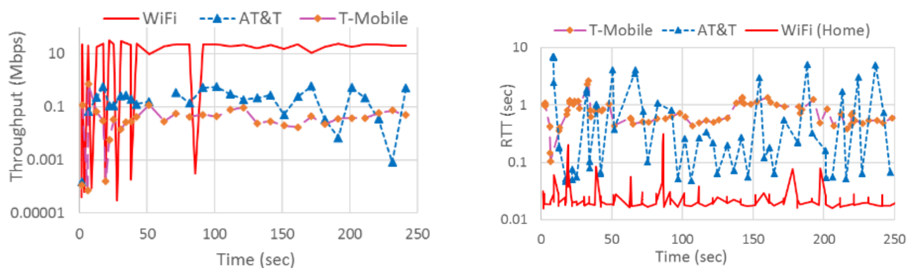


Fig. 8. (a) Downlink Throughput (left), and (b) RTT over time (right) in 3 interface MPTCP - WiFi, AT&T, T-Mobile

answer whether aggregating more interfaces always provides better throughput and better QoE. Our experiment shows that the optimal number of paths and the quality improvement over a different number of paths primarily depends on the path network characteristics.

Fig. 7 shows QoS and QoE of video streaming over MPTCP with one, two, and three interfaces from ServerVA, which offers the maximum resolution of 1080p. AT&T delivered 621p quality and T-Mobile 678p and was unable to achieve HD quality video streaming in a single path. However, when combined, AT&T and T-Mobile cellular interfaces deliver 744p, higher than HD quality. VzW (CS) alone delivers 1080p, the maximum resolution offered. WiFi (CS) and WiFi (home) both brought close to maximum resolution offered. However the quality gets worse when certain paths are combined: WiFi and AT&T (home) gives only 76% of WiFi throughput and rebuffering and initial start time also increases. WiFi, AT&T, and T-Mobile (home) gets only 66% of the throughput and QoE gets worse. As discussed in Section III-C, among these scenarios, the best quality is obtained using single path. The optimal number of sub-flows to achieve the maximum quality depends on target quality, network throughput and delays.

E. Effect of Shortest RTT scheduling

MPTCP uses Shortest RTT scheduling by default. When streaming over MPTCP using WiFi, AT&T, and T-Mobile, as shown in the downlink throughput (Fig. 8a), T-Mobile has downlink close to zero. T-Mobile has 19 times higher average delay than WiFi and 2.1 times higher than AT&T (Fig. 8b). Because the scheduling algorithm uses the shortest RTT, it uses the WiFi, then next lowest RTT AT&T, then back to WiFi after WiFi becomes available to send more packets. So T-Mobile path is hardly used for data transmission. Fig. 6a shows RTT differences is about 1.57 seconds. When the 3rd interface has very high network delay compared to the other two interfaces, we observe that overall RTT increases and the 3rd interface does not contribute to increase DL throughout. And it is better without using the 3rd interface. We recommend the scheduling algorithm to evaluate the RTT to determine whether it will schedule to transmit packets or wait for the high performing path freed up, instead of picking the next shortest RTT path regardless of the capacity and delays. Furthermore, due to fluctuation in wireless network capacities and delays, a wireless link can show much higher RTT during the streaming than initial delay measured. This results in the quality degradation even when configured with the recommended receive buffer size.

IV. Related Work

Although TCP is not ideal for delay sensitive applications [15], most current video applications such as Youtube, MPEG-DASH, Apple HLS, MS Smooth Streaming and Skype use TCP because of its compatibility with firewalls and routers. Thus, many studies have focused on adapting TCP for real-time applications [15]–[17].

There has been increasing efforts in MPTCP performance measurement studies in various areas: mobile device energy consumption [18], [19], handover between cellular and WiFi network [20], [21], and application performance measurement [17], [22], [23]. Studies show download performance over MPTCP using applications such as Web browsing, gaming, FTP over wireless network [17], [22] and analyzed how different file sizes impacts the TCP performance [22].

Recent studies measure video streaming performance over MPTCP [17], [24], [25]. Quality evaluation includes PSNR, delays, and goodput [24] or download time and delay [17]. To improve the performance, various approaches also have been proposed: To utilize Forward Error Correction and rate allocation [24], to decide which content to send and prioritize video packets [26], or to minimize using cellular network by assigning preference in MPTCP path selection [25].

We focus on the user perceived quality of experience of video streaming that impacts the user engagement [11]. We compare the quality over MPTCP and TCP, analyze the correlation between QoE and network characteristics and propose alternatives to the MPTCP path connection and scheduling algorithm based on the network conditions.

V. Conclusion

In this paper, we evaluate the performance of MPTCP in the context of video streaming and compare it with TCP. We use quality measurement metrics such as video resolution, startup time and rebuffering time of videos. We conclude the following:

- Increasing the number of interfaces with similar bandwidth and network delays improves video streaming QoE over MPTCP.
- The QoE of video streaming reduces with an increase in the heterogeneity of network interfaces. If the interfaces are very heterogeneous, it even performs worse than TCP.
- Adding an extra wireless interface to MPTCP does not benefit if the interface has much higher network delays and lower network capacity compared to the other path(s).

From our studies, we conclude that the variance in wireless network largely impacts the quality of Internet video streaming over MPTCP, delay sensitive application in general. To better handle the network fluctuation, instead of full mesh, we plan to extend MPTCP such that connection is established only if it is within a target range of bandwidth and delays variance. Instead of scheduling next shortest RTT path even when path has high delay difference compared to the other paths, we plan to extend the scheduling algorithm by evaluating throughput and RTT of individual path to determine whether it should schedule transmitting packets.

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