Parallel HTTP for Video Streaming in Wireless Networks

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***Abstract*—To stream video using HTTP, a client device sequentially requests and receives chunks of the video file from the server over a TCP connection. It is well-known that TCP performs poorly in networks with high latency and packet loss such as wireless networks. On mobile devices, in particular, using a single TCP connection for video streaming is not efficient, and thus, the user may not receive the highest video quality possible. In this paper, we design and analyze a system called *ParS* that uses parallel TCP connections to stream video on mobile devices. Our system uses parallel connections to fetch each chunk of the video file using HTTP range requests. We present measurement results to characterize the performance of ParS under various network conditions in terms of latency, loss rate and bandwidth. Given the limited communication and computational resources of mobile devices, we then focus on determining the minimum number of TCP connections required to achieve high utilization of the wireless bandwidth. We develop a simple model and study its accuracy using ns-3 simulations which confirm the utility of our model for estimating the minimum number of TCP connections required to fully utilize the available bandwidth.**

I. Introduction

A. Motivation

Video streaming over HTTP has become extremely popular and is adopted by major online streaming services such as YouTube and Netflix. Internet video streaming now takes the majority of worldwide Internet traffic and its share will continue to grow. It is estimated that, globally, Internet video traffic will be 80 percent of all consumer Internet traffic in 2019, up from 64 percent in 2014 [1].

There are several HTTP-based video streaming services implemented by different organizations. Adobe's HTTP Dynamic Streaming [2], Apple's HTTP Live Streaming [3] and Microsoft's Smooth Streaming [4] are a few examples of such services. Meanwhile, Dynamic Adaptive Streaming over HTTP (DASH) is being developed as an international standard to unify HTTP-based video streaming over the Internet [5].

The DASH standard is composed of two main parts. One part defines the Media Presentation Description (MPD) that is used by the client to discover the URLs for accessing the video content. The other part defines the format of the video content. In a DASH system, the video streaming server sends video content to clients via the HTTP protocol. Before transmission, a video is encoded into different bit rates on the server. The encoded video file at every bit rate is fragmented into small *chunks*, each chunk contains only several seconds (*e.g.*, 10 seconds) of the video. A client sends HTTP requests to the server to download video chunks sequentially. At the time of streaming, a client can dynamically change the target video chunk's bit rate based on its available resources such as available bandwidth and remaining battery [6].

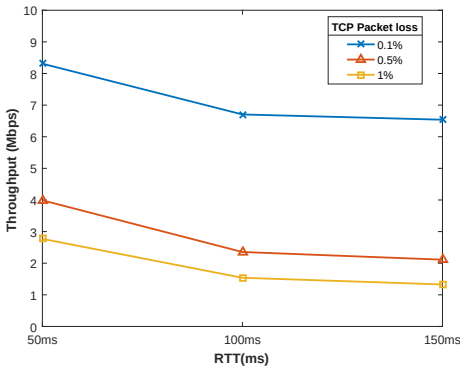
HTTP video streaming systems, including DASH, rely on TCP which has poor performance in networks with high latency and packet loss. In such networks, TCP and consequently the video streaming application is unable to fully utilize the available bandwidth leading to lower quality of experience for users [7]. In Fig. 1(a), we have plotted the measured throughput of a single TCP connection[1] over a 10 Mbps link. We increase round-trip-time (RTT) under different packet loss rates and measure the throughput. When RTT and packet loss are set to 50 milliseconds and 0.1%, respectively, TCP can utilize nearly 86% of the available bandwidth. By increasing RTT and packet loss the throughput drops drastically to a point where, when RTT is set to 150 milliseconds and packet loss is 1%, TCP can only utilize 13% of the bandwidth.

Wireless networks often suffer from high latency and packet loss [8]. Moreover, wireless bandwidth is generally more limited compared to wired bandwidth. This means that HTTP-based video streaming over mobile devices is negatively affected by TCP's inability to fully utilize the wireless bandwidth, which could result in lower video quality and buffering delays during video playback. To improve TCP throughput, the use of parallel connections has been considered. For example, GridFTP [9] is an extension of traditional FTP in which multiple TCP connections are used to transfer data resulting in significant improvements over single connection FTP. This concept has been also applied to video streaming. For instance, Parallel TCP connections have been used in [10] and [7] to improve the quality of video by using multiple connections to download multiple video chunks from the server simultaneously.

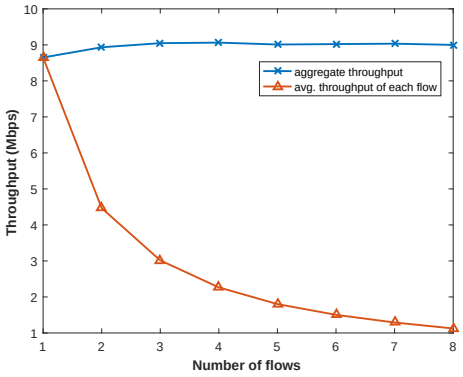
B. Our Work

In this work, we also employ parallel TCP connections to improve video streaming quality on mobile devices. However, different from the existing works (as in [10] and [7]), rather than using multiple connections to fetch different video chunks, we use multiple connections to download the *same chunk*. We note that when multiple connections are used, the available bandwidth between the client and server is divided

[1]We use the terms "flow" and "connection" interchangeably.



(a) Single TCP connection.



(b) Parallel TCP connections with loss probability of 0.1% and RTT of 50 ms.

Fig. 1: Measured TCP throughput over a link of 10 Mbps bandwidth: (a) one connection is not able to fully utilize the link, (b) while the aggregate throughput increases with the number of connections, the per-connection throughput decreases.

among the connections. This means that, while the aggregate throughput increases, the throughput of each individual connection actually *decreases* as depicted in Fig. 1(b). For instance, when there are 5 parallel connections the throughput of each connection on average is less than 2 Mbps.

In wired networks, where the bandwidth between the client and server is plenty, even the reduced per-connection bandwidth is large enough to sustain video streaming at a good quality. Thus, it makes sense to use multiple connections to download different video chunks. In wireless networks however, the bandwidth is limited. Thus, when using multiple connections, the share of each individual connection may be so low that it cannot sustain video playback at a reasonable quality resulting in playback stalls. This problem happens whenever a chunk is needed *as soon as possible* to avoid a playback stall. For example, consider the first chunk of the video. It takes 3 *times longer* to download the first chunk and start the video when using parallel connections to download different chunks (*i.e.*, existing works) compared to downloading the same chunk (*i.e.*, our work). In wireless networks, when the available bandwidth fluctuates significantly over time, this situation (*i.e.*, needing chunks quickly to avoid stalls) would be even more likely to happen compared to wired networks.

The work presented in this paper has two parts. In the first part, we focus on using multiple connections for video streaming as described above. We design and evaluate *Parallel Streaming* (ParS) that employs parallel HTTP for video streaming in low bandwidth and high loss networks. In the second part, we turn our attention to determining the relationship between the aggregate throughput and the number of parallel TCP connections. As discussed later, a critical problem in all systems that use parallel TCP connections (including the above mentioned works) is to decide how many connections are needed to achieve the best video playback quality. We show that as the number of connections increases, the aggregate throughput also increases up to a certain point. Beyond this point, increasing the number of connections actually results in lower throughput due to increased competition among the connections and increased processing and computation overhead. To this end, we develop a simple model to estimate the number of connections required to achieve high utilization of the available bandwidth in different scenarios.

Our contributions in this paper can be summarized as follows:

- We develop a protocol to use parallel TCP connections to download each chunk of a video file from the server using HTTP range requests.
- We develop a prototype system based on our protocol called Parallel Streaming (ParS) and deploy it on a testbed. We then conduct measurement experiments to analyze the impact of RTT, packet loss, bandwidth and chunk size on the throughput performance of our protocol.
- We develop a simple model to determine the number of parallel connections required to fully utilize a network link.
- We conduct ns-3 simulations to study the accuracy and utility of our model.

C. Related Work

There is a vast amount of literature on video streaming (please refer to [11] for a recent survey). The following works are more specific to our work which is on the use of parallel HTTP for enhancing the quality of video streaming.

1) Parallel TCP for File Transfer: One of the main application area of parallel TCP connections is in file transfer applications. GridFTP [9] uses parallel TCP to improve the performance of File Transfer Protocol (FTP). Many download manager softwares such as Internet Download Manager (IDM) [12] and Download Accelerator Plus (DAP) [13] use this concept for increasing download speed.

2) Parallel TCP for Video Streaming: In the area of HTTP streaming, the authors in [10] used parallel HTTP connections and suggested to gradually increase the number of connections based on the available bandwidth and network conditions. Their approach is based on initiating a separate connection for each video chunk. The work in [14] uses content centric

networking (CCN) and multiple network interfaces typically available on a mobile device to implement parallel streaming. It uses multiple connections to download the same video content. However, at any point in time, it uses the connection which delivers data at the highest rate. The work in [7] uses multiple connections to reduce throughput fluctuations in networks with high packet loss and RTT. It uses 10 parallel connections to overcome the poor performance of TCP in such networks, though the main focus is to maintain fairness among connections with favorable and poor network conditions in terms of packet loss and RTT. The authors in [15] use multiple TCP connections to enable users to play different parts of the video without having to wait for buffering of the entire video. An finally, [16] uses multiple connections to download different layers of the video with scalable video coding.

3) Models for TCP Throughput: A notable work on characterizing the effect of parallel connections on aggregate TCP throughput is [17] which provides informative measurements but no throughput formula. The work in [18] provides a closed-form expression for the throughput of parallel TCP connections assuming no synchronization among the connections, no transmission losses and no buffering at the bottleneck router. Under these assumptions, it is shown that the effect of parallel connections on the aggregate throughput diminishes inversley proportional to the number of connections. In the asymptotic regime of a large number of connections, [19] and [20] study the throughput behavior of parallel connections and show a linear relation between the aggregate throughput and the number of connections. None of the models developed in these works are applicable to our scenario where the number of connections is limited and the size of the buffer at the router is arbitrary.

D. Paper Organization

The rest of the paper is organized as follows. Section II is dedicated to presenting our approach and includes measurement results obtained from experiments on our testbed. Section III presents our model for determining how many parallel connections are required to fully utilize a network link. A summary of our ns-3 simulation results is presented in Section IV. Section V concludes the paper.

II. ParS: Parallel Streaming

A. overview

The problem with existing work is that, multiple connections are used to download distinct segments of the video file in parallel. This approach will improve the overall download time and throughput specially when the bottleneck link is in the core of the network. To see this, consider the network topology depicted in Fig. 2, where the link between the router and server is assumed to be the bottleneck link. Assume the bottleneck link has capacity C. If the current number of TCP flows through the link is M and the client starts N parallel

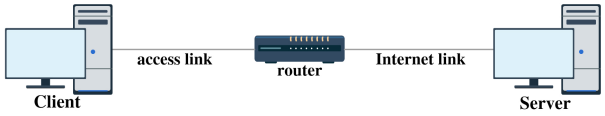


Fig. 2: Network topology used for modeling and simulations.

TCP connections, then the aggregate throughput denoted by X_N is approximately given by:

$$X_N = \frac{N}{N+M}C, \tag{1}$$

which is due to fairness property of TCP. Clearly, we are assuming a homogeneous case where all flows have similar characteristics (*e.g.*, same round trip time). Notice that, in this case, X_N is an increasing function of N. However, as N increases, the increase in X_N becomes negligible. Assume measurements for X_1 and X_2 are available. It is obtained that,

$$X_1 = \frac{1}{1+M}C, \qquad X_2 = \frac{2}{2+M}C, \tag{2}$$

which can be used to obtain the following estimates for M and C,

$$C = \frac{X_1 X_2}{2X_1 - X_2}, \qquad M = \frac{2(X_2 - X_1)}{2X_1 - X_2}. \tag{3}$$

Two observations are made about the behavior of X_N:

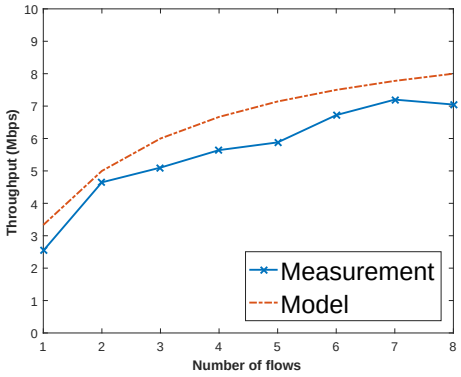
- If $N \gg M$ then, $X_N \approx C$.
- If $N \ll M$ then, $X_N \approx NX_1$.

Fig. 3 shows the throughput achieved with different number of background and parallel TCP flows, when the capacity of the bottleneck Internet link is set to 10 Mbps, 50 ms RTT and 0.1% packet loss. As can be seen, by increasing the number of parallel connections, the aggregate throughput increases. Now, let us consider the per-connection throughput. From (1) we have,

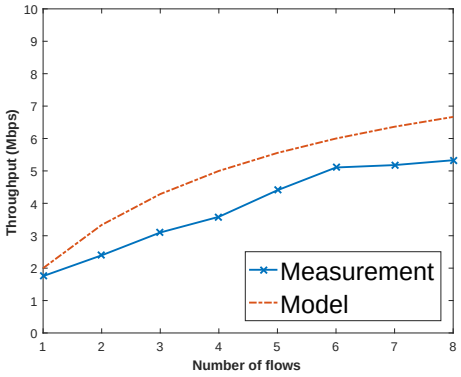
$$\frac{X_N}{N} = \frac{C}{N+M}, \tag{4}$$

which indicates that as the number of parallel connections N increases, the per-connection throughput decreases. As mentioned earlier, there are two competing effects in play:

1) **Aggregate throughput:** Aggregate throughput can be increased by using parallel connections. This is specially true when the bottleneck link is shared by other background traffic (*e.g.*, the link is at the core of the network). In this case, increasing the number of parallel connections effectively *steals* bandwidth from the background traffic. If the link is not shared with other users (*i.e.*, no background traffic) then using multiple connections helps improve the link utilization.
2) **Per-connection throughput:** Per-connection throughput decreases as the number of parallel connections increases. If the bandwidth of the bottleneck link is sufficiently high then the share of each connection of the bandwidth could still be high enough to support smooth video streaming at high quality. However, if the capacity of the bottleneck



(a) $M = 2$ background flows.

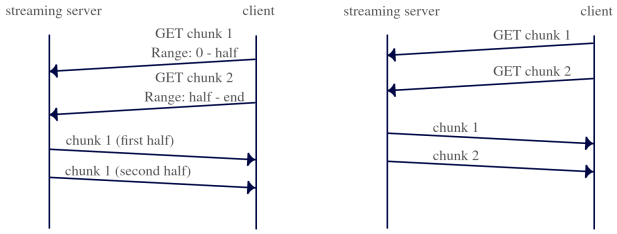


(b) $M = 4$ background flows.

Fig. 3: Aggregate TCP throughput with increasing number of parallel connections obtained from the measurements and model. The Internet bottleneck link has 10 Mbps bandwidth.

link is low, when each connection is downloading a distinct video chunk, this approach may lead to playback stalls depending on the level of synchronization among the connections (more on this later in the paper).

When streaming video on mobile devices, the low bandwidth wireless access link is usually the bottleneck. In this case, using multiple connections to download separate video chunks increases the download time of some video chunks. As discussed in the Introduction section, these are the chunks that need to be downloaded as fast as possible to avoid stalls. Our solution to this problem is to have multiple TCP connections downloading one video chunk at a time. This way, the main focus of our approach is to download the next chunk as soon as possible. This decreases the chance of stalls in video playback and speed ups the playback startup. Fig. 4 depicts the difference between existing approaches and our proposed approach (ParS) when using two parallel connections.



(a) ParS approach. (b) Existing approach.

Fig. 4: Video streaming using two parallel connections.

B. System Design

With ParS, to fetch a chunk, the client first obtains information about the size of the chunk by sending an HTTP-HEAD request to the streaming server[2]. Then it divides the chunk size by the number of parallel connections to find the range of each request. The client then sends multiple HTTP-GET requests each requesting a different byte-range (using *Range* attribute in HTTP request header) of the chunk, where each request is sent over a separate TCP connection. Then the client waits for the streaming server to send the chunk parts in HTTP responses. As the chunk parts arrive at the client, it reconstructs the original chunk. Algorithm 1 illustrates the steps for downloading a video chunk in ParS. This algorithm is repeated for each chunk sequentially.

[2]This step could be eliminated by embedding the size information in the MPD file when using DASH.

Algorithm 1 Parallel Streaming (ParS)

1: $n \leftarrow$ number of parallel TCP connections
2: Send `HTTP-HEAD` request for the chunk
3: $chunksize \leftarrow$ chunk size from `HTTP-HEAD` response
4: $range \leftarrow chunksize/n$
5: **for** $i = 0$ **to** $n - 1$ **do**
6: $start \leftarrow i \times range$
7: $end \leftarrow (i + 1) \times range$
8: Send `HTTP-GET` with `Range` set to $[start, end]$
9: **end for**
10: **for** $i = 0$ **to** $n - 1$ **do**
11: Receive HTTP response
12: Store partial chunk in buffer
13: **end for**
14: Restore the chunk

C. Testbed Specification

To study the effectiveness of our approach, we built a simple testbed consisting of a client and server running Ubuntu 14.04 LTS (with 8 GB RAM and Dual core 2.4 GHz CPU) connected via a D-link switch with 1 Gbps Ethernet cables. All TCP related parameters (*e.g.*, socket buffer size) are the default values in the Linux kernel. We use wired links in order to have the ability to accurately control RTT, packet loss and bandwidth without any random fluctuations due to wireless interference or collisions. However, various network parameters in our experiments (such as bandwidth, packet loss,

and RTT) are set so that we create a network behavior that is close to what would be observed in wireless networks.

To achieve this, we used *tc* and *netem* applications for simulating packet loss, RTT, and controlling transmission rate. Traffic control (*tc*) is part of the Linux kernel which allows the user to access networking features. It has three main features: monitoring the system, traffic classification, and traffic manipulation. The utility *netem* is an extension of *tc*, which emulates packet loss and delay on a Linux system. The server runs the Apache web server and hosts video chunk files. Chunks are stored in multiple directories based on their bitrates according to the MDP file[3].

In the following subsections, the default parameters of our network are as follows. The bandwidth between the client and server is set to 10 Mbps, packet loss is set to 1%, and RTT is set to 50 ms. Note that the size of each chunk depends on the throughput obtained when downloading the preceding chunk. If the measured throughput is high, then the client requests a chunk with higher bitrate which also has a larger size. This form of rate adaption is at the core of *adaptive* video streaming such as in DASH.

The throughput estimate is obtained by computing the average throughput of each chunk during the transmission of the entire video file from the server to client. The video file is divided into 40 chunks each of which contains 15 seconds of the actual video. The throughput estimate is then obtained by calculating the average of 40 throughputs computed for the chunks.

D. Measurement Results and Discussion

1) Effect of Packet Loss: Fig. 5 shows the effect of loss probability on the aggregate throughput when using parallel connections. As discussed earlier, the use of multiple connections alleviates the negative effect of packet loss on TCP throughput. It is observed that as the number of parallel connections increases, the aggregate throughput converges to a value close to the link capacity regardless of the actual loss probability. In particular, when the loss probability is high (as in wireless networks), the use of parallel connections is more effective.

2) Effect of Round Trip Time: It is well-known that as RTT increases the TCP throughput decreases. Fig. 6 shows the effect of RTT on the aggregate throughput when using parallel connection. Again, as the number of connections increases so does the aggregate throughput though the amount of improvement somewhat depends on the value of RTT. In particular, in scenarios when the RTT is large (*e.g.*, in high-latency networks) more connections are required to achieve high utilization of the link. The reason is that TCP performs so poorly in such scenarios that there is a lot of room for improvement by adding more connections.

3) Effect of Bandwidth: The effect of the available bandwidth between the client and server on the aggregate throughput is depicted in Fig. 7. It is observed that using parallel connections helps utilize most of the available bandwidth as opposed to a single connection, which is oblivious to the link bandwidth in this scenario. It is also observed that as the available bandwidth increases, the number of parallel connections required to utilize the bandwidth increases as well. This is expected based on the analysis presented earlier for the relation between the number of flows and the aggregate throughput. As the link bandwidth increases, TCP throughput becomes bounded by the loss probability and RTT. Thus, a single connection is not able to achieve higher throughput even if the link bandwidth increases beyond a limit. In this scenario, parallel connections are necessary to utilize the full bandwidth.

4) Effect of Chunk Size: Fig. 8 depicts the average time to download a chunk for different chunk sizes. As expected, increasing the number of parallel connections results in lower download time. However, as can be seen, only a few connections (*e.g.*, two) are enough to achieve most of the benefits of parallel connections. An important observation is that as the chunk size increases using parallel connections becomes more effective. Note that when streaming video at high quality, each chunk has a larger size. Thus, using parallel connections is particularly useful when streaming high quality video. A lower download time means less buffering delay for video playback, which directly translates to a better quality of experience.

III. Number of TCP Connections

So far, we have shown that it is feasible to use parallel TCP connections to download each chunk of a video file, and that this approach results in higher throughput and lower chunk download time. An important question then is how many parallel connections are required under different network conditions (*e.g.*, RTT, loss rate, *etc*.). Increasing the number of TCP connections adds complexity to the client application. In particular, mobile devices are usually less powerful compared to desktop machines in terms of computational power. Most web browsers on mobile devices actually limit the number of parallel connections per-server for this reason (*e.g.*, Google Chrome limits the number of parallel connections to 6). Therefore, we like to have as few parallel connections as possible. In general, it is difficult to find the precise number of connections required since it depends on dynamic network conditions, the interplay between TCP congestion control and buffer management policies at routers, and specific TCP implementation details such as RTO estimation and so on.

In this section, we develop a simple model to estimate the number of parallel connections required to fully utilize the bottleneck link of the end-to-end path. We also present ns-3 simulation results to verify the accuracy of our model in various network scenarios. Since our focus is on wireless networks, we consider a scenario where the bottleneck link is at the edge of the network (*i.e.*, the wireless access link is the bottleneck) as depicted in Fig. 2. We consider two cases in the following subsections depending on the size of the transmit queue of the router that is attached to the bottleneck link. Note that in order to forward packets on its outgoing link, the router

[3]The video file and dataset is obtained from http://www-itec.uni-klu.ac.at/ftp/datasets/DASHDataset2014/BigBuckBunny/

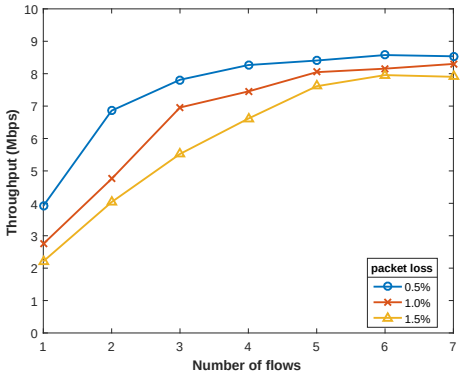


Fig. 5: Effect of packet loss on aggregate throughput.

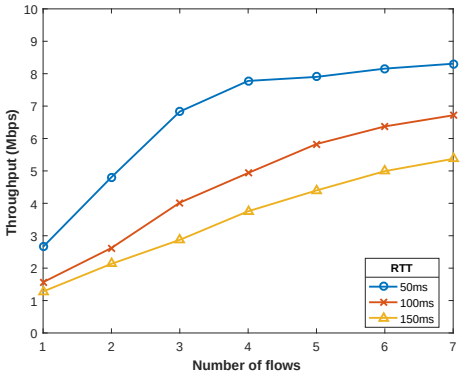


Fig. 6: Effect of RTT on aggregate throughput.

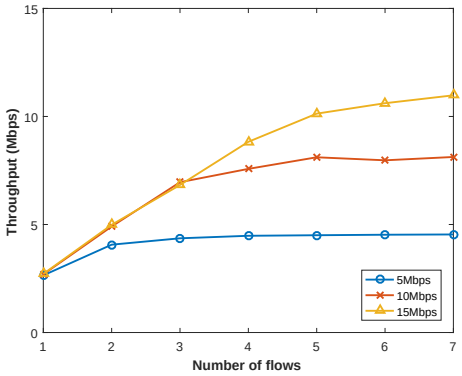


Fig. 7: Effect of link bandwidth on aggregate throughput.

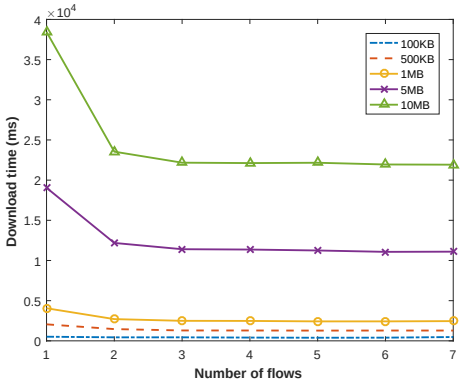


Fig. 8: Effect of chunk size on download time.

first copies the packets to the transmit queue associate with that link. Once the outgoing link becomes available then the packet at the head of the queue is transmitted.

A. Small Transmit Queue

In this case, congestion losses due to buffer[4] overflow at the router dominate the packet loss process seen by TCP. We consider two models to approximate the aggregate throughput of parallel connections in this case.

1) Fluid Approximation: Assume that there is only one flow between the client and server. A commonly used approximation for the TCP throughput is given by $X = \frac{3}{4}C$, where C denotes the capacity of the bottleneck access link [21]. Clearly, a single flow is not able to fully utilize the bottleneck link in this case. To determine the effect of multiple flows, we make the following (optimistic) approximation about the aggregate throughput (thus, the result is an upper bound). We assume that the first flow utilizes $3/4$ of the link bandwidth and the additional flows utilize the remaining $1/4$ of the bandwidth. Thus, the aggregate throughput with N parallel connection, denoted by X_N, is given by,

$$X_N \approx X_{N-1} + \frac{3}{4}(C - X_{N-1}) \quad (5)$$

$$= \frac{3}{4}C + \frac{1}{4}X_{N-1}, \quad (6)$$

which yields,

$$X_N \approx (1 - \frac{1}{4^N})C\,.$$

As can be seen from the above equation, while multiple flows improves the aggregate throughput, only a few flows are sufficient to fully utilize the bottleneck link.

The fluid approximation is valid for large number of flows (*i.e.*, the asymptotic fluid regime). For small number of flows however, it is inaccurate. Specifically, if the size of the transmit queue is 0, in reality, the achieved throughput of a single flow will be 0, and not 75% of C as predicted by this model. However, in practice, routers do have non-negligible buffers. In fact, most edge routers (such as the WiFi APs) suffer from the bufferbloat problem [22] due to excessive buffering at the router. Thus, the fluid approximation can be applied, and later, using ns-3 simulations we show that it is a reasonable approximation in some scenarios.

2) Packet Approximation: Consider a router with transmit queue of size 1 packet. The time to transmit a packet over the bottleneck link is given by L/C, where L denotes the length of the packet. For the sake of argument, assume that the system operates in a discrete time manner where time is divided into time slots of length L/C (long enough to transmit one full packet). In each time slot, zero or more packets arrive at the router but at most one is transmitted and the rest are dropped.

[4]We use the terms "buffer" and "queue" interchangeably.

Let

$$K = \frac{R}{L/C}, \tag{7}$$

denote the number of time slots in an RTT. Consider the following two cases:

1) *One Flow Between Client and Server*
   In this case, the server transmits one packet in the first RTT, receives the corresponding ACK and increases its window to two packets. In the second RTT, it sends two packets back-to-back. The first packet arrives at the router and is immediately scheduled for transmission. The second packet arrives while the first one is being transmitted (recall that the bottleneck access link is slower than the Internet link) and hence is dropped. After some time, the server receives an ACK for the first packet and increases its windows size to 2.5 packets. However, it never receives an ACK for the second packet which results in a time-out. It follows from this argument that the flow throughput is bounded by:

$$X = \frac{L}{R + L/C + RTO}\,. \tag{8}$$

   In this equation, RTO denotes the retransmission time-out which typically has a minimum value of 1 second in practice [23] (but 200 ms in ns-3).
2) *Multiple Flows Between Client and Server*
   In this case, the aggregate throughput is clearly a function of the degree of synchronization among the flows. If they are fully synchronized, then the aggregate throughput is equal to the throughput achieved by a single flow. If they are fully desynchronized, then the bottleneck link could be fully utilized if there are sufficient number of flows (at least K flows). In this case, the aggregate throughput increases linearly with the number of flows until it reaches the bottleneck link capacity C. In the following, we consider the case when N flows are randomly synchronized. For ease of exposition, ignore RTO and packet transmission time L/C. With "random synchronization", we assume that each flow transmits in a time slot that is uniformly chosen at random in one RTT, which has K time slots. The probability that a given time slot remains idle (no flow transmits during that time slot) is given by $(1-\frac{1}{K})^N$. Thus, the expected aggregate throughput is given by:

$$X_N = \Big(1 - \big(1 - \frac{1}{K}\big)^N\Big)C\,. \tag{9}$$

   Since K is constant, as N increases, the aggregate throughput approaches C quickly, which is consistent with the fluid model's prediction.

B. Large Transmit Queue

In this case, even one connection could fully utilize the bottleneck link if a sufficiently large transmit queue is installed on the router (in contrast to what is predicted by some existing models described in the Introduction section). Given a large queue, the packet loss process seen by TCP is dominated by *random transmission losses*. For a TCP connection with round trip time R and transmission loss probability p, the average throughput (in units of "segments") is given by [21]:

$$X = \frac{\phi}{R\sqrt{p}}, \tag{10}$$

where, ϕ is a constant that depends on the specific congestion control algorithm used. For TCP New Reno, we have $\phi \approx 0.3$. Our goal is to fully utilize the access link. Let Q denote the average queue length that is required to fully utilize the link. Also, let τ denote the two-way propagation delay between the client and server. We have, $R = \tau + Q/C$, and thus,

$$(1-p)X = C, \tag{11}$$

After substitution for X, it yields,

$$(1-p)\frac{\phi}{(\tau + Q/C)\sqrt{p}} = C\,. \tag{12}$$

Therefore,

$$\begin{aligned} \frac{\phi(1-p)}{C\sqrt{p}} &= (\tau + Q/C) \\ \Rightarrow Q &= \frac{\phi(1-p)}{\sqrt{p}} - \tau C\,. \end{aligned} \tag{13}$$

In order to have $Q \geq 0$, the following condition should be satisfied,

$$\begin{aligned} \frac{\phi(1-p)}{\sqrt{p}} &\geq \tau C \\ \overset{\approx}{\Rightarrow} p &\leq \left(\frac{\phi}{\tau C}\right)^2\,. \end{aligned} \tag{14}$$

In other words, if the packet loss probability *p* is very small, then even a single flow can fully utilize the bottleneck link. However, if the packet loss probability p is large (specifically, $p > (\phi/(\tau C))^2$) then, one flow cannot fully utilize the link. In this case, multiple flows are required to achieve an aggregate throughput of C. The number of flows can be computed as follows:

$$X_N = (1-p)NX = C\,. \tag{15}$$

Thus, we obtain that,

$$\frac{N\phi(1-p)}{R\sqrt{p}} = C, \tag{16}$$

and, consequently,

$$Q = \frac{N\phi(1-p)}{\sqrt{p}} - \tau C\,. \tag{17}$$

Given that $Q > 0$, we have,

$$\frac{N\phi(1-p)}{\sqrt{p}} \geq \tau C, \tag{18}$$

which yields the following relation for the number of flows,

$$N \geq \frac{(\tau C)\sqrt{p}}{\phi(1-p)}\,. \tag{19}$$

IV. Simulation Results

To have full control over network parameters such as the transmit queue size of the router, we use ns-3 to simulate the network topology depicted in Fig. 2 and study the accuracy of our model presented in Section III. We note that ns-3 is widely used in the literature on TCP and is believed to simulate TCP behavior accurately. With this setup, we can conveniently change the size of the router's transmit queue to arbitrary values to simulate *small queue* and *large queue* scenarios.

The network topology in both small queue and large queue scenarios, is the same topology depicted in Fig. 2. The access link's bandwidth is set to 10 Mbps and its delay is set to 20 ms. The Internet link's bandwidth is set to 100 Mbps and it has a delay of 5 ms. Thus RTT between the client and server is 50 ms. This setup makes the bottleneck link to be at the edge of the network. It is set in such a way that it recreates the common setup of a wireless network where the access link (the link between user's device and access point) has high latency and a relatively low bandwidth. The transmit queue is set on the access link side of the router where packets from the server are being sent to the client. By default, the packet loss probability in small and large queue scenarios is set to zero and 0.1%, respectively, unless otherwise specified. The TCP segment size in all simulations is set to 1448 bytes.

Each simulation run starts with a given number of parallel TCP connections and lasts for 100 seconds. During the simulation, the server continuously sends data to the client. If the packet loss is set to a number other than zero, then simulations are repeated for 10 times, each time with a different seed. The throughput is then obtained by calculating the average throughput over the 10 simulation runs.

A. Small Transmit Queue

We consider both fluid and packet models, as described earlier.

1) Fluid Approximation: We set the transmit queue size to 1, 5 and 10 packets and compare our fluid model with ns-3 simulations. The results are presented in Fig. 9. As explained in the previous section, the fluid model is not an accurate approximation when the queue size is very small. This is clearly visible in the figure for queue size of 1, where the model is quite far from the simulation. However, as soon as we increase the queue size slightly, the model becomes more accurate. In particular, the model always provides an upperbound on the aggregate throughput as expected. Moreover, the model is able to predict the behavior of the aggregate throughput with respect to the number of flows reasonably well specially for large number of flows.

2) Packet Approximation: Recall that when only one flow passes through the router, the TCP window size is bounded by 2.5 packets. Fig. 10 depicts the time series of the TCP window size for one flow when the router transmit queue is set to 1 packet. It can be seen from the figure that once the connection stabilizes, the window size is range bound between 1 and 2.5 packets. Using our model, the expected throughput in this case is 0.046 Mbps. From simulations, we obtain the throughput of 0.05 Mbps, which is remarkably close to the value predicted by the model.

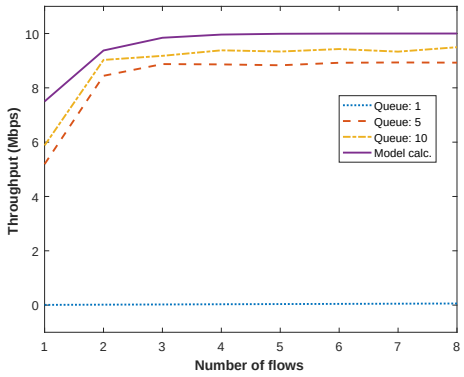


Fig. 9: Small transmit queue: Comparison of the model and simulation when the packet loss probability due to transmission errors is set to zero. The model provides an upperbound, which becomes more accurate as the queue size increases.

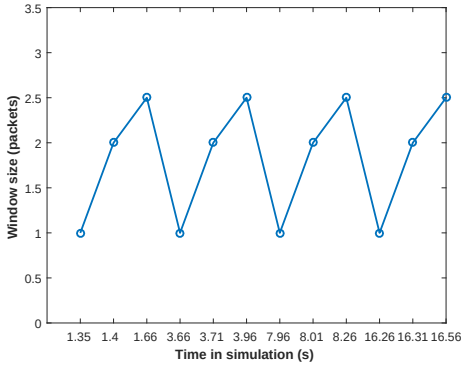


Fig. 10: Small transmit queue: Trace of TCP window size when queue is one packet.

When there are multiple flows, as discussed in Section III, the degree of synchronization among the flows is critical in determining the aggregate throughput. In real word scenarios, the synchronization degree is dynamic and varies over time. Since it is difficult to simulate scenarios in which flows are either completely synchronized or desynchronized, we simulate two scenarios where: 1) all connections are started at the same time, or 2) connections are started with a delay from each other in a staggered manner. The results are presented in Fig. 11. As expected, as the degree of synchronizing among flows decreases, the aggregate throughput increases. It would be interesting to develop a throughput model for multiple flows that could consider the synchronization degree among the flows as a parameter.

B. Large Transmit Queue

In this case, we set the size of the transmit queue at the router to 100 packets. Table I presents the number of flows required to fully utilize the access link as computed using the model presented in Section III. According to the table, for

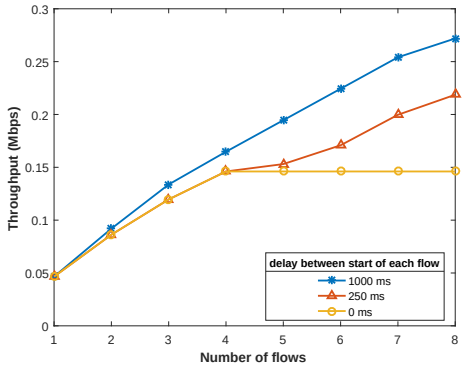


Fig. 11: Small transmit queue: Aggregate throughput increases as the gap between the start time of connections increases.

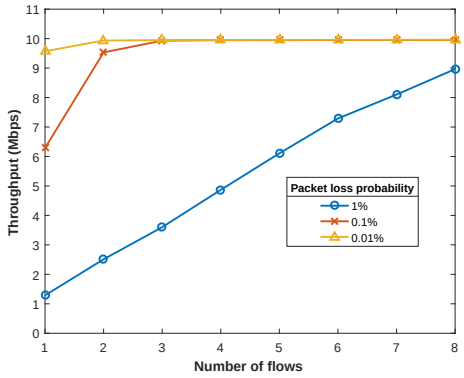


Fig. 12: Large transmit queue: Aggregate throughput under different loss probabilities.

instance, at least 8 flows are required to fully utilize the link when packet loss probability is set to 1%.

TABLE I: Number of TCP flows from the model.

Loss probability	Number of flows
1%	8
0.1%	2
0.01%	1

Using the same parameters as the ones specified for the model, we have computed the aggregate throughput achieved for different loss probabilities and number of flows. The results are presented in Fig. 12. It can be seen that for 1% loss probability, using 8 flows can achieve over 90% link utilization. Similarly, for 0.1% and 0.01% loss probabilities, using 2 and 1 flows can achieve high link utilization of over 95%.

Recall that our model also provides an upper bound on the loss probability that allows full link utilization. To verify this, we have conducted a simulation study with one TCP connection. From the model, we obtain that the loss probability should be less than 0.019 to allow full utilization. The simulation results are presented in Fig. 13. The figure shows that when the packet loss probability is 0.02, the link utilization

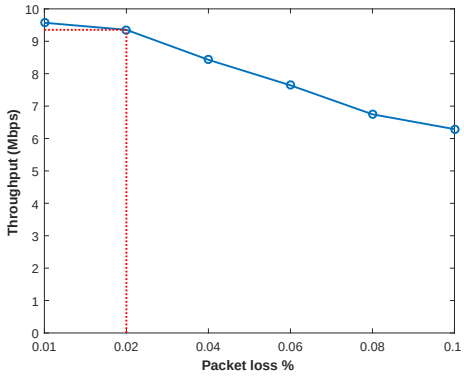


Fig. 13: Large transmit queue: When packet loss is 0.02, the link utilization is over 93%. Beyond this, the link utilization drops.

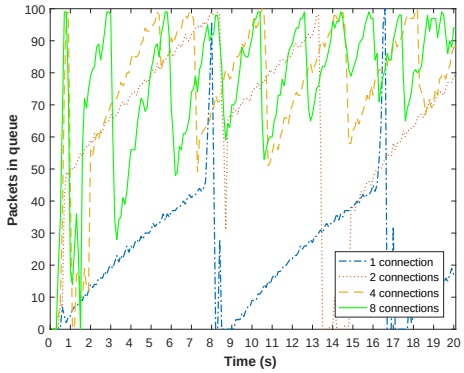


Fig. 14: Transmit queue occupancy with different number of flows. The size of the queue is set to 100 packets.

reaches over 93%.

C. Transmit Queue Occupancy

The transmit queue of the router plays a critical role in determining the utilization of the bottleneck link. A higher occupancy for the queue indicates a higher link utilization in general. To show the effect of multiple flows on the transmit queue occupancy, we run a simulation experiment with transmi queue size set to 100 packets.

Transmit queue traces over a period of 10 seconds are presented in Fig. 14. It can be seen that, while the queue occupancy fluctuates over time, having multiple flows generally results in higher occupancy rates. In Table II, we show the average number of packets in the queue (*i.e.*, the average queue occupancy) for different number of flows along with the achieved aggregate throughput. As expected, increasing the number of parallel flows results in increased average queue occupancy. Consequently, as the queue occupancy increases so does the aggregate throughput.

V. Conclusion

In this work, we introduced ParS, a video streaming system that uses parallel TCP connections in conjunction with HTTP range requests to speedup downloading video chunks from the

TABLE II: Average queue occupancy from simulations.

No. flows	Tput (Mbps)	Avg. packets in queue
1	8.839	22.62
2	9.623	64.49
4	9.785	74.64
8	9.837	76.45

server. We implemented a prototype of the system on a testbed and conducted various measurement experiments to study its performance. We then developed a model to estimate the minimum number of parallel connections required to achieve high utilization of the available bandwidth in the network. Simulation results conducted using ns-3 demonstrated the accuracy and utility of our model. In future, we plan to extend our implementation to mobile devices and conduct live measurements on WiFi and LTE networks.

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