Beyond Bufferbloat: End-to-End Congestion Control Cannot Avoid Latency Spikes

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Abstract—End-to-end congestion control is the main method of congestion control in the Internet, and achieving consistent low queuing latency with end-to-end methods is a very active area of research. Even so, achieving consistent low queuing latency in the Internet still remains an unsolved problem. Therefore, we ask “What are the fundamental limits of end-to-end congestion control?” We find that the unavoidable queuing latency for best-case end-to-end congestion control is on the order of hundreds of milliseconds under conditions that are common in the Internet. Our argument depends on two things: The latency of congestion signaling – at minimum the speed of light – and the fact that link capacity may change rapidly for an end-to-end path in the Internet.

Index Terms—Computer network performance, Traffic control (communications), Transport protocols

I. INTRODUCTION

End-to-end congestion control methods, such as those used by TCP and QUIC, are the main ways of avoiding congestion in the Internet. All end-to-end congestion control methods function through the interaction of two main mechanisms: The first is congestion signaling, which is how congestion signals are generated and how signals are sent to the traffic sources. The other mechanism is rate adaptation, which is how the traffic sources react to congestion signals. The measure of success for a congestion control scheme was originally that all competing flows achieved the same average throughput [1]. Recently proposed methods have focused on achieving both fair bandwidth sharing and low loss as well as low queuing delay [2] [3].

In this work we present a fundamental limitation of end-to-end congestion control (CC). We have reviewed two recent survey papers on low latency congestion control research [3] [4]. A striking feature of the surveys is that all of the surveyed work on low-latency CC has focused on algorithms that converge to a steady state where queuing delays are small. Latency during the time before an equilibrium is reached (the transient period), is ignored despite the fact that short-lived latency spikes are bad for application performance and user experience. None of the surveys mention the issue of transient latency spikes, and to our knowledge there are no works in the congestion control literature which describe the signaling constraint we present.

Fundamental limitations are highly relevant to the performance analysis of new methods because they serve as a theoretically optimal benchmark which new methods can be measured against. The latency spike problem described in this work may seem very simple, yet we have been unable to find a discussion of the specific limitation we describe in any of the recent surveys on the topic, so it seems clear that the issue requires attention.

The contributions of this paper are twofold. First, we analyze a theoretically optimal end-to-end congestion control algorithm where both congestion signaling and sender rate adaptation are perfect. Second, we show that under conditions that are common in the Internet, our theoretically optimal end-to-end congestion controller creates large transient delays.

II. RELATED WORK

Improving latency and packet loss in the Internet through improved end-to-end congestion control has been discussed in the literature since the conception of the Internet and is still a very active area of research. In this section we give an overview of what we consider as the most important results.

A. Improvements to congestion signaling

The original way of signaling congestion in the Internet was to drop packets that arrive at a full queue [1]. Nichols and Van Jacobson later proposed CoDel, an algorithm that improves congestion signaling by dropping head-of-line packets when queuing delay exceeds a threshold for too long [5]. In 2013, Cisco introduced PIE. PIE utilizes a PI-style controller to randomly drop packets based on the queue delay and rate of packet departures from the queue [6].
L4S and SCE are two methods for signaling congestion in the Internet protocol without dropping packets, proposed by Briscoe et al. and Morton et al. respectively. The two methods are variations of the same idea: When a network element notices that load exceeds capacity, the element starts marking packets by setting the Explicit Congestion Notification (ECN) bit in the packet headers. ECN is a more elegant congestion signal than packet loss because it avoids losing packets. The difference between L4S and SCE is in the details of when packets are tagged and how the traffic sender interprets the ECN signal.

In section III we analyse a theoretical congestion signaling method. In fact, our method is “impossibly perfect” because we assume the congestion is detected immediately, before a queue starts to build, and that the congestion signal includes perfect information about the capacity of the link.

B. Rate adaptation at the sender

The original way to do rate adaptation in TCP is described by Van Jacobson in [1]. This method has been widely successful. It has been thoroughly tested, expanded and improved upon [8, 3]. Common to variations of this method is that the sender decides on a transmit rate based on feedback received either through the TCP ACKs, the observed round-trip delay, or other signals from the network element where congestion occurs.

In section III we analyse the performance of a theoretically optimal rate adaptation method. We assume the sender will receive perfect information directly from the bottleneck interface, and that it immediately configures the perfect sending rate upon receiving a congestion notification. This is of course practically unrealistic, but we make these best-case assumptions so that our analysis is firmly a performance bound — no real-world implementation can possibly do better.

C. Overviews of the state of the art

Two recent surveys of end-to-end congestion control algorithms for 4G/5G networks by Haile et al. and Lorincz et al. both conclude that nobody have been able to solve the problem of reliable low-latency congestion control for cellular networks without sacrificing utilization. The authors cite large capacity variations, especially in 5G mmWave frequencies, as a main reason why this problem is so hard. Both surveys argue that the magnitude of capacity variations are likely to increase with the deployment of 5G, and the bandwidth vs. latency trade-off and the bufferbloat problem are listed as future challenges.

D. Bufferbloat

Bufferbloat has been identified as a common source of latency in the Internet. The exact definition of Bufferbloat is somewhat unclear. Nichols and Van Jacobson define Bufferbloat as “unnecessary queuing” [5]. Unnecessary queuing is caused by poor congestion signaling leading to a standing queue at bottleneck interfaces [9]. On the other hand, according to Gettys, any delay above the speed of light is too much [9]. Bufferbloat as defined by Nichols and Van Jacobson can be solved with better end-to-end congestion control, and it is partly solved by the CoDel algorithm [5]. However, to get close to Gettys’ ideal speed-of-light network, removing standing queues is not enough. In section III we show that transient queues, the “good queues” of [5], are large enough to contribute significantly to performance problems.

E. Measuring latency in the Internet

Høiland-Jørgensen et al. analyzed a large dataset of latency measurements in the Internet. It is not clear from measurements studies like this how much of the latency is caused by standing queues, and how much is caused by transients. What is clear, however, is that latency is large, highly variable and that this poses a barrier to certain use-cases for the Internet.

F. Fundamental limits of end-to-end congestion control

Chong et al. was the first to point out that end-to-end congestion control converges at a rate inversely proportional to the round-trip time. Johari and Tan develop a model of end-to-end congestion control with propagation delays. Their main conclusion is that end-to-end congestion control can be used to guarantee bounded load, and that they quantify the requirements that end-nodes and network elements must obey for the result to hold.

The results of Chong et al. and Johari and Tan define some limitations on end-to-end congestion control methods. Whereas their work focuses on the rate of convergence and criteria for stability, our work focuses on the magnitude of latency in the transient period.

III. Analysis

In this section we describe and analyze a theoretically optimal end-to-end congestion control algorithm.

Consider a sender, $A$, transmitting a capacity-seeking flow to a receiver $B$ through a bottleneck interface $X$. $A$ aims to use all of the available capacity of interface $X$, and we therefore assume that interface $X$ is fully loaded with traffic from $A$. Figures 1 and 2 show how packets flow in this setup.

Our analysis centers on what happens if the capacity of $X$ is suddenly and unpredictably reduced.

To make the analysis as simple as possible, we make several simplifications, some of which are “supernatural” in the sense that they break the laws of physics. By over-estimating the capabilities of the end-to-end controller we ensure our results can safely be regarded as best case performance for any real world implementation. In other words, we are modelling the most favourable case for end-to-end congestion control, and
thereby produce an upper bound on the performance of end-to-end congestion control algorithms in general.

In a typical real-world network, the interface \( X \) does not know how to talk to \( A \). Congestion signals are therefore normally sent by dropping or tagging packets and waiting for that information to pass through \( B \) before going through the reverse path from \( B \) to \( A \) as shown in Figure 2b. This is how TCP and QUIC does congestion signaling. In our analysis we assume \( X \) can talk directly to \( A \), as shown in Figure 2c, because this is the best-case scenario for any congestion signaling method.

We make the following simplifying assumptions about the end-to-end congestion control method:

- A congestion signal is created at \( X \) the instant capacity is reduced. The signal includes information about the new capacity of \( X \).
- The congestion signal travels from \( X \) to \( A \) in the shortest possible time \( d \).
- On receiving the signal, \( A \) reacts immediately by reducing load to exactly match the new capacity at \( X \).
- There is no queue at \( X \) when the capacity drops.
- There is infinite buffer space at \( X \) such that no traffic is lost.
- For simplicity, we ignore the fact that \( A \) must reduce load below the capacity at \( X \) to allow the queue to empty.

Now, what happens if the capacity of interface \( X \) is suddenly reduced?

### A. A step change in capacity

Assume \( X \) has an original capacity of \( T \) Mbit/s. The capacity of \( X \) is now instantaneously reduced by a factor \( C \), such that the new capacity, \( T_{\text{new}} \) is given by equation (1).

\[
T_{\text{new}} \text{ Mbit/s} = \frac{T \text{ Mbit/s}}{C}
\]  

(1)

When the capacity of \( X \) is reduced, \( X \) immediately sends a congestion signal towards \( A \). Figure 2c shows the situation moments before \( A \) receives the congestion signal (shown in green). The earliest time at which \( A \) knows about the capacity reduction is after \( d \) ms, where \( d \) is the minimum delay from \( X \) to \( A \). The reduction in capacity implies that the per-packet processing time at \( X \) increases by a factor \( C \). \( X \) now needs \( C \) milliseconds to transmit the amount of data that took 1 millisecond before the capacity reduction. Therefore, the arrival rate at \( X \) is now \( C \) ms of traffic every millisecond. A queue is now building up at \( X \) at a rate of \( (C - 1) \) ms/ms. Because \( d \) milliseconds pass before \( A \) receives the congestion signal and reduces the rate, the size of the queue at \( X \) will peak at \( (C - 1) \times d \) ms.

The peak transient delay, \( Q \), produced as a result of instantaneous capacity reduction is given by equation (2). Figure 3 illustrates how the queue delay can be visualized as the area spanned out by \( d \) and \( (C - 1) \). Notice that \( Q \) is independent of the throughput of \( X \).

\[
Q = (C - 1) \times d
\]  

(2)

FIGURE 2: A visualization of end-to-end congestion control

| Technology (Channel BW, # of antennas) | Note | Rate        |
|----------------------------------------|------|-------------|
| WiFi 802.11b                           |      | Min rate 1 Mbit/s |
| WiFi 802.11b                           |      | Max rate 11 Mbit/s |
| WiFi 4 (20MHz, 2x2)                    |      | Min rate 14.4 Mbit/s |
| WiFi 4 (20MHz, 2x2)                    |      | Max rate 144.4 Mbit/s |
| WiFi 5 (20MHz, 2x2)                    |      | Max rate 173.3 Mbit/s |
| WiFi 5 (40MHz, 2x2)                    |      | Max rate 400 Mbit/s |
| WiFi 5 (80MHz, 2x2)                    |      | Max rate 866.7 Mbit/s |

TABLE I: Some of the possible capacity levels in WiFi

Figure 4 shows how this function scales as \( d \) and \( C \) changes. The value of \( Q \) grows to several hundred milliseconds for the selected ranges of \( d \) and \( C \).

### IV. Discussion

What are some typical values for \( C \) and \( d \) in the Internet?

#### A. The data center

Low latency and low loss end-to-end congestion control can work well in a data center because the short physical distances between machines mean signaling delay \( d \) is very small. As long as \( C \) is also kept small enough, transient queuing delays can be kept at a manageable level.

#### B. Magnitude of capacity changes in wireless technologies

In wireless technologies, the link rate is dynamically adapted based on channel conditions. Table II shows some of the possible link rates for different generations of the WiFi protocol.
Channel conditions depend on many factors, such as link distance, the reflection and absorption properties of the things the radio signal encounters, competing traffic sources, antenna placement, antenna quality, etc. The number of factors is much too large to make an exhaustive list here. The consequence of all these factors is that link rate can change frequently, rapidly, and in large jumps.

The possible modulation and coding schemes (MCS) of WiFi span from 1 Mbit/s to several thousand Mbit/s. In theory, $C$ can therefore have a value of several thousands. In addition to the throughput variation caused by MCS rate adaptation, the WiFi channel may be in use by more than one radio. The resulting contention for the wireless channel is another source of capacity variation [15].

To summarize, values of $C$ in the range 1-3 must be regarded as routine in wireless networks because they can be caused by something as common as a competing wireless radio initiating use of the channel. $C$ values in the range 3-10 are very plausible for wireless networks with many users or fast-moving radios. In theory, $C$ can take values of several thousands without the link actually going down.

C. Worked example for the Internet

At the speed of light, information travels from Dublin to New York in 17 milliseconds. Let us assume we are having a transatlantic video conference using WiFi at both ends, where step changes in capacity of magnitude 10 may occur. Using equation $2$, we can calculate the relationship between signaling delay $d$ and transient queuing delay $Q$ for our imagined video call. Table II shows theoretical estimates and actual measurements of one-way delay for an Internet connection from Dublin to New York. We have also listed the theoretically optimal transient queue delays $Q$, which shows that some lag-spikes must be expected for the video conference whenever the WiFi link capacity drops.

D. What if the capacity reduction is not instant?

The reduction in link capacity may sometimes be more gradual, for instance if the reduction is due to competing traffic gradually ramping up. In this case, the transient queue delay $Q$ will not be as large as shown in Figure 4. But as we shall see, the transient delays are still large unless the capacity reduction happens over a relatively long period. Inspecting figures 3 and 5, it is easy to see that if the capacity drop is linear over a duration $d$, then the transient queue delay is reduced by 50% compared to the instant capacity drop. Similarly, if the capacity is reduced linearly over a period of $2d$, the transient queue is reduced by 75%. In general, if the capacity is linearly reduced over a period $d_{\text{ramp}}$, the peak transient delay, $Q$, is given by equation $3$.

$$Q = \begin{cases} \frac{(C-1)(2d-d_{\text{ramp}})}{2d_{\text{ramp}}} & d_{\text{ramp}} \leq d \\ \frac{(C-1)d^2}{2d_{\text{ramp}}} & d_{\text{ramp}} \geq d \end{cases}$$

Table II: Dublin to New York one-way delays

|                      | $d$     | $Q$ for $C = 10$ |
|----------------------|---------|-----------------|
| Speed of light       | 17 ms   | 135 ms          |
| Theoretically Optimal LEO Satellite [13] | 20.07 ms | 180.63 ms       |
| Theoretical Optical Terrestrial Cable [13] | 25.07 ms | 225.63 ms       |
| Internet measurements [14] | $>38.5$ ms | $>346.5$ ms     |

Equation 3 describes a scaling property of end-to-end congestion control methods. Let us assume a network where all

![Fig. 3: Instant capacity drop](image1)

![Fig. 4: Minimum queuing delay with end-to-end congestion control and no packet loss](image2)

![Fig. 5: Linear capacity drop](image3)
drops in capacity are due to competing traffic ramping up. The magnitude of latency spikes in this network will be decided by the worst-case signaling delay $d$, and the number of flows that ramp up at the same time (because this determines the maximum value of $C$). Knowing $d$ and $C$ we can find the relationship between $d_{\text{ramp}}$ and $Q$. For a world-wide network, we may somewhat generously assume that the worst-case signaling delay is 100 ms since the speed of light half-way around the world is approximately 67 ms. This assumption yields the graphs shown in Figure 6 for various choices of $C$.

While interesting for understanding end-to-end congestion control schemes, the implications of equation 3 are moot in the current Internet. Rapid changes in capacity frequently occur in the Internet as a result of things such as configuration changes, routing changes, and, for wireless links; varying channel conditions. Therefore, we argue that equation 2 is the most appropriate analysis for the current Internet.

V. Conclusion

In a world-wide network where large and rapid drops in capacity are likely to happen, it is impossible for end-to-end congestion control to avoid large spikes in latency. We have shown how the magnitude of the latency spikes is determined by the signaling delay of congestion notifications and the scale and speed of the capacity variations. A solution to this problem has to reliably achieve low latency end-to-end communication over the internet, including the ability to avoid latency spikes by reacting quickly enough to capacity variations. Research on improving latency and packet loss in the Internet must take this fundamental limitation of end-to-end congestion control into account. Our results are especially relevant for work on congestion control for wireless links, including WiFi and 5G, because of the large and frequent capacity variations in wireless communications.

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