

ACP: An End-to-End Transport Protocol for Delivering Fresh Updates in the Internet-of-Things

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ABSTRACT

The next generation of networks must support billions of connected devices in the Internet-of-Things (IoT). To support IoT applications, sources sense and send their measurement updates over the Internet to a monitor (control station) for real-time monitoring and actuation. Ideally, these updates would be delivered at a high rate, only constrained by the sensing rate supported by the sources. However, given network constraints, such a rate may lead to delays in delivery of updates at the monitor that make the freshest update at the monitor unacceptably old for the application.

We propose a novel transport layer protocol, namely the Age Control Protocol (ACP), that enables timely delivery of such updates to monitors, in a network-transparent manner. ACP allows the source to adapt its rate of updates to dynamic network conditions such that the average age of the sensed information at the monitor is minimized. We detail the protocol and the proposed control algorithm. We demonstrate its efficacy using extensive simulations and real-world experiments, which have a source send its updates over the Internet to a monitor on another continent.

1. INTRODUCTION

The availability of inexpensive embedded devices with the ability to sense and communicate has led to the proliferation of a relatively new class of real-time monitoring systems for applications such as health care, smart homes, transportation, and natural environment monitoring. Devices repeatedly sense various physical attributes of a region of interest, for example, traffic flow at an intersection. This results in a device (the *source*) generating a sequence of packets (*updates*) containing measurements of the attributes. A more recently generated update contains a more current measurement. The updates are communicated over the Internet to a *monitor* that processes them and decides on any actuation that may be required.

For such applications, it is desirable that freshly sensed information is available at monitors. However, as we will see, simply generating and sending updates at a high rate over the Internet is detrimental to this goal. In fact, freshness at a monitor is optimized by the source

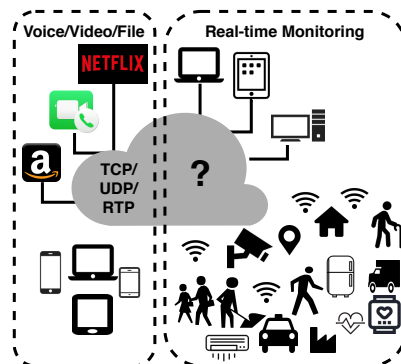


Figure 1: Unlike the traditional applications like voice, video or file download; real-time monitoring applications are highly loss resilient and care for freshness of an update. Existing transport protocols like TCP and RTP are not best suited for such applications.

smartly choosing an update rate, as a function of the end-to-end network conditions. Freshness at the monitor may suffer significantly when a too small or a too large rate of updates is chosen by the source. In this work, we propose the Age Control Protocol (ACP), which in a network-transparent manner regulates the rate at which updates from a source are sent over its end-to-end connection to the monitor. This rate is such that the average age, where the age of an update is the time elapsed since its generation by the source, of sensed information at the monitor is kept to a minimum, given the network conditions. Based on feedback from the monitor, ACP adapts its suggested update rate to the perceived congestion in the Internet. Consequently, ACP also limits congestion that would otherwise be introduced by sources sending to their monitors at unnecessarily fast update rates. The requirement of freshness is not akin to requirements of other pervasive real-time applications like voice and video. For these applications, the rate at which packets are sent is determined by the codec being used. This also determines the network bandwidth used and the obtained voice/video quality. Often, such applications adapt to network conditions

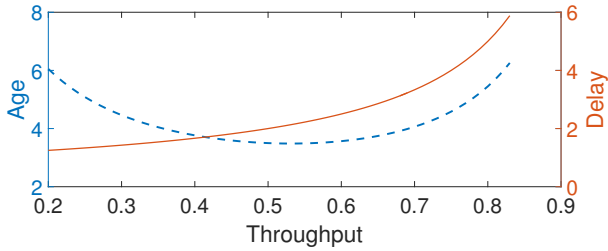


Figure 2: Example interplay of the networking metrics of delay (solid line), throughput (normalized by service rate) and age. Shown for a M/M/1 queue [29] with service rate of 1. The age curve was generated using the analysis for a M/M/1 queue in [20].

by choosing an appropriate code rate. These applications, while resilient to packet drops to a certain degree, require end-to-end packet delays to lie within known limits and would like small end-to-end jitter. Monitoring applications may achieve a low update packet delay by simply choosing a low rate at which the source sends updates. This, however, may be detrimental to freshness, as a low rate of updates can lead to a large *age* of sensed information at the monitor, simply because updates from the source are infrequent. More so than voice/video, monitoring applications are exceptionally loss resilient. Specifically, they don't benefit from the source retransmitting old updates that were not received by the monitor. Instead, the source should continue sending new updates at its configured rate.

At the other end of the spectrum are applications like that of file transfer that require reliable transport and high throughputs but are delay tolerant. Such applications use the transmission control protocol (TCP) for end-to-end delivery of application packets. TCP via its congestion control mechanism attempts to keep as many application bytes in transit over the end-to-end connection as is possible without packet drops due to network congestion. For updating applications, TCP is detrimental to optimizing freshness as it tries by design to fill the network pipe. This increases throughput but also increases packet delay because of queueing. Such a strategy would correspond to the source sending updates at a high rate. While the monitor would receive a steady stream of updates at the high rate, each update would have a high age, when received by the monitor, as a result of it having experienced a large network delay.

Figure 2 broadly captures the behavior of the metrics of delay and age as a function of throughput. Under light and moderate loads when packet dropping is negligible, throughput (average network utilization) increases linearly in the rate of updates. This leads to an increase in the average packet delay. Large packet delays coincide with large average age. Large age is also

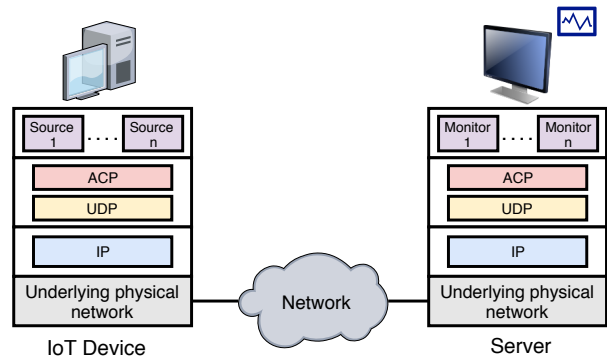


Figure 3: The ACP end-to-end connection.

seen for small throughputs (and corresponding small rate of updates). *At a low update rate, the monitor receives updates infrequently, and this increases the average age (staleness) of its most fresh update.* Finally, observe that there exists a sending rate (and corresponding throughput) at which age is minimized.

As noted in Section 2, many works have analyzed age as a Quality-of-Service metric for monitoring applications. Often such works have employed queue theoretic abstractions of networks. More recently in [28] the authors proposed a deep Q-learning based approach to optimize age over a given but unknown network topology. We believe our work is the first to investigate age control at the transport layer of the networking stack, that is over an end-to-end connection in an IP network and in a manner that is transparent to the application.

Our specific contributions are the following.

1. We propose the Age Control Protocol, a novel transport layer protocol for real-time monitoring applications that wish to deliver fresh updates over IP networks.
2. We argue that such a protocol, unlike other transport protocols like TCP and RTP, must have just the right number of update packets in transit at any given time.
3. We propose a novel control algorithm for ACP that regulates the rate at which a status updating source sends its updates to a monitor over its end-to-end connection in a manner that is application independent and makes the network transparent to the source.
4. We provide an extensive evaluation of the protocol using network simulations and real world experiments in which a source sends packets to a monitor over an inter-continental end-to-end IP connection.
5. We show that ACP adapts the source update rate to make effective use of a fast end-to-end path

with multiple hops from the source to the monitor. Over a connection with a round-trip-time of about 200 msec, ACP achieves a significant reduction in median age of about 80 msec ($\approx 28\%$ improvement) over age achieved by a protocol that sends one update every round-trip-time.

The rest of the paper is organized as follows. In the next section, we describe related work. In Section 3, we detail the Age Control Protocol, how it interfaces with a source and a monitor, and the protocol’s timeline. In Section 3 we define the age control problem. In Section 5, we use simple queueing models to intuit a good age control protocol and discuss a few challenges. Section 6 details the control algorithm that is a part of ACP. This is followed by details on the evaluation methodology in Section 7. We discuss simulation results in Section 8 and results from real-world experiments in Section 9. We conclude in Section 10.

2. RELATED WORK

The need for timely updates arises in many fields, including, for example, vehicular updating [19], real time databases [33], data warehousing [18], and web caching [7, 35].

For sources sending updates to monitors, there has been growing interest in the age of information (AoI) metric that was first analyzed for elementary queues in [20]. To evaluate AoI for a single source sending updates through a network cloud [14] or through an M/M/k server [15, 16], out-of-order packet delivery was the key analytical challenge. Packet deadlines are found to improve AoI in [17]. AoI in the presence of errors is evaluated in [6]. Distributional properties of the age process have also been analyzed for the D/G/1 queue under First Come First Served (FCFS) [5], as well as single server FCFS and LCFS queues [10]. There have also been studies of energy-constrained updating [1, 2, 8, 25, 30, 34].

There has also been substantial efforts to evaluate and optimize age for multiple sources sharing a communication link [11, 13, 21, 23, 26]. In particular, near-optimal scheduling based on the Whittle index has been explored in [9, 12, 13]. When multiple sources employ wireless networks subject to interference constraints, AoI has been analyzed under a variety of link scheduling methods [22, 31]. AoI analysis for multihop networks has also received attention [32]. Notably, optimality properties of a Last Generated First Served (LGFS) service when updates arrive out of order are found in [3].

While the early work [19] explored practical issues such as contention window sizes, the subsequent AoI literature has primarily been focused on analytically tractable simple models. Moreover, a model for the system is typically assumed to be known. In this work, our objective has been to develop end-to-end updating

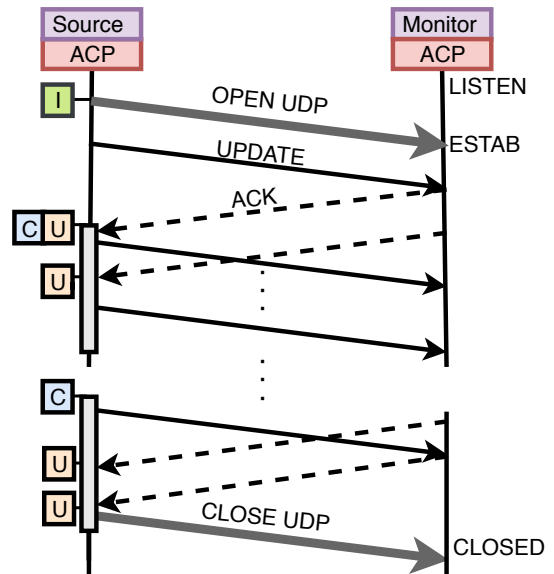


Figure 4: Illustration of the timeline of an ACP connection. The box I marks the beginning of the initialization phase of ACP. The box C denotes the ACP control algorithm (Figure 9) that is executed at the source every control epoch. The box U (Figure 10) is executed when an ACK is received and it updates \overline{RTT} and \overline{Z} .

schemes that perform reasonably well without assuming a particular network configuration or model. This approach attempts to learn (and adapt to time variations in) the condition of the network links from source to monitor. This is similar in spirit to hybrid ARQ based updating schemes [4, 24] that learn the wireless channel. The chief difference is that hybrid ARQ occurs on the short timescale of a single update delivery while ACP learns what the network supports over many delivered updates.

3. THE AGE CONTROL PROTOCOL

The Age Control Protocol resides in the transport layer of the TCP/IP networking stack and operates only on the end hosts. An end host that runs ACP could, for example, be an Internet-of-Things (IoT) device with one or more sources (or a gateway that has sensors connected to it), or it could be a server that hosts one or more monitoring applications. Figure 3 shows an *end-to-end connection* between two hosts, an IoT device, and a server, over the Internet. A source opens an ACP connection to its monitor. Multiple sources may connect to the same monitor. Much like the real-time transport protocol (RTP) [27] that supports voice/video applications, ACP also uses the unreliable transport provided by the user datagram protocol (UDP) for sending of updates generated by the sources. This is in line with the requirements of fresh delivery of updates. Retrans-

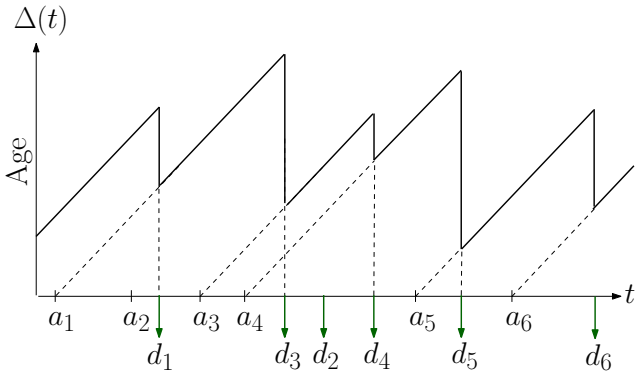


Figure 5: A sample function of the age process $\Delta(t)$. Updates are indexed $1, 2, \dots$. The timestamp of update i is a_i . The time at which update i is received by the monitor is d_i . Since update 2 is received out-of-sequence, it doesn't reset the age process.

missions make an update stale and also compete with fresh updates for network resources.

The source ACP appends a header to an update from a source. The header contains a *timestamp* field that stores the time the update was generated. The source ACP suggests to the source the rate at which it must generate updates. To be able to calculate the rate, the source ACP must estimate network conditions over the end-to-end path to the monitor ACP. This is achieved by having the monitor ACP acknowledge each update packet received from the source ACP by sending an ACK packet in return. The ACK contains the timestamp of the update being acknowledged. The ACK(s) allow the source ACP to keep an estimate of the age of sensed information at the monitor. We say that an ACK is received *out-of-sequence* if it is received after an ACK corresponding to a more recent update packet. An *out-of-sequence* ACK is discarded by the source ACP. Similarly, an update that is received *out-of-sequence* is discarded by the monitor. This is because the monitor has already received a more recent measurement from the source.

Figure 4 shows a timeline of a typical ACP connection. For an ACP connection to take place, the monitor ACP must be listening on a previously advertised UDP port. The ACP source first establishes a UDP connection with the monitor. This is followed by an *initialization* phase during which the source sends an update and waits for an ACK or for a suitable timeout to occur, and repeats this process for a few times, with the goal of probing the network to set an initial update rate. Following this phase, the ACP connection may be described by a sequence of *control epochs*. The end of the *initialization* phase marks the start of the first control epoch. At the beginning of each control epoch, ACP sets the rate at which updates generated

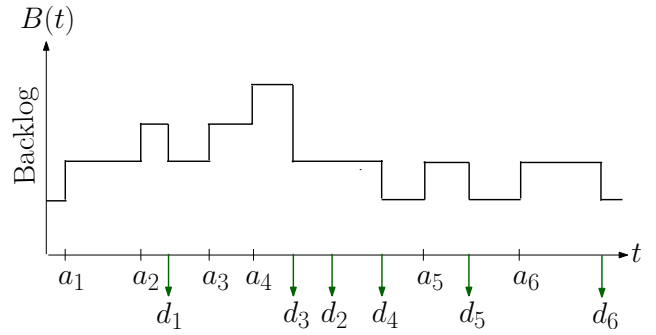


Figure 6: A sample function of the backlog process $B(t)$. Updates are indexed $1, 2, \dots$. The timestamp of update i is a_i . The time at which update i is received by the monitor is d_i . Since update 3 is received before 2, backlog is reduced by 2 packets at d_3 . Also, there is no change in $B(t)$ at $d_2 > d_3$.

from the source are sent until the beginning of the next epoch. ACP may do this in different ways. For example, ACP may simply communicate the rate to the source. This however, would require an appropriate interface between ACP and the source. An alternative to the above is a mechanism wherein updates generated by the source gets queued in the transport layer send buffer assigned to it. The updates waiting to be serviced in the transport layer send buffer are sent by the ACP in a Last Generated First Served (LGFS) manner. The older updates are discarded which is in line with the *freshness* requirement.

An ACP end-to-end connection is closed when the source closes its corresponding UDP socket.

4. THE AGE CONTROL PROBLEM

We will formally define the age of sensed information at a monitor. To simplify presentation, in this section, we will assume that the source and monitor are time synchronized, although the functioning of ACP doesn't require the same. Let $u(t)$ be the timestamp of the freshest update received by the monitor up to time t . Recall that this is the time the update was generated by the source.

The age at the monitor is $\Delta(t) = t - u(t)$ of the freshest update available at the monitor at time t . An example sample function of the age stochastic process is shown in Figure 5. The figure shows the timestamps a_1, a_2, \dots, a_6 of 6 packets generated by the source. Packet i is received by the monitor at time d_i . At time d_i , packet i has age $d_i - a_i$. The age $\Delta(t)$ at the monitor increases linearly in between reception of updates received in the correct sequence. Specifically, it is reset to the age $d_i - a_i$ of packet i , in case packet i is the freshest packet (one with the most recent timestamp) at the monitor at time d_i . For example, when update

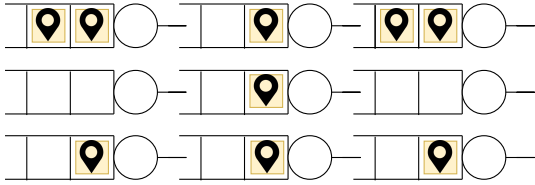


Figure 7: A snapshot of how updates may be seen transiting through a 3 queue network. The top two correspond to a high and low rate of updating, respectively. The third one corresponds to an optimal rate of updating.

3 is received at the monitor, the only other update received by the monitor until then was update 1. Since update 1 was generated at time $a_1 < a_3$, the reception of 3 resets the age to $d_3 - a_3$ at time d_3 . On the other hand, while update 2 was sent at a time $a_2 < a_3$, it is delivered out-of-order at a time $d_2 > d_3$. So packet 2 is discarded by the monitor ACP and the age at the monitor stays unchanged at time d_2 .

We want to choose the rate λ (updates/second) that minimizes the expected value $\lim_{t \rightarrow \infty} E[\Delta(t)]$ of age at the monitor, where the expectation is over any randomness introduced by the network. Note that in the absence of a priori knowledge of a network model, as is the case with the end-to-end connection over which ACP runs, this expectation is unknown to both source and monitor and must be estimated using measurements. Lastly, we would like to dynamically adapt the rate λ to nonstationarities in the network.

5. GOOD AGE CONTROL BEHAVIOR AND CHALLENGES

ACP must suggest a rate λ updates/second at which a source must send fresh updates to its monitor. ACP must adapt this rate to network conditions. To build intuition, let's suppose that the end-to-end connection is well described by an idealized setting that consists of a single FCFS queue that serves each update in constant time. An update generated by the source enters the queue, waits for previously queued updates, and then enters service. The monitor receives an update once it completes service. Note that every update must age at least by the (constant) time it spends in service, before it is received by the monitor. It may age more if it ends up waiting for one or more other updates to complete service.

In this idealized setting, one would want a new update to arrive as soon as the last generated update finishes service. To ensure that the age of each update received at the monitor is the minimum, one must choose a rate λ such that new updates are generated in a periodic manner with the period set to the time an update spends in

service. Also, update generation must be synchronized with service completion instants so that a new update enters the queue as soon as the last update finishes service. In fact, such a rate λ is age minimizing even when updates pass through a sequence of $Q > 1$ such queues in tandem [29]. The update is received by the monitor when it leaves the last queue in the sequence. The rate λ will ensure that a generated packet ages exactly Q times the time it spends in the server of any given queue. At any given time, there will be exactly Q update packets in the network, one in each server. An illustration shown in Figure 7.

Of course, the assumed network is a gross idealization. We assumed a series of similar constant service facilities and that the time spent in service and instant of service completion were known exactly. We also assumed lack of any other traffic. However, as we will see further, the resulting intuition is significant. Specifically, *a good age control algorithm must strive to have as many update packets in transit as possible while simultaneously ensuring that these updates avoid waiting for other previously queued updates.*

Before we detail our proposed control method, we will make a few salient observations using analytical results for simple queueing models that capture stochastic service and generation of updates. These will help build on our intuition and also elucidate the challenges of age control over a priori unknown and likely non-stationary end-to-end network conditions.

We will consider two queueing models. One is the $M/M/1$ FCFS queue with an infinite buffer in which a source sends update packets at a rate λ to a monitor via a single queue, which services packets at a rate μ updates per second. The updates are generated as a Poisson process of rate λ and packet service times are exponentially distributed with $1/\mu$ as the average time it takes to service a packet. In the other model, updates travel through two queues in tandem. Specifically, they enter the first queue that is serviced at the rate μ_1 . On finishing service in the first queue, they enter the second queue that services packets at a rate of μ_2 . As before, updates arrive to the first queue as a Poisson process and packet service times are exponentially distributed. The average age for the case of a single $M/M/1$ queue was analyzed in [20]. We extend their analysis to obtain analytical expressions of average age as a function of λ , μ_1 and μ_2 for the two queue case, by using the well known result that updates also enter the second queue as a Poisson process of rate λ [29].

On the impact of non-stationarity and transient network conditions: Figure 8a shows the expected value (average) of age as a function of λ when the queueing systems are in *steady state*. It is shown for three single $M/M/1$ queues, each with a different service rate, and for two queues in tandem with both servers having the

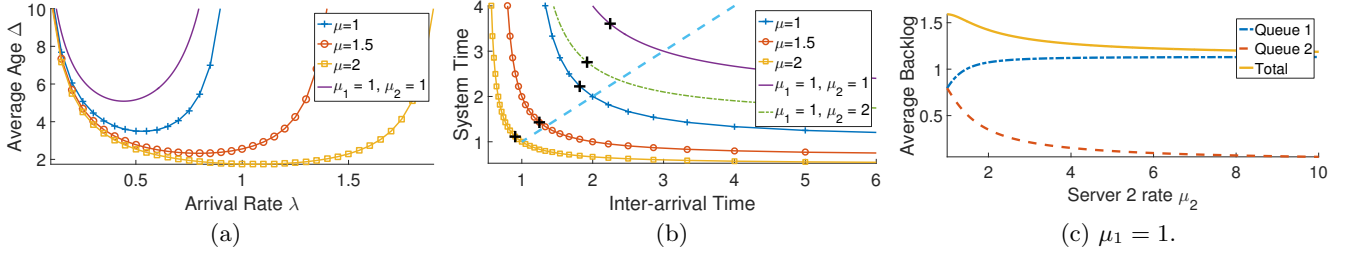


Figure 8: (a) Expected value of age as a function of update rate λ is shown for different queueing networks. (b) Average update packet system time is shown as a function of inter-arrival time ($1/\lambda$). The green dashed line is the 45° line. The + mark the age minimizing λ . (c) The backlog in each queue and the sum is shown as the service rate of the second queue increases from 1 to 5. Service rate of the first queue is $\mu_1 = 1$.

same unit service rate. Observe that all the age curves have a bowl-like shape that captures the fact that a too small or a too large λ leads to large age. Such behavior has been observed in non-preemptive queueing disciplines in which updates can't preempt other older updates. A reasonable strategy to find the optimal rate thus seems to be one that starts at a certain initial λ and changes λ in a direction such that a smaller expected age is achieved.

In practice, the absence of a network model (unknown service distributions and expectations), would require Monte-Carlo estimates of the expected value of age for every choice of λ . Getting these estimates, however, would require averaging over a large number of instantaneous age samples and would slow down adaptation. This could lead to updates experiencing excessive waiting times when λ is too large. Worse, transient network conditions (a run of bad luck) and non-stationarities, for example, because of introduction of other traffic flows, could push these delays to even higher values, leading to an even larger backlog of packets in transit. Figure 8a, illustrates how changes in network conditions (service rate μ and number of hops (queues)) can lead to large changes in the expected age.

It is desirable for a good age control algorithm to not allow the end-to-end connection to drift into a high backlog state. As we describe in the next section, ACP tracks changes in the average number of backlogged packets and average age over short intervals, and in case backlog and age increase, ACP acts to rapidly reduce the backlog.

On Optimal Average Backlogs: Figure 8b plots the average packet system times, where the system time of a packet is the time that elapses between its arrival and completion of its service, as a function of inter-arrival time ($1/\lambda$) for three single queue $M/M/1$ networks and two networks that have two queues in tandem. As expected, increase in inter-arrival time reduces the system time. As inter-arrival times become large, packets wait

less often for others to complete service. As a result, as inter-arrival time increases, the system times converge to the average service time of a packet. For each queueing system, we also mark on its plot the inter-arrival time that minimizes age. It is instructive to note that for the three single queue systems this inter-arrival time is only slightly smaller than the system time. However, for the two queues in tandem with service rates of 1 each, the inter-arrival time is a lot smaller than the system time. The implication being that on an average it is optimal to send slightly more than one (≈ 1.2) packet every system time for the single queue system. However, for the two queue network with the similar servers, we want to send a larger number (≈ 1.6) of packets every system time. For the two queue network where the second queue is served by a faster server, this number is smaller (≈ 1.43). As we observe next, as one of the servers becomes faster, the two queue network becomes more akin to a single queue network with the slower server.

Note that these numbers are in fact the optimal (age minimizing) average number of packets in the system. Figure 8c shows how this optimal average backlog varies as a function of μ_2 for a given μ_1 . The observations stay the same on swapping μ_1 and μ_2 . As μ_2 increases, that is as the second server becomes faster than the first ($\mu_1 = 1$), we see that the average backlog increases in queue 1 and reduces in queue 2, while the sum backlog gets closer to the optimal backlog for the single queue case. Specifically, as queue 1 becomes a larger bottleneck relative to queue 2, optimal λ must adapt to the bottleneck queue. The backlog in the faster queue is governed by the resulting choice of λ . For when the rates μ_1 and μ_2 are similar, they see similar backlogs. However, as is seen for when $\mu_1, \mu_2 = 1$, the backlog per queue is smaller than a network with only a single such queue. However, the sum backlog (≈ 1.6) is larger.

To summarize, one would want a good age control algorithm to have a larger number of packets simulta-

neously in transit in a network with a larger number of hops (queues).

6. THE ACP CONTROL ALGORITHM

Let the control epochs of ACP (Section 3) be indexed $1, 2, \dots$. Control epoch k starts at time t_k . The first control epoch marks the end of the *initialization* phase of ACP. At time t_1 the update rate λ_1 is set to the inverse of the average packet round-trip-times (RTT) obtained at the end of the initialization phase. At time $t_k, k > 1$, the update rate is set to λ_k . The source transmits updates at a fixed period of $1/\lambda_k$ in the interval (t_k, t_{k+1}) .

Let $\bar{\Delta}_k$ be the estimate at the source ACP of the time average update age at the monitor at time t_k . This average is calculated over (t_{k-1}, t_k) . To calculate it, the source ACP must construct its estimate of the age sample function (see Figure 5), over the interval, at the monitor. It knows the time a_i a source sent a certain update i . However, it needs the time d_i at which update i was received by the monitor, which it approximates by the time the ACK for packet i was received. On receiving the ACK, it resets its estimate of age to the resulting round-trip-time (RTT) of packet i .

Note that this value is an overestimate of the age of the update packet when it was received at the monitor, since it includes the time taken to send the ACK over the network. The time average $\bar{\Delta}_k$ is obtained simply by calculating the area under the resulting age curve over (t_{k-1}, t_k) and dividing it by the length $t_k - t_{k-1}$ of the interval.

Let \bar{B}_k be the time average of backlog calculated over the interval (t_{k-1}, t_k) . This is the time average of the instantaneous backlog $B(t)$ over the interval. The instantaneous backlog increases by 1 when the source sends a new update. When an ACK corresponding to an update i is received, update i and any unacknowledged updates older than i are removed from the instantaneous backlog. Figure 6 shows the instantaneous backlog as a function of time corresponding to the age sample function in Figure 5.

In addition to using RTT(s) of updates for age estimation, we also use them to maintain an exponentially weighted moving average (EWMA) $\overline{\text{RTT}}$ of RTT. We update $\overline{\text{RTT}} = (1 - \alpha)\overline{\text{RTT}} + \alpha\text{RTT}$ on reception of an ACK that corresponds to a round-trip-time of RTT.

The source ACP also uses an estimate of the average time between consecutive update arrivals at the monitor. Specifically, it estimates the inter-update arrival times at the monitor and the corresponding EWMA \bar{Z} . The inter-update arrival times are approximated by the corresponding inter-ACK arrival times. The length \bar{T} of a control epoch is set as an integral multiple of $\mathcal{T} = \min(\overline{\text{RTT}}, \bar{Z})$. This ensures that the length of a control epoch is never too large and allows for fast

enough adaptation. Note that at sufficiently low rate λ_k of sending updates \bar{Z} is large and at a sufficiently high update rate $\overline{\text{RTT}}$ is large. At time t_k we set $t_{k+1} = t_k + \bar{T}$. In all our evaluation we have used $\bar{T} = 10\mathcal{T}$. The resulting length of \bar{T} was observed to be long enough to see desired changes in average backlog and age in response to a choice of source update rate at the beginning of an epoch. The source updates $\overline{\text{RTT}}$, \bar{Z} , and \bar{T} every time an ACK is received. Figure 10 summarizes the updates.

At the beginning of control epoch $k > 1$, at time t_k , the source ACP calculates the difference $\delta_k = \bar{\Delta}_k - \bar{\Delta}_{k-1}$ in average age measured over intervals (t_{k-1}, t_k) and (t_{k-2}, t_{k-1}) respectively. Similarly, it calculates $b_k = \bar{B}_k - \bar{B}_{k-1}$.

ACP at the source chooses an action u_k at the k^{th} epoch that targets a change b_{k+1}^* in average backlog over an interval of length \mathcal{T} with respect to the k^{th} interval. The actions, may be broadly classified into (a) additive increase (INC), additive decrease (DEC), and multiplicative decrease (MDEC). MDEC corresponds to a set of actions MDEC(γ), where $\gamma = 1, 2, \dots$. We have

$$\begin{aligned} \text{INC: } b_{k+1}^* &= \kappa, \\ \text{DEC: } b_{k+1}^* &= -\kappa, \\ \text{MDEC}(\gamma): b_{k+1}^* &= -(1 - 2^{-\gamma})B_k, \end{aligned} \quad (1)$$

where $\kappa > 0$ is a step size parameter. Later we evaluate selections of step size ranging from 0.25 to 2.

ACP attempts to achieve b_{k+1}^* by setting λ_k appropriately. The estimate of \bar{Z} at the source ACP of the average inter-update arrival time at the monitor gives us the rate $1/\bar{Z}$ at which updates sent by the source arrive at the monitor. This and λ_k allow us to estimate the average change in backlog over \mathcal{T} as $(\lambda_k - (1/\bar{Z}))\mathcal{T}$. Therefore, to achieve a change of b_{k+1}^* requires choosing

$$\lambda_k = \frac{1}{\bar{Z}} + \frac{b_{k+1}^*}{\mathcal{T}}. \quad (2)$$

Figure 9 summarizes how ACP chooses its action u_k as a function of b_k and δ_k .

The source ACP targets a reduction in average backlog over the next control interval in case (a) $b_k > 0, \delta_k > 0$ or (b) $b_k < 0, \delta_k < 0$. The first condition indicates that the update rate is such that update packets are experiencing larger than optimal delays. ACP attempts to reduce backlog multiplicatively to reduce congestion delays and in the process reduce age quickly. Every consecutive occurrence of this case (tracked by increasing γ by 1 every time) attempts to decrease backlog even more aggressively, which is by a larger power of 2.

The second condition $b_k < 0, \delta_k < 0$ captures a reduction in both age and backlog. ACP greedily aims at reducing backlog further hoping that age will reduce too. It attempts multiplicative decrease if the previous

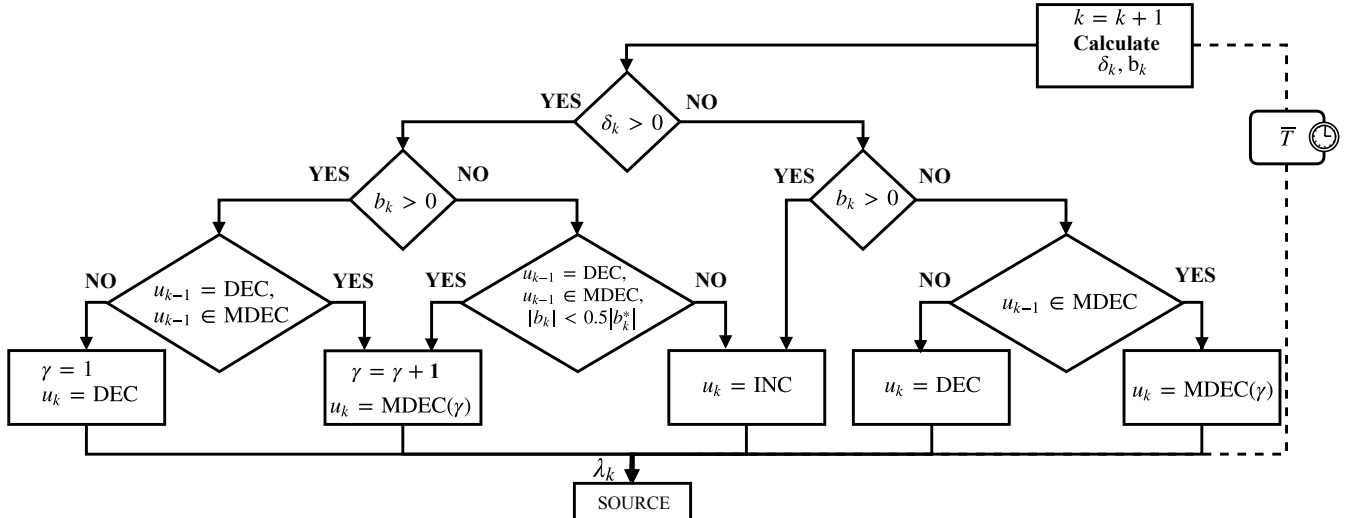


Figure 9: The control algorithm at the source ACP.

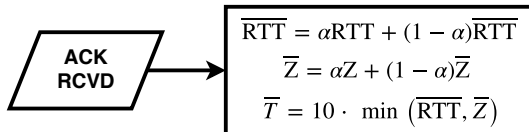


Figure 10: Update of $\overline{\text{RTT}}$, \overline{Z} , and \overline{T} , which takes place every time an ACK is received.

action did so. Else, it attempts an additive decrease.

The source ACP targets an increase in average backlog over the next control interval in case (a) $b_k < 0, \delta_k > 0$ or (b) $b_k > 0, \delta_k < 0$. The first condition hints at too low an update rate causing an increase in age. So, ACP additively increases backlog. On the occurrence of the second condition ACP greedily increases backlog.

When the condition $b_k < 0, \delta_k > 0$ occurs, we check if the previous action attempted to reduce the backlog. If yes, and if the actual change in backlog was much smaller than the desired, we reduce backlog multiplicatively. This helps counter situations where the increase in age is in fact because of increasing congestion. Specifically, increasing congestion in the network may cause the inter-update arrival rate $1/\overline{Z}$ at the monitor to reduce during the epoch. As a result, despite the attempted multiplicative decrease in backlog, it may change very little. Clearly, in such a situation, even if the backlog reduced a little, the increase in age was not caused because the backlog was low. The above check ensures that ACP successfully reduces backlog to desired levels. In the above case, if instead ACP ignores the much smaller than desired change, it will end up increasing the rate of updates, which will only further increase backlog and age.

7. EVALUATION METHODOLOGY

We evaluated ACP using network topologies simulated using the network simulator ns3¹ (Section 8) and by conducting real experiments over an inter-continental end-to-end path over the Internet (Section 9).

The simulated environment while limited in scale and its vagaries allows us to evaluate ACP in a controlled and repeatable setting. It also allows us to understand how the behavior of ACP matches up to that of a desired age control algorithm. In simulations, we compare ACP with two baselines namely *Optimal* and *Basic*.

The *Optimal* baseline is obtained by empirically calculating Monte-Carlo estimates of average age for a selection of update rates with sufficiently fine granularity, for the network being simulated. For each selection of a *fixed* update rate λ , we estimate the average age as would be done by ACP, which is by using ACK packets obtained in return of update packets sent. *Optimal* chooses the update rate that gives the smallest estimate of average age for the network.

The other baseline *Basic* uses the average round-trip-time of an update in a very lightly loaded network, that is for a very small selection of update rate, to set its update rate. Specifically, the average age achieved by *Basic* is the Monte-Carlo estimate of the expected value of age for a constant update rate that is the inverse of the round-trip-time as calculated above. We choose such a baseline as we know from analysis of single queue networks that such a rate is close to optimal (marked by + in Figure 8b). For networks shown in the figure, this rate is the inverse of the inter-arrival time at the point where the 45° line intersects the plot for the chosen network. In fact, as can be seen in the figure, even

¹<https://www.nsnam.org/>

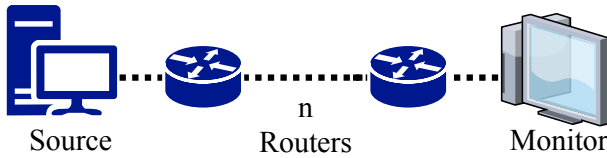


Figure 11: The source is connected to the monitor via multiple routers.

when there are two queues in tandem, if one of them is a relative bottleneck, the choice of rate by *Basic* is close to optimal.

To baseline real-world performance of ACP, we use a modified version of *Basic*. We will call it *Lazy*. *Lazy*, like ACP, also adapts the update rate to network conditions. However, it is very conservative and keeps the average number of update packets in transit small. Specifically, it updates the EWMA of RTT every time an ACK is received and sets the current update rate to the inverse of the EWMA of RTT. As a result, it aims at maintaining an average backlog of 1, given the network conditions.

8. SIMULATION SETUP AND RESULTS

We simulated IP networks with multiple hops between the source and the monitor. The physical and the link layer of each hop was simulated by a simple point-to-point full duplex link whose bit rate could be set to a desired value. Each hop was assigned a link rate of either 1 Mbps or 5 Mbps. We simulated all possible combinations of these rates for a given number of hops. This allows us to simulate conditions in which one or more links are rate bottlenecks relative to others in the network.

We denote a network in which there is only one hop by its corresponding link rate. So if the link rate is 1 Mbps, the network is denoted by 1. Similarly, a two hop network is denoted by 1 – 5 in case the link that connects the source and the router has a rate of 1 Mbps and the link that connects the router and the monitor has rate 5 Mbps.

Figure 11 shows an illustration of the simulated network. We simulated two kinds of update applications, one which generated packets of size 1040 bytes and the other that generated packets of size chosen uniformly and randomly over $\{560, 561, \dots, 1560\}$ bytes.

For every selection of hop count and link rates per hop, and update application type, we evaluated ACP under the following conditions.

- *Lightly loaded network*: The only packets being processed by the network are the source packets and the ACK(s).
- *Link errors*: Lots of prior works have considered

the impact of link errors on the congestion control mechanism of TCP. Motivated by this, we simulated packet drops by links to understand how ACP adapts to them. The hops are not occupied by other traffic. However, each hop drops packets with positive probability.

- *Heavily loaded network*: There is a competing constant bit rate UDP flow at each hop that takes a significant fraction of the rate that is available at the point-to-point links. In all our simulations with ACP, the network transitions from the state of being lightly loaded for 500 seconds to being heavily loaded for 1000 seconds and back to being lightly loaded for 1500 seconds. This allows us to verify the ability of ACP to (a) quickly counter the rapid increase in backlogged update packets due to sudden changes in network conditions and (b) maintain the backlog at a desired level under the new conditions.
- *Varying step size*: We repeat every simulation for different values of step size. We simulate for $\kappa = 0.25, 0.5$, and 1.

8.1 Lightly Loaded Network

We show results for a step size $\kappa = 0.25$ and constant packet sizes. The results are qualitatively similar for uniform random packet sizes and we don't show them. Figure 12 compares the performance of ACP with *Optimal* and *Basic* in terms of the metrics of average age, backlog, and RTT. We look at a single hop network with link rate set to 1 Mbps and two 2-hop networks, one in which both the links have a rate of 1 Mbps and the other in which one link has a rate of 5 Mbps and the other 1 Mbps. As is seen from Figure 12a, the ages obtained by all the mechanisms are within a few milliseconds of each other. This is a small fraction of the packet round-trip-times which are about 15 – 30 msec. ACP, given its control algorithm, sees a slightly larger average age for the networks 1 and 1 – 5 as it looks to change the backlog in the network with the hope of reducing age. *Optimal* benefits from having good Monte-Carlo estimates of age as a function of λ . *Basic* gets lucky that in both the networks the optimal backlog is influenced by a single link with rate 1 Mbps. As is seen in Figure 12b, the average backlog in the network when using ACP is larger than when using *Basic*.

ACP's strategy gains over *Basic*, albeit in a modest manner (28.3 msec for ACP and 29.5 for *Basic*), in the two hop network with links rates of 1 Mbps. Also, note that the average backlog due to ACP is almost $1.5\times$ as large as *Basic*. This ability of ACP to better populate the existing hops with updates, as we will see later, is especially beneficial in the Internet, where end hosts are likely to have many hops between them.

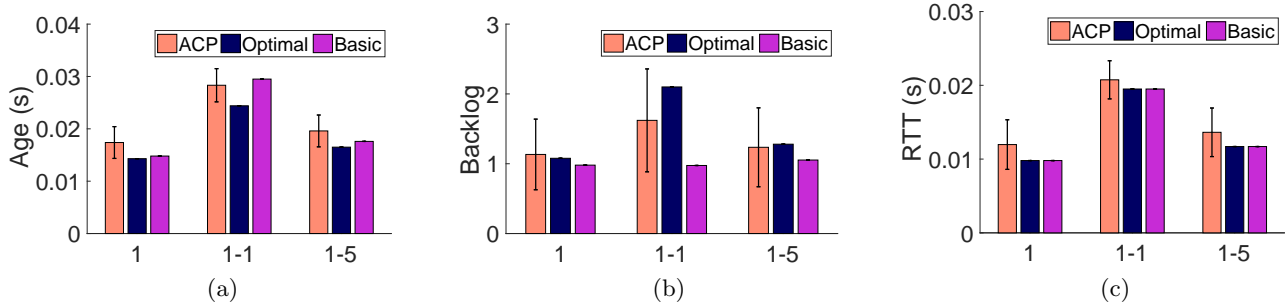


Figure 12: We compare the average (a) Age (b) Backlog and (c) RTT that results from using ACP with averages that result from *Optimal* and *Basic*. Simulated networks had one or two hops. The application generated fixed size updates and no other traffic was sharing the network.

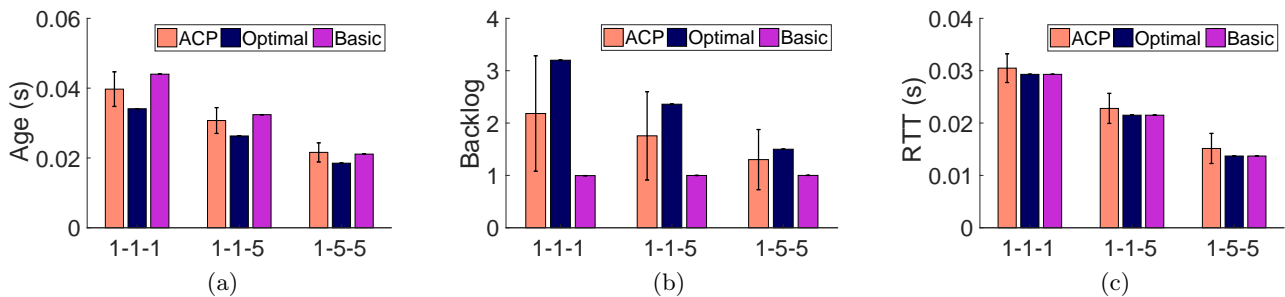


Figure 13: We compare the average (a) Age (b) Backlog and (c) RTT that results from using ACP with averages that result from *Optimal* and *Basic*. Simulated networks had three hops. The application generated fixed size updates and no other traffic was sharing the network.

Figure 12c compares the average RTT(s). As one would expect, given its larger backlogs, ACP results in larger RTT(s) than *Basic*. That said, larger average backlogs don't necessarily cause larger RTT(s). See, for example, the backlogs corresponding to 1 – 1 for ACP and *Optimal*. While *Optimal* has larger backlogs, it sees smaller RTT. This is explained by the fact that the network stays at a steady backlog when using *Optimal*. Also, given constant packet sizes, the updates don't experience any waiting. On the other hand, ACP constantly attempts changing the backlog. An increase in backlog can lead to updates temporarily experience large RTT. This may increase age and makes ACP rapidly reduce backlog. As a result, ACP sees smaller average backlogs and larger RTT.

Figure 13 compares performance for three hop networks. As is seen, ACP achieves a smaller age than *Basic* (Figure 13a) especially when the network has a larger number of bottlenecks. Also, observe that ACP maintains larger average backlogs (Figure 13b) in the network than *Basic*. Finally, for reasons explained above for two hop networks, ACP ends up with larger RTT (Figure 13c) even when *Optimal* has larger backlogs.

8.2 Link Errors

Figures 14a, 14b, and 14c, show the impact of link errors on ACP for one hop, two hop (1 – 1), and three hop (1 – 1 – 1) networks, respectively. We consider packet error probabilities of 0.1 and 0.3 for each link. As is seen from the figures, ACP's choice of update rate is not affected by errors introduced in the one hop network. However, for larger numbers of hops (thus, a larger probability of an update being dropped because of an error), unlike *Basic* and *Optimal*, ACP reduces its rate by large amounts with respect to the rate it uses for when there is no error ($p = 0$ in the figures). This is because it, like TCP, confuses the link errors with congestion. Note that packet drops, for a given λ , lead to an increase in average age and average backlog. ACP tries to recover from this situation by aggressively reducing λ .

8.3 Heavily Loaded Network

We show results for a step size $\kappa = 0.25$, uniformly random packet sizes, and three hop networks. A UDP flow of 0.6 Mbps was introduced at each hop. As is

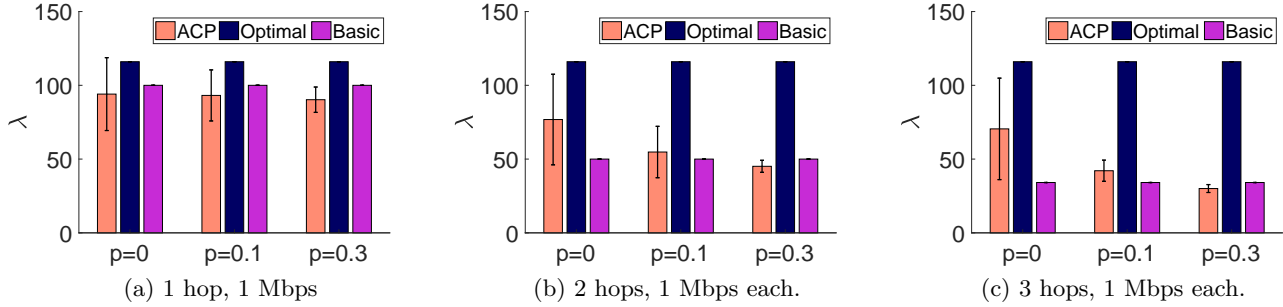


Figure 14: For a network with (a) one, (b) two and (c) three hops we show how ACP adapts to link errors that cause loss of update packets.

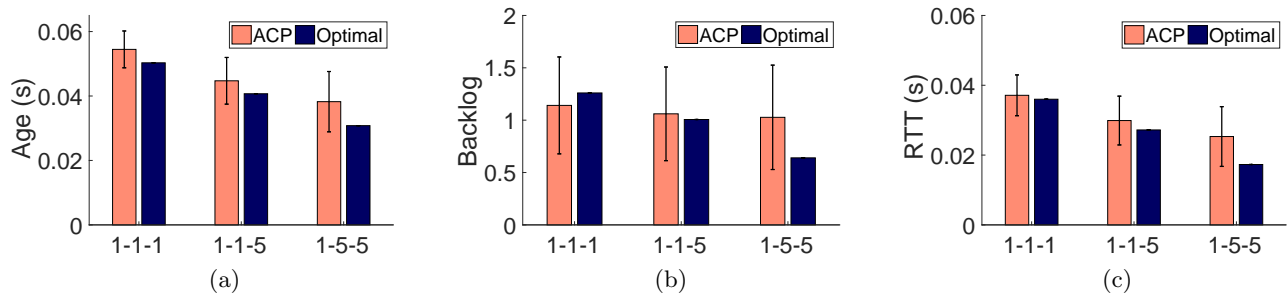


Figure 15: We compare the average (a) Age (b) Backlog and (c) RTT obtained by ACP with that obtained by *Optimal* and *Basic*. Simulated networks had three hops. The application generated random sized packets and each hop was shared by a 0.6 Mbps UDP flow.

seen from the comparisons in Figure 15, ACP performs very well in the presence of competing UDP traffic. Its performance is especially close for networks with larger numbers of similar hops. The results for constant packet sizes and networks with one or two hops are qualitatively similar.

8.4 On Selection of Step Sizes

An appropriate selection of step size is crucial to the proper functioning of ACP. A very small step size may lead to negligible changes in the update rate λ , which may further not lead to the desired changes in backlog and age. Worse, the control algorithm may start oscillating over a small range of λ , which could lead to an average age far from optimal. Our simulations and real-world experiments have shown that step size must also be cognizant of the round-trip-times. In simulations where the RTT was in the range of tens of milliseconds, a step size of 0.25 was found to be sufficiently large. However, in real-world experiments over an inter-continental end-to-end path, with about 20 hops, which we describe in the following section, RTT values were in the range of 200 msec. Here a step size of 0.25 turned out to be too small. A step size of 1 worked very well

for the inter-continental path.

Large step sizes ensure that ACP does not get stuck oscillating in a small region around a sub-par selection of the update rate. However, they cause large swings in instantaneous backlogs and hence RTT(s). This may lead to larger average age. To exemplify, a step size $\kappa = 0.25$ gives an average age of 0.0174 sec for a single hop network with link rate 1 Mbps. This is close to the average 0.0143 sec obtained by *Optimal*. In comparison, a step size of $\kappa = 1$ results in an average age of 0.0351 sec.

9. UPDATES FROM ANOTHER CONTINENT

We had a source and a monitor on different continents communicating over the Internet. The monitor was on a machine with a global IP and the source was running on a machine behind a firewall. We did about 50 experiments over a span of few days. Using the traceroute utility, we observed that the number of hops varied between 15 – 20 during the course of our experiments. The source and the monitor took turns communicating using ACP and *Lazy*. During each turn, 1000 update packets were sent by the source to the monitor. This

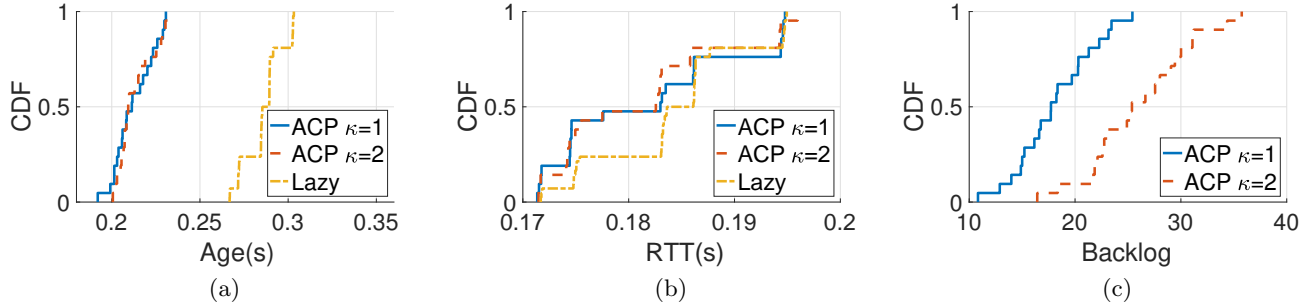


Figure 16: We compare the CDF(s) of average (a) Age (b) RTT and (c) Backlog obtained over 20 runs each of *Lazy* and ACP with step size choices of $\kappa = 1, 2$.

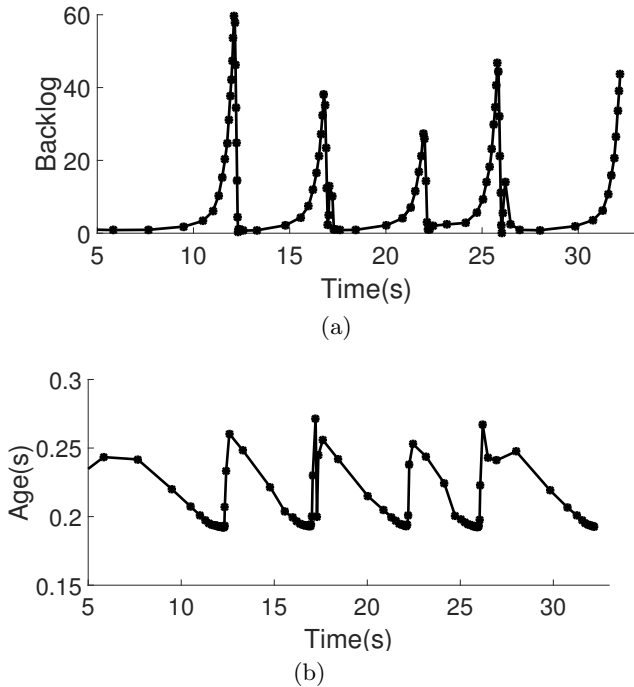


Figure 17: The evolution of average age and backlog that resulted from ACP running over the Internet.

ensured that both ACP and *Lazy* experienced similar network conditions.

Figure 16 summarizes the comparison of ACP and *Lazy*. Figure 16a shows the cumulative distribution functions (CDF) of the average age obtained by ACP and *Lazy* over the experiments. ACP used a step size of $\kappa = 1$ for half the experiments and $\kappa = 2$ for the rest. As is seen for the figure, ACP outperforms *Lazy* and obtains a median improvement of about 80 msec in age ($\approx 28\%$ over age obtained using *Lazy*). This over an end-to-end connection with RTT of about 200 msec. Also, observe from Figure 16b that the median RTT(s) for both ACP and *Lazy* are almost the same.

Lastly, consider a comparison of the CDF of average backlogs shown in Figure 16c. ACP exploits very well the fast end-to-end connection with multiple hops and achieves a very high median average backlog of about 18 when using a step size of 1 and that of 25 when using a step size of 2. On the other hand *Lazy* achieves a backlog of about 1 (not shown in figure).

We will end by showing snippets of ACP in action over the end-to-end path. Figures 17a and 17b show the time evolution of average backlog and average age, as calculated at control epochs. ACP increases backlog in small steps over a large range followed by a rapid decrease in backlog. The increase coincides with reduction in average age and the rapid decrease is initiated once age increases. Also observe that age decreases very slowly (dense regions of points low on the age curve) with increase in backlog just before it increases rapidly. It seems that the region of slow decrease is around where, ideally, backlog must be set to keep age to a minimum.

10. CONCLUSIONS

We proposed the Age Control Protocol, which is a novel transport layer protocol for real-time monitoring applications that desire the freshness of information communicated over the Internet. ACP works in an application-independent manner. It provides a source the ability to regulate its update rate in a network-transparent manner. We detailed ACP’s control algorithm that adapts the rate so that the age of the updates at the monitor is minimized. Via an extensive evaluation using network simulations and real-world experiments, we showed that ACP adapts the source update rate well to make an effective use of network resources available to the end-to-end connection between the source and the monitor. For example, over a connection with a round-trip-time of about 200 msec, ACP achieved a significant reduction in age of about 80 msec over age achieved by a protocol that sends one update

every round-trip-time.

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