ATP: a Datacenter Approximate Transmission Protocol

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Abstract

Many datacenter applications such as machine learning and streaming systems do not need the complete set of data to perform their computation. Current approximate applications in datacenters run on a reliable network layer like TCP. To improve performance, they either let sender select a subset of data and transmit them to the receiver or transmit all the data and let receiver drop some of them. These approaches are network oblivious and unnecessarily transmit more data, affecting both application runtime and network bandwidth usage. On the other hand, running approximate application on a lossy network with UDP cannot guarantee the accuracy of application computation.

We propose to run approximate applications on a lossy network and to allow packet loss in a controlled manner. Specifically, we designed a new network protocol called Approximate Transmission Protocol, or ATP, for datacenter approximate applications. ATP opportunistically exploits available network bandwidth as much as possible, while performing a loss-based rate control algorithm to avoid bandwidth waste and retransmission. It also ensures bandwidth fair sharing across flows and improves accurate applications' performance by leaving more switch buffer space to accurate flows. We evaluated ATP with both simulation and real implementation using two macro-benchmarks and two real applications, Apache Kafka and Flink. Our evaluation results show that ATP reduces application runtime by 13.9% to 74.6% compared to a TCP-based solution that drops packets at sender, and it improves accuracy by up to 94.0% compared to UDP.

1 Introduction

Many applications running in modern datacenters are approximate by nature — their computation can tolerate incomplete or inaccurate data. For example, data analytics and machine learning jobs do not need all the data to find the most accurate analysis results; images and videos can be stored and processed in an approximate manner [40, 49, 51]. To leverage the approximate nature for better performance or energy efficiency, there have been a host of approximate systems at the programming language level [19, 23, 50, 51], hardware level [51], storage level [51], and application level [10, 17, 25, 27, 33, 39, 42, 44, 48].

Most datacenter approximate applications run on a distributed set of machines and involve network communication. While performing inaccurate computation or operating on inaccurate data at host machines, current approximate systems send data across machines using a reliable network. As a result, these systems often send all data (with possible retransmission) to a host, which will then discard certain data before performing approximate computation. Few systems [33] first drop data and then send the remaining over the network. Although such approaches reduces network consumption, they are network oblivious and cannot fully utilize available network bandwidth or adapt to network congestion. For example, senders can drop data when the network is lightly loaded while sending more data when the network is congested.

This paper explores the missing opportunity of utilizing and co-designing network layer for approximate computing. We propose ATP, a transport-layer protocol designed for approximate applications in datacenters. The basic idea of ATP is to exploit approximate applications' tolerance of incomplete data to *allow or even create losses* in network in return for better application performance. Although the basic idea of ATP is simple, designing and implementing a full-fledged protocol that can meet datacenter approximate applications' needs is not easy.

Our strawman design of ATP (§4) aggressively sends data as fast as possible, *i.e.*, at NIC line rate, to opportunistically exploit all available network bandwidth. When the network has abundant bandwidth, doing so can complete an application job as early as possible. With less available bandwidth, packets lost start to happen, since ATP continues to send at line rate. ATP allows packet losses as long as the amount of loss is within what the approximate application can tolerate. When the loss rate is beyond what applications can accept, ATP retransmits failed packets.

In addition to the above *sender-based* approach which allows loss to happen, we further propose a simple *switch-based* technique to *create* more losses. We use very small queues (*e.g.*, 5 packets) at switches for approximate flows, leaving most switch buffer space for accurate flows. Doing so can improve the performance of accurate flows without sacrificing the performance of approximate flow much, as long as the amount of packets dropped by the switch (when its queue for approximate flow is full) is acceptable to the approximate applications.

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The strawman `ATP` design improves application performance and reduces switch cost. However, it has three major limitations. It can trigger significant amount of retransmission when the network is congested; it cannot guarantee the bandwidth fair sharing across flows; and it can cause bandwidth waste where the data sent at a high rate from the sender only gets dropped by later switches along its path to the receiver.

A fundamental reason behind these limitations is the lack of adaptation to network status change in the strawman design. In response, we propose a set of novel algorithms and techniques on top of the strawman design. First, we designed a *loss-based rate control* algorithm that can quickly adapt to dynamic network status and minimize retransmissions (§5.1). Second, we employ a priority-based packet-tagging mechanism to let switches treat packets of different priorities with different dropping probabilities (§5.2), which in turn guarantees fair sharing of network bandwidth across different flow. Third, we propose a method to use a low-priority sub-flow to opportunistically harness bandwidth left in the network after sending most of the data through a main flow with higher priority (§5.3). Finally, we design a new algorithm to consider application-level message size in scheduling packet sending (§5.4).

With these optimization methods, `ATP` can achieve good application performance, guaranteed accuracy, efficient network bandwidth utilization, small queuing delay, bandwidth fair sharing, improved performance of co-running accurate flows, no hardware changes, and minimal application modification. Moreover, `ATP` requires no changes in existing switches; all our techniques are implemented either at host or by changing simple configurations of switches.

We implemented `ATP` with both simulation and real implementation. Our simulation is based on the ns2 simulator framework [5]. We use our modified ns2 simulator to evaluate `ATP` with large Fat-Tree [11] and Leaf-Spine topology and two real workloads, a Facebook key-value store workload [18] and a data mining workload [34]. We compared `ATP` to UDP, DCTCP [14], pFabric [16], and two sender-drop approaches, one that drops packets randomly and one that considers network bandwidth when deciding on packet drop. Our real implementation of `ATP` includes a host-side library on Linux and a software switch on Soft-NIC [38]. We ported two real datacenter applications to `ATP`, Apache Kafka [3] and Apache Flink [2].

Our evaluation results show that `ATP` improves application job completion time by up to 74.6% with our large-scale simulation experiments and by up to 75% with our real experiments. Meanwhile, `ATP`'s measured loss rate is small (at most 9%) and always below application-specified max loss rate. Moreover, it does not affect other accurate applications when running together with approximate applications.

This paper makes the following contribution.

- We identified the limitations of existing approximation systems that are network oblivious.

- As far as we know, we are the first to propose the incorporation of approximate computing at the network layer and we identified the advantages and challenges of doing so.

- We proposed a new transport-layer protocol for approximate computing and designed set of optimization techniques to facilitate it.

- We implemented `ATP` with both simulation and real implementation and ported two real datacenter applications to `ATP`.

We will make our simulation and real implementation source code publicly available soon.

2 Background and Motivation

This section briefly introduces approximate computation in datacenters and motivates the need for a network-aware approximate solution.

2.1 Datacenter Approximate Computing

Data analytics and machine learning applications that aim to extract deep insight from vast amount of "big data" have gained great traction from both academic and industry in recent years. These applications require large computation, memory, and storage resources and need to run on a distributed set of machines in datacenters. One way to improve the performance of these applications or reduce their energy and resource consumption is through *approximation*, by performing inaccurate, approximate computation or using inaccurate, partial data. For example, to improve Hadoop performance, ApproxHadoop [33] approximates the received data blocks from HDFS at the mappers by dropping part of data blocks. In addition to data analytics and machine learning, other application domains such as video and image processing, streaming, and certain probabilistic databases can all benefit from approximation [40, 48, 49, 51]. For example, StreamApprox [44] approximates the received data stream before processing them by dropping part of records.

2.2 Limitations of Existing Solutions

Although there are many proposals of performing approximate computing, most of them are all confined in a single-node environment — at programming language [19, 23, 50, 51], hardware [51], storage [51], and application layers [10, 17, 25, 27, 33, 39, 42, 44]. They are network oblivious and only perform approximation either at the sender or receiver host machines.

At the same time, these applications often run on a reliable network such as DCTCP [14], which guarantees all sent data is received by the receiver.

Both receiver-side approximation and sender-side approximation have their limitations. For receiver-side approximation approaches like StreamApprox and ApproxHadoop, all sender's data is transmitted via a reliable network to its receiver, which then drops part of its received messages when performing approximation computation. These dropped packets unnecessarily consume network bandwidth and postpone the completion time of application jobs.

An alternative solution is to first drop messages at the sender and only send the remaining data to the receiver, for example, by dropping data randomly [33]. Such sender-based approximation approaches (which we call *SD*) reduce network bandwidth consumption and could potentially improve application performance. However, they cannot achieve best application performance or network bandwidth utilization because they are not aware of and do not utilize network status. For example, a sender could have sent more data without dropping (and thus completing an application job sooner) if it knows that the network load is low at the time. Instead, a network-oblivious *SD* approach drops data blindly. Similarly, when the network is congested, a network-oblivious sender may not be able to drop enough data, causing high retransmission rate of the remaining data (*e.g.*, over TCP).

3   `ATP` Abstraction and Design Overview

We believe that distributed approximate computing should exploit and incorporate the network layer. Doing so could improve approximate applications' performance and reduce network load in a way that network-oblivious approximate approaches cannot.

We propose *ATP*, a transport layer protocol designed for approximate workloads in datacenters. The basic idea of `ATP` is to let sender send data as fast as it can without causing retransmission and to let switches drop packets by using a small queue size for approximate flows. `ATP` consists of three parts deployed at sender hosts (as user-level library), receiver hosts (as user-level library), and switches; they collectively perform network-aware approximate communication.

Sender library controls the packet sending rate. It initially sets an aggressive sending rate and quickly adapts the rate to fit network status and application loss requirements (§ 4.1 and § 5.1). The sender also tags packets with priorities for fair bandwidth sharing (§ 5.2). Receiver library receives packets and acknowledges the sender in a new way we designed for approximate applications (§ 4.1). Switches in `ATP` schedule transmissions according to their priority tags and lever-age weighted fair sharing to isolate accurate flows from approximate flows (§ 5.2). For approximate flows, `ATP` switches drop packets when the queue occupancy exceeds a small threshold and use packet spray to spread packets (§ 4.2).

An alternative solution to `ATP` is to (only) have senders drop data based on network congestion, which senders could model based on bandwidth estimations [24] and ECN [14]. As a comparison point to `ATP`, we implemented such an approach on top of DCTCP and found that this approach could actually cause more congestion and affect the performance of both approximate flows and other accurate flows. This is because senders cannot react to network status change as quickly as switches; they can keep sending data and causing more congestion before getting (delayed) feedback. By just changing switch configuration to use a tiny queue size for approximate flows and dropping approximate flow packets at switches, `ATP` can not only adapt to network congestion quickly but also save significantly more switch buffer space for accurate flows.

Applications use `ATP` by linking `ATP` library and calling largely unmodified network APIs (*e.g.*, socket). The only change they need is to specify a *maximum loss rate*, or *MLR*, for each flow (we define each application send request as a separate flow). MLR defines the maximum percentage of messages that an application can lose in a flow, and each flow can have its own MLR. Message is the unit that `ATP` guarantees atomic delivery (either whole message being dropped or delivered). Our current design of `ATP` does not differentiate the importance of messages in a flow, *i.e.*, any of them can be dropped as long as the total amount is under MLR. Applications that require certain messages to be received can place them in separate flows and set their MLR to zero. This abstraction is easy to use and eases `ATP` implementions, while still being flexible and expressive enough for most datacenter approximate applications.

Overall, `ATP` achieves the following design goals.

- *Guaranteed application accuracy.* `ATP` delivers at least $1 - $ MLR messages for each flow.

- *Great application performance.* `ATP` minimizes job completion time (*JCT*) by opportunistically sending data as early as possible.

- *Improved performances of accurate applications.* `ATP`'s switch design leaves more buffer space for accurate flows and thus improves accurate application performance.

- *Bandwidth fair sharing.* `ATP` fairly shares network bandwidth across flows by their priorities.

- *Minimal change to applications and switches.* `ATP` applications only need to specify flows

and their `MLR`, and `ATP` requires no changes to switches by leveraging features that exist in most commodity switches.

4   `ATP` Strawman Design

Before presenting a full design of our final version of `ATP`, we first present a strawman design of `ATP` we initially had and analyze its limitations. We call the strawman `ATP`, `ATP_Base`. To simplify the discussion of this and next section, we further assume that every user-specified message contains only one network packet. We will discuss the case where one message contains multiple packets in §5.4.

4.1   End-Host Design

`ATP_Base` opportunistically sends as much data as possible to minimize the completion time of an approximate flow. Specifically, it sends at either the line rate of sender NIC or the arrival rate of workload, whichever is smaller. If a flow's `MLR` cannot be satisfied after sending all buffered incoming messages, `ATP_Base` retransmits the lost messages until the `MLR` is satisfied.

For every received packet, the receiver sends an ACK with N_{ack} computed as follows: let N denote the number of messages received, we have $N_{ack} = N/(1 - $ `MLR`$)$. N_{ack} is an indicator to the sender on the total number of messages that the sender can safely discard. N_{ack} is bigger than the actual number of messages received by the receiver when `MLR` is greater than zero. The sender stops sending new messages when N_{ack} is larger than the total amount of messages the sender needs to send.

Since `ATP_Base` always sends data at maximum rate, it may results in too much data loss (*i.e.*, above `MLR`) when the network is congested. To remedy this violation, `ATP_Base` employs a simple retransmission mechanism. The sender adds a packet to the tail of a FIFO retransmission queue for possible retransmissions before sending the packet out. The sender starts retransmitting messages in the retransmission queue (in FIFO order) when it has sent out all new messages and N_{ack} is smaller than the total amount of messages sent out. This situation indicates that more than `MLR` messages have been lost.

We use a simple mechanism to determine the packets that got lost (and thus the candidates for retransmission). When sending ACK, the receiver also sends back the sequence numbers of received packets by filling N_{seq} and *data_len*. The sender then locates these packets in the retransmission queue and remove them from the queue. To determine whether a sent packet is lost, we use a similar method as conventional TCP: if the sender receives *dupAck* = 3 ACK packets acknowledging data packets sent after an unacknowledged packet, the unacknowledged packet is determined to be lost.

In contrast to the approach of having sender drop packets at a fixed rate (*e.g.*, at `MLR`), `ATP_Base` always sends as many packets as possible in the beginning of a flow and stops sending (*i.e.*, completing the flow) when the receiver has received enough packets. Essentially, `ATP_Base` "drops" packets that arrive late in a flow to complete a flow as soon as possible.

4.2   Switch Design

We now present the switch design of `ATP_Base`. Note that we only configure functionalities that already exist in modern switches and requires no hardware changes.
Small queuing for approximate flows. A key insight we make with approximate application is that switches do not need and should not use big queues for approximate flows. First, approximate flows can allow packet loss when switch drop packets with small queues. `ATP_Base` aggressively sends data at a rate that is often higher than available link bandwidth. In such situation, a large queue is not useful. Second, configuring a large queue size increases the queue delay, which deteriorates the JCT of flows contending at the same queue (§ 2.2).

Thus, we configure queues for approximate flows to be very small at all switches. A switch drops packets of approximate flows if the corresponding queue overflows. In our large-scale simulation, we found the queue size of five packets to be a good value for all our workloads (§ 7.1.5).

By using tiny queues for approximated flows, switches can use most of its buffer space for accurate flows. Doing so improves accurate flows' completion time (*FCT*) and enables switches to handle more accurate flows.
Multi-path with packet spray. In today's datacenters which often deploy topologies like Fat-Tree [11] and CLOS [34, 36], there are multiple paths of equal distance between any given pair of hosts. To exploit abundant bandwidth provided by multi-path [9, 45], we utilize the packet-spraying feature available in many commodity switches to forward approximated packets [1, 28, 16]. With packet spray, a switch spreads packets uniformly across the set of available routes and can reduce congestion in the network. In a traditional network that runs accurate applications, a drawback of packet spray is that packets received from different paths need to be reordered at the receiver to reconstruct a flow. However, this problem can be largely avoided in `ATP_Base`. Since the queue size for approximate flows is extremely small, the RTTs along different paths between the same source-destination pair are mostly similar. Thus, out-of-order packets will happen much less frequently and the reordering problem in the original

packet-spraying can be largely alleviated. Therefore, we do not need sophisticated path-level scheduling (as in Fastpass [43] and MPTCP [46]) or detailed packet scheduling in the core switches (as in pFabric [16]).

4.3 Limitations

Before further discussion, we first use a simple evaluation result of ATP_Base to illustrate its benefit (we will present our full-fledged evaluation results in § 7, which also describes our ns2-based simulator). This simple test uses a synthetic workload with a simple topology of one sender and one receiver connecting to a switch and the bottleneck link in the network has 0.5 Gbps bandwidth. The sender sends 1000 messages in one flow at 1 Gbps to the receiver. ATP_Base reduces the FCT by half, when the flow's MLR is 0.5. The available bandwidth in the network can satisfy the MLR without retransmission, and the sender stops sending when 500 messages have been received.

This illustrating example demonstrates that ATP_Base can maximize bandwidth efficiency and improve FCT, when there is enough bandwidth in a network to sustain the flow's MLR. However, ATP_Base has three main limitations.

Limitation 1: retransmission. When available bandwidth in a network cannot sustain application MLR ATP_Base will trigger retransmission, causing redundant packets to go across network and consuming more bandwidth.

We evaluate the number of bytes sent with an approximate flow using a simple topology whose bottleneck link is 500 Mbps. The flow consists of 1000 1460 B messages (a total of 1.4 MB) and is sent from one sender to one receiver. Under ATP_Base, the sender sends at the line rate of 1 Gbps. With the bottleneck link only having half of the capacity compared to the line rate, ATP_Base triggers significant retransmission, causing the total bytes sent to be 2.1 MB, a 1.25× increase in bandwidth consumption.

Limitation 2: unfair sharing. ATP_Base's sending rate is determined by the minimum of workload arrival rate and link line rate. When a flow's arrival rate is smaller than another flow and the line rate, it cannot get same bandwidth share as the other flow. For example, suppose a job contains two approximate flows, one with an arrival rate of 990 Mbps and the other with 10 Mbps. When they compete at a bottleneck link of 100 Mbps, only 1 Mbps will be allocated to the second flow. The first flow will significantly increase the FCT of the second flow. To ensure fair sharing, ATP_Base requires switches to implement another bandwidth fair sharing scheduling.

Limitation 3: bandwidth waste. ATP_Base works well for single-hop topologies but can cause unnecessary bandwidth consumption in multi-hop topologies like Fat-Tree [11]. This is because when later hops are congested and earlier hops are not, packets will go through earlier hops and be dropped later on. The bandwidth consumed at earlier hops is then wasted. Instead, a better scheme could have dropped these packets at earlier hops or reduce the sending rate at the sender.

5 ATP Improved Design

To address the limitations of ATP_Base, we employed a set of novel techniques, including an adaptive, loss-based rate control algorithm that minimizes retransmissions and improves bandwidth efficiency (limitations 1 and 3), a priority-based mechanism to guarantee fair sharing (limitation 2), and a method to use a low-priority sub-flow to opportunistically harness bandwidth left in the network (limitation 3). This section presents these techniques we added on top of ATP_Base. We also describe our algorithm to incorporate application-level message size. The resulting system is the full design of ATP, ATP_Full.

5.1 Loss-Based Rate Control

The cause of ATP_Base's first limitation is its lack of sending-rate adaptation to network status. To remedy this issue, we propose a loss-based rate control algorithm that adapts to network status. We call the resulting system *ATP_RC*; ATP_RC adds rate control to ATP_Base. ATP_RC's rate control mechanism is lightweight and scales well with large network. Specifically, it avoids any broadcast or multicast and only reuse ATP_Base's header fields to pass information from receiver to sender.

ATP_RC sender changes sending rate based on the message loss rate in a time window, T_δ. Since an ATP_RC receiver acknowledges the sequence number and data length of every received packet, the sender can count the number of received packets within T_δ, denoted by n_{rcv}. The sender also calculates the number of packets it has sent during T_δ, denoted as n_{sent}. Thus, the packet loss rate is simply $(n_{sent} - n_{rcv})/n_{sent}$. We denote the loss rate within a time window T_δ^j as l_j.

ATP_RC uses a *target loss rate*, or *TLR*, to determine when to increase or decrease the sending rate (denoted by R_j). If the measured loss rate is smaller than the target loss rate (*i.e.*, $l_j \leq$ TLR), ATP_RC increases its sending rate more aggressively.

$$R_{j+1} = (1-m)R_{j+1} + mR_{max} \tag{1}$$

where R_{max} denotes the maximum sending rate, *i.e.*, the line rate. m controls the speed of rate increasing; it is a tradeoff between convergence speed and rate stability.

When the measured loss rate is larger than TLR, ATP

cuts its sending rate by a factor of 2.

$$R_{j+1} = R_j(1 - l_j/2) \qquad (2)$$

Our send rate reduction algorithm is similar to the one from DCTCP which estimates the fraction of packets that are marked as Congestion Encountered (CE) codepoint [14, 30]. However, `ATP_RC` uses the fraction of packets lost in the network. Packet lost rate is more accurate than the fraction of packets that are marked CE codepoint, since one packet can be marked as CE in multiple switches in a multi-hop topology [11, 34, 36] when these switches experience congestion.

The target loss rate `TLR` is a tradeoff between the bandwidth efficiency and unnecessary bandwidth consumption. With a large `TLR`, `ATP_RC` sends data more aggressively and exploit more available bandwidth in the network. However, it can cause more unnecessary bandwidth consumption in a multi-hop topology (packets are sent by one switch but are dropped in the subsequent switches). But it is more likely to efficiently utilize the bandwidth, as the sending rate becomes higher than the link capacity with a larger target loss rate. Under a small `TLR`, `ATP_RC`'s unnecessary bandwidth consumption reduces with a tradeoff in bandwidth efficiency. Our evaluation results on the performance impact of `TLR` in Section 7 show that a `TLR` of 0.1 can efficiently utilize the bandwidth, while causes little bandwidth consumption. We leave automatic adjusting `TLR` to future work.

A final challenge in designing `ATP_RC` is the scenario where the network is congested. With a congested network, the measured loss rate will be high and `ATP_RC` would use a low sending rate to send very few packets per RTT, *e.g.*, 1 packet per RTT. In such cases, we may not be able to calculate loss rate accurately, since it is possible for all packets sent in an RTT to be dropped. Without receiving acknowledgment of any packets, the sender cannot estimate the loss rate or determine the sending rate in the next time window.

To address this issue, the sender decreases the sending rate multiplicatively when it does not receive any acknowledgment from the receiver.

$$R_{j+1} = R_j(1 - \beta) \qquad (3)$$

where β is a *decreasing factor* (in our evaluation, we set $\beta = 0.1$ by default).

5.2 Priority-Based Fair Sharing

In cloud and datacenter environments, it is important to isolate the performance of different applications, and ensure that they have their *fair share* of resources. However, it is difficult to guarantee network bandwidth fair sharing with `ATP_RC`. To deliver ideal fair sharing, switches should drop packets of different flows with equal probability. However, since `ATP_RC` uses tiny queue size for approximate, it is difficult to ensure fair dropping when multiple queues are full. When a switch queue is full, the order of packet arrival determines which packets get dropped first. Subtle timing issue can easily result in a skewed packet drop with drop-tail queues [26].

In fact, our evaluation results show that `ATP_RC` cannot guarantee the fairness of 1024 concurrent flows when they compete on a 1 Gbps bottleneck link. `ATP_RC`'s fairness is even worse when every flow can send no more than one packet in an RTT, since flows with lower sending rates has the equal probability to be dropped as the ones with higher sending rates. To elaborate on this situation further, let us consider the situation when a switch drops arriving packets uniformly across two flows and one flow with low sending rate has not sent any data in the previous RTT and the other flow has sent data in the previous RTT. In the current RTT, a packet in the first flow can be dropped at equal probability as one from the second flow, unfairly penalizing the flow with low sending rate. Ideally, the flows with the lower sending rate should receive lower drop probability.

The root cause behind this issue is that the switch treats all packets equally without considering the sending rate of every flow. We address the unfair issue of `ATP_RC` by differentiating flows according to their sending rate and let switch treat them differently. We propose a packet priority tagging mechanism to improve fairness over `ATP_RC` (we call the resulting system *ATP_Pri*).

`ATP_Pri` leverages the priority queues that are common in commodity switches to treat packets differently. Switches favor high-priority packets and will drop low-priority ones more. `ATP_Pri` tags packets with different priority at the sender according to its sending rate. When a new flow is initialized, the sender tags its packets with a priority according to its initial sending rate. As the sending rate changes, `ATP_Pri` changes the priority assigned to the packets in that flow.

Suppose there are K priorities P_1 to P_K, where $P_1 > P_2 \cdots > P_K$. We use $K-1$ sending rate thresholds to α_1 to α_{K-1} separate packets into K priorities. Since we assign high priorities to packets with lower sending rate, we have $\alpha_1 \le \alpha_2 \cdots \le \alpha_{K-1}$. At the end of T_δ^j, `ATP_Pri` re-assigns priority; it assigns priority P_m to a flow with sending rate R_j, if $\alpha_{m-1} \le R_j < \alpha_m$.

`ATP_Pri` adapts priority and sending rate dynamically in a feedback-loop manner. When assigning a high priority to a packet, `ATP_Pri` increases its chance of being received, lowering the loss rate of the flow the packet belongs to. Our loss-based rate control algorithm will then increase the sending rate in the next

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time window. This in turn will cause `ATP_Pri` to lower the priority of the flow. Similarly, those flows with the lower priority is more likely to have a higher loss rate and thus are assigned a higher priority in the next update intervals. This adaptive adjustment can quickly change to fit network and workload status. It also avoids a potential starvation problem where high-priority flows starve low-priority ones. Similar method was also proposed in RC3 [41] and QJump [35] but in a different context. For example, RC3 uses serveral priorities to improve the ramp-up speed during the very first RTT of a TCP flow.

5.3 Using Backup Flows

As discussed in Sections 5.1 and 5.2, our loss-based rate control and priority tagging in `ATP_Pri` trades bandwidth efficiency (*i.e.*, how much a flow can fully-utilize remaining available bandwidth in the network) for reduced bandwidth consumption, rate stability, and fairness. To improve bandwidth efficiency of `ATP_Pri`, we propose a novel method that uses an additional flow to opportunistically harness available bandwidth in the network.

We assigns two *sub-flows* to every approximate flow. a primary sub-flow that runs the loss-based rate control and priority tagging in `ATP_Pri`, and a backup sub-flow that only sends data with the lowest priority. Backup sub-flows can utilize the remaining network bandwidth left by primary sub-flows. Since switches first discard packets with the lowest priority, backup sub-flows will not affect primary sub-flows within the same flow or primary sub-flows of other flows.

5.4 Message-Size-Aware Scheduling

The discussion so far assumes every message only consists of one packet. In reality, a message can contain multiple packets (when its payload is larger than the MTU). A message is only successfully received when all its packets have been received. Larger messages with more packets are more likely to be lost than smaller messages. When all messages have equal importance in a flow, it is more efficient to send smaller messages and drop larger ones.

With this basic idea, we develop a message-size-aware scheduling protocol in `ATP_Full`. Inspired by Shortest Job First scheduling, our basic idea is to send packets of the message that has the *minimal remaining data* first (*MRDF*). Specifically, every sender maintains a sorted list of messages at the sender according to their remaining size. This list is updated whenever a new message arrives or when the sender gets the acknowledgment of packets in an existing message. Whenever sending a packet, the sender chooses a packet that belongs to the message with minimal remaining size.

The above algorithm implements exact MRDF scheduling, but it has to maintain a sorted list of messages. Another alternative of implementing exact MRDF is to not maintain a sorted list but to traverse all the messages at every packet sending time. Both these methods can lead to high performance overhead. Instead, we choose to implement an inexact MRDF scheduler, where we divide message size into *K* categories and then maintain a sorted list of *K* bins, each containing messages whose remaining size fall into the corresponding category. Maintaining this inexact sorted list is much more efficient than maintaining a fully sorted list of messages, but still able to largely improve application JCT compared to `ATP` without being message-size-aware.

6 Deployment and Implementation

This section discusses how to deploy `ATP` in existing datacenters, how we implemented `ATP` with real systems and adapted real applications to `ATP`.

6.1 Fitting to Current Datacenters

`ATP` places no requirements on network topology and can work with any datacenter topology such as Fat-Tree [11] and Leaf-Spine. `ATP` can co-exist with other transport protocols such as DCTCP [14] and pFabric [16]. Accurate applications can either use datacenters' own choice of a reliable transport-layer protocol or use `ATP` by setting `MLR` to 0. `ATP` properly isolates accurate flows from approximate flows and in fact, improves the performance of accurate flows (see § 7.1.4).

Deployment of `ATP` at switches is also simple and requires no hardware changes. To configure a switch to use `ATP`, we can assign different switch queues to accurate flows, different priorities of approximate flows, and the approximate backup flow. We further configuration occupancy thresholds for different queues to control switches' dropping aggressiveness.

6.2 Real Implementation

We implemented the host logics of `ATP` in a user-level library that applications can link to dynamically. `ATP` intercepts several Linux socket system calls, including connect, accept, write, writev, and read, to executes its sender/receiver logics.

We implemented `ATP`'s switch functionalities as a module in SoftNIC [38], a framework that allows developers to extend NIC functionalities and to send/receive data directly to/from a NIC (through DPDK) in the user level. We implemented `ATP` switch functionalities as a module in SoftNIC. To emulate packet spray, we distribute packets received by an `ATP` switch module to multiple `ATP` switch modules in Round Robin.

For a switch whose ports each has eight FIFO

queues, we assign queue 0 to accurate traffic and queue 1 to 7 to approximate traffic. Queue 1 to 7 have a decreasing priority level, where queue 7 is assigned to backup flows. At every round, the switch selects either queue 0 or one of the queues from 1 to 7 to send packets, according to the quantum assigned to accurate/approximate traffic. Quantum is the number of bytes of a traffic type can be sent in every round and we set quantum to be 1.5 KB. Approximated packets from queue 1 to 7 is scheduled according to their priority.

We use RED [29] to control switch queue occupancy. We set the *max* queue occupancy threshold in RED to five by default for queue 1 to 6. `ATP` uses a more aggressive drop criteria for backup flows, thus we set queue 7 threshold to one (*i.e.*, dropping an incoming packet as long as the queue is not empty). The *min* threshed of RED for queue 1 to 7 are set to 1 by default. RED drops incoming packets probabilistically to mitigate packet synchronization, when the queue occupancy is between 1 and 5.

6.3 Application Adaptation

To use `ATP`, approximate applications need to identify the group of messages that have the same approximation requirement (messages having the same `MLR` and any of them can be dropped) and form them as a flow. Applications then inform `ATP` about the scope of a flow and its `MLR` in an extended socket API.

To demonstrate `ATP`'s ease of use and to evaluate `ATP` with real applications, we ported two streaming systems, Kafka [3] and Flink [2], to `ATP`. Kafka is a distributed stream injection platform that supports multiple publishers to publish (produce) streams of records under different topics to a Kafka broker, which can be consumed by a set of subscribers. Flink is a distributed stream processing engine, which can take Kafka subscriber as input data source. We chose these applications because stream injection and stream analysis can both be approximated.

Porting Kafka and Flink to `ATP` is easy (a total of 177 and 261 SLOC changes respectively and two student days each). To send a message to a Flink consumer, a Kafka broker calls two *writev* syscalls, one for sending the metadata of the message (*e.g.*, message size) and one for sending the actual message. Since a message is meaningless without its metadata and a dangling metadata is also useless, we change Kafka to combine both metadata and data into the same message. The receiving side, Flink consumer, will either get the whole combined message or none of it. In the former case, Flink consumer will pass the combined message to obtain the metadata and actual message.

7 Evaluation

This section presents our evaluation of `ATP` in a simulator and on real Linux machines with two datacenter streaming applications. When not specified, we use `ATP` and `ATP_Full` interchangeably.

7.1 Simulation Results

We first use simulation to evaluate `ATP`. Simulation allows us to evaluate `ATP` with large scale and controlled, complex topology.

7.1.1 Methodology and Workloads

We implemented our simulation of `ATP` based on ns-2, a packet-level simulator [5], by extending it with `ATP`'s end-host and switch functionalities.

Protocols in comparison. We compare `ATP` with five schemes: unmodified **DCTCP** [14] and **UDP**, two network-oblivious sender-drop approaches, DCTCP with random packet dropping at sender (**DCTCP-SD**) and modified **pFabric** [16], and a network-aware sender-drop approach on DCTCP (**DCTCP-BW**). Note that DCTCP-SD and DCTCP-BW are our own strawmen, and not variants from the original DCTCP paper. We use DCTCP and UDP as two extreme points of comparison: one that is completely accurate and one that is lossy with no control over accuracy. As `ATP` is aggressive in sending rate, we compare it with another aggressively sending transport protocol, pFabric. We modify it to always send at max rate but completes a flow as soon as its `MLR` is met. DCTCP-BW uses the congestion window (*CWnd*) in DCTCP to estimate network status and sends as much data as possible when it determines that the network is not congested. Otherwise, DCTCP-BW decides whether to send data according to `MLR`. We employ Equal Cost Multi Path (ECMP) [12] for load balancing. We use the standard marking threshold for DCTCP, which is 65 packets, *i.e.*, an arriving packet is marked with the CE code point [14] if a switch queue occupancy exceeds 65.

Topology. We evaluated `ATP` on two topologies: a Fat-Tree [11] topology that consists of 8 core switches, 16 aggregate switches, 32 top-of-rack (ToR) switches, and 192 hosts, and a 144-host leaf-spine topology with 12 leaf (ToR) switches and 12 spine (Core) switches. The Fat-Tree topology has an over-subscription ratio of 3:1 at ToR switches. Each leaf switch in the leaf-spine topology has 12 1 Gbps links to hosts and 12 1 Gbps links to spines.

We use two network bandwidth setting, one with 1 Gbps links and one with 10 Gbps links. We set the buffer capacity at each switch to be 1.54 MB (1,000 MTUs) shared by 8 FIFO queues for both networks, and configure switches according to § 6.2. We follow prior work [22, 20, 21] to set all network link delay to

$1\mu s$ and host delay to $10\mu s$, which results in a maximum base RTT of $28\mu s$ between a pair of nodes (when there is no queuing delay).

The results with the leaf-spine topology are consistent with the results of the Fat-Tree topology. Because of space constraints, we only present the results of the Fat-Tree topology in the rest of this section.

Workloads. We use two workloads to evaluate ATP with simulation, a Facebook key-value store workload [18] and a data mining workload [34]. The Facebook workload's request size is all below 10KB, and most messages only have one packet. The data mining (DM) workload has more larger messages, 9% DM requests are above 1MB, while 78% of them are below 10KB. The DM workload specifies a Poisson distribution of the inter-arrival time of its messages and the Facebook workload specifies its own inter-arrival time distribution. To model different traffic load, we change the inter-arrival time proportionally from a ratio of 8 to 1 (thus the adapted traffic load is $0.125\times$ to $1\times$ of the original traffic load).

Each run in our simulation evaluation sends a total of 100,000 messages, and we assign these messages uniformly to all the hosts. Every host (a sender) sends messages to a randomly selected host (a receiver).

7.1.2 Application Performance, Accuracy, and Bandwidth Usage

We first evaluate the performance of the Facebook and DM workloads under the six schemes. We measure the total job completion time (JCT) of each workload across all flows as MLR increases (Figure 1) and as traffic load increases (Figure 2). We calculate the JCT of UDP simply as when a sender has sent out all the data (with no confirmation of receiving). UDP serves as an upper bound of sending rate and thus has the best JCT. However, there is no control over message drop rate in UDP and applications cannot use UDP to achieve desired accuracy, as will be discussed soon. Comparing ATP with the other four schemes is more informative and we have the following findings.

First, ATP constantly outperforms DCTCP-SD and DCTCP. DCTCP performs the worst because it always sends all data over the network. ATP and DCTCP-SD both send only partial data with the knowledge of approximation. ATP outperforms DCTCP-SD because DCTCP-SD drops packets at the sender host at a constant rate (*e.g.*, at MLR) even when the network has more available bandwidth. ATP's sending rate is adaptive to the network status. When available network bandwidth is high, ATP will adaptively send more data — beyond the application specified target receive rate. With this more aggressive sending rate, ATP can send more messages in a period than DCTCP-SD does,

yielding the better JCT.

Second, ATP also outperforms DCTCP-BW. Although DCTCP-BW is network-aware, it reacts to network congestion after one feedback delay, during which a large queue can be built at switch. This queuing effect could delay other approximate flows, which further increases network congestion. Figure 1 and Figure 2 shows that DCTCP-BW even performs worse than DCTCP-SD in 1 Gbps network, which demonstrates that DCTCP-BW can create more congestion when network does not have abundant bandwidth. However, DCTCP-BW performs better than DCTCP-SD and approaches ATP in 10 Gbps network with Facebook workload, because the network has abundant bandwidth.

Third, pFabric can obtain similar performance as ATP with high network available bandwidth. But it is the worst in the 1 Gbps network, since it always sends at line rate without congestion control.

Finally, as expected, as MLR increases, JCT decreases with ATP, since ATP sends data more aggressively with larger MLR and completes a job sooner. DCTCP-SD's JCT also decreases with larger MLR when link bandwidth is low. However, JCT stays the same for DCTCP-SD when link bandwidth is high.

Next, we measure the actual packet loss rate of ATP and UDP. Figure 3 plots the total loss rate of the Facebook workload as MLR increases. In this experiment, we set the ATP target loss rate to 10% in our rate control algorithm. The measured loss rate of ATP is always under 10%, meeting application-specified MLR and our target loss rate. In contrast, UDP does not control packet loss and can easily exceed application-specified MLR (with measured loss rate up to 55%). The accuracy results confirm that rate adjustment is crucial to ensure application data quality.

We also compare different workload loads by varying arrival rate. With the increasing traffic load, ATP is more likely to experience packet loss in the network.

Finally, we characterize the bandwidth usage of the network by measuring the average receiving throughput of ATP flows. As the maximum loss rate increases, the bandwidth usage increases as flows can be finished faster, thus the bandwidth can be used by even less ATP flows, thus every flow can use more bandwidth. DCTCP-SD drops messages at the sender and its bandwidth usage is limited by the workload arrival rate. As the traffic load increases, the bandwidth usage of every flow decreases as more traffic is injected in the network. We have the similar observation for peak bandwidth usage of the network by measuring the maximum receiving throughput of ATP flows. Because of space constraints, we do not include figures for the bandwidth usage results here.
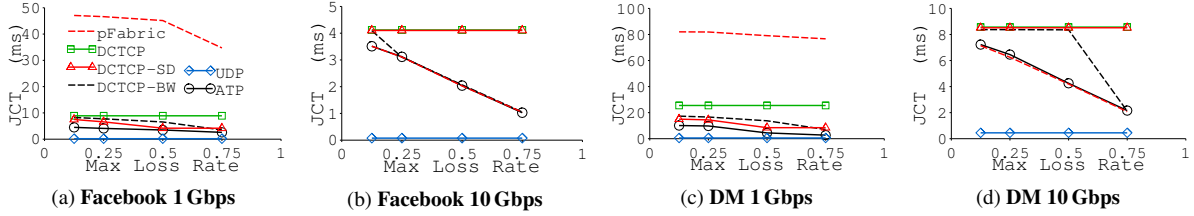
(a) **Facebook 1 Gbps** (b) **Facebook 10 Gbps** (c) **DM 1 Gbps** (d) **DM 10 Gbps**

Figure 1: **JCT with Different MLR** *We ran Facebook and data mining (DM) workload over a 192 hosts Fat-Tree topology with 1 Gbps ((a) and (c)) and 10 Gbps ((b) and (d)) link capacity.*
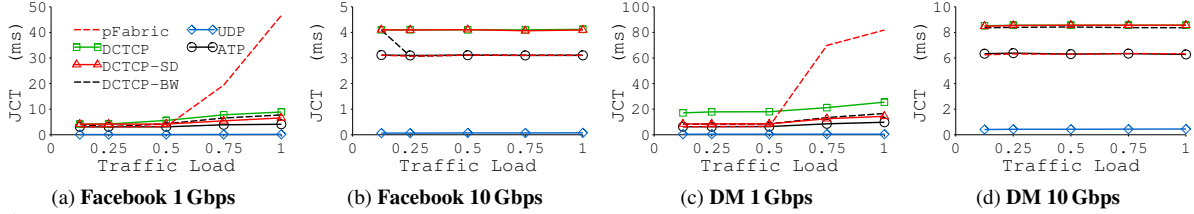

(a) **Facebook 1 Gbps** (b) **Facebook 10 Gbps** (c) **DM 1 Gbps** (d) **DM 10 Gbps**

Figure 2: **JCT with Different Message Arrival Rate** *Facebook and data mining workloads running on 1 Gbps ((a) and (c)) and 10 Gbps ((b) and (d)) network.*

7.1.3 Effect of ATP Techniques

To understand where ATP's performance gain comes from, we evaluate the effect of various ATP design by adding them on top of the strawman ATP_Base. Specifically, we compare ATP_Base with ATP_Full to evaluate our ATP improved design. We compared all optimization techniques we add on top of ATP_Base and found rate control to be the most effective one. We further found that backup flows are more effective in an asymmetric topology than a symmetric one. Because of space constraint, we only present ATP_Base and ATP_Full.

We also compare ATP_Full which uses packet spray to ATP_Full that uses multi-path instead. In the latter, every sender opens two ATP flows going along two different paths. We use the Facebook workload and two traffic loads in this set of experiments, where every host sends 100 messages. Figure 4 plots the JCT of ATP's three modes, DCTCP-SD, and pFabric across all flows, when MLR increases.

As expected, ATP_Base performs the worst among the three modes, especially for small MLR. With small MLR, ATP_Base is even worse than DCTCP-SD and pFabric. ATP_Base does not control the sending rate according to available bandwidth, and employs a smaller queue occupancy threshold at switches, both of which can result in significant packet losses and retransmission. With lighter traffic load, the amount of unnecessary packet drop in the network reduces. Therefore, the performance of ATP_Base improves. For example, with 0.5 traffic load, the JCT reduces from more than 6 ms to around 2 ms.

When enabling loss-based rate control, JCT im-

proves significantly, by up to 67%. ATP_Full with multi-path performs similarly as ATP_Full with packet spray, but at the cost of increasing host complexity.

7.1.4 Impact on Accurate Flows

We now evaluate whether or not ATP affects co-running accurate flows. We separate the Facebook workload into two halves: one half running as approximate flows on ATP or DCTCP-SD and the other half running as accurate flows on DCTCP at the same time. Figure 5 plots the JCT of the accurate flows as the traffic load of the approximate flows increases (Figure 5a for MLR 0.05 and Figure 5b for MLR 0.15). For all settings, DCTCP-SD has a higher impact on accurate flows' performance than ATP.

We also evaluate the effect of switch buffer size using two sizes (250 packets and 1000 packets). With smaller switch buffer, the performance of the accurate flows drops further with DCTCP-SD. In contrast, with ATP, the performance of accurate flows is not affected by switch buffer size, as ATP leaves most buffer for accurate flows.

7.1.5 Parameter Sensitivity

Switch queue size. ATP uses small switch queue size for approximate flows and leaves most switch buffer to accurate flow. Switches drop packets in approximate flows when its queues for approximate flows are full. To evaluate the effect of switch queue size in ATP, we measure the goodput (Figures 6a) and JCT (Figures 6b) of the Facebook workload with different MLR and queue sizes. We also use two flow sizes to evaluate the impact of short and long flows.
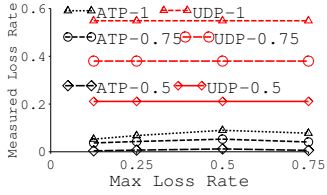
10

Figure 3: **Measured Loss Rate of ATP and UDP** *We meausred the average loss rates of the ATP flow with diffrenet MLR over a 192 hosts Fat-Tree topology.*
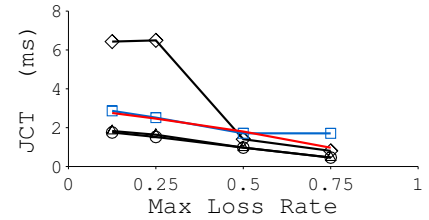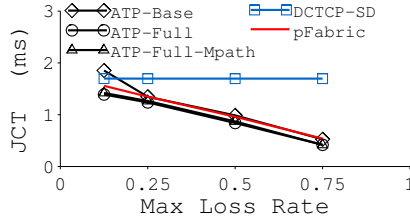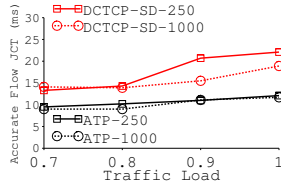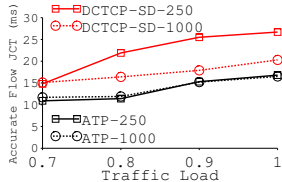

(a) **Traffic Load 0.5**

(b) **Traffic Load 1.0**

Figure 4: **Performance Effect of ATP Techniques** *We compared ATP_Base to ATP_Full to evaluate ATP improved design. The Facebook workload is used over a 192-host Fat-Tree topology.*

(a) **MLR 0.05**

(b) **MLR 0.15**

Figure 5: **Performance Impact on Accurate Flows** *Running half approximate flows and half accurate flows. Y axis shows the JCT of accurate flows when approximate flows run on SD and ATP. Switch buffer size is 250 and 1000 packets (-250 and - 1000).*
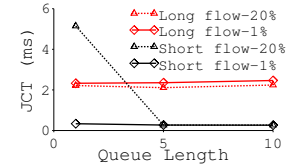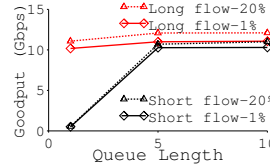
(a) **Effect on Goodput**

(b) **Effect on JCT**

Figure 6: **Impact of Queue Length** *Impact of queue length on JCT and overall goodput using short and long flows. Short flow contains 10 messages, and long flow contains 100 messages.*
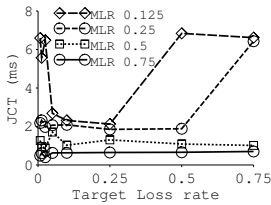

Figure 7: **Performance Effect of Target Loss Rate** *The performance effect of different target loss rate used in the rate control algorithm.*
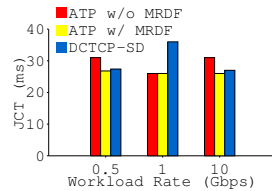

Figure 8: **Performance Effect of MRDF Algorithm** *We compared the performance of ATP with MRDF algorithm with the ATP without MRDF using microbenchmark.*

For short flows, queue size affects goodput, even when MLR is more than 0.2. This is because more packets can be dropped when the queue size is small, which causes ATP to retransmit more data and add at least one more RTT to complete a flow. For long flows, even a queue size of one is sufficient to sustain a high goodput and JCT. When the queue size increases to 5, the performance of short flows improves significantly and is similar to the performance of long flows. We also evaluate the sensitivity of the queue size to ATP performance under real workload, we found that JCT increases with short flows when the queue size is 1. Similarly, when the the queue size increases to 5, JCT becomes similar to the one using larger queue sizes. Therefore, we consider 5 packets queue size to be sufficient under common cases and set it as the default queue size for ATP; users can configure the queue size based on their appli-

cation and network properties.

Message size. ATP uses MRDF algorithm to prioritize the transmissions of the packets from the message that has the minimal remaining size not received. To illustrate the effect of the MRDF algorithm, we use a microbenchmark that lets one sender send messages at a constant rate to a receiver through a bottleneck link. Every message contains 3 MTUs. The bottleneck link is 0.5 Gbps. Figure 8 plots the JCT with and without MRDF (MLR is 0.5 for both). Using MRDF, ATP minimizes the effect of message size and outperforms DCTCP-SD for all traffic loads. ATP without MRDF performs even worse than DCTCP-SD for traffic load 0.5 Gbps and 10 Gbps, which shows that the message size does affect the performance of ATP.

Target loss rate. Target loss rate (TLR) is an important configuration in ATP. We now evaluate the effect of TLR on JCT. We change TLR from 0.0075 to 0.75 and plots JCT of approximate flows with various MLR in Figure 7. When ATP uses a large target loss rate, e.g., 0.75, JCT increases. The reason is that a high TLR results in significant packet loss which consumes bandwidth unnecessarily. When ATP uses a small target loss rate, e.g., 0.0075, JCT also increases, because the loss-rate-based rate control cannot efficiently utilize the bandwidth with a small TLR. From our result, we recommend setting TLR between 0.05 and 0.25.
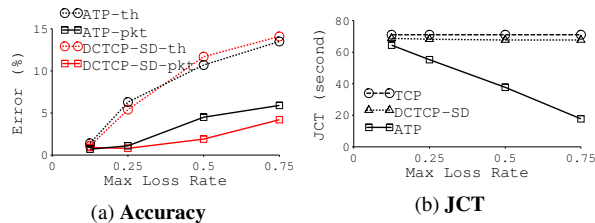
(a) **Accuracy**   (b) **JCT**

Figure 9: **Performance and Accuracy of Nework Traffic Workload**
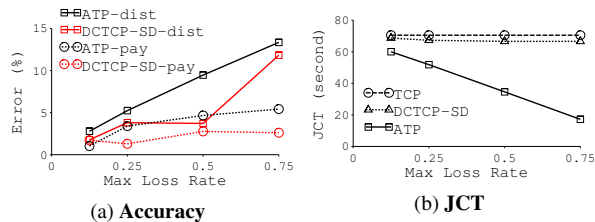

(a) **Accuracy**   (b) **JCT**

Figure 10: **Performance and Accuracy using Taxi Ride Workload**

7.2 Real Implementation Results

We now present our evaluation results using real implementation and adapted Kafka and Flink.

7.2.1 Environments and Workloads

We consider the following typical setup in our real implementation: one Kafka producer produces messages and sends to a Kafka broker. The Flink consumer receives messages for analysis from the Kafka broker. First, Flink consumer opens a persistent TCP connection to Kafka broker, and sends control message to probe any available messages in the broker. The broker forwards messages through the same connection.

We setup a testbed in CloudLab [4] with four producer nodes, one broker node, four consumer nodes, and two emulated ATP switches. Each switch has a buffer capacity of 1000 packets The two emulated ATP switches run on a dual-NIC server with our extension of SoftNIC. This machine connects the consumers and the broker machine via a physical switch respectively and captures packets from both NICs. The link capacity of both NICs are 1 Gbps. The producer machines connect to a physical switch that connects to the broker via 1 Gbps NICs. To avoid unnecessary packet drop at the physical switch, each producer is rate-limited at 250 Mbps using the Traffic Control (TC) utility in Linux [8]. To emulate different link capacity available to the ATP switches, we rate limit the outgoing message rate of the ATP switch modules to 0.9 Gbps (0.45 Gbps per ATP switch).

We evaluated application performance with ATP using real workloads. We developed a producer application atop of Kafka that can send messages at different rate. We also developed a Flink consumer atop of Flink framework that can receive messages and analyze the message payload. We used two real workloads: a UDP network traffic trace collected in the Internet [6] and a taxi ride traffic that describes the information of every ride in NYC [7].

7.2.2 Application Performance

We evaluate the JCT of all four ATP flows from the four hosts using the two real workloads. the producer reads the content from the trace file and sends each line of

the trace as a message to the broker. As shown in Figures 9b and 10b, ATP achieves the best JCT compared to DCTCP and DCTCP-SD.

7.2.3 Impact on Application Accuracy

Finally, we evaluate ATP's impact on actual application accuracy using simple data analytics (calculating average UDP throughput and packet size from the network traffic trace and calculating average taxi distance and payment from the taxi trace). Figures 9a and 10a illustrate the impact of approximation on application accuracy. The application accuracy of both the network traffic and taxi rides worsens as MLR increases, but the error is not significant, e.g., 0.13 inaccuracy is introduced for ATP with 0.75 MLR.

8 Related Work

This section discusses related works to ATP.

Given the advantages of making a tradeoff between accuracy and efficiency, researchers applied approximate techniques to various domains, such as programming language [50, 19], hardware [51], query and database systems [10, 25, 48]. They also applied approximation on distributed systems such as Hadoop, MapReduce and data streaming [27, 42, 39, 17, 33, 44], which are running in datacenters. They focus on reducing the application running time by applying approximations on application algorithms. In contract, ATP applies approximations in network transmissions to reduces the communication delay in terms of JCT.

A large body of work focuses on flow scheduling [16, 21, 32, 43, 15, 13, 54], and congestion control [52, 53, 37, 26] in datacenter networks to minimize JCT and Flow Completion Time (FCT). However, they all target accurate flows, and have the same problem as SD when applying for approximated traffic. Express-Pass [26] shares the idea of using a loss-rate-based rate control algorithm to control the sending rate of the small-size credit packets, which is used to schedule the transmissions of the accurate flows. To the best of our knowledge, only two works [47, 31] applied approximation on network. They both relax the lower layer integreity checking on data bit errors when trasnmitting over wireless and optical networks, which tradeoffs the

better performance in terms of better resouce efficiency and lower latency. Their approaches might not work well over datacenter networks, as the bit error rate is not as significant as wireless and optical networks.

9 Conclusion

This paper presents a datacenter approximate trasnmission protocol `ATP` that allows network to drop packets to achieve approximation. `ATP` tries to utilize the bandwidth available to finish the job earlier, while results in the minimal bandwidth consumption. Compared to contventional approximation approaches `SD` and DCTCP, `ATP` achieves better JCT/bandwidth efficiency than `SD` and DCTCP and lower bandwidth consumption than DCTCP. We evaluated `ATP` with extensive large-scale simulations and ported `ATP` to two real datacenter applications. Both our simualation and real implementation results verified `ATP`'s benefits.

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