Transport Area working group (tsvwg)
Internet-Draft
Intended status: Experimental
Expires: April 25, 2019

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October 22, 2018

DualQ Coupled AQMs for Low Latency, Low Loss and Scalable Throughput (L4S)
draft-ietf-tsvwg-aqm-dualq-coupled-07

Abstract

Data Centre TCP (DCTCP) was designed to provide predictably low queuing latency, near-zero loss, and throughput scalability using explicit congestion notification (ECN) and an extremely simple marking behaviour on switches. However, DCTCP does not co-exist with existing TCP traffic---DCTCP is so aggressive that existing TCP algorithms approach starvation. So, until now, DCTCP could only be deployed where a clean-slate environment could be arranged, such as in private data centres. This specification defines `DualQ Coupled Active Queue Management (AQM)' to allow scalable congestion controls like DCTCP to safely co-exist with classic Internet traffic. The Coupled AQM ensures that a flow runs at about the same rate whether it uses DCTCP or TCP Reno/Cubic, but without inspecting transport layer flow identifiers. When tested in a residential broadband setting, DCTCP achieved sub-millisecond average queuing delay and zero congestion loss under a wide range of mixes of DCTCP and `Classic' broadband Internet traffic, without compromising the performance of the Classic traffic. The solution also reduces network complexity and eliminates network configuration.

Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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This Internet-Draft will expire on April 25, 2019.

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1. Introduction

1.1. Problem and Scope

Latency is becoming the critical performance factor for many (most?) applications on the public Internet, e.g. interactive Web, Web services, voice, conversational video, interactive video, interactive remote presence, instant messaging, online gaming, remote desktop, cloud-based applications, and video-assisted remote control of machinery and industrial processes. In the developed world, further increases in access network bit-rate offer diminishing returns, whereas latency is still a multi-faceted problem. In the last decade or so, much has been done to reduce propagation time by placing caches or servers closer to users. However, queuing remains a major component of latency.

The Diffserv architecture provides Expedited Forwarding [RFC3246], so that low latency traffic can jump the queue of other traffic. However, on access links dedicated to individual sites (homes, small enterprises or mobile devices), often all traffic at any one time will be latency-sensitive and, if all the traffic on a link is marked as EF, Diffserv cannot reduce the delay of any of it. In contrast, the Low Latency Low Loss Scalable throughput (L4S) approach removes the causes of any unnecessary queuing delay.

The bufferbloat project has shown that excessively-large buffering (`bufferbloat') has been introducing significantly more delay than the underlying propagation time. These delays appear only intermittently--only when a capacity-seeking (e.g. TCP) flow is long enough for the queue to fill the buffer, making every packet in other flows sharing the buffer sit through the queue.

Active queue management (AQM) was originally developed to solve this problem (and others). Unlike Diffserv, which gives low latency to some traffic at the expense of others, AQM controls latency for _all_ traffic in a class. In general, AQMs introduce an increasing level of discard from the buffer the longer the queue persists above a shallow threshold. This gives sufficient signals to capacity-seeking (aka. greedy) flows to keep the buffer empty for its intended purpose: absorbing bursts. However, RED [RFC2309] and other

algorithms from the 1990s were sensitive to their configuration and hard to set correctly. So, AQM was not widely deployed.

More recent state-of-the-art AQMs, e.g. fq_CoDel [RFC8290], PIE [RFC8033], Adaptive RED [ARED01], are easier to configure, because they define the queuing threshold in time not bytes, so it is invariant for different link rates. However, no matter how good the AQM, the sawtoothing rate of TCP will either cause queuing delay to vary or cause the link to be under-utilized. Even with a perfectly tuned AQM, the additional queuing delay will be of the same order as the underlying speed-of-light delay across the network. Flow-queuing can isolate one flow from another, but it cannot isolate a TCP flow from the delay variations it inflicts on itself, and it has other problems - it overrides the flow rate decisions of variable rate video applications, it does not recognise the flows within IPSec VPN tunnels and it is relatively expensive to implement.

It seems that further changes to the network alone will now yield diminishing returns. Data Centre TCP (DCTCP [RFC8257]) teaches us that a small but radical change to TCP is needed to cut two major outstanding causes of queuing delay variability:

1. the `sawtooth' varying rate of TCP itself;

2. the smoothing delay deliberately introduced into AQMs to permit bursts without triggering losses.

The former causes a flow's round trip time (RTT) to vary from about 1 to 2 times the base RTT between the machines in question. The latter delays the system's response to change by a worst-case (transcontinental) RTT, which could be hundreds of times the actual RTT of typical traffic from localized CDNs.

Latency is not our only concern:

3. It was known when TCP was first developed that it would not scale to high bandwidth-delay products.

Given regular broadband bit-rates over WAN distances are already [RFC3649] beyond the scaling range of `classic' TCP Reno, `less unscalable' Cubic [RFC8312] and Compound [I-D.sridharan-tcpm-ctcp] variants of TCP have been successfully deployed. However, these are now approaching their scaling limits. Unfortunately, fully scalable TCPs such as DCTCP cause `classic' TCP to starve itself, which is why they have been confined to private data centres or research testbeds (until now).

This document specifies a `DualQ Coupled AQM' extension that solves the problem of coexistence between scalable and classic flows, without having to inspect flow identifiers. The AQM is not like flow-queuing approaches [RFC8290] that classify packets by flow identifier into numerous separate queues in order to isolate sparse flows from the higher latency in the queues assigned to heavier flow. In contrast, the AQM exploits the behaviour of scalable congestion controls like DCTCP so that every packet in every flow sharing the queue for DCTCP-like traffic can be served with very low latency.

This AQM extension can be combined with any single queue AQM that generates a statistical or deterministic mark/drop probability driven by the queue dynamics. In many cases it simplifies the basic control algorithm, and requires little extra processing. Therefore it is believed the Coupled AQM would be applicable and easy to deploy in all types of buffers; buffers in cost-reduced mass-market residential equipment; buffers in end-system stacks; buffers in carrier-scale equipment including remote access servers, routers, firewalls and Ethernet switches; buffers in network interface cards, buffers in virtualized network appliances, hypervisors, and so on.

The overall L4S architecture is described in [I-D.ietf-tsvwg-l4s-arch]. The supporting papers [PI2] and [DCttH15] give the full rationale for the AQM's design, both discursively and in more precise mathematical form.

1.2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119]. In this document, these words will appear with that interpretation only when in ALL CAPS. Lower case uses of these words are not to be interpreted as carrying RFC-2119 significance.

The DualQ Coupled AQM uses two queues for two services. Each of the following terms identifies both the service and the queue that provides the service:

Classic (denoted by subscript C): The `Classic' service is intended for all the behaviours that currently co-exist with TCP Reno (TCP Cubic, Compound, SCTP, etc).

Low-Latency, Low-Loss and Scalable (L4S, denoted by subscript L): The `L4S' service is intended for a set of congestion controls with scalable properties such as DCTCP (e.g. Relentless [Mathis09]).

Either service can cope with a proportion of unresponsive or less-responsive traffic as well (e.g. DNS, VoIP, etc), just as a single queue AQM can. The DualQ Coupled AQM behaviour is similar to a single FIFO queue with respect to unresponsive and overload traffic.

1.3. Features

The AQM couples marking and/or dropping across the two queues such that a flow will get roughly the same throughput whichever it uses. Therefore both queues can feed into the full capacity of a link and no rates need to be configured for the queues. The L4S queue enables scalable congestion controls like DCTCP to give stunningly low and predictably low latency, without compromising the performance of competing 'Classic' Internet traffic. Thousands of tests have been conducted in a typical fixed residential broadband setting. Typical experiments used base round trip delays up to 100ms between the data centre and home network, and large amounts of background traffic in both queues. For every L4S packet, the AQM kept the average queuing delay below 1ms (or 2 packets if serialization delay is bigger for slow links), and no losses at all were introduced by the AQM. Details of the extensive experiments will be made available [PI2] [DCttH15].

Subjective testing was also conducted using a demanding panoramic interactive video application run over a stack with DCTCP enabled and deployed on the testbed. Each user could pan or zoom their own high definition (HD) sub-window of a larger video scene from a football match. Even though the user was also downloading large amounts of L4S and Classic data, latency was so low that the picture appeared to stick to their finger on the touchpad (all the L4S data achieved the same ultra-low latency). With an alternative AQM, the video noticeably lagged behind the finger gestures.

Unlike Diffserv Expedited Forwarding, the L4S queue does not have to be limited to a small proportion of the link capacity in order to achieve low delay. The L4S queue can be filled with a heavy load of capacity-seeking flows like DCTCP and still achieve low delay. The L4S queue does not rely on the presence of other traffic in the Classic queue that can be 'overtaken'. It gives low latency to L4S traffic whether or not there is Classic traffic, and the latency of Classic traffic does not suffer when a proportion of the traffic is L4S. The two queues are only necessary because DCTCP-like flows cannot keep latency predictably low and keep utilization high if they are mixed with legacy TCP flows,

The experiments used the Linux implementation of DCTCP that is deployed in private data centres, without any modification despite its known deficiencies. Nonetheless, certain modifications will be

necessary before DCTCP is safe to use on the Internet, which are recorded in Appendix A of [I-D.ietf-tsvwg-ecn-l4s-id]. However, the focus of this specification is to get the network service in place. Then, without any management intervention, applications can exploit it by migrating to scalable controls like DCTCP, which can then evolve _while_ their benefits are being enjoyed by everyone on the Internet.

2. DualQ Coupled AQM

There are two main aspects to the approach:

- the Coupled AQM that addresses throughput equivalence between Classic (e.g. Reno, Cubic) flows and L4S (e.g. DCTCP) flows

- the Dual Queue structure that provides latency separation for L4S flows to isolate them from the typically large Classic queue.

2.1. Coupled AQM

In the 1990s, the `TCP formula' was derived for the relationship between TCP's congestion window, cwnd, and its drop probability, p. To a first order approximation, cwnd of TCP Reno is inversely proportional to the square root of p.

TCP Cubic implements a Reno-compatibility mode, which is the only relevant mode for typical RTTs under 20ms as long as the throughput of a single flow is less than about 500Mb/s. Therefore it can be assumed that Cubic traffic behaves similarly to Reno (but with a slightly different constant of proportionality), and the term 'Classic' will be used for the collection of Reno-friendly traffic including Cubic in Reno mode.

The supporting paper [PI2] includes the derivation of the equivalent rate equation for DCTCP, for which cwnd is inversely proportional to p (not the square root), where in this case p is the ECN marking probability. DCTCP is not the only congestion control that behaves like this, so the term 'L4S' traffic will be used for all similar behaviour.

In order to make a DCTCP flow run at roughly the same rate as a Reno TCP flow (all other factors being equal), the drop or marking probability for Classic traffic, p_C has to be distinct from the marking probability for L4S traffic, p_L (in contrast to RFC3168 which requires them to be the same). To remain stable, Classic traffic needs p_C to change relatively slowly, whereas L4S traffic needs to be controlled rapidly by a probability p_L that track the instantaneous queue. It is necessary to make the Classic drop

probability p_C proportional to the square of a variable we shall call p_CL, which is an input to the instantaneous L4S marking probability p_L but changes as slowly as p_C. This makes the Reno flow rate roughly equal the DCTCP flow rate, because it squares the square root of p_C in the Reno rate equation to make it proportional to the smoothed value of p_L used in the DCTCP rate equation.

Stating this as a formula, the relation between Classic drop probability, p_C, and the input variable p_CL to the L4S marking probability p_L needs to take the form:

$$p_C = (p_CL / k)^2 \quad (1)$$

where k is the constant of proportionality.

2.2. Dual Queue

Classic traffic typically builds a large queue to prevent under-utilization. Therefore a separate queue is provided for L4S traffic, and it is scheduled with priority over Classic. Priority is conditional to prevent starvation of Classic traffic.

Nonetheless, coupled marking ensures that giving priority to L4S traffic still leaves the right amount of spare scheduling time for Classic flows to each get equivalent throughput to DCTCP flows (all other factors such as RTT being equal). The algorithm achieves this without having to inspect flow identifiers.

2.3. Traffic Classification

Both the Coupled AQM and DualQ mechanisms need an identifier to distinguish L and C packets. A separate draft [I-D.ietf-tsvwg-ecn-l4s-id] recommends using the ECT(1) codepoint of the ECN field as this identifier, having assessed various alternatives. An additional process document has proved necessary to make the ECT(1) codepoint available for experimentation [RFC8311].

For policy reasons, an operator might choose to steer certain packets (e.g. from certain flows or with certain addresses) out of the L queue, even though they identify themselves as L4S by their ECN codepoints. In such cases, the classifier MUST NOT alter the ECN field, so that it is preserved end-to-end. The aim is that each operator can choose how it treats L4S traffic locally, but an individual operator does not alter the identification of L4S packets, which would prevent other operators downstream from making their own choices on how to treat L4S traffic.

In addition, other identifiers could be used to classify certain additional packet types into the L queue, that are deemed not to risk harming the L4S service. For instance addresses of specific applications or hosts (see [I-D.ietf-tsvwg-ecn-l4s-id]), specific Diffserv codepoints such as EF (Expedited Forwarding) and Voice-Admit service classes (see [I-D.briscoe-tsvwg-l4s-diffserv]) or certain protocols (e.g. ARP, DNS).

Note that the DualQ Coupled AQM only reads these classifiers, it MUST NOT re-mark or alter these identifiers (except for marking the ECN field with the CE codepoint - with increasing frequency to indicate increasing congestion).

2.4. Overall DualQ Coupled AQM Structure

Figure 1 shows the overall structure that any DualQ Coupled AQM is likely to have. This schematic is intended to aid understanding of the current designs of DualQ Coupled AQMs. However, it is not intended to preclude other innovative ways of satisfying the normative requirements in Section 2.5 that minimally define a DualQ Coupled AQM.

The classifier on the left separates incoming traffic between the two queues (L and C). Each queue has its own AQM that determines the likelihood of marking or dropping (p_L and p_C). It has been proved [PI2] that it is preferable to control TCP with a linear PI controller, then square the output before applying it as a drop probability to TCP. So, the AQM for Classic traffic needs to be implemented in two stages: i) a base stage that outputs an internal probability p' (pronounced p-prime); and ii) a squaring stage that outputs p_C, where

$$p_C = (p')^2. \qquad (2)$$

Substituting for p_C in Eqn (1) gives:

$$p' = p_CL / k$$

So we get our slow-moving input to ECN marking in the L queue as:

$$p_CL = k*p', \qquad (3)$$

where k is the constant coupling factor (see Appendix C).

It can be seen that these two transformations of p' implement the required coupling given in equation (1) earlier. Substituting for p' from equation (3) into (2):

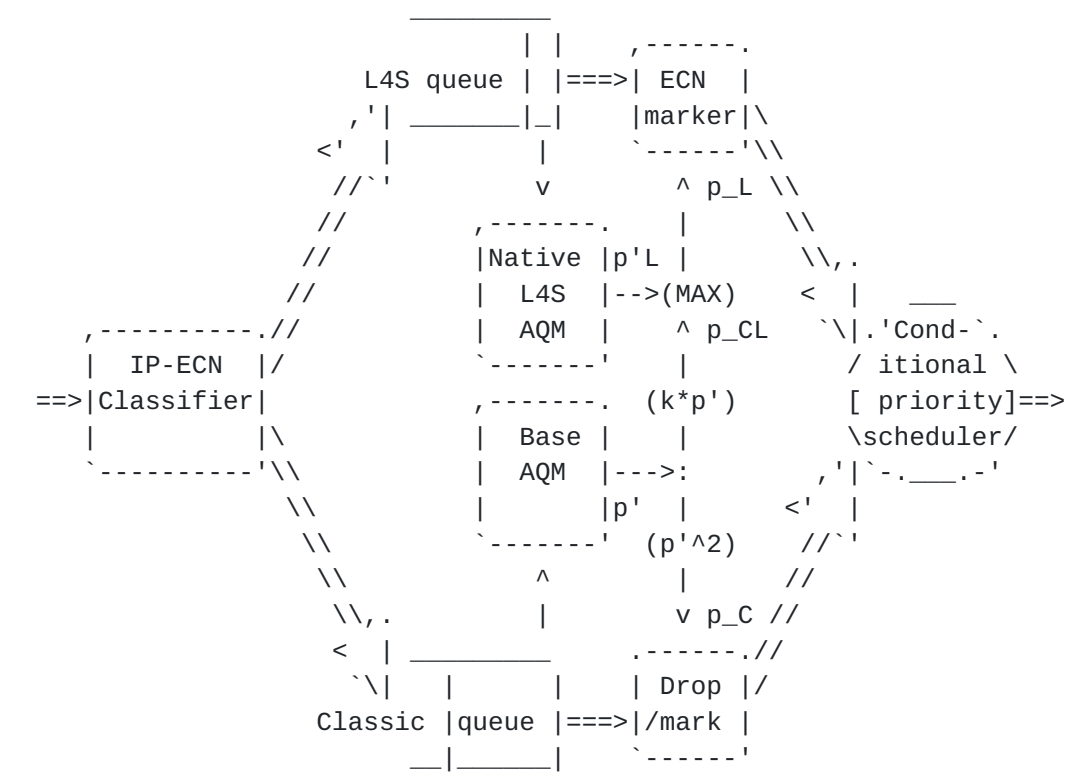
$$p_C = (p_{CL} / k)^2.$$

The actual probability p_L that we apply to the L queue needs to track the immediate L queue delay, as well as track p_CL under stationary conditions. So we use a native AQM in the L queue that calculates a marking probability p'L as a function of the instantaneous L queue. And, given the L queue has conditional strict priority over the C queue, whenever the L queue grows, we should apply marking probability p'_L, but p_L should not fall below p_CL. This suggests:

$$p_L = \max(p'_L, p_{CL}),$$

which has also been found to work very well in practice.

This allows p_L to be coupled to p_C by marking L4S traffic proportionately to the intermediate output from the first stage. Specifically, the output of the base AQM is coupled across to the L queue in proportion to the output of the base AQM



Legend: ===> traffic flow; ---> control dependency.

Figure 1: DualQ Coupled AQM Schematic

After the AQMs have applied their dropping or marking, the scheduler forwards their packets to the link, giving priority to L4S traffic. Priority has to be conditional in some way (see Section 4.1). Simple strict priority is inappropriate otherwise it could lead the L4S queue to starve the Classic queue. For example, consider the case where a continually busy L4S queue blocks a DNS request in the Classic queue, arbitrarily delaying the start of a new Classic flow.

Example DualQ Coupled AQM algorithms called DualPI2 and Curvy RED are given in Appendix A and Appendix B. Either example AQM can be used to couple packet marking and dropping across a dual Q.

DualPI2 uses a Proportional-Integral (PI) controller as the Base AQM. Indeed, this Base AQM with just the squared output and no L4S queue can be used as a drop-in replacement for PIE [RFC8033], in which case we call it just PI2 [PI2]. PI2 is a principled simplification of PIE that is both more responsive and more stable in the face of dynamically varying load.

Curvy RED is derived from RED [RFC2309], but its configuration parameters are insensitive to link rate and it requires less operations per packet. However, DualPI2 is more responsive and stable over a wider range of RTTs than Curvy RED. As a consequence, DualPI2 has attracted more development attention than Curvy RED, leaving the Curvy RED design incomplete and not so fully evaluated.

Both AQMs regulate their queue in units of time not bytes. As already explained, this ensures configuration can be invariant for different drain rates. With AQMs in a dualQ structure this is particularly important because the drain rate of each queue can vary rapidly as flows for the two queues arrive and depart, even if the combined link rate is constant.

It would be possible to control the queues with other alternative AQMs, as long as the normative requirements (those expressed in capitals) in Section 2.5 are observed.

2.5. Normative Requirements for a DualQ Coupled AQM

The following requirements are intended to capture only the essential aspects of a DualQ Coupled AQM. They are intended to be independent of the particular AQMs used for each queue.

2.5.1. Functional Requirements

In the Dual Queue, L4S packets MUST be given priority over Classic, although priority MUST be bounded in order not to starve Classic traffic.

Whatever identifier is used for L4S experiments, [I-D.ietf-tsvwg-ecn-l4s-id] defines the meaning of an ECN marking on L4S traffic, relative to drop of Classic traffic. In order to prevent starvation of Classic traffic by scalable L4S traffic, it says, "The likelihood that an AQM drops a Not-ECT Classic packet (p_C) MUST be roughly proportional to the square of the likelihood that it would have marked it if it had been an L4S packet (p_L)." In other words, in any DualQ Coupled AQM, the power to which p_L is raised in Eqn. (1) MUST be 2. The term 'likelihood' is used to allow for marking and dropping to be either probabilistic or deterministic.

The constant of proportionality, k, in Eqn (1) determines the relative flow rates of Classic and L4S flows when the AQM concerned is the bottleneck (all other factors being equal). [I-D.ietf-tsvwg-ecn-l4s-id] says, "The constant of proportionality (k) does not have to be standardised for interoperability, but a value of 2 is RECOMMENDED."

Assuming scalable congestion controls for the Internet will be as aggressive as DCTCP, this will ensure their congestion window will be roughly the same as that of a standards track TCP congestion control (Reno) [RFC5681] and other so-called TCP-friendly controls, such as TCP Cubic in its TCP-friendly mode.

{ToDo: The requirements for scalable congestion controls on the Internet (termed the TCP Prague requirements) [I-D.ietf-tsvwg-ecn-l4s-id] are not necessarily final. If the aggressiveness of DCTCP is not defined as the benchmark for scalable controls on the Internet, the recommended value of k will also be subject to change.}

The choice of k is a matter of operator policy, and operators MAY choose a different value using Table 1 and the guidelines in Appendix C.

If multiple users share capacity at a bottleneck (e.g. in the Internet access link of a campus network), the operator's choice of k will determine capacity sharing between the flows of different users. However, on the public Internet, access network operators typically isolate customers from each other with some form of layer-2 multiplexing (TDM in DOCSIS, CDMA in 3G) or L3 scheduling (WRR in DSL), rather than relying on TCP to share capacity between customers [RFC0970]. In such cases, the choice of k will solely affect relative flow rates within each customer's access capacity, not between customers. Also, k will not affect relative flow rates at any times when all flows are Classic or all L4S, and it will not affect small flows.

2.5.1.1. Requirements in Unexpected Cases

The flexibility to allow operator-specific classifiers (Section 2.3) leads to the need to specify what the AQM in each queue ought to do with packets that do not carry the ECN field expected for that queue. It is recommended that the AQM in each queue inspects the ECN field to determine what sort of congestion notification to signal, then decides whether to apply congestion notification to this particular packet, as follows:

- If a packet that does not carry an ECT(1) or CE codepoint is classified into the L queue:
  - if the packet is ECT(0), the L AQM SHOULD apply CE-marking using a probability appropriate to Classic congestion control and appropriate to the target delay in the L queue
  - if the packet is Not-ECT, the appropriate action depends on whether some other function is protecting the L queue from misbehaving flows (e.g. per-flow queue protection or latency policing):
    - If separate queue protection is provided, the L AQM SHOULD ignore the packet and forward it unchanged, meaning it should not calculate whether to apply congestion notification and it should neither drop nor CE-mark the packet (for instance, the operator might classify EF traffic that is unresponsive to drop into the L queue, alongside responsive L4S-ECN traffic)
    - if separate queue protection is not provided, the L AQM SHOULD apply drop using a drop probability appropriate to Classic congestion control and appropriate to the target delay in the L queue
- If a packet that carries an ECT(1) codepoint is classified into the C queue:
  - the C AQM SHOULD apply CE-marking using the coupled AQM probability p_CL (= k*p').

If the DualQ Coupled AQM has detected overload, it will signal congestion solely using drop, irrespective of the ECN field.

The above requirements are worded as "SHOULDs", because operator-specific classifiers are for flexibility, by definition. Therefore, alternative actions might be appropriate in the operator's specific circumstances. An example would be where the operator knows that

certain legacy traffic marked with one codepoint actually has a congestion response associated with another codepoint.

2.5.2. Management Requirements

By default, a DualQ Coupled AQM SHOULD NOT need any configuration for use at a bottleneck on the public Internet [RFC7567]. The following parameters MAY be operator-configurable, e.g. to tune for non-Internet settings:

- Optional packet classifier(s) to use in addition to the ECN field (see Section 2.3);
- Expected typical RTT (a parameter for typical or target queuing delay in each queue might be configurable instead);
- Expected maximum RTT (a stability parameter that depends on maximum RTT might be configurable instead);
- Coupling factor, k;
- The limit to the conditional priority of L4S (scheduler-dependent, e.g. the scheduler weight for WRR, or the time-shift for time-shifted FIFO);
- The maximum Classic ECN marking probability, p_Cmax, before switching over to drop.

An experimental DualQ Coupled AQM SHOULD allow the operator to monitor the following operational statistics:

- Bits forwarded (total and per queue per sample interval), from which utilization can be calculated
- Q delay (per queue over sample interval) {ToDo: max per interval, histogram with configurable edges (from which percentile(s) can be derived), not incl. medium access delay}
- Total packets arriving, enqueued and dequeued (per queue per sample interval)
- ECN packets marked, non-ECN packets dropped, ECN packets dropped (per queue per sample interval), from which marking and dropping probabilities can be calculated
- Time and duration of each overload event.

The type of statistics produced for variables like Q delay (mean, percentiles, etc.) will depend on implementation constraints.

3. IANA Considerations

This specification contains no IANA considerations.

4. Security Considerations

4.1. Overload Handling

Where the interests of users or flows might conflict, it could be necessary to police traffic to isolate any harm to the performance of individual flows. However it is hard to avoid unintended side-effects with policing, and in a trusted environment policing is not necessary. Therefore per-flow policing needs to be separable from a basic AQM, as an option under policy control.

However, a basic DualQ AQM does at least need to handle overload. A useful objective would be for the overload behaviour of the DualQ AQM to be at least no worse than a single queue AQM. However, a trade-off needs to be made between complexity and the risk of either traffic class harming the other. In each of the following three subsections, an overload issue specific to the DualQ is described, followed by proposed solution(s).

Under overload the higher priority L4S service will have to sacrifice some aspect of its performance. Alternative solutions are provided below that each relax a different factor: e.g. throughput, delay, drop. Some of these choices might need to be determined by operator policy or by the developer, rather than by the IETF. {ToDo: Reach consensus on which it is to be in each case.}

4.1.1. Avoiding Classic Starvation: Sacrifice L4S Throughput or Delay?

Priority of L4S is required to be conditional to avoid total throughput starvation of Classic by heavy L4S traffic. This raises the question of whether to sacrifice L4S throughput or L4S delay (or some other policy) to mitigate starvation of Classic:

Sacrifice L4S throughput: By using weighted round robin as the conditional priority scheduler, the L4S service can sacrifice some throughput during overload to guarantee a minimum throughput service for Classic traffic. The scheduling weight of the Classic queue should be small (e.g. 1/16). Then, in most traffic scenarios the scheduler will not interfere and it will not need to - the coupling mechanism and the end-systems will share out the capacity across both queues as if it were a single pool. However,

because the congestion coupling only applies in one direction (from C to L), if L4S traffic is over-aggressive or unresponsive, the scheduler weight for Classic traffic will at least be large enough to ensure it does not starve.

In cases where the ratio of L4S to Classic flows (e.g. 19:1) is greater than the ratio of their scheduler weights (e.g. 15:1), the L4S flows will get less than an equal share of the capacity, but only slightly. For instance, with the example numbers given, each L4S flow will get (15/16)/19 = 4.9% when ideally each would get 1/20=5%. In the rather specific case of an unresponsive flow taking up a large part of the capacity set aside for L4S, using WRR could significantly reduce the capacity left for any responsive L4S flows.

Sacrifice L4S Delay: To control milder overload of responsive traffic, particularly when close to the maximum congestion signal, the operator could choose to control overload of the Classic queue by allowing some delay to 'leak' across to the L4S queue. The scheduler can be made to behave like a single First-In First-Out (FIFO) queue with different service times by implementing a very simple conditional priority scheduler that could be called a "time-shifted FIFO" (see the Modifier Earliest Deadline First (MEDF) scheduler of [MEDF]). This scheduler adds tshift to the queue delay of the next L4S packet, before comparing it with the queue delay of the next Classic packet, then it selects the packet with the greater adjusted queue delay. Under regular conditions, this time-shifted FIFO scheduler behaves just like a strict priority scheduler. But under moderate or high overload it prevents starvation of the Classic queue, because the time-shift (tshift) defines the maximum extra queuing delay of Classic packets relative to L4S.

The example implementation in Appendix A can implement either policy.

4.1.2. Congestion Signal Saturation: Introduce L4S Drop or Delay?

To keep the throughput of both L4S and Classic flows roughly equal over the full load range, a different control strategy needs to be defined above the point where one AQM first saturates to a probability of 100% leaving no room to push back the load any harder. If k>1, L4S will saturate first, but saturation can be caused by unresponsive traffic in either queue.

The term 'unresponsive' includes cases where a flow becomes temporarily unresponsive, for instance, a real-time flow that takes a while to adapt its rate in response to congestion, or a TCP-like flow that is normally responsive, but above a certain congestion level it

will not be able to reduce its congestion window below the minimum of 2 segments, effectively becoming unresponsive. (Note that L4S traffic ought to remain responsive below a window of 2 segments (see [I-D.ietf-tsvwg-ecn-l4s-id]).

Saturation raises the question of whether to relieve congestion by introducing some drop into the L4S queue or by allowing delay to grow in both queues (which could eventually lead to tail drop too):

Drop on Saturation: Saturation can be avoided by setting a maximum threshold for L4S ECN marking (assuming k>1) before saturation starts to make the flow rates of the different traffic types diverge. Above that the drop probability of Classic traffic is applied to all packets of all traffic types. Then experiments have shown that queueing delay can be kept at the target in any overload situation, including with unresponsive traffic, and no further measures are required.

Delay on Saturation: When L4S marking saturates, instead of switching to drop, the drop and marking probabilities could be capped. Beyond that, delay will grow either solely in the queue with unresponsive traffic (if WRR is used), or in both queues (if time-shifted FIFO is used). In either case, the higher delay ought to control temporary high congestion. If the overload is more persistent, eventually the combined DualQ will overflow and tail drop will control congestion.

The example implementation in Appendix A applies only the "drop on saturation" policy.

4.1.3. Protecting against Unresponsive ECN-Capable Traffic

Unresponsive traffic has a greater advantage if it is also ECN-capable. The advantage is undetectable at normal low levels of drop/marking, but it becomes significant with the higher levels of drop/marking typical during overload. This is an issue whether the ECN-capable traffic is L4S or Classic.

This raises the question of whether and when to switch off ECN marking and use solely drop instead, as required by both Section 7 of [RFC3168] and Section 4.2.1 of [RFC7567].

Experiments with the DualPI2 AQM (Appendix A) have shown that introducing 'drop on saturation' at 100% L4S marking addresses this problem with unresponsive ECN as well as addressing the saturation problem. It leaves only a small range of congestion levels where unresponsive traffic gains any advantage from using the ECN capability, and the advantage is hardly detectable [DualQ-Test].

5. Acknowledgements

Thanks to Anil Agarwal, Sowmini Varadhan's and Gabi Bracha for detailed review comments particularly of the appendices and suggestions on how to make our explanation clearer. Thanks also to Greg White and Tom Henderson for insights on the choice of schedulers and queue delay measurement techniques.

The authors' contributions were originally part-funded by the European Community under its Seventh Framework Programme through the Reducing Internet Transport Latency (RITE) project (ICT-317700). Bob Briscoe's contribution was also part-funded by the Research Council of Norway through the TimeIn project. The views expressed here are solely those of the authors.

6. References

6.1. Normative References

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, DOI 10.17487/RFC2119, March 1997, <https://www.rfc-editor.org/info/rfc2119>.

6.2. Informative References

[ARED01] Floyd, S., Gummadi, R., and S. Shenker, "Adaptive RED: An Algorithm for Increasing the Robustness of RED's Active Queue Management", ACIRI Technical Report , August 2001, <http://www.icir.org/floyd/red.html>.

[CoDel] Nichols, K. and V. Jacobson, "Controlling Queue Delay", ACM Queue 10(5), May 2012, <http://queue.acm.org/issuedetail.cfm?issue=2208917>.

[CRED_Insights] Briscoe, B., "Insights from Curvy RED (Random Early Detection)", BT Technical Report TR-TUB8-2015-003, July 2015, <http://www.bobbriscoe.net/projects/latency/credi_tr.pdf>.

[DCttH15] De Schepper, K., Bondarenko, O., Briscoe, B., and I. Tsang, "`Data Centre to the Home': Ultra-Low Latency for All", 2015, <http://www.bobbriscoe.net/projects/latency/dctth_preprint.pdf>.

(Under submission)

[DualQ-Test]
Steen, H., "Destruction Testing: Ultra-Low Delay using Dual Queue Coupled Active Queue Management", Masters Thesis, Dept of Informatics, Uni Oslo , May 2017.

[I-D.briscoe-tsvwg-l4s-diffserv]
Briscoe, B., "Interactions between Low Latency, Low Loss, Scalable Throughput (L4S) and Differentiated Services", draft-briscoe-tsvwg-l4s-diffserv-00 (work in progress), March 2018.

[I-D.ietf-tsvwg-ecn-l4s-id]
Schepper, K., Briscoe, B., and I. Tsang, "Identifying Modified Explicit Congestion Notification (ECN) Semantics for Ultra-Low Queuing Delay", draft-ietf-tsvwg-ecn-l4s-id-02 (work in progress), March 2018.

[I-D.ietf-tsvwg-l4s-arch]
Briscoe, B., Schepper, K., and M. Bagnulo, "Low Latency, Low Loss, Scalable Throughput (L4S) Internet Service: Architecture", draft-ietf-tsvwg-l4s-arch-02 (work in progress), March 2018.

[I-D.sridharan-tcpm-ctcp]
Sridharan, M., Tan, K., Bansal, D., and D. Thaler, "Compound TCP: A New TCP Congestion Control for High-Speed and Long Distance Networks", draft-sridharan-tcpm-ctcp-02 (work in progress), November 2008.

[Mathis09]
Mathis, M., "Relentless Congestion Control", PFLDNeT'09 , May 2009, <http://www.hpcc.jp/pfldnet2009/Program_files/1569198525.pdf>.

[MEDF] Menth, M., Schmid, M., Heiss, H., and T. Reim, "MEDF - a simple scheduling algorithm for two real-time transport service classes with application in the UTRAN", Proc. IEEE Conference on Computer Communications (INFOCOM'03) Vol.2 pp.1116-1122, March 2003.

[PI2] De Schepper, K., Bondarenko, O., Briscoe, B., and I. Tsang, "PI2: A Linearized AQM for both Classic and Scalable TCP", ACM CoNEXT'16 , December 2016, <https://riteproject.files.wordpress.com/2015/10/pi2_conext.pdf>.

(To appear)

[RFC0970] Nagle, J., "On Packet Switches With Infinite Storage", RFC 970, DOI 10.17487/RFC0970, December 1985, <https://www.rfc-editor.org/info/rfc970>.

[RFC2309] Braden, B., Clark, D., Crowcroft, J., Davie, B., Deering, S., Estrin, D., Floyd, S., Jacobson, V., Minshall, G., Partridge, C., Peterson, L., Ramakrishnan, K., Shenker, S., Wroclawski, J., and L. Zhang, "Recommendations on Queue Management and Congestion Avoidance in the Internet", RFC 2309, DOI 10.17487/RFC2309, April 1998, <https://www.rfc-editor.org/info/rfc2309>.

[RFC3168] Ramakrishnan, K., Floyd, S., and D. Black, "The Addition of Explicit Congestion Notification (ECN) to IP", RFC 3168, DOI 10.17487/RFC3168, September 2001, <https://www.rfc-editor.org/info/rfc3168>.

[RFC3246] Davie, B., Charny, A., Bennet, J., Benson, K., Le Boudec, J., Courtney, W., Davari, S., Firoiu, V., and D. Stiliadis, "An Expedited Forwarding PHB (Per-Hop Behavior)", RFC 3246, DOI 10.17487/RFC3246, March 2002, <https://www.rfc-editor.org/info/rfc3246>.

[RFC3649] Floyd, S., "HighSpeed TCP for Large Congestion Windows", RFC 3649, DOI 10.17487/RFC3649, December 2003, <https://www.rfc-editor.org/info/rfc3649>.

[RFC5681] Allman, M., Paxson, V., and E. Blanton, "TCP Congestion Control", RFC 5681, DOI 10.17487/RFC5681, September 2009, <https://www.rfc-editor.org/info/rfc5681>.

[RFC7567] Baker, F., Ed. and G. Fairhurst, Ed., "IETF Recommendations Regarding Active Queue Management", BCP 197, RFC 7567, DOI 10.17487/RFC7567, July 2015, <https://www.rfc-editor.org/info/rfc7567>.

[RFC8033] Pan, R., Natarajan, P., Baker, F., and G. White, "Proportional Integral Controller Enhanced (PIE): A Lightweight Control Scheme to Address the Bufferbloat Problem", RFC 8033, DOI 10.17487/RFC8033, February 2017, <https://www.rfc-editor.org/info/rfc8033>.

[RFC8034] White, G. and R. Pan, "Active Queue Management (AQM) Based on Proportional Integral Controller Enhanced PIE) for Data-Over-Cable Service Interface Specifications (DOCSIS) Cable Modems", RFC 8034, DOI 10.17487/RFC8034, February 2017, <https://www.rfc-editor.org/info/rfc8034>.

[RFC8257] Bensley, S., Thaler, D., Balasubramanian, P., Eggert, L., and G. Judd, "Data Center TCP (DCTCP): TCP Congestion Control for Data Centers", RFC 8257, DOI 10.17487/RFC8257, October 2017, <https://www.rfc-editor.org/info/rfc8257>.

[RFC8290] Hoeiland-Joergensen, T., McKenney, P., Taht, D., Gettys, J., and E. Dumazet, "The Flow Queue CoDel Packet Scheduler and Active Queue Management Algorithm", RFC 8290, DOI 10.17487/RFC8290, January 2018, <https://www.rfc-editor.org/info/rfc8290>.

[RFC8311] Black, D., "Relaxing Restrictions on Explicit Congestion Notification (ECN) Experimentation", RFC 8311, DOI 10.17487/RFC8311, January 2018, <https://www.rfc-editor.org/info/rfc8311>.

[RFC8312] Rhee, I., Xu, L., Ha, S., Zimmermann, A., Eggert, L., and R. Scheffenegger, "CUBIC for Fast Long-Distance Networks", RFC 8312, DOI 10.17487/RFC8312, February 2018, <https://www.rfc-editor.org/info/rfc8312>.

Appendix A. Example DualQ Coupled PI2 Algorithm

As a first concrete example, the pseudocode below gives the DualPI2 algorithm. DualPI2 follows the structure of the DualQ Coupled AQM framework in Figure 1. A simple step threshold (in units of queuing time) is used for the Native L4S AQM, but a ramp is also described as an alternative. And the PI2 algorithm [PI2] is used for the Classic AQM. PI2 is an improved variant of the PIE AQM [RFC8033].

We will introduce the pseudocode in two passes. The first pass explains the core concepts, deferring handling of overload to the second pass. To aid comparison, line numbers are kept in step between the two passes by using letter suffixes where the longer code needs extra lines.

A full open source implementation for Linux is available at: https://github.com/olgabo/dualpi2.

A.1. Pass #1: Core Concepts

The pseudocode manipulates three main structures of variables: the packet (pkt), the L4S queue (lq) and the Classic queue (cq). The pseudocode consists of the following four functions:

- initialization code (Figure 2) that sets parameter defaults (the API for setting non-default values is omitted for brevity)

- enqueue code (Figure 3)
- dequeue code (Figure 4)
- code to regularly update the base probability (p) used in the dequeue code (Figure 5).

It also uses the following functions that are not shown in full here:

- scheduler(), which selects between the head packets of the two queues; the choice of scheduler technology is discussed later;
- cq.len() or lq.len() returns the current length (aka. backlog) of the relevant queue in bytes;
- cq.time() or lq.time() returns the current queuing delay (aka. sojourn time or service time) of the relevant queue in units of time;

Queuing delay could be measured directly by storing a per-packet time-stamp as each packet is enqueued, and subtracting this from the system time when the packet is dequeued. If time-stamping is not easy to introduce with certain hardware, queuing delay could be predicted indirectly by dividing the size of the queue by the predicted departure rate, which might be known precisely for some link technologies (see for example [RFC8034]).

In our experiments so far (building on experiments with PIE) on broadband access links ranging from 4 Mb/s to 200 Mb/s with base RTTs from 5 ms to 100 ms, DualPI2 achieves good results with the default parameters in Figure 2. The parameters are categorised by whether they relate to the Base PI2 AQM, the L4S AQM or the framework coupling them together. Variables derived from these parameters are also included at the end of each category. Each parameter is explained as it is encountered in the walk-through of the pseudocode below.

```
1:  dualpi2_params_init(...) {         % Set input parameter defaults
2:    % PI2 AQM parameters
3:    target = 15 ms              % PI AQM Classic queue delay target
4:    Tupdate = 16 ms           % PI Classic queue sampling interval
5:    alpha = 10 Hz^2                              % PI integral gain
6:    beta = 100 Hz^2                          % PI proportional gain
7:    p_Cmax = 1/4                       % Max Classic drop/mark prob
8:    % Constants derived from PI2 AQM parameters
9:    alpha_U = alpha *Tupdate % PI integral gain per update interval
10:   beta_U = beta * Tupdate  % PI prop'nal gain per update interval
11:
12:   % DualQ Coupled framework parameters
13:   k = 2                                          % Coupling factor
14:   % scheduler weight or equival't parameter (scheduler-dependent)
15:   limit = MAX_LINK_RATE * 250 ms               % Dual buffer size
16:
17:   % L4S AQM parameters
18:   T_time = 1 ms                   % L4S marking threshold in time
19:   T_len = 2 * MTU            % Min L4S marking threshold in bytes
20:   % Constants derived from L4S AQM parameters
21:   p_Lmax = min(k*sqrt(p_Cmax), 1)          % Max L4S marking prob
22: }
```

Figure 2: Example Header Pseudocode for DualQ Coupled PI2 AQM

The overall goal of the code is to maintain the base probability (p), which is an internal variable from which the marking and dropping probabilities for L4S and Classic traffic (p_L and p_C) are derived. The variable named p in the pseudocode and in this walk-through is the same as p' (p-prime) in Section 2.4. The probabilities p_L and p_C are derived in lines 3, 4 and 5 of the dualpi2_update() function (Figure 5) then used in the dualpi2_dequeue() function (Figure 4). The code walk-through below builds up to explaining that part of the code eventually, but it starts from packet arrival.

```
1:  dualpi2_enqueue(lq, cq, pkt) { % Test limit and classify lq or cq
2:    if ( lq.len() + cq.len() > limit )
3:      drop(pkt)                     % drop packet if buffer is full
4:    else {                                    % Packet classifier
5:      if ( ecn(pkt) modulo 2 == 1 )       % ECN bits = ECT(1) or CE
6:        lq.enqueue(pkt)
7:      else                           % ECN bits = not-ECT or ECT(0)
8:        cq.enqueue(pkt)
9:    }
10: }
```

Figure 3: Example Enqueue Pseudocode for DualQ Coupled PI2 AQM

```
1:  dualpi2_dequeue(lq, cq, pkt) {     % Couples L4S & Classic queues
2:    while ( lq.len() + cq.len() > 0 )
3:      if ( scheduler() == lq ) {
4:        lq.dequeue(pkt)                      % Scheduler chooses lq

{ToDo: Generalize 5-7 for any L AQM (see email to Tom 9-Aug-18)}

5:        if ( ((lq.time() > T_time)              % step marking ...
6:              AND (lq.len() > T_len))
7:             OR (p_CL > rand()) )             % ...or linear marking
8:          mark(pkt)
9:      } else {
10:       cq.dequeue(pkt)                      % Scheduler chooses cq
11:       if ( p_C > rand() ) {               % probability p_C = p^2
12:         if ( ecn(pkt) == 0 ) {           % if ECN field = not-ECT
13:           drop(pkt)                                % squared drop
14:           continue        % continue to the top of the while loop
15:         }
16:         mark(pkt)                                  % squared mark
17:       }
18:     }
19:     return(pkt)                     % return the packet and stop
20:   }
21:   return(NULL)                             % no packet to dequeue
22: }
```

Figure 4: Example Dequeue Pseudocode for DualQ Coupled PI2 AQM

When packets arrive, first a common queue limit is checked as shown in line 2 of the enqueuing pseudocode in Figure 3. Note that the limit is deliberately tested before enqueue to avoid any bias against larger packets (so depending whether the implementation stores a packet while testing whether to drop it from the tail, it might be necessary for the actual buffer memory to be one MTU larger than limit).

Line 2 assumes an implementation where lq and cq share common buffer memory. An alternative implementation could use separate buffers for each queue, in which case the arriving packet would have to be classified first to determine which buffer to check for available space. The choice is a trade off; a shared buffer can use less memory whereas separate buffers isolate the L4S queue from tail-drop due to large bursts of Classic traffic (e.g. a Classic TCP during slow-start over a long RTT).

Returning to the shared buffer case, if limit is not exceeded, the packet will be classified and enqueued to the Classic or L4S queue dependent on the least significant bit of the ECN field in the IP

header (line 5). Packets with a codepoint having an LSB of 0 (Not-ECT and ECT(0)) will be enqueued in the Classic queue. Otherwise, ECT(1) and CE packets will be enqueued in the L4S queue. Optional additional packet classification flexibility is omitted for brevity (see [I-D.ietf-tsvwg-ecn-l4s-id]).

The dequeue pseudocode (Figure 4) is repeatedly called whenever the lower layer is ready to forward a packet. It schedules one packet for dequeuing (or zero if the queue is empty) then returns control to the caller, so that it does not block while that packet is being forwarded. While making this dequeue decision, it also makes the necessary AQM decisions on dropping or marking. The alternative of applying the AQMs at enqueue would shift some processing from the critical time when each packet is dequeued. However, it would also add a whole queue of delay to the control signals, making the control loop very sloppy.

All the dequeue code is contained within a large while loop so that if it decides to drop a packet, it will continue until it selects a packet to schedule. Line 3 of the dequeue pseudocode is where the scheduler chooses between the L4S queue (lq) and the Classic queue (cq). Detailed implementation of the scheduler is not shown (see discussion later).

- If an L4S packet is scheduled, lines 5 to 8 mark the packet if either the L4S threshold (T_time) is exceeded, or if a random marking decision is drawn according to p_CL (maintained by the dualpi2_update() function discussed below). This logical 'OR' on a per-packet basis implements the max() function shown in Figure 1 to couple the outputs of the two AQMs together. The L4S threshold is usually in units of time (default T_time = 1 ms). However, on slow links the packet serialization time can approach the threshold T_time, so line 6 sets a floor of T_len (=2 MTU) to the threshold, otherwise marking is always too frequent on slow links.

- If a Classic packet is scheduled, lines 10 to 17 drop or mark the packet based on the squared probability p_C.

There is some concern that using a step function for the Native L4S AQM requires end-systems to smooth the signal for a lot longer - until its fidelity is sufficient. The latency benefits of a ramp are being investigated as a simple alternative to the step. This ramp would be similar to the RED algorithm, with the following differences:

- The min and max of the ramp are defined in units of queuing delay, not bytes, so that configuration remains invariant as the queue departure rate varies.

- It uses instantaneous queueing delay without smoothing (smoothing is done in the end-systems).

- Determinism is being experimented with instead of randomness; to reduce the delay necessary to smooth out the noise of randomness from the signal. For each packet, the algorithm would accumulate p'_L in a counter and mark the packet that took the counter over 1, then subtract 1 from the counter and continue.

- The ramp rises linearly directly from 0 to 1, not to a an intermediate value of p'_L as RED would, because there is no need to keep ECN marking probability low.

This ramp algorithm would require two configuration parameters (min and max threshold in units of queuing time), in contrast to the single parameter of a step.

```
1:  dualpi2_update(lq, cq, target) {         % Update p every Tupdate
2:    curq = cq.time()  % use queuing time of first-in Classic packet
3:    p = p + alpha_U * (curq - target) + beta_U * (curq - prevq)
4:    p_CL = p * k   % Coupled L4S prob = base prob * coupling factor
5:    p_C = p^2                        % Classic prob = (base prob)^2
6:    prevq = curq
7:  }
```

Figure 5: Example PI-Update Pseudocode for DualQ Coupled PI2 AQM

The base probability (p) is kept up to date by the core PI algorithm in Figure 5, which is executed every Tupdate.

Note that p solely depends on the queuing time in the Classic queue. In line 2, the current queuing delay (curq) is evaluated from how long the head packet was in the Classic queue (cq). The function cq.time() (not shown) subtracts the time stamped at enqueue from the current time and implicitly takes the current queuing delay as 0 if the queue is empty.

The algorithm centres on line 3, which is a classical Proportional-Integral (PI) controller that alters p dependent on: a) the error between the current queuing delay (curq) and the target queuing delay ('target' - see [RFC8033]); and b) the change in queuing delay since the last sample. The name 'PI' represents the fact that the second factor (how fast the queue is growing) is _P_roportional to load while the first is the _I_ntegral of the load (so it removes any standing queue in excess of the target).

The two 'gain factors' in line 3, alpha_U and beta_U, respectively weight how strongly each of these elements ((a) and (b)) alters p.

They are in units of 'per second of delay' or Hz, because they transform differences in queueing delay into changes in probability.

alpha_U and beta_U are derived from the input parameters alpha and beta (see lines 5 and 6 of Figure 2). These recommended values of alpha and beta come from the stability analysis in [PI2] so that the AQM can change p as fast as possible in response to changes in load without over-compensating and therefore causing oscillations in the queue.

alpha and beta determine how much p ought to change if it was updated every second. It is best to update p as frequently as possible, but the update interval (Tupdate) will probably be constrained by hardware performance. For link rates from 4 - 200 Mb/s, we found Tupdate=16ms (as recommended in [RFC8033]) is sufficient. However small the chosen value of Tupdate, p should change by the same amount per second, but in finer more frequent steps. So the gain factors used for updating p in Figure 5 need to be scaled by (Tupdate/1s), which is done in lines 9 and 10 of Figure 2). The suffix '_U' represents 'per update time' (Tupdate).

In corner cases, p can overflow the range [0,1] so the resulting value of p has to be bounded (omitted from the pseudocode). Then, as already explained, the coupled and Classic probabilities are derived from the new p in lines 4 and 5 as p_CL = k*p and p_C = p^2.

Because the coupled L4S marking probability (p_CL) is factored up by k, the dynamic gain parameters alpha and beta are also inherently factored up by k for the L4S queue, which is necessary to ensure that Classic TCP and DCTCP controls have the same stability. So, if alpha is 10 Hz^2, the effective gain factor for the L4S queue is k*alpha, which is 20 Hz^2 with the default coupling factor of k=2.

Unlike in PIE [RFC8033], alpha_U and beta_U do not need to be tuned every Tupdate dependent on p. Instead, in PI2, alpha_U and beta_U are independent of p because the squaring applied to Classic traffic tunes them inherently. This is explained in [PI2], which also explains why this more principled approach removes the need for most of the heuristics that had to be added to PIE.

{ToDo: Scaling beta with Tupdate and scaling both alpha & beta with RTT}

A.2. Pass #2: Overload Details

Figure 6 repeats the dequeue function of Figure 4, but with overload details added. Similarly Figure 7 repeats the core PI algorithm of

Figure 5 with overload details added. The initialization and enqueue functions are unchanged.

In line 7 of the initialization function (Figure 2), the default maximum Classic drop probability p_Cmax = 1/4 or 25%. This is the point at which it is deemed that the Classic queue has become persistently overloaded, so it switches to using solely drop, even for ECN-capable packets. This protects the queue against any unresponsive traffic that falsely claims that it is responsive to ECN marking, as required by [RFC3168] and [RFC7567].

Line 21 of the initialization function translates this into a maximum L4S marking probability (p_Lmax) by rearranging Equation (1). With a coupling factor of k=2 (the default) or greater, this translates to a maximum L4S marking probability of 1 (or 100%). This is intended to ensure that the L4S queue starts to introduce dropping once marking saturates and can rise no further. The 'TCP Prague' requirements [I-D.ietf-tsvwg-ecn-l4s-id] state that, when an L4S congestion control detects a drop, it falls back to a response that coexists with 'Classic' TCP. So it is correct that the L4S queue drops packets proportional to p^2, as if they are Classic packets.

Both these switch-overs are triggered by the tests for overload introduced in lines 4b and 12b of the dequeue function (Figure 6). Lines 8c to 8g drop L4S packets with probability p^2. Lines 8h to 8i mark the remaining packets with probability p_CL. If p_Lmax = 1, which is the suggested default configuration, all remaining packets will be marked because, to have reached the else block at line 8b, p_CL >= 1.

Lines 2c to 2d in the core PI algorithm (Figure 7) deal with overload of the L4S queue when there is no Classic traffic. This is necessary, because the core PI algorithm maintains the appropriate drop probability to regulate overload, but it depends on the length of the Classic queue. If there is no Classic queue the naive algorithm in Figure 5 drops nothing, even if the L4S queue is overloaded - so tail drop would have to take over (lines 3 and 4 of Figure 3).

If the test at line 2a finds that the Classic queue is empty, line 2d measures the current queue delay using the L4S queue instead. While the L4S queue is not overloaded, its delay will always be tiny compared to the target Classic queue delay. So p_L will be driven to zero, and the L4S queue will naturally be governed solely by threshold marking (lines 5 and 6 of the dequeue algorithm in Figure 6). But, if unresponsive L4S source(s) cause overload, the DualQ transitions smoothly to L4S marking based on the PI algorithm.

And as overload increases, it naturally transitions from marking to dropping by the switch-over mechanism already described.

```
1:  dualpi2_dequeue(lq, cq) { % Couples L4S & Classic queues, lq & cq
2:    while ( lq.len() + cq.len() > 0 )
3:      if ( scheduler() == lq ) {
4a:       lq.dequeue(pkt)
4b:       if ( p_CL < p_Lmax ) {      % Check for overload saturation
5:          if ( ((lq.time() > T_time)             % step marking ...
6:                AND (lq.len > T_len))
7:               OR (p_CL > rand()) )           % ...or linear marking
8a:            mark(pkt)
8b:       } else {                                % overload saturation
8c:         if ( p_C > rand() ) {             % probability p_C = p^2
8e:           drop(pkt)      % revert to Classic drop due to overload
8f:           continue        % continue to the top of the while loop
8g:         }
8h:         if ( p_CL > rand() )           % probability p_CL = k * p
8i:           mark(pkt)         % linear marking of remaining packets
8j:       }
9:      } else {
10:       cq.dequeue(pkt)
11:       if ( p_C > rand() ) {               % probability p_C = p^2
12a:        if ( (ecn(pkt) == 0)                % ECN field = not-ECT
12b:             OR (p_C >= p_Cmax) ) {       % Overload disables ECN
13:           drop(pkt)                      % squared drop, redo loop
14:           continue        % continue to the top of the while loop
15:         }
16:         mark(pkt)                                 % squared mark
17:       }
18:     }
19:     return(pkt)                     % return the packet and stop
20:   }
21:   return(NULL)                            % no packet to dequeue
22: }
```

Figure 6: Example Dequeue Pseudocode for DualQ Coupled PI2 AQM (Including Integer Arithmetic and Overload Code)

```
1:  dualpi2_update(lq, cq, target) {         % Update p every Tupdate
2a:   if ( cq.len() > 0 )
2b:     curq = cq.time() %use queuing time of first-in Classic packet
2c:   else                                      % Classic queue empty
2d:     curq = lq.time()    % use queuing time of first-in L4S packet
3:    p = p + alpha_U * (curq - target) + beta_U * (curq - prevq)
4:    p_CL = p * k   % Coupled L4S prob = base prob * coupling factor
5:    p_C = p^2                        % Classic prob = (base prob)^2
6:    prevq = curq
7:  }
```

Figure 7: Example PI-Update Pseudocode for DualQ Coupled PI2 AQM (Including Overload Code)

The choice of scheduler technology is critical to overload protection (see Section 4.1).

- A well-understood weighted scheduler such as weighted round robin (WRR) is recommended. The scheduler weight for Classic should be low, e.g. 1/16.

- Alternatively, a time-shifted FIFO could be used. This is a very simple scheduler, but it does not fully isolate latency in the L4S queue from uncontrolled bursts in the Classic queue. It works by selecting the head packet that has waited the longest, biased against the Classic traffic by a time-shift of tshift. To implement time-shifted FIFO, the "if (scheduler() == lq)" test in line 3 of the dequeue code would simply be replaced by "if (lq.time() + tshift >= cq.time())". For the public Internet a good value for tshift is 50ms. For private networks with smaller diameter, about 4*target would be reasonable.

- A strict priority scheduler would be inappropriate, because it would starve Classic if L4S was overloaded.

Appendix B. Example DualQ Coupled Curvy RED Algorithm

As another example of a DualQ Coupled AQM algorithm, the pseudocode below gives the Curvy RED based algorithm we used and tested. Although we designed the AQM to be efficient in integer arithmetic, to aid understanding it is first given using real-number arithmetic. Then, one possible optimization for integer arithmetic is given, also in pseudocode. To aid comparison, the line numbers are kept in step between the two by using letter suffixes where the longer code needs extra lines.

```
1:  dualq_dequeue(lq, cq) {  % Couples L4S & Classic queues, lq & cq
2:    if ( lq.dequeue(pkt) ) {
3a:     p_L = cq.sec() / 2^S_L
3b:     if ( lq.byt() > T )
3c:       mark(pkt)
3d:     elif ( p_L > maxrand(U) )
4:        mark(pkt)
5:      return(pkt)               % return the packet and stop here
6:    }
7:    while ( cq.dequeue(pkt) ) {
8a:     alpha = 2^(-f_C)
8b:     Q_C = alpha * pkt.sec() + (1-alpha)* Q_C    % Classic Q EWMA
9a:     sqrt_p_C = Q_C / 2^S_C
9b:     if ( sqrt_p_C > maxrand(2*U) )
10:       drop(pkt)                       % Squared drop, redo loop
11:     else
12:       return(pkt)             % return the packet and stop here
13:   }
14:   return(NULL)                          % no packet to dequeue
15: }

16: maxrand(u) {               % return the max of u random numbers
17:     maxr=0
18:     while (u-- > 0)
19:         maxr = max(maxr, rand())              % 0 <= rand() < 1
20:     return(maxr)
21: }
```

Figure 8: Example Dequeue Pseudocode for DualQ Coupled Curvy RED AQM

Packet classification code is not shown, as it is no different from Figure 3. Potential classification schemes are discussed in Section 2.3. The Curvy RED algorithm has not been maintained to the same degree as the DualPI2 algorithm. Some ideas used in DualPI2 would need to be translated into Curvy RED, such as i) the conditional priority scheduler instead of strict priority ii) the time-based L4S threshold; iii) turning off ECN as overload protection; iv) Classic ECN support. These are not shown in the Curvy RED pseudocode, but would need to be implemented for production. {ToDo}

At the outer level, the structure of dualq_dequeue() implements strict priority scheduling. The code is written assuming the AQM is applied on dequeue (Note 1) . Every time dualq_dequeue() is called, the if-block in lines 2-6 determines whether there is an L4S packet to dequeue by calling lq.dequeue(pkt), and otherwise the while-block in lines 7-13 determines whether there is a Classic packet to dequeue, by calling cq.dequeue(pkt). (Note 2)

In the lower priority Classic queue, a while loop is used so that, if the AQM determines that a classic packet should be dropped, it continues to test for classic packets deciding whether to drop each until it actually forwards one. Thus, every call to dualq_dequeue() returns one packet if at least one is present in either queue, otherwise it returns NULL at line 14. (Note 3)

Within each queue, the decision whether to drop or mark is taken as follows (to simplify the explanation, it is assumed that U=1):

L4S: If the test at line 2 determines there is an L4S packet to dequeue, the tests at lines 3a and 3c determine whether to mark it. The first is a simple test of whether the L4S queue (lq.byt() in bytes) is greater than a step threshold T in bytes (Note 4). The second test is similar to the random ECN marking in RED, but with the following differences: i) the marking function does not start with a plateau of zero marking until a minimum threshold, rather the marking probability starts to increase as soon as the queue is positive; ii) marking depends on queuing time, not bytes, in order to scale for any link rate without being reconfigured; iii) marking of the L4S queue does not depend on itself, it depends on the queuing time of the _other_ (Classic) queue, where cq.sec() is the queuing time of the packet at the head of the Classic queue (zero if empty); iv) marking depends on the instantaneous queuing time (of the other Classic queue), not a smoothed average; v) the queue is compared with the maximum of U random numbers (but if U=1, this is the same as the single random number used in RED).

Specifically, in line 3a the marking probability p_L is set to the Classic queueing time qc.sec() in seconds divided by the L4S scaling parameter 2^S_L, which represents the queuing time (in seconds) at which marking probability would hit 100%. Then in line 3d (if U=1) the result is compared with a uniformly distributed random number between 0 and 1, which ensures that marking probability will linearly increase with queueing time. The scaling parameter is expressed as a power of 2 so that division can be implemented as a right bit-shift (>>) in line 3 of the integer variant of the pseudocode (Figure 9).

Classic: If the test at line 7 determines that there is at least one Classic packet to dequeue, the test at line 9b determines whether to drop it. But before that, line 8b updates Q_C, which is an exponentially weighted moving average (Note 5) of the queuing time in the Classic queue, where pkt.sec() is the instantaneous queueing time of the current Classic packet and alpha is the EWMA constant for the classic queue. In line 8a, alpha is represented as an integer power of 2, so that in line 8 of the integer code

the division needed to weight the moving average can be implemented by a right bit-shift (>> f_C).

Lines 9a and 9b implement the drop function. In line 9a the averaged queuing time Q_C is divided by the Classic scaling parameter 2^S_C, in the same way that queuing time was scaled for L4S marking. This scaled queuing time is given the variable name sqrt_p_C because it will be squared to compute Classic drop probability, so before it is squared it is effectively the square root of the drop probability. The squaring is done by comparing it with the maximum out of two random numbers (assuming U=1). Comparing it with the maximum out of two is the same as the logical `AND' of two tests, which ensures drop probability rises with the square of queuing time (Note 6). Again, the scaling parameter is expressed as a power of 2 so that division can be implemented as a right bit-shift in line 9 of the integer pseudocode.

The marking/dropping functions in each queue (lines 3 & 9) are two cases of a new generalization of RED called Curvy RED, motivated as follows. When we compared the performance of our AQM with fq_CoDel and PIE, we came to the conclusion that their goal of holding queuing delay to a fixed target is misguided [CRED_Insights]. As the number of flows increases, if the AQM does not allow TCP to increase queuing delay, it has to introduce abnormally high levels of loss. Then loss rather than queuing becomes the dominant cause of delay for short flows, due to timeouts and tail losses.

Curvy RED constrains delay with a softened target that allows some increase in delay as load increases. This is achieved by increasing drop probability on a convex curve relative to queue growth (the square curve in the Classic queue, if U=1). Like RED, the curve hugs the zero axis while the queue is shallow. Then, as load increases, it introduces a growing barrier to higher delay. But, unlike RED, it requires only one parameter, the scaling, not three. The diadvantage of Curvy RED is that it is not adapted to a wide range of RTTs. Curvy RED can be used as is when the RTT range to support is limited otherwise an adaptation mechanism is required.

There follows a summary listing of the two parameters used for each of the two queues:

Classic:

S_C : The scaling factor of the dropping function scales Classic queuing times in the range [0, 2^(S_C)] seconds into a dropping probability in the range [0,1]. To make division efficient, it is constrained to be an integer power of two;

f_C : To smooth the queuing time of the Classic queue and make multiplication efficient, we use a negative integer power of two for the dimensionless EWMA constant, which we define as alpha = 2^(-f_C).

L4S :

S_L (and k'): As for the Classic queue, the scaling factor of the L4S marking function scales Classic queueing times in the range [0, 2^(S_L)] seconds into a probability in the range [0,1]. Note that S_L = S_C + k', where k' is the coupling between the queues. So S_L and k' count as only one parameter; k' is related to k in Equation (1) (Section 2.1) by k=2^k', where both k and k' are constants. Then implementations can avoid costly division by shifting p_L by k' bits to the right.

T : The queue size in bytes at which step threshold marking starts in the L4S queue.

{ToDo: These are the raw parameters used within the algorithm. A configuration front-end could accept more meaningful parameters and convert them into these raw parameters.}

From our experiments so far, recommended values for these parameters are: S_C = -1; f_C = 5; T = 5 * MTU for the range of base RTTs typical on the public Internet. [CRED_Insights] explains why these parameters are applicable whatever rate link this AQM implementation is deployed on and how the parameters would need to be adjusted for a scenario with a different range of RTTs (e.g. a data centre) {ToDo incorporate a summary of that report into this draft}. The setting of k depends on policy (see Section 2.5 and Appendix C respectively for its recommended setting and guidance on alternatives).

There is also a cUrviness parameter, U, which is a small positive integer. It is likely to take the same hard-coded value for all implementations, once experiments have determined a good value. We have solely used U=1 in our experiments so far, but results might be even better with U=2 or higher.

Note that the dropping function at line 9 calls maxrand(2*U), which gives twice as much curviness as the call to maxrand(U) in the marking function at line 3. This is the trick that implements the square rule in equation (1) (Section 2.1). This is based on the fact that, given a number X from 1 to 6, the probability that two dice throws will both be less than X is the square of the probability that one throw will be less than X. So, when U=1, the L4S marking function is linear and the Classic dropping function is squared. If

U=2, L4S would be a square function and Classic would be quartic. And so on.

The maxrand(u) function in lines 16-21 simply generates u random numbers and returns the maximum (Note 7). Typically, maxrand(u) could be run in parallel out of band. For instance, if U=1, the Classic queue would require the maximum of two random numbers. So, instead of calling maxrand(2*U) in-band, the maximum of every pair of values from a pseudorandom number generator could be generated out-of-band, and held in a buffer ready for the Classic queue to consume.

```
1:  dualq_dequeue(lq, cq) {  % Couples L4S & Classic queues, lq & cq
2:     if ( lq.dequeue(pkt) ) {
3:        if ((lq.byt() > T) || ((cq.ns() >> (S_L-2)) > maxrand(U)))
4:           mark(pkt)
5:        return(pkt)              % return the packet and stop here
6:     }
7:     while ( cq.dequeue(pkt) ) {
8:         Q_C += (pkt.ns() - Q_C) >> f_C           % Classic Q EWMA
9:        if ( (Q_C >> (S_C-2) ) > maxrand(2*U) )
10:          drop(pkt)                    % Squared drop, redo loop
11:       else
12:          return(pkt)           % return the packet and stop here
13:    }
14:    return(NULL)                           % no packet to dequeue
15: }
```

Figure 9: Optimised Example Dequeue Pseudocode for Coupled DualQ AQM using Integer Arithmetic

Notes:

1. The drain rate of the queue can vary if it is scheduled relative to other queues, or to cater for fluctuations in a wireless medium. To auto-adjust to changes in drain rate, the queue must be measured in time, not bytes or packets [CoDel]. In our Linux implementation, it was easiest to measure queuing time at dequeue. Queuing time can be estimated when a packet is enqueued by measuring the queue length in bytes and dividing by the recent drain rate.

2. An implementation has to use priority queueing, but it need not implement strict priority.

3. If packets can be enqueued while processing dequeue code, an implementer might prefer to place the while loop around both queues so that it goes back to test again whether any L4S packets arrived while it was dropping a Classic packet.

4. In order not to change too many factors at once, for now, we keep the marking function for DCTCP-only traffic as similar as possible to DCTCP. However, unlike DCTCP, all processing is at dequeue, so we determine whether to mark a packet at the head of the queue by the byte-length of the queue _behind_ it. We plan to test whether using queuing time will work in all circumstances, and if we find that the step can cause oscillations, we will investigate replacing it with a steep random marking curve.

5. An EWMA is only one possible way to filter bursts; other more adaptive smoothing methods could be valid and it might be appropriate to decrease the EWMA faster than it increases.

6. In practice at line 10 the Classic queue would probably test for ECN capability on the packet to determine whether to drop or mark the packet. However, for brevity such detail is omitted. All packets classified into the L4S queue have to be ECN-capable, so no dropping logic is necessary at line 3. Nonetheless, L4S packets could be dropped by overload code (see Section 4.1).

7. In the integer variant of the pseudocode (Figure 9) real numbers are all represented as integers scaled up by 2^32. In lines 3 & 9 the function maxrand() is arranged to return an integer in the range 0 <= maxrand() < 2^32. Queuing times are also scaled up by 2^32, but in two stages: i) In lines 3 and 8 queuing times cq.ns() and pkt.ns() are returned in integer nanoseconds, making the values about 2^30 times larger than when the units were seconds, ii) then in lines 3 and 9 an adjustment of -2 to the right bit-shift multiplies the result by 2^2, to complete the scaling by 2^32.

Appendix C. Guidance on Controlling Throughput Equivalence

RTT_C / RTT_L	Reno	Cubic
1	k'=1	k'=0
2	k'=2	k'=1
3	k'=2	k'=2
4	k'=3	k'=2
5	k'=3	k'=3

Table 1: Value of k' for which DCTCP throughput is roughly the same as Reno or Cubic, for some example RTT ratios

k' is related to k in Equation (1) (Section 2.1) by k=2^k'.

To determine the appropriate policy, the operator first has to judge whether it wants DCTCP flows to have roughly equal throughput with Reno or with Cubic (because, even in its Reno-compatibility mode, Cubic is about 1.4 times more aggressive than Reno). Then the operator needs to decide at what ratio of RTTs it wants DCTCP and Classic flows to have roughly equal throughput. For example choosing k'=0 (equivalent to k=1) will make DCTCP throughput roughly the same as Cubic, _if their RTTs are the same_.

However, even if the base RTTs are the same, the actual RTTs are unlikely to be the same, because Classic (Cubic or Reno) traffic needs a large queue to avoid under-utilization and excess drop, whereas L4S (DCTCP) does not. The operator might still choose this policy if it judges that DCTCP throughput should be rewarded for keeping its own queue short.

On the other hand, the operator will choose one of the higher values for k', if it wants to slow DCTCP down to roughly the same throughput as Classic flows, to compensate for Classic flows slowing themselves down by causing themselves extra queuing delay.

The values for k' in the table are derived from the formulae, which was developed in [DCttH15]:

$$2^{k'} = 1.64 \, (RTT_{reno} / RTT_{dc}) \quad (2)$$

$$2^{k'} = 1.19 \, (RTT_{cubic} / RTT_{dc}) \quad (3)$$

For localized traffic from a particular ISP's data centre, we used the measured RTTs to calculate that a value of k'=3 (equivalant to k=8) would achieve throughput equivalence, and our experiments verified the formula very closely.

For a typical mix of RTTs from local data centres and across the general Internet, a value of k'=1 (equivalent to k=2) is recommended as a good workable compromise.

Appendix D. Open Issues

Most of the following open issues are also tagged '{ToDo}' at the appropriate point in the document:

Operational guidance to monitor L4S experiment

PI2 appendix: scaling of alpha & beta, esp. dependence of beta_U on Tupdate

Curvy RED appendix: complete the unfinished parts

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