Transport Area working group (tsvwg)
Internet-Draft
Intended status: Experimental
Expires: September 10, 2020

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March 9, 2020

DualQ Coupled AQMs for Low Latency, Low Loss and Scalable Throughput (L4S)
draft-ietf-tsvwg-aqm-dualq-coupled-11

Abstract

The Low Latency Low Loss Scalable Throughput (L4S) architecture allows data flows over the public Internet to achieve consistent low queuing latency, generally zero congestion loss and scaling of per-flow throughput without the scaling problems of standard TCP Reno-friendly congestion controls. To achieve this, L4S data flows have to use one of the family of 'Scalable' congestion controls (TCP Prague and Data Center TCP are examples) and a form of Explicit Congestion Notification (ECN) with modified behaviour. However, until now, Scalable congestion controls did not co-exist with existing Reno/Cubic traffic --- Scalable controls are so aggressive that 'Classic' (e.g. Reno-friendly) algorithms sharing an ECN-capable queue would drive themselves to a small capacity share. Therefore, until now, L4S controls could only be deployed where a clean-slate environment could be arranged, such as in private data centres (hence the name DCTCP). This specification defines `DualQ Coupled Active Queue Management (AQM)', which enables Scalable congestion controls that comply with the Prague L4S requirements to co-exist safely with Classic Internet traffic.

Analytical study and implementation testing of the Coupled AQM have shown that Scalable and Classic flows competing under similar conditions run at roughly the same rate. It achieves this indirectly, without having to inspect transport layer flow identifiers. When tested in a residential broadband setting, DCTCP also achieves 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 has low complexity and requires no configuration for the public Internet.

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1. Introduction

This document specifies a framework for DualQ Coupled AQMs, which is the network part of the L4S architecture [I-D.ietf-tsvwg-l4s-arch]. L4S enables both ultra-low queuing latency (sub-millisecond on average) and high throughput at the same time, for ad hoc numbers of capacity-seeking applications all sharing the same capacity.

1.1. Outline of the Problem

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 intermittent component of latency.

Traditionally ultra-low latency has only been available for a few selected low rate applications, that confine their sending rate within a specially carved-off portion of capacity, which is prioritized over other traffic, e.g. Diffserv EF [RFC3246]. Up to now it has not been possible to allow any number of low latency, high throughput applications to seek to fully utilize available capacity, because the capacity-seeking process itself causes too much queuing delay.

To reduce this queuing delay caused by the capacity seeking process, changes either to the network alone or to end-systems alone are in progress. L4S involves a recognition that both approaches are yielding diminishing returns:

- Recent state-of-the-art active queue management (AQM) in the network, e.g. fq_CoDel [RFC8290], PIE [RFC8033], Adaptive RED [ARED01]) has reduced queuing delay for all traffic, not just a select few applications. However, no matter how good the AQM, the capacity-seeking (sawtoothing) rate of TCP-like congestion controls represents a lower limit that will either cause queuing delay to vary or cause the link to be under-utilized. These AQMs are tuned to allow a typical capacity-seeking Reno-friendly flow to induce an average queue that roughly doubles the base RTT, adding 5-15 ms of queuing on average (cf. 500 microseconds with L4S for the same mix of long-running and web traffic). However, for many applications low delay is not useful unless it is consistently low. With these AQMs, 99th percentile queuing delay is 20-30 ms (cf. 2 ms with the same traffic over L4S).

- Similarly, recent research into using e2e congestion control without needing an AQM in the network (e.g.BBRv1 [BBRv1]) seems to have hit a similar lower limit to queuing delay of about 20ms on average (and any additional BBRv1 flow adds another 20ms of queuing) but there are also regular 25ms delay spikes due to bandwidth probes and 60ms spikes due to flow-starts.

L4S learns from the experience of Data Center TCP [RFC8257], which shows the power of complementary changes both in the network and on end-systems. DCTCP teaches us that two small but radical changes to congestion control are needed to cut the two major outstanding causes of queuing delay variability:

1. Far smaller rate variations (sawteeth) than Reno-friendly congestion controls;

2. A shift of smoothing and hence smoothing delay from network to sender.

Without the former, a 'Classic' (e.g. Reno-friendly) flow's round trip time (RTT) varies between roughly 1 and 2 times the base RTT between the machines in question. Without the latter a 'Classic' flow's response to changing events is delayed by a worst-case (transcontinental) RTT, which could be hundreds of times the actual smoothing delay needed for the RTT of typical traffic from localized CDNs.

These changes are the two main features of the family of so-called 'Scalable' congestion controls (which includes DCTCP). Both these changes only reduce delay in combination with a complementary change in the network and they are both only feasible with ECN, not drop, for the signalling:

1. The smaller sawteeth allow an extremely shallow ECN packet-marking threshold in the queue.

2. And no smoothing in the network means that every fluctuation of the queue is signalled immediately.

Without ECN, either of these would lead to very high loss levels. But, with ECN, the resulting high marking levels are just signals, not impairments.

However, until now, Scalable congestion controls (like DCTCP) did not co-exist well in a shared ECN-capable queue with existing ECN-capable TCP Reno [RFC5681] or Cubic [RFC8312] congestion controls --- Scalable controls are so aggressive that these 'Classic' algorithms would drive themselves to a small capacity share. Therefore, until now, L4S controls could only be deployed where a clean-slate environment could be arranged, such as in private data centres (hence the name DCTCP).

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. It is not like flow-queuing approaches [RFC8290] that classify packets by flow identifier into separate queues in order to isolate sparse flows from the higher latency in the queues assigned to heavier flows. If a flow needs both low delay and high throughput, having a queue to itself does not isolate it from the harm it causes to itself. In contrast, L4S addresses the root cause of the latency problem --- it is an enabler for the smooth low latency scalable behaviour of Scalable congestion controls, so that every packet in every flow can enjoy very low

latency, then there is no need to isolate each flow into a separate queue.

1.2. Scope

L4S involves complementary changes in the network and on end-systems:

Network: A DualQ Coupled AQM (defined in the present document);

End-system: A Scalable congestion control (defined in Section 2.1).

Packet identifier: The network and end-system parts of L4S can be deployed incrementally, because they both identify L4S packets using the experimentally assigned explicit congestion notification (ECN) codepoints in the IP header: ECT(1) and CE [RFC8311] [I-D.ietf-tsvwg-ecn-l4s-id].

Data Center TCP (DCTCP [RFC8257]) is an example of a Scalable congestion control that has been deployed for some time in Linux, Windows and FreeBSD operating systems and Relentless TCP [Mathis09] is another example. During the progress of this document through the IETF a number of other Scalable congestion controls were implemented, e.g. TCP Prague [PragueLinux], QUIC Prague and the L4S variant of SCREAM for real-time media [RFC8298]. (Note: after the v3.19 Linux kernel, bugs were introduced into DCTCP's scalable behaviour and not all the patches applied for L4S evaluation had been applied to the mainline Linux kernel, which was at v5.5 at the time of writing. TCP Prague includes these patches and is available for all these Linux kernels).

The focus of this specification is to enable deployment of the network part of the L4S service. Then, without any management intervention, applications can exploit this new network capability as their operating systems migrate to Scalable congestion controls, which can then evolve _while_ their benefits are being enjoyed by everyone on the Internet.

The DualQ Coupled AQM framework can incorporate any AQM designed for a single queue that generates a statistical or deterministic mark/drop probability driven by the queue dynamics. Pseudocode examples of two different DualQ Coupled AQMs are given in the appendices. In many cases the framework 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.

For the public Internet, nearly all the benefit will typically be achieved by deploying the Coupled AQM into either end of the access link between a 'site' and the Internet, which is invariably the bottleneck. Here, the term 'site' is used loosely to mean a home, an office, a campus or mobile user equipment.

Latency is not the only concern of L4S:

- The 'Low Loss" part of the name denotes that L4S generally achieves zero congestion loss (which would otherwise cause retransmission delays), due to its use of ECN.

- The "Scalable throughput" part of the name denotes that the per-flow throughput of Scalable congestion controls should scale indefinitely, avoiding the imminent scaling problems with 'TCP-Friendly' congestion control algorithms [RFC3649].

The former is clearly in scope of this AQM document. However, the latter is an outcome of the end-system behaviour, and therefore outside the scope of this AQM document, even though the AQM is an enabler.

The overall L4S architecture [I-D.ietf-tsvwg-l4s-arch] gives more detail, including on wider deployment aspects such as backwards compatibility of Scalable congestion controls in bottlenecks where a DualQ Coupled AQM has not been deployed. The supporting papers [PI2] and [DCttH15] give the full rationale for the AQM's design, both discursively and in more precise mathematical form.

1.3. 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] when, and only when, they appear in all capitals, as shown here.

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 service/queue: The Classic service is intended for all the congestion control behaviours that co-exist with Reno [RFC5681] (e.g. Reno itself, Cubic [RFC8312], TFRC [RFC5348]).

Low-Latency, Low-Loss Scalable throughput (L4S) service/queue: The 'L4S' service is intended for traffic from scalable congestion control algorithms, such as Data Center TCP [RFC8257]. The L4S service is for more general traffic than just DCTCP--it allows the set of congestion controls with similar scaling properties to DCTCP to evolve (e.g. Relentless TCP [Mathis09], TCP Prague [PragueLinux] and the L4S variant of SCREAM for real-time media [RFC8298]).

Classic Congestion Control: A congestion control behaviour that can co-exist with standard TCP Reno [RFC5681] without causing significantly negative impact on its flow rate [RFC5033]. With Classic congestion controls, as flow rate scales, the number of round trips between congestion signals (losses or ECN marks) rises with the flow rate. So it takes longer and longer to recover after each congestion event. Therefore control of queuing and utilization becomes very slack, and the slightest disturbance prevents a high rate from being attained [RFC3649].

Scalable Congestion Control: A congestion control where the average time from one congestion signal to the next (the recovery time) remains invariant as the flow rate scales, all other factors being equal. This maintains the same degree of control over queueing and utilization whatever the flow rate, as well as ensuring that high throughput is robust to disturbances. For instance, DCTCP averages 2 congestion signals per round-trip whatever the flow rate. For the public Internet a Scalable transport has to comply with the requirements in Section 4 of [I-D.ietf-tsvwg-ecn-l4s-id] (aka. the 'Prague L4S requirements').

C: Abbreviation for Classic, e.g. when used as a subscript.

L: Abbreviation for L4S, e.g. when used as a subscript.

The terms Classic or L4S can also qualify other nouns, such as 'codepoint', 'identifier', 'classification', 'packet', 'flow'. For example: an L4S packet means a packet with an L4S identifier sent from an L4S congestion control.

Both Classic and L4S queues can cope with a proportion of unresponsive or less-responsive traffic as well (e.g. DNS, VoIP, game sync datagrams), just as a single queue AQM can if this traffic makes minimal contribution to queuing. The DualQ Coupled AQM behaviour is defined to be similar to a single FIFO queue with respect to unresponsive and overload traffic.

Reno-friendly: The subset of Classic traffic that excludes unresponsive traffic and excludes experimental congestion controls

intended to coexist with Reno but without always being strictly friendly to it (as allowed by [RFC5033]). Reno-friendly is used in place of 'TCP-friendly', given that the TCP protocol is used with many different congestion control behaviours.

Classic ECN: The original Explicit Congestion Notification (ECN) protocol [RFC3168], which requires ECN signals to be treated the same as drops, both when generated in the network and when responded to by the sender.

The names used for the four codepoints of the 2-bit IP-ECN field are as defined in [RFC3168]: Not ECT, ECT(0), ECT(1) and CE, where ECT stands for ECN-Capable Transport and CE stands for Congestion Experienced.

1.4. Features

The AQM couples marking and/or dropping from the Classic queue to the L4S queue in such a way 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 or TCP Prague to give ultra-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. Experiments used a range of base round trip delays up to 100ms and link rates up to 200 Mb/s between the data centre and home network, with varying amounts of background traffic in both queues. For every L4S packet, the AQM kept the average queuing delay below 1ms (or 2 packets where serialization delay exceeded 1ms on slower links), with 99th percentile no worse than 2ms. No losses at all were introduced by the L4S AQM. Details of the extensive experiments are available [PI2] [DCttH15].

Subjective testing was also conducted by multiple people all simultaneously using very demanding high bandwidth low latency applications over a single shared access link [L4Sdemo16]. In one application, each user could use finger gestures to pan or zoom their own high definition (HD) sub-window of a larger video scene generated on the fly in 'the cloud' from a football match. Another user wearing VR goggles was remotely receiving a feed from a 360-degree camera in a racing car, again with the sub-window in their field of vision generated on the fly in 'the cloud' dependent on their head movements. Even though other users were also downloading large amounts of L4S and Classic data, playing a gaming benchmark and watchings videos over the same 40Mb/s downstream broadband link,

latency was so low that the football picture appeared to stick to the user's finger on the touchpad and the experience fed from the remote camera did not noticeably lag head movements. All the L4S data (even including the downloads) achieved the same ultra-low latency. With an alternative AQM, the video noticeably lagged behind the finger gestures and head movements.

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 (TCP Prague etc.) 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:

- the large variations (sawteeth) of Classic flows need roughly a base RTT of queuing delay to ensure full utilization

- Scalable flows do not need a queue to keep utilization high, but they cannot keep latency predictably low if they are mixed with Classic traffic,

The L4S queue has latency priority, but the coupling from the Classic to the L4S AQM (explained below) ensures that it does not have bandwidth priority over the Classic queue.

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 flows (that satisfy the Prague L4S requirements).

- 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 the steady-state congestion window, cwnd, and the drop probability, p of standard Reno congestion control [RFC5681] . To a first order approximation, the steady-state cwnd of Reno is inversely proportional to the square root of p.

The design focuses on Reno as the worst case, because if it does no harm to Reno, it will not harm Cubic or any traffic designed to be friendly to Reno. TCP Cubic implements a Reno-compatibility mode, which is relevant for typical RTTs under 20ms as long as the throughput of a single flow is less than about 700Mb/s. In such cases it can be assumed that Cubic traffic behaves similarly to Reno (but with a slightly different constant of proportionality). The term 'Classic' will be used for the collection of Reno-friendly traffic including Cubic and potentially other experimental congestion controls intended not to significantly impact the flow rate of Reno.

A 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 'Scalable' will be used for all similar congestion control behaviours (see examples in Section 1.2). The term 'L4S' is also used for traffic driven by a Scalable congestion control that also complies with the additional 'Prague L4S' requirements [I-D.ietf-tsvwg-ecn-l4s-id].

For safe co-existence, under stationary conditions, a Scalable flow has to run at roughly the same rate as a Reno TCP flow (all other factors being equal). So the drop or marking probability for Classic traffic, p_C has to be distinct from the marking probability for L4S traffic, p_L. The original ECN specification [RFC3168] required these probabilities to be the same, but [RFC8311] updates RFC 3168 to enable experiments in which these probabilities are different.

Also, to remain stable, Classic sources need the network to smooth p_C so it changes relatively slowly. It is hard for a network node to know the RTTs of all the flows, so a Classic AQM adds a _worst-case_ RTT of smoothing delay (about 100-200 ms). In contrast, L4S shifts responsibility for smoothing ECN feedback to the sender, which only delays its response by its _own_ RTT, as well as allowing a more immediate response if necessary.

The Coupled AQM achieves safe coexistence by making the Classic drop probability p_C proportional to the square of the coupled L4S probability p_CL. p_CL is an input to the instantaneous L4S marking probability p_L but it changes as slowly as p_C. This makes the Reno flow rate roughly equal the DCTCP flow rate, because the squaring of p_CL counterbalances the square root of p_C in the 'TCP formula' of Classic Reno congestion control.

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

$$p_C = (p_{CL} / k)^2 \quad (1)$$

where k is the constant of proportionality, which is termed the coupling factor.

2.2. Dual Queue

Classic traffic needs to build a large queue to prevent under-utilization. Therefore a separate queue is provided for L4S traffic, and it is scheduled with priority over the Classic queue. 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).

2.3. Traffic Classification

Both the Coupled AQM and DualQ mechanisms need an identifier to distinguish L4S (L) and Classic (C) packets. Then the coupling algorithm can achieve coexistence without having to inspect flow identifiers, because it can apply the appropriate marking or dropping probability to all flows of each type. A separate specification [I-D.ietf-tsvwg-ecn-l4s-id] requires the network to treat the ECT(1) and CE codepoints 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, [I-D.ietf-tsvwg-ecn-l4s-id] says that the device "MUST NOT alter the end-to-end L4S ECN identifier", 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, an operator could use other identifiers to classify certain additional packet types into the L queue that it deems will not risk harm to 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]), the Non-Queue-Building (NQB) per-hop behaviour [I-D.ietf-tsvwg-nqb] or

certain protocols (e.g. ARP, DNS). Note that the mechanism only reads these identifiers. [I-D.ietf-tsvwg-ecn-l4s-id] says it "MUST NOT alter these non-ECN identifiers". Thus, the L queue is not soley an L4S queue, it can be consider more generally as a low latency queue.

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 load with a linear controller, then square the output before applying it as a drop probability to Reno-friendly traffic (because Reno congestion control decreases its load proportional to the square-root of the increase in drop). 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 the slow-moving input to ECN marking in the L queue (the coupled L4S probability) is:

$$p_{CL} = k*p'. \qquad (3)$$

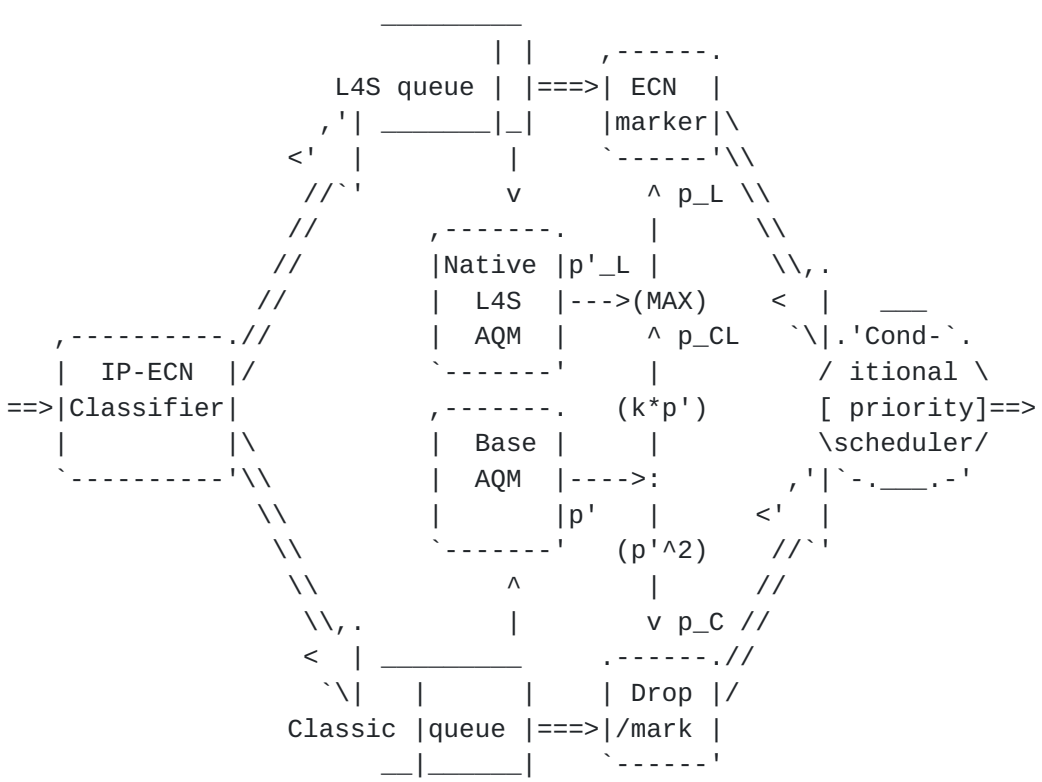
The actual ECN marking probability p_L that is applied to the L queue needs to track the immediate L queue delay under L-only congestion conditions, as well as track p_CL under coupled congestion conditions. So the L queue uses a native AQM that calculates a probability p'_L as a function of the instantaneous L queue delay. And, given the L queue has conditional priority over the C queue, whenever the L queue grows, the AQM ought to apply marking probability p'_L, but p_L ought not to fall below p_CL. This suggests:

$$p_L = \max(p'_L, p_{CL}), \qquad (4)$$

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

The two transformations of p' in equations (2) and (3) implement the required coupling given in equation (1) earlier.

The constant of proportionality or coupling factor, k, in equation (1) determines the ratio between the congestion probabilities (loss or marking) experienced by L4S and Classic traffic. Thus k indirectly determines the ratio between L4S and Classic flow rates, because flows (assuming they are responsive) adjust their rate in response to congestion probability. Appendix C.2 gives guidance on the choice of k and its effect on relative flow rates.



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. Even though the scheduler gives priority to the L queue, it is not as strong as the coupling from the C queue. This is because, as the C queue grows, the base AQM applies more congestion signals to L traffic (as well as C). As L flows

reduce their rate in response, they use less than the scheduling share for L traffic. So, because the scheduler is work preserving, it schedules any C traffic in the gaps.

Giving priority to the L queue has the benefit of very low L queue delay, because the L queue is kept empty whenever L traffic is controlled by the coupling. Also there only has to be a coupling in one direction - from Classic to L4S. Priority has to be conditional in some way to prevent the C queue starving under overload conditions (see Section 4.1). With normal responsive traffic simple strict priority would work, but it would make new Classic traffic wait until its queue activated the coupling and L4S flows had in turn reduced their rate enough to drain the L queue so that Classic traffic could be scheduled. Giving a small weight or limited waiting time for C traffic improves response times for short Classic messages, such as DNS requests and improves Classic flow startup because immediate capacity is available.

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 it is just called 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 and evaluation attention than Curvy RED, leaving the Curvy RED design incomplete and not so fully evaluated.

Both AQMs regulate their queue in units of time rather than 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

A Dual Queue Coupled AQM implementation MUST utilize two queues, each with an AQM algorithm. The two queues can be part of a larger queuing hierarchy [I-D.briscoe-tsvwg-l4s-diffserv].

The AQM algorithm for the low latency (L) queue MUST be able to apply ECN marking to ECN-capable packets.

The scheduler draining the two queues MUST give L4S packets priority over Classic, although priority MUST be bounded in order not to starve Classic traffic. The scheduler SHOULD be work-conserving.

[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 ensure coexistence of Classic and 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)." The term 'likelihood' is used to allow for marking and dropping to be either probabilistic or deterministic.

For the current specification, this translates into the following requirement. A DualQ Coupled AQM MUST apply ECN marking to traffic in the L queue that is no lower than that derived from the likelihood of drop (or ECN marking) in the Classic queue using Eqn. (1).

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 Reno congestion control (Reno) [RFC5681] and other Reno-friendly controls, such as TCP Cubic in its Reno-compatibility mode.

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.2.

If multiple customers or 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 customers. However, on the public Internet, access network operators typically isolate customers from each other with some form of layer-2 multiplexing (OFDM(A) in DOCSIS3.1, CDMA in 3G, SC-FDMA in LTE) or L3 scheduling (WRR in DSL), rather than relying on host congestion controls 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 flows are L4S, and it will not affect the relative throughput of 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 expected that the AQM in each queue will inspect the ECN field to determine what sort of congestion notification to signal, then it will decide 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 [I-D.briscoe-docsis-q-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').

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.

If the DualQ Coupled AQM has detected overload, it SHOULD signal congestion solely using drop, irrespective of the ECN field. Switching to drop if ECN marking is persistently high is required by Section 7 of [RFC3168] and Section 4.2.1 of [RFC7567].

2.5.2. Management Requirements

2.5.2.1. Configuration

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, which can be used to determine the queuing delay of the Classic AQM at its operating point, in order to prevent typical lone flows from under-utilizing capacity. For example:

  * for the PI2 algorithm (Appendix A) the queuing delay target is set to the typical RTT;

  * for the Curvy RED algorithm (Appendix B) the queuing delay at the desired operating point of the curvy ramp is configured to encompass a typical RTT;

  * if another Classic AQM was used, it would be likely to need an operating point for the queue based on the typical RTT, and if so it SHOULD be expressed in units of time.

  An operating point that is manually calculated might be directly configurable instead, e.g. for links with large numbers of flows where under-utilization by a single flow would be unlikely.

- Expected maximum RTT, which can be used to set the stability parameter(s) of the Classic AQM. For example:

  * for the PI2 algorithm (Appendix A), the gain parameters of the PI algorithm depend on the maximum RTT.

  * for the Curvy RED algorithm (Appendix B) the smoothing parameter is chosen to filter out transients in the queue within a maximum RTT.

  Stability parameter(s) that are manually calculated assuming a maximum RTT might be directly configurable instead.

- Coupling factor, k (see Appendix C.2);

- A limit to the conditional priority of L4S. This is scheduler-dependent, but it SHOULD be expressed as a relation between the max delay of a C packet and an L packet. For example:

  * for a WRR scheduler a weight ratio between L and C of w:1 means that the maximum delay to a C packet is w times that of an L packet.

  * for a time-shifted FIFO (TS-FIFO) scheduler (see Section 4.1.1) a time-shift of tshift means that the maximum delay to a C packet is tshift greater than that of an L packet. tshift could be expressed as a multiple of the typical RTT rather than as an absolute delay.

- The maximum Classic ECN marking probability, p_Cmax, before switching over to drop.

2.5.2.2. Monitoring

An experimental DualQ Coupled AQM SHOULD allow the operator to monitor each of the following operational statistics on demand, per queue and per configurable sample interval, for performance monitoring and perhaps also for accounting in some cases:

- Bits forwarded, from which utilization can be calculated;

- Total packets in the three categories: arrived, presented to the AQM, and forwarded. The difference between the first two will measure any non-AQM tail discard. The difference between the last two will measure proactive AQM discard;

- ECN packets marked, non-ECN packets dropped, ECN packets dropped, which can be combined with the three total packet counts above to calculate marking and dropping probabilities;

- Queue delay (not including serialization delay of the head packet or medium acquisition delay) - see further notes below.

  Unlike the other statistics, queue delay cannot be captured in a simple accumulating counter. Therefore the type of queue delay statistics produced (mean, percentiles, etc.) will depend on implementation constraints. To facilitate comparative evaluation of different implementations and approaches, an implementation SHOULD allow mean and 99th percentile queue delay to be derived (per queue per sample interval). A relatively simple way to do this would be to store a coarse-grained histogram of queue delay. This could be done with a small number of bins with configurable edges that represent contiguous ranges of queue delay. Then, over a sample interval, each bin would accumulate a count of the number of packets that had fallen within each range. The maximum queue delay per queue per interval MAY also be recorded.

2.5.2.3. Anomaly Detection

An experimental DualQ Coupled AQM SHOULD asynchronously report the following data about anomalous conditions:

- Start-time and duration of overload state.

  A hysteresis mechanism SHOULD be used to prevent flapping in and out of overload causing an event storm. For instance, exit from overload state could trigger one report, but also latch a timer. Then, during that time, if the AQM enters and exits overload state any number of times, the duration in overload state is accumulated but no new report is generated until the first time the AQM is out of overload once the timer has expired.

2.5.2.4. Deployment, Coexistence and Scaling

[RFC5706] suggests that deployment, coexistence and scaling should also be covered as management requirements. The raison d'etre of the DualQ Coupled AQM is to enable deployment and coexistence of Scalable congestion controls - as incremental replacements for today's Reno-friendly controls that do not scale with bandwidth-delay product.

Therefore there is no need to repeat these motivating issues here given they are already explained in the Introduction and detailed in the L4S architecture [I-D.ietf-tsvwg-l4s-arch].

The descriptions of specific DualQ Coupled AQM algorithms in the appendices cover scaling of their configuration parameters, e.g. with respect to RTT and sampling frequency.

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 (e.g. [I-D.briscoe-docsis-q-protection]) 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. These choices need to be made either by the developer or by operator policy, rather than by the IETF.

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

Priority of L4S is required to be conditional to avoid total 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. This can either be thought of as

guaranteeing a minimum throughput service for Classic traffic, or as guaranteeing a maximum delay for a packet at the head of the Classic queue.

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 just less than the capacity set aside for L4S (e.g. 14/16 in the above example), using WRR could significantly reduce the capacity left for any responsive L4S flows.

The scheduling weight of the Classic queue should not be too small, otherwise a C packet at the head of the queue could be excessively delayed by a continually busy L queue. For instance if the Classic weight is 1/16, the maximum that a Classic packet at the head of the queue can be delayed by L traffic is the serialization delay of 15 MTU-sized packets.

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 implementations in Appendix A and Appendix B could both be implemented with 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, even though saturation could 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 standard Reno flow that is normally responsive, but above a certain congestion level it will not be able to reduce its congestion window below the allowed minimum of 2 segments [RFC5681], 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 [DualQ-Test].

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 solely applies the "drop on saturation" policy. The DOCSIS specification of a DualQ Coupled AQM [DOCSIS3.1] also implements the 'drop on saturation' policy with a very shallow L buffer. However, the addition of DOCSIS per-flow Queue Protection [I-D.briscoe-docsis-q-protection] turns this into

'delay on saturation' by redirecting some packets of the flow(s) most responsible for L queue overload into the C queue, which has a higher delay target. If overload continues, this again becomes 'drop on saturation' as the level of drop in the C queue rises to maintain the target delay of the C queue.

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, Gabi Bracha, Nicolas Kuhn, Greg Skinner, Tom Henderson and David Pullen for detailed review comments particularly of the appendices and suggestions on how to make the explanations clearer. Thanks also to Tom Henderson for insights on the choice of schedulers and queue delay measurement techniques.

The early contributions of Koen De Schepper, Bob Briscoe, Olga Bondarenko and Inton Tsang were 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 Comcast Innovation Fund and the Research Council of Norway through the TimeIn project. The views expressed here are solely those of the authors.

6. Contributors

The following contributed implementations and evaluations that validated and helped to improve this specification:

Olga Albisser <olga@albisser.org> of Simula Research Lab, Norway (Olga Bondarenko during early drafts) implemented the prototype DualPI2 AQM for Linux with Koen De Schepper and conducted extensive evaluations as well as implementing the live performance visualization GUI [L4Sdemo16].

Olivier Tilmans <olivier.tilmans@nokia-bell-labs.com> of Nokia Bell Labs, Belgium prepared and maintains the Linux implementation of DualPI2 for upstreaming.

Tom Henderson <tomh@tomh.org> of CableLabs, US implemented various DualQ Coupled AQMs for ns3, including DualPI2 and DualPIE over point to point and DOCSIS 3.1 link models and conducted extensive evaluations.

Ing Jyh (Inton) Tsang of Nokia, Belgium built the End-to-End Data Centre to the Home broadband testbed on which DualQ Coupled AQM implementations were tested.

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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 ramp function (configured in units of queuing time) with unsmoothed ECN marking is used for the Native L4S AQM. The ramp can also be configured as a step function. The PI2 algorithm [PI2] is used for the Classic AQM. PI2 is an improved variant of the PIE AQM [RFC8033].

The pseudocode will be introduced 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.

All variables are assumed to be floating point in their basic units (size in bytes, time in seconds, rates in bytes/second, alpha and beta in Hz, and probabilities from 0 to 1. Constants expressed in k (kilo), M (mega), G (giga), u (micro), m (milli) , %, ... are assumed to be converted to their appropriate multiple or fraction to represent the basic units. A real implementation that wants to use integer values needs to handle appropriate scaling factors and allow accordingly appropriate resolution of its integer types (including temporary internal values during calculations).

A full open source implementation for Linux is available at: https://github.com/L4STeam/sch_dualpi2_upstream and explained in [DualPI2Linux]. The specification of the DualQ Coupled AQM for DOCSIS cable modems and CMTSs is available in [DOCSIS3.1] and explained in [LLD].

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 six functions:

- the initialization function dualpi2_params_init(...) (Figure 2) that sets parameter defaults (the API for setting non-default values is omitted for brevity)

- the enqueue function dualpi2_enqueue(lq, cq, pkt) (Figure 3)

- the dequeue function dualpi2_dequeue(lq, cq, pkt) (Figure 4)

- recur(q, likelihood) for de-randomized ECN marking (shown at the end of Figure 4).

- the L4S AQM function laqm(qdelay) (Figure 5) used to calculate the ECN-marking probability for the L4S queue

- the base AQM function that implements the PI algorithm dualpi2_update(lq, cq) (Figure 6) used to regularly update the base probability (p'), which is squared for the Classic AQM as well as being coupled across to the L4S queue.

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 (see Note a);

- mark(pkt) and drop(pkt) for ECN-marking and dropping a packet;

In 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. Constants and 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:    % DualQ Coupled framework parameters
5:    limit = MAX_LINK_RATE * 250 ms               % Dual buffer size
3:    k = 2                                         % Coupling factor
4:    % NOT SHOWN % scheduler-dependent weight or equival't parameter
6:
7:    % PI2 AQM parameters
8:    RTT_max = 100 ms                      % Worst case RTT expected
9:    RTT_typ = 15 ms                                  % Typical RTT
11:   % PI2 constants derived from above PI2 parameters
10:   p_Cmax = min(1/k^2, 1)             % Max Classic drop/mark prob
12:   target = RTT_typ            % PI AQM Classic queue delay target
13:   Tupdate = min(RTT_typ, RTT_max/3)        % PI sampling interval
14:   alpha = 0.1 * Tupdate / RTT_max^2      % PI integral gain in Hz
15:   beta = 0.3 / RTT_max               % PI proportional gain in Hz
16:
17:   % L4S ramp AQM parameters
18:   minTh = 800 us        % L4S min marking threshold in time units
19:   range = 400 us                % Range of L4S ramp in time units
20:   Th_len = 2 * MTU           % Min L4S marking threshold in bytes
21:   % L4S constants incl. those derived from other parameters
22:   p_Lmax = 1                                  % Max L4S marking prob
23:   floor = Th_len / MIN_LINK_RATE
24:   if (minTh < floor) {
25:     % Shift ramp so minTh >= serialization time of 2 MTU
26:     minTh = floor
27:   }
28:   maxTh = minTh+range   % L4S max marking threshold in time units
29: }
```

Figure 2: Example Header Pseudocode for DualQ Coupled PI2 AQM

The overall goal of the code is to maintain the base probability (p', p-prime as in Section 2.4), which is an internal variable from which the marking and dropping probabilities for L4S and Classic traffic (p_L and p_C) are derived, with p_L in turn being derived from p_CL. The probabilities p_CL and p_C are derived in lines 4 and 5 of the dualpi2_update() function (Figure 6) then used in the dualpi2_dequeue() function where p_L is also derived from p_CL at line 6 (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() + MTU > limit)
 3:      drop(pkt)                    % drop packet if buffer is full
 4:    timestamp(pkt)                 % attach arrival time to packet
 5:    % Packet classifier
 6:    if ( ecn(pkt) modulo 2 == 1 )        % ECN bits = ECT(1) or CE
 7:      lq.enqueue(pkt)
 8:    else                            % ECN bits = not-ECT or ECT(0)
 9:      cq.enqueue(pkt)
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
 5:        p'_L = laqm(lq.time())                    % Native L4S AQM
 6:        p_L = max(p'_L, p_CL)                % Combining function
 7:        if ( recur(lq, p_L) )                    % Linear marking
 8:          mark(pkt)
 9:      } else {
10:        cq.dequeue(pkt)                     % Scheduler chooses cq
11:        if ( recur(cq, p_C) ) {           % 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: }

23: recur(q, likelihood) {   % Returns TRUE with a certain likelihood
24:    q.count += likelihood
25:    if (q.count > 1) {
26:      q.count -= 1
27:      return TRUE
28:    }
29:    return FALSE
30: }
```

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. This assumes a shared buffer for the two queues (Note b discusses the merits of separate buffers). In order to avoid any bias against larger packets, 1 MTU of space is always allowed and the limit is deliberately tested before enqueue.

If limit is not exceeded, the packet is timestamped in line 4. This assumes that queue delay is measured using the sojourn time technique (see Note a for alternatives).

At lines 5-9, the packet is classified and enqueued to the Classic or L4S queue dependent on the least significant bit of the ECN field in the IP header (line 6). 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 sloppier (for a typical RTT it would double the Classic queue's feedback delay).

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, in lines 7 and 8 the packet is ECN-marked with likelihood p_L. The recur() function at the end of Figure 4 is used, which is preferred over random marking because it avoids delay due to randomization when interpreting congestion signals, but it still desynchronizes the saw-teeth of the flows. Line 6 calculates p_L as the maximum of the coupled L4S probability p_CL and the probability from the native L4S AQM p'_L. This implements the max() function shown in Figure 1 to couple the outputs of the two AQMs together. Of the two probabilities input to p_L in line 6:

  * p'_L is calculated per packet in line 5 by the laqm() function (see Figure 5),

  * whereas p_CL is maintained by the dualpi2_update() function which runs every Tupdate (Tupdate is set in line 13 of Figure 2. It defaults to 16 ms in the reference Linux implementation because it has to be rounded to a multiple of 4 ms).

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

The Native L4S AQM algorithm (Figure 5) is a ramp function, similar to the RED algorithm, but simplified as follows:

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

- It uses instantaneous queueing delay, which avoids the complexity of smoothing, but also avoids embedding a worst-case RTT of smoothing delay in the network (see Section 2.1).

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

- Marking does not have to be randomized. Determinism is used instead of randomness; to reduce the delay necessary to smooth out the noise of randomness from the signal.

The ramp function requires two configuration parameters, the minimum threshold (minTh) and the width of the ramp (range), both in units of queuing time), as shown in lines 18 & 19 of the initialization function in Figure 2. The ramp function can be configured as a step (see Note c).

Although the DCTCP paper [Alizadeh-stability] recommends an ECN marking threshold of 0.17*RTT_typ, it also shows that the threshold can be much shallower with hardly any worse under-utilization of the link (because the amplitude of DCTCP's sawteeth is so small). Based on extensive experiments, for the public Internet the default minimum ECN marking threshold in Figure 2 is considered a good compromise, even though it is significantly smaller fraction of RTT_typ.

A minimum marking threshold parameter (Th_len) in transmission units (default 2 MTU) is also necessary to ensure that the ramp does not trigger excessive marking on slow links. The code in lines 24-27 of

the initialization function (Figure 2) converts 2 MTU into time units and shifts the ramp so that the min threshold is no shallower than this floor.

```
1:  laqm(qdelay) {              % Returns native L4S AQM probability
2:    if (qdelay >= maxTh)
3:      return 1
4:    else if (qdelay > minTh)
5:      return (qdelay - minTh)/range  % Divide could use a bit-shift
6:    else
7:      return 0
8:  }
```

Figure 5: Example Pseudocode for the Native L4S AQM

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

(Clamping p' within the range [0,1] omitted for clarity - see text)

Figure 6: Example PI-Update Pseudocode for DualQ Coupled PI2 AQM

The coupled marking probability, p_CL depends on the base probability (p'), which is kept up to date by the core PI algorithm in Figure 6 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 (see Note a) 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 and beta, 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 (assuming probability has a value from 0 to 1).

alpha and beta determine how much p' ought to change after each update interval (Tupdate). For smaller Tupdate, p' should change by the same amount per second, but in finer more frequent steps. So alpha depends on Tupdate (see line 14 of the initialization function in Figure 2). It is best to update p' as frequently as possible, but Tupdate will probably be constrained by hardware performance. As shown in line 13, the update interval should be at least as frequent as once per the RTT of a typical flow (RTT_typ) as long as it does not exceed roughly RTT_max/3. For link rates from 4 - 200 Mb/s, a target RTT of 15ms and a maximum RTT of 100ms, it has been verified through extensive testing that Tupdate=16ms (as recommended in [RFC8033]) is sufficient.

The choice of alpha and beta also determines the AQM's stable operating range. The AQM ought to change p' as fast as possible in response to changes in load without over-compensating and therefore causing oscillations in the queue. Therefore, the values of alpha and beta also depend on the RTT of the expected worst-case flow (RTT_max).

Recommended derivations of the gain constants alpha and beta can be approximated for Reno over a PI2 AQM as: alpha = 0.1 * Tupdate / RTT_max^2; beta = 0.3 / RTT_max, as shown in lines 14 & 15 of Figure 2. These are derived from the stability analysis in [PI2]. For the default values of Tupdate=16 ms and RTT_max = 100 ms, they result in alpha = 0.16; beta = 3.2 (discrepancies are due to rounding). These defaults have been verified with a wide range of link rates, target delays and a range of traffic models with mixed and similar RTTs, short and long flows, etc.

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 of Figure 6 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. So, the effective gain factor for the L4S queue is k*alpha (with defaults alpha = 0.16 Hz and k=2, effective L4S alpha = 0.32 Hz).

Unlike in PIE [RFC8033], alpha and beta do not need to be tuned every Tupdate dependent on p'. Instead, in PI2, alpha and beta 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.

Nonetheless, an implementer might wish to add selected heuristics to either AQM. For instance the Linux reference DualPI2 implementation includes the following:

- Prior to enqueuing an L4S packet, if the L queue contains <2 packets, the packet is flagged to suppress any native L4S AQM marking at dequeue (which depends on sojourn time);

- Classic and coupled marking or dropping (i.e. based on p_C and p_CL from the PI controller) is only applied to a packet if the respective queue length in bytes is > 2 MTU (prior to enqueueing the packet or after dequeuing it, depending on whether the AQM is configured to be applied at enqueue or dequeue);

- In the WRR scheduler, the 'credit' indicating which queue should transmit is only changed if there are packets in both queues (i.e. if there is actual resource contention). This means that a properly paced L flow might never be delayed by the WRR. The WRR credit is reset in favour of the L queue when the link is idle.

An implementer might also wish to add other heuristics, e.g. burst protection [RFC8033] or enhanced burst protection [RFC8034].

Notes:

a. 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 needs to be measured in time, not bytes or packets [AQMmetrics] [CoDel]. 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]).

b. Line 2 of the dualpi2_enqueue() function (Figure 3) 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 Reno TCP during slow-start over a long RTT).

c. There has been some concern that using the step function of DCTCP for the Native L4S AQM requires end-systems to smooth the signal for an unnecessarily large number of round trips to ensure sufficient fidelity. A ramp is no worse than a step in initial experiments with existing DCTCP. Therefore, it is recommended that a ramp is configured in place of a step, which will allow congestion control algorithms to investigate faster smoothing algorithms.

   A ramp is more general that a step, because an operator can effectively turn the ramp into a step function, as used by DCTCP, by setting the range to zero. There will not be a divide by zero problem at line 5 of Figure 5 because, if minTh is equal to maxTh, the condition for this ramp calculation cannot arise.

A.2. Pass #2: Overload Details

Figure 7 repeats the dequeue function of Figure 4, but with overload details added. Similarly Figure 8 repeats the core PI algorithm of Figure 6 with overload details added. The initialization, enqueue, L4S AQM and recur functions are unchanged.

In line 10 of the initialization function (Figure 2), the maximum Classic drop probability p_Cmax = min(1/k^2, 1) or 1/4 for the default coupling factor k=2. p_Cmax is the point at which it is deemed that the Classic queue has become persistently overloaded, so it switches to using drop, even for ECN-capable packets. ECT packets that are not dropped can still be ECN-marked.

In practice, 25% has been found to be a good threshold to preserve fairness between ECN capable and non ECN capable traffic. This protects the queues against both temporary overload from responsive flows and more persistent overload from any unresponsive traffic that falsely claims to be responsive to ECN.

When the Classic ECN marking probability reaches the p_Cmax threshold (1/k^2), the marking probability coupled to the L4S queue, p_CL will always be 100% for any k (by equation (1) in Section 2). So, for readability, the constant p_Lmax is defined as 1 in line 22 of the initialization function (Figure 2). This is intended to ensure that the L4S queue starts to introduce dropping once ECN-marking saturates at 100% and can rise no further. The 'Prague L4S' 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' Reno congestion control. So it is correct that, when the L4S queue drops packets, it drops them 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 7). Lines 8c to 8g drop L4S packets with probability p'^2. Lines 8h to 8i mark the remaining packets with probability p_CL. Given p_Lmax = 1, 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 8) 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 PI update function in Figure 6 would drop nothing, even if the L4S queue were overloaded - so tail drop would have to take over (lines 2 and 3 of Figure 3).

Instead, the test at line 2a of the full PI update function in Figure 8 keeps delay on target using drop. If the test at line 2a of Figure 8 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_CL will be driven to zero, and the L4S queue will naturally be governed solely by p'_L from the native L4S AQM (lines 5 and 6 of the dequeue algorithm in Figure 7). But, if unresponsive L4S source(s) cause overload, the DualQ transitions smoothly to L4S marking based on the PI algorithm. If overload increases further, it naturally transitions from marking to dropping by the switch-over mechanism already described.

```
1:  dualpi2_dequeue(lq, cq, pkt) {     % Couples L4S & Classic queues
2:    while ( lq.len() + cq.len() > 0 ) {
3:      if ( scheduler() == lq ) {
4a:       lq.dequeue(pkt)                          % L4S scheduled
4b:       if ( p_CL < p_Lmax ) {      % Check for overload saturation
5:          p'_L = laqm(lq.time())                  % Native L4S AQM
6:          p_L = max(p'_L, p_CL)                % Combining function
7:          if ( recur(lq, p_L) )                   % Linear marking
8a:           mark(pkt)
8b:       } else {                               % overload saturation
8c:         if ( recur(lq, p_C) ) {          % 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 ( recur(lq, p_CL) )        % probability p_CL = k * p'
8i:           mark(pkt)         % linear marking of remaining packets
8j:       }
9:      } else {
10:       cq.dequeue(pkt)                          % Classic scheduled
11:       if ( recur(cq, p_C) ) {           % 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 7: Example Dequeue Pseudocode for DualQ Coupled PI2 AQM (Including Overload Code)

```
1:  dualpi2_update(lq, cq) {                % 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 * (curq - target) + beta * (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 8: 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. As long as the scheduler weight for Classic is small (e.g. 1/16), its exact value is unimportant because it does not normally determine capacity shares. The weight is only important to prevent unresponsive L4S traffic starving Classic traffic. This is because capacity sharing between the queues is normally determined by the coupled congestion signal, which overrides the scheduler, by making L4S sources leave roughly equal per-flow capacity available for Classic flows.

- Alternatively, a time-shifted FIFO (TS-FIFO) could be used. 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 scheduler() function in line 3 of the dequeue code would simply be implemented as the scheduler() function at the bottom of Figure 10 in Appendix B. For the public Internet a good value for tshift is 50ms. For private networks with smaller diameter, about 4*target would be reasonable. TS-FIFO is a very simple scheduler, but complexity might need to be added to address some deficiencies (which is why it is not recommended over WRR):

  * TS-FIFO does not fully isolate latency in the L4S queue from uncontrolled bursts in the Classic queue;

  * TS-FIFO is only appropriate if time-stamping of packets is feasible;

  * Even if time-stamping is supported, the sojourn time of the head packet is always stale. For instance, if a burst arrives at an empty queue, the sojourn time will only measure the delay

of the burst once the burst is over, even though the queue knew about it from the start. At the cost of more operations and more storage, a 'scaled sojourn time' metric of queue delay can be used, which is the sojourn time of a packet scaled by the ratio of the queue sizes when the packet departed and arrived [SigQ-Dyn].

- 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. Although the AQM was designed to be efficient in integer arithmetic, to aid understanding it is first given using floating point arithmetic (Figure 10). Then, one possible optimization for integer arithmetic is given, also in pseudocode (Figure 11). To aid comparison, the line numbers are kept in step between the two by using letter suffixes where the longer code needs extra lines.

B.1. Curvy RED in Pseudocode

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

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

- the dequeue function cred_dequeue(lq, cq, pkt) (Figure 4);

- the scheduling function scheduler(), which selects between the head packets of the two queues.

It also uses the following functions that are either shown elsewhere, or not shown in full here:

- the enqueue function, which is identical to that used for DualPI2, dualpi2_enqueue(lq, cq, pkt) in Figure 3;

- mark(pkt) and drop(pkt) for ECN-marking and dropping a packet;

- 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 (see Note a in Appendix A.1).

Because Curvy RED was evaluated before DualPI2, certain improvements introduced for DualPI2 were not evaluated for Curvy RED. In the pseudocode below, the straightforward improvements have been added on the assumption they will provide similar benefits, but that has not been proven experimentally. They are: i) a conditional priority scheduler instead of strict priority ii) a time-based threshold for the native L4S AQM; iii) ECN support for the Classic AQM. A recent evaluation has proved that a minimum ECN-marking threshold (minTh) greatly improves performance, so this is also included in the pseudocode.

Overload protection has not been added to the Curvy RED pseudocode below so as not to detract from the main features. It would be added in exactly the same way as in Appendix A.2 for the DualPI2 pseudocode. The native L4S AQM uses a step threshold, but a ramp like that described for DualPI2 could be used instead. The scheduler uses the simple TS-FIFO algorithm, but it could be replaced with WRR.

The Curvy RED algorithm has not been maintained or evaluated to the same degree as the DualPI2 algorithm. In initial experiments on broadband access links ranging from 4 Mb/s to 200 Mb/s with base RTTs from 5 ms to 100 ms, Curvy RED achieved good results with the default parameters in Figure 9.

The parameters are categorised by whether they relate to the Classic AQM, the L4S AQM or the framework coupling them together. Constants and variables derived from these parameters are also included at the end of each category. These are the raw input parameters for the algorithm. A configuration front-end could accept more meaningful parameters (e.g. RTT_max and RTT_typ) and convert them into these raw parameters, as has been done for DualPI2 in Appendix A. Where necessary, parameters are explained further in the walk-through of the pseudocode below.

```
 1:  cred_params_init(...) {           % Set input parameter defaults
 2:    % DualQ Coupled framework parameters
 3:    limit = MAX_LINK_RATE * 250 ms              % Dual buffer size
 4:    k' = 1                        % Coupling factor as a power of 2
 5:    tshift = 50 ms                % Time shift of TS-FIFO scheduler
 6:    % Constants derived from Classic AQM parameters
 7:    k = 2^k'                    % Coupling factor from Equation (1)
 6:
 7:    % Classic AQM parameters
 8:    g_C = 5            % EWMA smoothing parameter as a power of 1/2
 9:    S_C = -1          % Classic ramp scaling factor as a power of 2
10:    minTh = 500 ms    % No Classic drop/mark below this queue delay
11:    % Constants derived from Classic AQM parameters
12:    gamma = 2^(-g_C)                    % EWMA smoothing parameter
13:    range_C = 2^S_C                        % Range of Classic ramp
14:
15:    % L4S AQM parameters
16:    T = 1 ms             % Queue delay threshold for native L4S AQM
17:    % Constants derived from above parameters
18:    S_L = S_C - k'        % L4S ramp scaling factor as a power of 2
19:    range_L = 2^S_L                             % Range of L4S ramp
20: }
```

Figure 9: Example Header Pseudocode for DualQ Coupled Curvy RED AQM

```
1:  cred_dequeue(lq, cq, pkt) {       % Couples L4S & Classic queues
2:    while ( lq.len() + cq.len() > 0 ) {
3:      if ( scheduler() == lq ) {
4:        lq.dequeue(pkt)                          % L4S scheduled
5a:       p_CL = (cq.time() - minTh) / range_L
5b:       if (  ( lq.time() > T )
5c:          OR ( p_CL > maxrand(U) ) )
6:          mark(pkt)
7:      } else {
8:        cq.dequeue(pkt)                        % Classic scheduled
9a:       Q_C = gamma * qc.time() + (1-gamma) * Q_C % Classic Q EWMA
10a:      sqrt_p_C = (Q_C - minTh) / range_C
10b:      if ( sqrt_p_C > maxrand(2*U) ) {
11:         if ( (ecn(pkt) == 0)  {           % ECN field = not-ECT
12:           drop(pkt)                    % Squared drop, redo loop
13:           continue       % continue to the top of the while loop
14:         }
15:         mark(pkt)
16:       }
17:     }
18:     return(pkt)               % return the packet and stop here
19:   }
20:   return(NULL)                           % no packet to dequeue
21: }

22: maxrand(u) {               % return the max of u random numbers
23:   maxr=0
24:   while (u-- > 0)
25:     maxr = max(maxr, rand())                  % 0 <= rand() < 1
26:   return(maxr)
27: }

28: scheduler() {
29:   if ( lq.time() + tshift >= cq.time() )
30:     return lq;
31:   else
32:     return cq;
33: }
```

Figure 10: Example Dequeue Pseudocode for DualQ Coupled Curvy RED AQM

The dequeue pseudocode (Figure 10) 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.

The code is written assuming the AQMs are applied on dequeue (Note 1). 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. If both queues are empty, the routine returns NULL at line 20. Line 3 of the dequeue pseudocode is where the conditional priority scheduler chooses between the L4S queue (lq) and the Classic queue (cq). The time-shifted FIFO scheduler is shown at lines 28-33, which would be suitable if simplicity is paramount (see Note 2).

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

L4S: If the test at line 3 determines there is an L4S packet to dequeue, the tests at lines 5b and 5c determine whether to mark it. The first is a simple test of whether the L4S queue delay (lq.time()) is greater than a step threshold T (Note 3). The second test is similar to the random ECN marking in RED, but with the following differences: ii) marking depends on queuing time, not bytes, in order to scale for any link rate without being reconfigured; ii) marking of the L4S queue does not depend on itself, it depends on the queuing time of the _other_ (Classic) queue, where cq.time() is the queuing time of the packet at the head of the Classic queue (zero if empty); iii) marking depends on the instantaneous queuing time (of the other Classic queue), not a smoothed average; iv) 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 5a the coupled marking probability p_CL is set to the excess of the Classic queueing delay qc.time() above the minimum queuing delay threshold (minTh) all divided by the L4S scaling parameter range_L. range_L represents the queuing delay (in seconds) added to minTh at which marking probability would hit 100%. Then in line 5c (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.

Classic: If the scheduler at line 3 chooses to dequeue a Classic packet and jumps to line 7, the test at line 10b determines whether to drop or mark it. But before that, line 9a updates Q_C, which is an exponentially weighted moving average (Note 4) of the

queuing time in the Classic queue, where qc.time() is the current instantaneous queueing time of the Classic queue and gamma is the EWMA constant (default 1/32, see line 12 of the initialization function).

Lines 10a and 10b implement the Classic AQM. In line 10a the averaged queuing time Q_C is divided by the Classic scaling parameter range_C, in the same way that queuing time was scaled for L4S marking. This scaled queuing time will be squared to compute Classic drop probability so, before it is squared, it is effectively the square root of the drop probability, hence it is given the variable name sqrt_p_C. 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.

The AQM functions in each queue (lines 5c & 10b) are two cases of a new generalization of RED called Curvy RED, motivated as follows. When the performance of this AQM was compared with fq_CoDel and PIE, their goal of holding queuing delay to a fixed target seemed misguided [CRED_Insights]. As the number of flows increases, if the AQM does not allow host congestion controllers 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 two parameters, not three. The disadvantage 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 be supported is limited, otherwise an adaptation mechanism is required.

From our limited experiments with Curvy RED so far, recommended values of these parameters are: S_C = -1; g_C = 5; T = 5 * MTU at the link rate (about 1ms at 60Mb/s) 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). The setting of k depends on policy (see Section 2.5 and Appendix C.2 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. Only U=1 has been used in experiments so far, but results might be even better with U=2 or higher.

Notes:

1. 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 sloppier (for a typical RTT it would double the Classic queue's feedback delay). On a platform where packet timestamping is feasible, e.g. Linux, it is also easiest to apply the AQMs at dequeue because that is where queuing time is also measured.

2. WRR better isolates the L4S queue from large delay bursts in the Classic queue, but it is slightly less simple than TS-FIFO. If WRR were used, a low default Classic weight (e.g. 1/16) would need to be configured in place of the time shift in line 5 of the initialization function (Figure 9).

3. A step function is shown for simplicity. A ramp function (see Figure 5 and the discussion around it in Appendix A.1) is recommended, because it is more general than a step and has the potential to enable L4S congestion controls to converge more rapidly.

4. 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, e.g. by using the minimum of the smoothed and instantaneous queue delays, min(Q_C, qc.time()).

B.2. Efficient Implementation of Curvy RED

Although code optimization depends on the platform, the following notes explain where the design of Curvy RED was particularly motivated by efficient implementation.

The Classic AQM at line 10b calls maxrand(2*U), which gives twice as much curviness as the call to maxrand(U) in the marking function at line 5c. 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. 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:  cred_dequeue(lq, cq, pkt) {       % Couples L4S & Classic queues
2:    while ( lq.len() + cq.len() > 0 ) {
3:      if ( scheduler() == lq ) {
4:        lq.dequeue(pkt)                          % L4S scheduled
5:        if ((lq.time() > T) OR (cq.ns() >> (S_L-2) > maxrand(U)))
6:          mark(pkt)
7:      } else {
8:        cq.dequeue(pkt)                       % Classic scheduled
9:        Q_C += (cq.ns() - Q_C) >> g_C            % Classic Q EWMA
10:       if ( (Q_C >> (S_C-2) ) > maxrand(2*U) ) {
11:         if ( (ecn(pkt) == 0)  {           % ECN field = not-ECT
12:           drop(pkt)                   % Squared drop, redo loop
13:           continue       % continue to the top of the while loop
14:         }
15:         mark(pkt)
16:       }
17:     }
18:     return(pkt)               % return the packet and stop here
19:   }
20:   return(NULL)                           % no packet to dequeue
21: }
```

Figure 11: Optimised Example Dequeue Pseudocode for Coupled DualQ AQM using Integer Arithmetic

The two ranges, range_L and range_C are expressed as powers of 2 so that division can be implemented as a right bit-shift (>>) in lines 5 and 10 of the integer variant of the pseudocode (Figure 11).

For the integer variant of the pseudocode, an integer version of the rand() function used at line 25 of the maxrand(function) in Figure 10 would be arranged to return an integer in the range 0 <= maxrand() < 2^32 (not shown). This would scale up all the floating point probabilities in the range [0,1] by 2^32.

Queuing delays are also scaled up by 2^32, but in two stages: i) In lines 5 and 10 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.

In line 8 of the initialization function, the EWMA constant gamma is represented as an integer power of 2, g_C, so that in line 9 of the integer code the division needed to weight the moving average can be implemented by a right bit-shift (>> g_C).

Appendix C. Choice of Coupling Factor, k

C.1. RTT-Dependence

Where Classic flows compete for the same capacity, their relative flow rates depend not only on the congestion probability, but also on their end-to-end RTT (= base RTT + queue delay). The rates of competing Reno [RFC5681] flows are roughly inversely proportional to their RTTs. Cubic exhibits similar RTT-dependence when in Reno-compatibility mode, but is less RTT-dependent otherwise.

Until the early experiments with the DualQ Coupled AQM, the importance of the reasonably large Classic queue in mitigating RTT-dependence had not been appreciated. Appendix A.1.5 of [I-D.ietf-tsvwg-ecn-l4s-id] uses numerical examples to explain why bloated buffers had concealed the RTT-dependence of Classic congestion controls before that time. Then it explains why, the more that queuing delays have reduced, the more that RTT-dependence has surfaced as a potential starvation problem for long RTT flows.

Given that congestion control on end-systems is voluntary, there is no reason why it has to be voluntarily RTT-dependent. Therefore [I-D.ietf-tsvwg-ecn-l4s-id] requires L4S congestion controls to be significantly less RTT-dependent than the standard Reno congestion control [RFC5681]. Following this approach means there is no need for network devices to address RTT-dependence, although there would be no harm if they did, which per-flow queuing inherently does.

At the time of writing, the range of approaches to RTT-dependence in L4S congestion controls has not settled. Therefore, the guidance on the choice of the coupling factor in Appendix C.2 is given against DCTCP [RFC8257], which has well-understood RTT-dependence. The guidance is given for various RTT ratios, so that it can be adapted to future circumstances.

C.2. 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

In the above appendices that give example DualQ Coupled algorithms, to aid efficient implementation, a coupling factor that is an integer power of 2 is always used. k' is always used to denote the power. k' is related to the coupling factor k in Equation (1) (Section 2.1) by k=2^k'.

To determine the appropriate coupling factor 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 roughly a typical base round trip of 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 below, which were developed in [DCttH15]:

$$2^{k'} = 1.64 \, (RTT_{reno} / RTT_{dc}) \quad (5)$$

$$2^{k'} = 1.19 \, (RTT_{cubic} / RTT_{dc}) \quad (6)$$

For localized traffic from a particular ISP's data centre, using the measured RTTs, it was calculated that a value of k'=3 (equivalant to k=8) would achieve throughput equivalence, and 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.

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