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**DetNet Bounded Latency**  
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Abstract

This document presents a parameterized timing model for Deterministic Networking (DetNet), so that existing and future standards can achieve the DetNet quality of service features of bounded latency and zero congestion loss. It defines requirements for resource reservation protocols or servers. It calls out queuing mechanisms, defined in other documents, that can provide the DetNet quality of service.

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## **[1.](#) Introduction**

The ability for IETF Deterministic Networking (DetNet) or IEEE 802.1 Time-Sensitive Networking (TSN, [[IEEE8021TSN](#)]) to provide the DetNet services of bounded latency and zero congestion loss depends upon A) configuring and allocating network resources for the exclusive use of DetNet/TSN flows; B) identifying, in the data plane, the resources to be utilized by any given packet, and C) the detailed behavior of those resources, especially transmission queue selection, so that latency bounds can be reliably assured. Thus, DetNet is an example of an IntServ Guaranteed Quality of Service [[RFC2212](#)]

As explained in [[I-D.ietf-detnet-architecture](#)], DetNet flows are characterized by 1) a maximum bandwidth, guaranteed either by the transmitter or by strict input metering; and 2) a requirement for a guaranteed worst-case end-to-end latency. That latency guarantee, in turn, provides the opportunity for the network to supply enough buffer space to guarantee zero congestion loss.

To be of use to the applications identified in [[I-D.ietf-detnet-use-cases](#)], it must be possible to calculate, before the transmission of a DetNet flow commences, both the worst-case end-to-end network latency, and the amount of buffer space required at each hop to ensure against congestion loss.

This document references specific queuing mechanisms, defined in other documents, that can be used to control packet transmission at each output port and achieve the DetNet qualities of service. This document presents a timing model for sources, destinations, and the DetNet transit nodes that relay packets that is applicable to all of those referenced queuing mechanisms. The parameters specified in this model:

- o Characterize a DetNet flow in a way that provides externally measurable verification that the sender is conforming to its promised maximum, can be implemented reasonably easily by a sending device, and does not require excessive over-allocation of resources by the network.
- o Enable reasonably accurate computation of worst-case end-to-end latency, in a way that requires as little detailed knowledge as possible of the behavior of the Quality of Service (QoS) algorithms implemented in each device, including queuing, shaping, metering, policing, and transmission selection techniques.



Using the model presented in this document, it should be possible for an implementor, user, or standards development organization to select a particular set of queuing mechanisms for each device in a DetNet network, and to select a resource reservation algorithm for that network, so that those elements can work together to provide the DetNet service.

This document does not specify any resource reservation protocol or server. It does not describe all of the requirements for that protocol or server. It does describe requirements for such resource reservation methods, and for queuing mechanisms that, if met, will enable them to work together.

## **2. Terminology and Definitions**

This document uses the terms defined in [\[I-D.ietf-detnet-architecture\]](#).

## **3. DetNet bounded latency model**

### **3.1. Flow creation**

There are two models for flow creation, static ([Section 3.1.1](#)) and dynamic ([Section 3.1.2](#)). Most of the mathematical analysis provided in this document is applicable to either flow creation model; any dependencies on the choice of flow creation model are pointed out in the text.

#### **3.1.1. Static flow creation**

The static problem:

Given a network and a set of DetNet flows, compute an end-to-end latency bound (if computable) for each flow, and compute the resources, particularly buffer space, required in each DetNet transit node to achieve zero congestion loss.

In this model, all of the DetNet flows are known before the calculation commences. This problem is of interest to relatively static networks, or static parts of larger networks. It gives the best possible worst-case behavior. The calculations can be extended to provide global optimizations, such as altering the path of one DetNet flow in order to make resources available to another DetNet flow with tighter constraints.

This calculation may be more difficult to perform than that of the dynamic model ([Section 3.1.2](#)), because the flows passing through one port on a DetNet transit node affect each others' latency. The effects can even be circular, from Flow A to B to C and back to A.



On the other hand, the static calculation can often accommodate queuing methods, such as transmission selection by strict priority, that are unsuitable for the dynamic calculation.

The static flow creation model is not limited only to static networks; the entire calculation for all flows can be repeated each time a new DetNet flow is created or deleted. If some already-established flow would be pushed beyond its latency requirements by the new flow, then either the new flow is refused, or some other suitable action taken.

### **3.1.2. Dynamic flow creation**

The dynamic problem:

Given a network whose maximum capacity for DetNet flows is bounded by a set of static configuration parameters applied to the DetNet transit nodes, and given just one DetNet flow, compute the worst-case end-to-end latency that can be experienced by that flow, no matter what other DetNet flows (within the network's configured parameters) might be created or deleted in the future. Also, compute the resources, particularly buffer space, required in each DetNet transit node to achieve zero congestion loss.

This model is dynamic, in the sense that flows can be added or deleted at any time, with a minimum of computation effort, and without affecting the guarantees already given to other flows.

The choice of queuing methods is critical to the applicability of the dynamic model. Some queuing methods (e.g. CQF, [Section 7.1](#)) make it easy to configure bounds on the network's capacity, and to make independent calculations for each flow. Other queuing methods (e.g., transmission selection by strict priority), make this calculation impossible, because the worst case for one flow cannot be computed without complete knowledge of all other flows. Other queuing methods (e.g. the credit-based shaper defined in [\[IEEE8021Q\] section 8.6.8.2](#)) can be used for dynamic flow creation, but yield poorer latency and buffer space guarantees than when that same queuing method is used for static flow creation ([Section 3.1.1](#)).

The dynamic flow creation model assumes the use of the following paradigm for provisioning DetNet flows:

1. Perform any configuration required by the DetNet transit nodes in the network for the classes of service to be offered, including one or more classes of DetNet service. This configuration is done beforehand, and not tied to any particular flow.





2. Characterize the new DetNet flow in IntServ terms ([Section 8](#)).
3. Establish the path that the DetNet flow will take through the network from the source to the destination(s). This can be a point-to-point or a point-to-multipoint path.
4. Select one of the DetNet classes of service for the DetNet flow.
5. Compute the worst-case end-to-end latency for the DetNet flow. In the process, determine whether sufficient resources are available for that flow to guarantee the required latency and to provide zero congestion loss.
6. Assuming that the resources are available, commit those resources to the flow. This may or may not require adjusting the parameters that control the queuing mechanisms at each hop along the flow's path.

This paradigm can be static and/or dynamic, and can be implemented using peer-to-peer protocols or using a central server model. In some situations, backtracking and recursing through this list may be necessary.

Issues such as un-provisioning a DetNet flow in favor of another when resources are scarce are not considered, but are left to the static flow creation model ([Section 3.1.1](#)). How the path to be taken by a DetNet flow is chosen is not considered in this document.

### **[3.2](#). Relay node model**

A model for the operation of a DetNet transit node is required, in order to define the latency and buffer calculations. In Figure 1 we see a breakdown of the per-hop latency experienced by a packet passing through a DetNet transit node, in terms that are suitable for computing both hop-by-hop latency and per-hop buffer requirements.



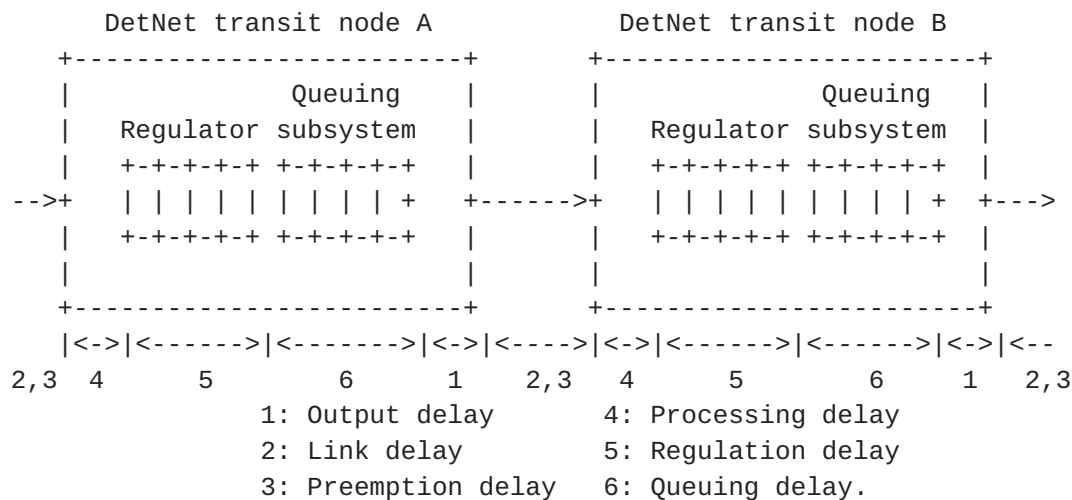


Figure 1: Timing model for DetNet or TSN

In Figure 1, we see two DetNet transit nodes (typically, bridges or routers), with a wired link between them. In this model, the only queues we deal with explicitly are attached to the output port; other queues are modeled as variations in the other delay times. (E.g., an input queue could be modeled as either a variation in the link delay [2] or the processing delay [4].) There are six delays that a packet can experience from hop to hop.

### 1. Output delay

The time taken from the selection of a packet for output from a queue to the transmission of the first bit of the packet on the physical link. If the queue is directly attached to the physical port, output delay can be a constant. But, in many implementations, the queuing mechanism in a forwarding ASIC is separated from a multi-port MAC/PHY, in a second ASIC, by a multiplexed connection. This causes variations in the output delay that are hard for the forwarding node to predict or control.

### 2. Link delay

The time taken from the transmission of the first bit of the packet to the reception of the last bit, assuming that the transmission is not suspended by a preemption event. This delay has two components, the first-bit-out to first-bit-in delay and the first-bit-in to last-bit-in delay that varies with packet size. The former is typically measured by the Precision Time Protocol and is constant (see [[I-D.ietf-detnet-architecture](#)]). However, a virtual "link" could exhibit a variable link delay.

### 3. Preemption delay



If the packet is interrupted in order to transmit another packet or packets, (e.g. [[IEEE8023](#)] clause 99 frame preemption) an arbitrary delay can result.

4. Processing delay

This delay covers the time from the reception of the last bit of the packet to the time the packet is enqueued in the regulator (Queuing subsystem, if there is no regulation). This delay can be variable, and depends on the details of the operation of the forwarding node.

5. Regulator delay

This is the time spent from the insertion of the last bit of a packet into a regulation queue until the time the packet is declared eligible according to its regulation constraints. We assume that this time can be calculated based on the details of regulation policy. If there is no regulation, this time is zero.

6. Queuing subsystem delay

This is the time spent for a packet from being declared eligible until being selected for output on the next link. We assume that this time is calculable based on the details of the queuing mechanism. If there is no regulation, this time is from the insertion of the packet into a queue until it is selected for output on the next link.

Not shown in Figure 1 are the other output queues that we presume are also attached to that same output port as the queue shown, and against which this shown queue competes for transmission opportunities.

The initial and final measurement point in this analysis (that is, the definition of a "hop") is the point at which a packet is selected for output. In general, any queue selection method that is suitable for use in a DetNet network includes a detailed specification as to exactly when packets are selected for transmission. Any variations in any of the delay times 1-4 result in a need for additional buffers in the queue. If all delays 1-4 are constant, then any variation in the time at which packets are inserted into a queue depends entirely on the timing of packet selection in the previous node. If the delays 1-4 are not constant, then additional buffers are required in the queue to absorb these variations. Thus:

- o Variations in output delay (1) require buffers to absorb that variation in the next hop, so the output delay variations of the previous hop (on each input port) must be known in order to calculate the buffer space required on this hop.



- o Variations in processing delay (4) require additional output buffers in the queues of that same DetNet transit node. Depending on the details of the queueing subsystem delay (6) calculations, these variations need not be visible outside the DetNet transit node.

## **4. Computing End-to-end Latency Bounds**

### **4.1. Non-queueing delay bound**

End-to-end latency bounds can be computed using the delay model in [Section 3.2](#). Here it is important to be aware that for several queueing mechanisms, the worst-case end-to-end delay is less than the sum of the per-hop worst-case delays. An end-to-end latency bound for one DetNet flow can be computed as

$$\text{end\_to\_end\_latency\_bound} = \text{non\_queueing\_latency} + \text{queueing\_latency}$$

The two terms in the above formula are computed as follows. First, at the  $h$ -th hop along the path of this DetNet flow, obtain an upper bound  $\text{per-hop\_non\_queueing\_latency}[h]$  on the sum of delays 1,2,3,4 of Figure 1. These upper-bounds are expected to depend on the specific technology of the DetNet transit node at the  $h$ -th hop but not on the T-SPEC of this DetNet flow. Then set  $\text{non\_queueing\_latency}$  = the sum of  $\text{per-hop\_non\_queueing\_latency}[h]$  over all hops  $h$ .

### **4.2. Queueing delay bound**

Second, compute  $\text{queueing\_latency}$  as an upper bound to the sum of the queueing delays along the path. The value of  $\text{queueing\_latency}$  depends on the T-SPEC of this flow and possibly of other flows in the network, as well as the specifics of the queueing mechanisms deployed along the path of this flow.

For several queueing mechanisms,  $\text{queueing\_latency}$  is less than the sum of upper bounds on the queueing delays (5,6) at every hop. This occurs with (1) per-flow queueing, and (2) per-class queueing with regulators, as explained in [Section 4.2.1](#), [Section 4.2.2](#), and [Section 6](#).

For other queueing mechanisms the only available value of  $\text{queueing\_latency}$  is the sum of the per-hop queueing delay bounds. In such cases, the computation of per-hop queueing delay bounds must account for the fact that the T-SPEC of a DetNet flow is no longer satisfied at the ingress of a hop, since burstiness increases as one flow traverses one DetNet transit node.





#### 4.2.1. Per-flow queuing mechanisms

With such mechanisms, each flow uses a separate queue inside every node. The service for each queue is abstracted with a guaranteed rate and a delay. For every flow the per-node delay bound as well as end-to-end delay bound can be computed from the traffic specification of this flow at its source and from the values of rates and latencies at all nodes along its path. Details of calculation for IntServ are described in [Section 6.5](#).

#### 4.2.2. Per-class queuing mechanisms

With such mechanisms, the flows that have the same class share the same queue. A practical example is the queuing mechanism in Time Sensitive Networking. One key issue in this context is how to deal with the burstiness cascade: individual flows that share a resource dedicated to a class may see their burstiness increase, which may in turn cause increased burstiness to other flows downstream of this resource. Computing latency upper bounds for such cases is difficult, and in some conditions impossible [[charny2000delay](#)][[bennett2002delay](#)]. Also, when bounds are obtained, they depend on the complete configuration, and must be recomputed when one flow is added.

A solution to deal with this issue is to reshape the flows at every hop. This can be done with per-flow regulators (e.g. leaky bucket shapers), but this requires per-flow queuing and defeats the purpose of per-class queuing. An alternative is the interleaved regulator, which reshapes individual flows without per-flow queuing ([[Specht2016UBS](#)], [[IEEE8021Qcr](#)]). With an interleaved regulator, the packet at the head of the queue is regulated based on its (flow) regulation constraints; it is released at the earliest time at which this is possible without violating the constraint. One key feature of per-flow or interleaved regulator is that, it does not increase worst-case latency bounds [[le boudec theory 2018](#)]. Specifically, when an interleaved regulator is appended to a FIFO subsystem, it does not increase the worst-case delay of the latter.

Figure 2 shows an example of a network with 5 nodes, per-class queuing mechanism and interleaved regulators as in Figure 1. An end-to-end delay bound for flow *f*, traversing nodes 1 to 5, is calculated as follows:

$$\text{end\_to\_end\_latency\_bound\_of\_flow\_f} = C_{12} + C_{23} + C_{34} + S_4$$

In the above formula,  $C_{ij}$  is a bound on the aggregate response time of queuing subsystem in node *i* and interleaved regulator of node *j*, and  $S_4$  is a bound on the response time of the queuing subsystem in



node 4 for flow  $f$ . In fact, using the delay definitions in [Section 3.2](#),  $C_{ij}$  is a bound on sum of the delays 1,2,3,6 of node  $i$  and 4,5 of node  $j$ . Similarly,  $S_4$  is a bound on sum of the delays 1,2,3,6 of node 4. A practical example of queuing model and delay calculation is presented [Section 6.4](#).

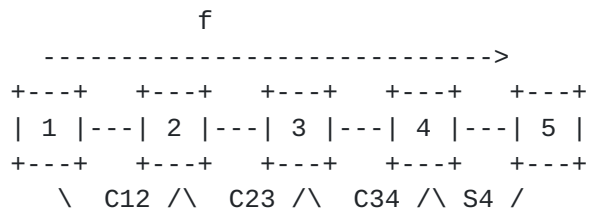


Figure 2: End-to-end latency computation example

REMARK: The end-to-end delay bound calculation provided here gives a much better upper bound in comparison with end-to-end delay bound computation by adding the delay bounds of each node in the path of a flow [[TSNwithATS](#)].

### 4.3. Ingress considerations

A sender can be a DetNet node which uses exactly the same queuing methods as its adjacent DetNet transit node, so that the latency and buffer calculations at the first hop are indistinguishable from those at a later hop within the DetNet domain. On the other hand, the sender may be DetNet unaware, in which case some conditioning of the flow may be necessary at the ingress DetNet transit node.

This ingress conditioning typically consists of a FIFO with an output regulator that is compatible with the queuing employed by the DetNet transit node on its output port(s). For some queuing methods, simply requires added extra buffer space in the queuing subsystem. Ingress conditioning requirements for different queuing methods are mentioned in the sections, below, describing those queuing methods.

### 4.4. Interspersed non-DetNet transit nodes

It is sometimes desirable to build a network that has both DetNet aware transit nodes and DetNet non-aware transit nodes, and for a DetNet flow to traverse an island of non-DetNet transit nodes, while still allowing the network to offer latency and congestion loss guarantees. This is possible under certain conditions.

In general, when passing through a non-DetNet island, the island causes delay variation in excess of what would be caused by DetNet nodes. That is, the DetNet flow is "lumpier" after traversing the non-DetNet island. DetNet guarantees for latency and buffer



requirements can still be calculated and met if and only if the following are true:

1. The latency variation across the non-DetNet island must be bounded and calculable.
2. An ingress conditioning function ([Section 4.3](#)) may be required at the re-entry to the DetNet-aware domain. This will, at least, require some extra buffering to accommodate the additional delay variation, and thus further increases the worst-case latency.

The ingress conditioning is exactly the same problem as that of a sender at the edge of the DetNet domain. The requirement for bounds on the latency variation across the non-DetNet island is typically the most difficult to achieve. Without such a bound, it is obvious that DetNet cannot deliver its guarantees, so a non-DetNet island that cannot offer bounded latency variation cannot be used to carry a DetNet flow.

## **5. Achieving zero congestion loss**

When the input rate to an output queue exceeds the output rate for a sufficient length of time, the queue must overflow. This is congestion loss, and this is what deterministic networking seeks to avoid.

### **5.1. A General Formula**

To avoid congestion losses, an upper bound on the backlog present in the regulator and queuing subsystem of Figure 1 must be computed during resource reservation. This bound depends on the set of flows that use these queues, the details of the specific queuing mechanism and an upper bound on the processing delay (4). The queue must contain the packet in transmission plus all other packets that are waiting to be selected for output.

A conservative backlog bound, that applies to all systems, can be derived as follows.

The backlog bound is counted in data units (bytes, or words of multiple bytes) that are relevant for buffer allocation. For every class we need one buffer space for the packet in transmission, plus space for the packets that are waiting to be selected for output. Excluding transmission and preemption times, the packets are waiting in the queue since reception of the last bit, for a duration equal to the processing delay (4) plus the queuing delays (5,6).

Let



- o `nb_classes` be the number of classes of traffic that may use this output port
- o `total_in_rate` be the sum of the line rates of all input ports that send traffic of any class to this output port. The value of `total_in_rate` is in data units (e.g. bytes) per second.
- o `nb_input_ports` be the number input ports that send traffic of any class to this output port
- o `max_packet_length` be the maximum packet size for packets of any class that may be sent to this output port. This is counted in data units.
- o `max_delay45` be an upper bound, in seconds, on the sum of the processing delay (4) and the queuing delays (5,6) for a packet of any class at this output port.

Then a bound on the backlog of traffic of all classes in the queue at this output port is

$$\text{backlog\_bound} = ( \text{nb\_classes} + \text{nb\_input\_ports} ) * \\ \text{max\_packet\_length} + \text{total\_in\_rate} * \text{max\_delay45}$$

## **6. Queuing model**

### **6.1. Queuing data model**

Sophisticated queuing mechanisms are available in Layer 3 (L3, see, e.g., [\[RFC7806\]](#) for an overview). In general, we assume that "Layer 3" queues, shapers, meters, etc., are precisely the "regulators" shown in Figure 1. The "queuing subsystems" in this figure are not the province solely of bridges; they are an essential part of any DetNet transit node. As illustrated by numerous implementation examples, some of the "Layer 3" mechanisms described in documents such as [\[RFC7806\]](#) are often integrated, in an implementation, with the "Layer 2" mechanisms also implemented in the same node. An integrated model is needed in order to successfully predict the interactions among the different queuing mechanisms needed in a network carrying both DetNet flows and non-DetNet flows.

Figure 3 shows the general model for the flow of packets through the queues of a DetNet transit node. Packets are assigned to a class of service. The classes of service are mapped to some number of regulator queues. Only DetNet/TSN packets pass through regulators. Queues compete for the selection of packets to be passed to queues in the queuing subsystem. Packets again are selected for output from the queuing subsystem.









transit node can utilize any of the following information to assign a packet to a particular class of service (queue):

- o Input port.
- o Selector based on a rotating schedule that starts at regular, time-synchronized intervals and has nanosecond precision.
- o MAC addresses, VLAN ID, IP addresses, Layer 4 port numbers, DSCP. ([[I-D.ietf-detnet-dp-sol-ip](#)], [[I-D.ietf-detnet-dp-sol-mpls](#)]) (Work items are expected to add MPC and other indicators.)
- o The Class of Service Assignment function can contain metering and policing functions.
- o MPLS and/or pseudowire ([[RFC6658](#)]) labels.

The "Transmission selection" function decides which queue is to transfer its oldest packet to the output port when a transmission opportunity arises.

## **6.2. Preemption**

In [[IEEE8021Q](#)] and [[IEEE8023](#)], the transmission of a frame can be interrupted by one or more "express" frames, and then the interrupted frame can continue transmission. This frame preemption is modeled as consisting of two MAC/PHY stacks, one for packets that can be interrupted, and one for packets that can interrupt the interruptible packets. The Class of Service (queue) determines which packets are which. Only one layer of preemption is supported -- a transmitter cannot have more than one interrupted frame in progress. DetNet flows typically pass through the interrupting MAC. Best-effort queues pass through the interruptible MAC, and can thus be preempted.

## **6.3. Time-scheduled queuing**

In [[IEEE8021Q](#)], the notion of time-scheduling queue gates is described in [section 8.6.8.4](#). Below every output queue (the lower row of queues in Figure 3) is a gate that permits or denies the queue to present data for transmission selection. The gates are controlled by a rotating schedule that can be locked to a clock that is synchronized with other DetNet transit nodes. The DetNet class of service can be supplied by queuing mechanisms based on time, rather than the regulator model in Figure 3. These queuing mechanisms are discussed in [Section 7](#), below.



#### **6.4. Time-Sensitive Networking with Asynchronous Traffic Shaping**

Consider a network with a set of nodes (DetNet transit nodes and hosts) along with a set of flows between hosts. Hosts are sources or destinations of flows. There are four types of flows, namely, control-data traffic (CDT), class A, class B, and best effort (BE) in decreasing order of priority. Flows of classes A and B are together referred to AVB flows. It is assumed a subset of TSN functions as described next.

It is also assumed that contention occurs only at the output port of a TSN node. Each node output port performs per-class scheduling with eight classes: one for CDT, one for class A traffic, one for class B traffic, and five for BE traffic denoted as BE0-BE4 (according to TSN standard). In addition, each node output port also performs per-flow regulation for AVB flows using an interleaved regulator (IR), called Asynchronous Traffic Shaper (ATS) in TSN. Thus, at each output port of a node, there is one interleaved regulator per-input port and per-class. The detailed picture of scheduling and regulation architecture at a node output port is given by Figure 4. The packets received at a node input port for a given class are enqueued in the respective interleaved regulator at the output port. Then, the packets from all the flows, including CDT and BE flows, are enqueued in a class based FIFO system (CBFS) [[TSNwithATS](#)].



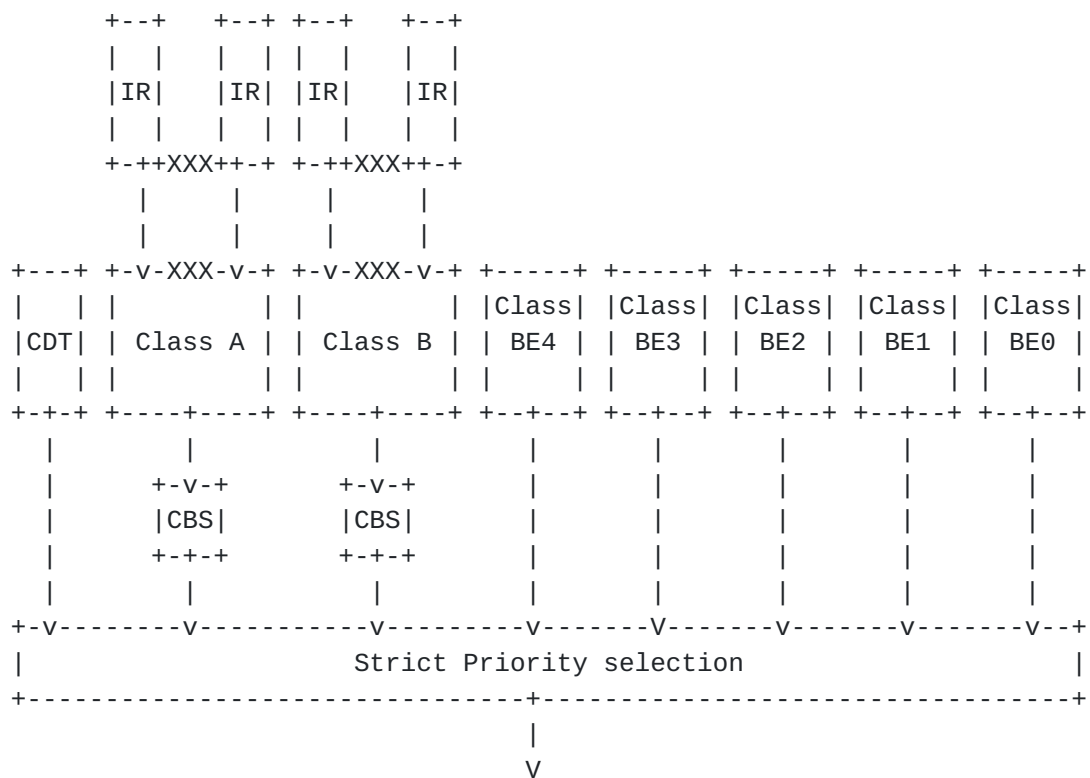


Figure 4: Architecture of a TSN node output port with interleaved regulators (IRs)

The CBFS includes two Credit-Based Shaper (CBS) subsystems, one for each class A and B. The CBS serves a packet from a class according to the available credit for that class. The credit for each class A or B increases based on the idle slope, and decreases based on the send slope, both of which are parameters of the CBS. The CDT and BE0-BE4 flows in the CBFS are served by separate FIFO subsystems. Then, packets from all flows are served by a transmission selection subsystem that serves packets from each class based on its priority. All subsystems are non-preemptive. Guarantees for AVB traffic can be provided only if CDT traffic is bounded; it is assumed that the CDT traffic has leaky bucket arrival curve with two parameters  $r_h$  as rate and  $b_h$  as bucket size, i.e., the amount of bits entering a node within a time interval  $t$  is bounded by  $r_h t + b_h$ .

Additionally, it is assumed that the AVB flows are also regulated at their source according to leaky bucket arrival curve. At the source hosts, the traffic satisfies its regulation constraint, i.e. the delay due to interleaved regulator at hosts is ignored.

At each DetNet transit node implementing an interleaved regulator, packets of multiple flows are processed in one FIFO queue; the packet at the head of the queue is regulated based on its leaky bucket





parameters; it is released at the earliest time at which this is possible without violating the constraint. The regulation parameters for a flow (leaky bucket rate and bucket size) are the same at its source and at all DetNet transit nodes along its path. A delay bound of CBFS for an AVB flow  $f$  of class A or B can be computed if the following condition holds:

sum of leaky bucket rates of all flows of this class at this node  
 $\leq R$ , where  $R$  is given below for every class.

If the condition holds, the delay bound is:

$$d_f = T + (b_t - L_{\min_f})/R - L_{\min_f}/c$$

where  $L_{\min_f}$  is the minimum packet length of flow  $f$ ;  $c$  is the output link transmission rate;  $b_t$  is the sum of the  $b$  term (bucket size) for all the flows having the same class as flow  $f$  at this node. Parameters  $R$  and  $T$  are calculated as follows for class A and class B, separately:

If  $f$  is of class A:

$$R = I_A (c - r_h) / c$$

$$T = L_{nA} + b_h + r_h L_n / c / (c - r_h)$$

where  $L_{nA}$  is the maximum packet length of class B and BE packets;  $L_n$  is the maximum packet length of classes A, B, and BE.

If  $f$  is of class B:

$$R = I_B (c - r_h) / c$$

$$T = (L_{BE} + L_A + L_{nA} I_A / (c_h - I_A) + b_h + r_h L_n / c) / (c - r_h)$$

where  $L_A$  is the maximum packet length of class A;  $L_{BE}$  is the maximum packet length of class BE.

Then, an end-to-end delay bound is calculated by the formula [Section 4.2.2](#), where for  $C_{ij}$ :

$$C_{ij} = \max(d_{f'})$$

where  $f'$  is any flow that shares the same CBFS class with flow  $f$  at node  $i$  and the same interleaved regulator as flow  $f$  at node  $j$ .

More information of delay analysis in such a DetNet transit node is described in [\[TSNwithATS\]](#).



#### 6.4.1. Flow Admission

The delay calculation requires some information about each node. For each node, it is required to know the idle slope of CBS for each class A and B ( $I_A$  and  $I_B$ ), as well as the transmission rate of the output link ( $c$ ). Besides, it is necessary to have the information on each class, i.e. maximum packet length of classes A, B, and BE. Moreover, the leaky bucket parameters of CDT ( $r_h, b_h$ ) should be known. To admit a flow/flows, their delay requirements should be guaranteed not to be violated. As described in [Section 3.1](#), the two problems static and dynamic are addressed separately. In either of the problems, the rate and delay should be guaranteed. Thus,

The static admission control:

The leaky bucket parameters of all flows are known, therefore, for each flow a delay bound can be calculated. The computed delay bound for every flow should not be more than its delay requirement. Moreover, the sum of the rate of each flow ( $r_f$ ) should not be more than the rate allocated to each class ( $R$ ). If these two conditions hold, the configuration is declared admissible.

The dynamic admission control:

For dynamic admission control, we allocate to every node and class A or B, static value for rate ( $R$ ) and maximum burstiness ( $b_t$ ). In addition, for every node and every class A and B, two counters are maintained:

$R_{acc}$  is equal to the sum of the leaky-bucket rates of all flows of this class already admitted at this node; At all times, we must have:

$$R_{acc} \leq R, \text{ (Eq. 1)}$$

$b_{acc}$  is equal to the sum of the bucket sizes of all flows of this class already admitted at this node; At all times, we must have:

$$b_{acc} \leq b_t. \text{ (Eq. 2)}$$

A new flow is admitted at this node, if Eqs. (1) and (2) continue to be satisfied after adding its leaky bucket rate



and bucket size to  $R_{acc}$  and  $b_{acc}$ . A flow is admitted in the network, if it is admitted at all nodes along its path. When this happens, all variables  $R_{acc}$  and  $b_{acc}$  along its path must be incremented to reflect the addition of the flow. Similarly, when a flow leaves the network, all variables  $R_{acc}$  and  $b_{acc}$  along its path must be decremented to reflect the removal of the flow.

The choice of the static values of  $R$  and  $b_t$  at all nodes and classes must be done in a prior configuration phase;  $R$  controls the bandwidth allocated to this class at this node,  $b_t$  affects the delay bound and the buffer requirement.  $R$  must satisfy the constraints given in Annex L.1 of [IEEE8021Q].

### 6.5. IntServ

Integrated service (IntServ) is an architecture that specifies the elements to guarantee quality of service (QoS) on networks. To satisfied guaranteed service, a flow must conform to a traffic specification (T-spec), and reservation is made along a path, only if routers are able to guarantee the required bandwidth and buffer.

Consider the traffic model which conforms to token bucket regulator  $(r, b)$ , with

- o Token bucket depth ( $b$ ).
- o Token bucket rate ( $r$ ).

The traffic specification can be described as an arrival curve:

$$\alpha(t) = b + rt$$

This token bucket regulator requires that, during any time window  $t$ , the number of bit for the flow is limited by  $\alpha(t) = b + rt$ .

If resource reservation on a path is applied, IntServ model of a router can be described as a rate-latency service curve  $\beta(t)$ .

$$\beta(t) = \max(0, R(t-T))$$

It describes that bits might have to wait up to  $T$  before being served with a rate greater or equal to  $R$ .

It should be noted that, the guaranteed service rate  $R$  is a share of link's bandwidth. The choice of  $R$  is related to the specification of flows which will transmit on this node. For example, in strict priority policy, considering a flow with priority  $j$ , its share of



bandwidth may be  $R=c-\text{sum}(r_i)$ ,  $i < j$ , where  $c$  is the link bandwidth,  $r_i$  is the token bucket rate for the flows with priority higher than  $j$ . The choice of  $T$  is also related to the specification of all the flows traversing this node. For example, in a generalized processor sharing (GPS) node,  $T = L / R + L_{\text{max}}/c$ , where  $L$  is the maximum packet size for the flow,  $L_{\text{max}}$  is the maximum packet size in the node across all flows. Other choice of  $R$  and  $T$  are also supported, according to the specific scheduling of the node and flows traversing this node.

As mentioned previously in this section, delay bound and backlog bound can be easily obtained by comparing arrival curve and service curve. Backlog bound, or buffer bound, is the maximum vertical derivation between curves  $\alpha(t)$  and  $\beta(t)$ , which is  $v=b+rT$ . Delay bound is the maximum horizontal derivation between curves  $\alpha(t)$  and  $\beta(t)$ , which is  $h = T+b/R$ . Graphical illustration of the IntServ model is shown in Figure 5.

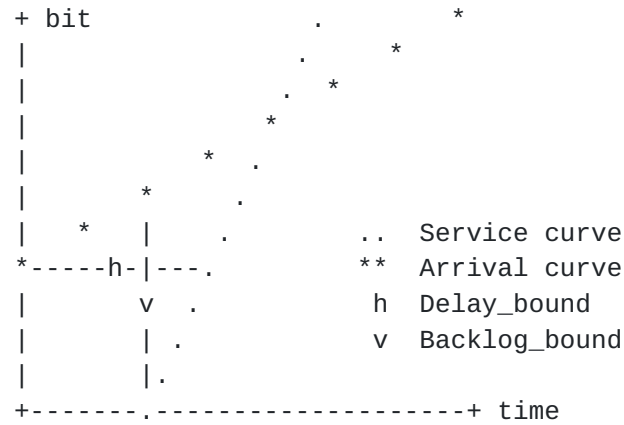


Figure 5: Computation of backlog bound and delay bound. Note that arrival and service curves are not necessary to be linear.

The output bound, or the next-hop arrival curve, is  $\alpha_{\text{out}}(t) = b + rT + rt$ , where burstiness of the flow is increased by  $rT$ , compared with the arrival curve.

We can calculate the end-to-end delay bound for a path including  $N$  nodes, among which the  $i$ -th node offers service curve  $\beta_{i-1}(t)$ ,

$$\beta_i(t) = \max(0, R_i(t-T_i)), \quad i=1, \dots, N$$

By concatenating these IntServ nodes, an end-to-end service curve can be computed as

$$\beta_{\text{e2e}}(t) = \max(0, R_{\text{e2e}}(t-T_{\text{e2e}}))$$









Figure 6 shows two DetNet transit nodes A and B, including three timelines for:

1. The output queues on port 1 in node A.
2. The input gate function ([[IEEE8021Q](#)], 8.6.5.1) that assigns packets received on port 1 of transit node B to output queues on port 2 of transit node B.
3. The output queues on port 2 of node B.

In this figure, the output ports on the two nodes are synchronized, and a new buffer starts transmitting at each tick, shown as 0, 1, 2, ... The output times shown for timelines 1 and 3 are the times at which packets are selected for output, which is the start point of the output time (1) of Figure 1. The queue assignments times on timeline 3 take place at the beginning of the queuing delay (6) of Figure 1. Time-based CQF, as described here, does not require any regulator queues. In the shown in the figure, the total time for delays 1 through 6 of Figure 1 is  $1.3T_c$ . Of course, any value is possible.

#### **7.1.1. CQF timing sequence**

In general, as shown in Figure 6, the windows for buffer assignment do not align perfectly with the windows for buffer transmission. The input gates (the center timeline in Figure 6) must switch from using one buffer to using another buffer in sync with the (delayed) received data, at times offset by the dead time from the output buffer switching (the bottom timeline in Figure 6).

If the dead time  $DT$  in Figure 6 is not excessive, then it is feasible to subtract the dead time from the cycle time  $T_c$ , and use the remainder as the input window. In the example in Figure 6, packets from node A buffer a can be transferred from the input port to node B's buffer "c" during the window shown by the upper row "VVVV...". Input must cease by time = 2.0, because that is when transit node B starts transmitting the contents of buffer c. In this case, only two output buffers are in use, one filling and one outputting.

If the dead time is too large (e.g., if the delays placed the middle timeline's switching points at  $n+0.9$ , instead of  $n+0.3$ ), three buffers are used by node B. This case is shown by the lower row "VVVV..." in Figure 6. In this case, node B places the data received from node A buffer a into node B buffer d between the times 1.3 and 2.3 in Figure 6. Buffer b starts outputting at time = 2.0, while buffer d is filling. Thus, three buffers are in use, one filling, one waiting, and one emptying.



### **7.1.2. Dead time computation**

The time for switching input packet buffer assignments is equal to the minimum possible offset from transmission selection in node A to buffer assignment in node B, which is the sum of the minimum values for all of the delays 1 through 5 in Figure 1 (the queue-to-queue delay). All packets must be received and assigned to an output buffer before the next switching point, which means that all must be transmitted in time for them to arrive at buffer assignment even if worst-case (longest delay) is encountered for the queue-to-queue delay. Thus, the minimum dead time for the 3-buffer case is the sum of the worst-case variation in the queue-to-queue delay, plus the worst-case difference between the two transit nodes' buffer switching clocks.

For the 2-buffer case, we must add the offset (shown as "DT" in Figure 6) from the end of node B's output switch to the end of node B's input switch.

### **7.1.3. Tc computation**

Given the dead time DT, there remains a transmit window of  $(T_c - DT - Int)$ . The DT was explained in ([Section 7.1.2](#)). "Int" is the worst-case interference with the start of transmission, when the output buffers switch, caused by lower-priority traffic. This is equal to one worst-case transmission time, which means that the size of the packets in all lower-priority queues must be bounded. If Ethernet preemption ([\[IEEE8023\]](#) clause 99) is employed for lower-priority queues, then this worst-case interference is reduced to the size of the largest unfragmentable Ethernet frame.

The bandwidth requirement of any given DetNet flow has to be translated to CQF terms, in order to determine whether that flow can be accommodated at each port. A flow has to be characterized as using a maximum number of bit times on the wire per cycle time  $T_c$ . For Ethernet, for example ([\[IEEE8023\]](#)), this includes the preamble (8 bytes), destination MAC address through CRC (minimum 64 bytes) and the inter-packet gap (12 bytes). The total bit times per cycle  $T_c$  required by all of the DetNet flows passing through a given port cannot exceed the available transmit window  $(T_c - DT - Int)$ .

### **7.1.4. CQF latency calculation**

The per-hop latency is trivially determined by the wire delay plus the queuing delay. Since the wire delay is either absorbed into the queuing delay (dead time is small and two buffers are used) or padded out to a whole cycle time  $T_c$  (three buffers are used) the per-



hop latency is always an integral number of cycle times  $T_c$ , with a latency variation at the output of the final hop of  $T_c$ .

Ingress conditioning may be required if the source of a DetNet flow does not, itself, employ CQF. See [Section 7.1.6](#).

#### **[7.1.5](#). CQF parameterization**

The transmit window for a given DetNet transit node running CQF, for example the transmit window for node A in Figure 6, depends on the interference (Int in section [Section 7.1.3](#)), the value of  $T_c$ , and the dead time required by the following node B. The size of the transmit window determines how many total bits can be reserved per period  $T_c$  by DetNet flows.

Part of the dead time derives from delays and delay variations such as output delay (1) and link delay (2) and preemption delay (3) of Figure 1, all of which are known to node A. However, the dead time also depends on the processing delay (4) of node B and the upon whether node B is using 2 or 3 output buffers, which is not necessarily known to node A.

The information in DetNet transit node B necessary to compute the dead time to be observed by transit node A must be known to the entity responsible for making reservation decisions, whether that is node A itself, or a central controller. A decision can be made, by the controller or by the node, whether to use the dead time and two buffers, in order to reduce the per-hop latency by one cycle time, or to use three buffers and eliminate the dead time and increase the total allocable bandwidth.

If the packet sizes of a DetNet flow are variable, or perhaps even unknown beyond the imposition of a maximum size, then some degree of overprovisioning is required. The measurement used to allocate bandwidth to a given DetNet flow is bit times in one cycle time  $T_c$ . Therefore, one extra maximum packet time (less one bit) has to be allocated to a flow per cycle time  $T_c$  in order to ensure that, no matter what mix of packet sizes are presented, the flow will get its guaranteed latency.

#### **[7.1.6](#). Ingress conditioning for CQF**

Assuming that a DetNet domain is using CQF, it is always possible that the previous node (or sender) may not support the queuing method of CQF, or may support CQF but not use the same configuration in accordance of the current DetNet domain. In this case, ingress conditioning is helpful to shape the flow according to the TSPEC in





the current DetNet domain and transmission cycle of CQF, thus control the burstiness and reduce overprovisioning.

A DetNet node running CQF satisfies that a maximum number of `max_number_packet_per_cycle` packets, each with length no larger than `max_packet_size`, can be transmitted during a CQF cycle  $T_c$ . Also, the dead time  $D_t$ , during which the former node cannot transmit to the later node (See [Section 7.1.1](#)), should be considered. Here we use the notation `max_number_packet_per_cycle`, `max_packet_size` to describe the maximum amount of transmitting data during the available transmit window  $T_c - D_t$ .

An ingress conditioner typically consists with a FIFO queue with an output regulator. Every incoming packet enters the FIFO queue, which passes it on if and only if the packet conforms to the CQF requirement. For this purpose, two criteria below is suggested to follow.

- o The incoming flow's average rate would be no more than the average output queue rate at ingress conditioner, to avoid overflow and congestion loss.
- o The queueing buffer at ingress conditioner is suggested to be large enough, to cover burstiness and jitter of bit rate from the previous node (or sender).

The output regulator controls the transmitting of packets, with credit function for instance. At the start of a CQF cycle, credit is set to the maximum bits to transmit in a cycle, `max_bit_per_cycle` = `max_number_packet_per_cycle` \* `max_packet_size`. When a packet is transmitted from the node, the credit is reduced by `length(packet)`. The operation of the output regulator can be described as below.



```

credit = max_bit_per_cycle; % Initial credit
t = 0; % Initial time offset within a cycle

outputRegulate(packet){
if (t>=0 && t< Tc-Dt) % during a transmit window
{
    if ~isEmpty(queue) % if queue is not empty
    {
        if credit >= length(packet);
        {
            dequeue(packet);
            credit = credit - length(packet);
        }
    }
}
elseif (t >= Tc) % when a cycle end, reset t and credit.
{
    t = t - Tc; % reset t to [0, Tc-Dt)
    credit = max_bit_per_cycle; % credit is refilled
}
}

```

Other instance of output regulator may also meet the CQF requirement.

#### **7.1.7. CQF ingress conditioning timing model**

Consider the input traffic conforms to variable bit rate (VBR) constraint  $(p, M, r, b)$ , which can be modeled as arrival curve:

$$\alpha(t) = \min(pt+M, rt+b)$$

where  $p$  is the peak rate,  $M$  is the maximum size of a packet,  $(r, b)$  stand for token bucket rate and burst. Note that, if  $N$  flows enter the ingress conditioner, each flow  $f_i$  conforms to token bucket  $(r_i, b_i)$ , the arrival curve parameters are of the superposition form:  $r = \sum(r_i)$ ,  $b = \sum(b_i)$ ,  $i=1, \dots, N$ .

Ingress conditioner, which transmit packets as soon as possible if conformance is satisfied, does not increase worst-case queuing latency, if we consider the latency from the input of ingress conditioner to the first output of CQF node. The CQF node, which transmits at most  $\text{max\_bit\_per\_cycle}$  bits during a cycle  $T_c$ , offers a service curve

$$\beta(t) = \text{max\_bit\_per\_cycle} * \text{floor}(t-Dt) + \min(p * \text{mod}(t-Dt, T_c), \text{max\_bit\_per\_cycle})$$

where  $Dt$  is the dead time (see [Section 7.1.1](#)),  $p$  is the peak rate,  $\text{max\_bit\_per\_cycle} = \text{max\_number\_packet\_per\_cycle} * \text{max\_packet\_size}$  is



the maximum transmitting data in a CQF cycle,  $\text{floor}(x)$  calculates the nearest integer less than or equal to  $x$ , and  $\text{mod}(x,y)=x-\text{floor}(x/y)$ .

According to network calculus, the worst-case queuing delay for ingress conditioner and CQF node, denoted  $\text{delay\_bound}$ , is the maximum horizontal distance between arrival curve and service curve.

Specifically, in general assumptions, ingress conditioner's average output rate  $r_2 = \text{max\_bit\_per\_cycle}/T_c$  is always no less than input traffic rate  $r$  (or else congestion loss would happen). In such conditions, the queuing delay up-bound for ingress conditioner and first CQF node is derived as  $\text{ingress\_delay\_bound} = \max(t : \beta(t) = (pb-rm) / (p-r))$ . Therefore, considering a path traverses ingress conditioner and  $H$  CQF nodes, the total delay up-bound is  $H \cdot T_c + \text{ingress\_delay\_bound}$ .

## 7.2. Time-scheduled queuing

[IEEE8021Q] [section 8.6.8.4](#) specifies a time-aware queue-draining procedure for transmission selection at egress port of a DetNet transit node, which supports up to eight traffic classes. Each traffic class has a separate queue, frame transmission from each queue is allowed or prevented by a time gate. This time gate controlled scheduling allows time-sensitive traffic classes to transmit on dedicate time slots. Within the time slots, the transmitting flows can be granted exclusive use of the transmission medium. Generally, this time-aware scheduling is a layer 2 time division multiplexing (TDM) technique.

Consider the static configuration of a deterministic network. To provide end-to-end latency guaranteed service, network nodes can support time-based behavior, which is determined by gate control list (GCL). GCL defines the gate operation, in open or closed state, with associated timing for each traffic class queue. A time slice with gate state "open" is called transmission window. The time-based traffic scheduling must be coordinated among the DetNet transit nodes along the path from sender to receiver, to control the transmission of time-sensitive traffic.

Ideally all network devices are time synchronized and static GCL configurations on all devices along the routed path are coordinated to ensure that length of transmission window fits the assigned frames, and no two time windows for DetNet traffic on the same port overlap. (DetNet flows' windows can overlap with best-effort windows, so that unused DetNet bandwidth is available to best-effort traffic.) The processing delay, link delay and output delay in transmitting are considered in GCL computation. Transmission window for a certain flow may require that a time offset on consecutive hops



be selected to reduce queueing delay as much as possible. In this case, TSN/DetNet frames transmit at the assigned transmission window at every node through the routed path, with zero congestion loss and bounded end-to-end latency. Then, the worst-case end-to-end latency of the flow can be derived from GCL configuration. For a TSN or DetNet frame, denote the transmission window on last hop closes at `gate_close_time_last_hop`. Assuming talker supports scheduled traffic behavior, it starts the transmission at `gate_open_time_on_talker`. Then worst case end-to-end delay of this flow is bounded by  $\text{gate\_close\_time\_last\_hop} - \text{gate\_open\_time\_on\_talker} + \text{link\_delay\_last\_hop}$ .

It should be noted that scheduled traffic service relies on a synchronized network and coordinated GCL configuration. Synthesis of GCL on multiple nodes in network is a scheduling problem considering all TSN/DetNet flows traversing the network, which is a non-deterministic polynomial-time hard (NP-hard) problem. Also, at this writing, scheduled traffic service supports no more than eight traffic classes, typically using up to seven priority classes and at least one best effort class.

## **8. Parameters for the bounded latency model**

The use of the TSPEC parameters defined in [RFC2212] and related documents for IntServ are well-established and adequate for DetNet purposes. The parameterization used by [IEEE8021Q] are somewhat different, as discussed above (Section 7.1.6). These parameters are maximum number of frames per interval, interval size, and maximum frame size. They are more suitable for the physical determination of compliance by a sender than for resource reservation purposes.

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