FavourQueue: a Stateless Active Queue Management to Speed Up Short TCP Flows (and others too!)

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***Index Terms*—Active Queue Management; TCP; Performance Evaluation; Simulation; Flow interaction.**

***Abstract*—This paper presents and analyses the implementation of a novel active queue management (AQM) named FavourQueue that aims to improve delay transfer of short lived TCP flows over a best-effort network. The idea is to dequeue in first packets that do not belong to a flow previously enqueued. The rationale is to mitigate the delay induced by long-lived TCP flows over the pace of short TCP data requests and to prevent dropped packets at the beginning of a connection and during recovery period. Although the main target of this AQM is to accelerate short TCP traffic, we show that FavourQueue does not only improve the performance of short TCP traffic but also improve the performance of all TCP traffic in terms of drop ratio and latency whatever the flow size. In particular, we demonstrate that FavourQueue reduces the loss of a retransmitted packet, decrease the RTO recovery ratio and improves the latency up to 30% compared to DropTail.**

I. Introduction

Internet is still dominated by web traffic running on top of short-lived TCP connections [1]. Indeed, as shown in [2], among 95% of the client TCP traffic and 70% of the server TCP traffic have a size lower than ten packets. This follows a common web design practice that is to keep viewed pages lightweight to improve interactive browsing in terms of response time [3]. In other words, the access to a webpage often triggers several short web traffics that allow to keep the downloaded page small and to speed up the display of the text content compared to other heavier components that might compose it[1] (e.g. pictures, multimedia content, design components). As a matter of fact and following the growth of the web content, we can still expect a large amount of short web traffic in the near future.

TCP performance suffers significantly in the presence of bursty, non-adaptive cross-traffic or when the congestion window is small (i.e. in the slow-start phase or when it operates in the small window regime). Indeed, bursty losses, or losses during the small window regime, may cause Retransmission Timeouts (RTO) which trigger a slow-start phase. In the context of short TCP flows, TCP fast retransmit cannot be triggered if not enough packets are in transit. As a result, the loss recovery is mainly done thanks to the TCP RTO and this strongly impacts the delay. Following this, in this study we seek to improve the performance of this pervasive short TCP traffic without impacting on long-lived TCP flows. We aim to exploit router capabilities to enhance the performance of short TCP flows over a best-effort network, by giving a higher priority to a TCP packet if no other packet belonging to the same flow is already enqueued inside a router queue. The rationale is that isolated losses (for instance losses that occur at the early stage of the connection) have a strong impact on the TCP flow performance than losses inside a large window. Then, we propose an AQM, called FavourQueue, which allows to better protect packet retransmission and short TCP traffic when the network is severely congested.

In order to give to the reader a clear view of the problem we tackle with our proposal, we lean on paper [2]. Figure 1 shows that the flow duration (or latency[2]) of short TCP traffic is strongly impacted by an initial lost packet which is recovered later by an RTO. Indeed, at the early stage of the connection, the number of packets exchanged is to small to allow an accurate RTO estimation. Thus, an RTO is triggered by the default time value which is set to two seconds by default [4]. In this figure, the authors also give the cumulative distribution function of TCP flow length and the probability density function of their completion time from an experimental measurement dataset obtained during one day on a ISP BRAS link which aggregates more than 30,000 users. We have reproduced a similar experiment with ns-2 (i.e. with a similar flow length CDF according to a Pareto distribution) and obtained a similar probability density function of the TCP flows duration as shown in Figure 2 for the DropTail queue curve. Both figures (1 and 2) clearly highlight a latency peak at $t = 3$ seconds which corresponds to this default RTO value [4]. In this experiment scenario, the RTO recovery ratio is equal to 56% (versus 70% in the experiments of [2]). As a matter of fact, these experiments show that the success of the TCP slow-start is a key performance indicator. The second curve in Figure 2, shows the result we obtain by using our proposal called FavourQueue. Clearly, the peak previously emphasized has disappeared. This means the initial losses that strongly

[1] See for instance: ”Best Practices for Speeding Up Your Web Site” from Yahoo developer network.

[2] The latency refers to the delay elapsed between the first sent and the last packet received.

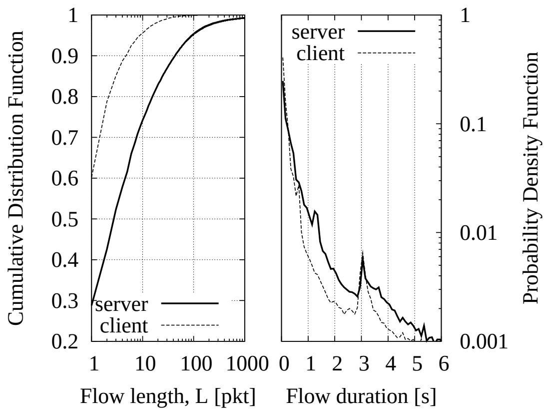

Fig. 1. TCP flow length distribution and latency (by courtesy of the authors of [2]).

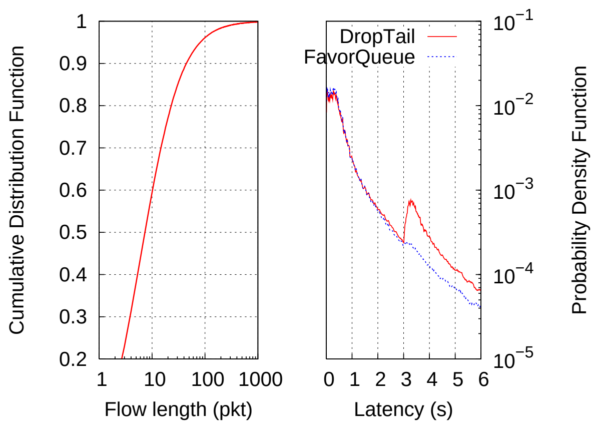

Fig. 2. TCP flow latency distribution from our simulation model.

impacted the TCP traffic performance have decreased.

An important contribution of this work is the demonstration that our scheme, by favouring isolated TCP packets, decreases the latency by decreasing the loss ratio of short TCP flows without impacting long TCP traffic. However, as FavourQueue does not discriminate short from long TCP flows, every flows take advantage of this mechanism when entering either the slow-start or a recovery phase. Our evaluations show that 58% of short TCP flows improve their latency and that 80% of long-lived TCP flows also take advantage of this AQM. For all sizes of flows, on average, the expected gain of the transfer delay is about 30%. This gain results from the decrease of the drop ratio of non opportunistic flows which are those that less occupy the queue. Furthermore, the more the queue is loaded, the more FavourQueue has an impact. Indeed, when there is no congestion, FavourQueue does not have any effect on the traffic. In other words, this proposal is activated only when the network is severely congested.

Finally, FavourQueue does not request any transport protocol modification. Although we talk about giving a priority to certain packet, there is no per-flow state needed inside the FavourQueue router. This mechanism must be seen as an extension of DropTail that greatly enhances TCP sources performance by favouring (more than prioritizing) certain TCP packets. The next Section II describes the design of the proposed scheme. Then, we presents in Section III the experimental methodology used in this paper. Sections IV and V dissects and analyses the performance of FavourQueue. Following these experiments and statistical analysis, we propose a stochastic model of the mechanism Section VI. We also present a related work in Section VII where we position FavourQueue with other propositions and in particular discuss how this AQM completes the action of recent proposals that aim to increase the TCP initial slow-start window. Finally, we propose to discuss the implementation and some security issues in Section VIII and conclude this work Section IX.

II. FavourQueue description

Short TCP flows usually carry short TCP requests such as HTTP requests or interactive SSH or Telnet commands. As a result, their delay performance are mainly driven by:

1) the end-to-end transfer delay. This delay can be reduced if the queueing delay of each router is low;
2) the potential losses at the beginning connection. The first packets lost at the beginning of a TCP connection (i.e. in the slow-start phase) are mainly recovered by the RTO mechanism. Furthermore, as the RTO is initially set to a high value, this greatly decreases the performance of short TCP flows.

The two main metrics on which we can act to minimize the end to end delay and protect from loss the first packets of a TCP connection and are respectively the queuing delay and the drop ratio. Consequently, the idea we develop with FavourQueue is to favor certain packet in order to accelerate the transfer delay by giving a preferential access to transmission and to protect them from drop.

This corresponds to implement a preferential access to transmission when a packet is enqueued and must be favoured (temporal priority) and a drop protection is provided when the queue is full (drop precedence) with push-out scheme that dequeue a standard packet in order to enqueue a favoured packet.

When a packet is enqueued, a check is done on the whole queue to seek another packet from the same flow. If no other packet is found, it becomes a favoured packet. The rationale is to decrease the loss of a retransmitted packet in order to decrease the RTO recovery ratio. The proposed algorithm (given in Algorithm 1) extends the one presented in [5] by adding a drop precedence to non-favoured packets in order to decrease the loss ratio of favoured packets. The selection of a favoured packet is done on a per-flow basis. As a result the complexity is as a function of the size of the queue which corresponds to the maximum number of state that the router must handle. The number of state is scalable considering today's routers capability to manage million of flows simultaneously [6]. However the selection decision is local and temporary as the state only exists when at least one packet is enqueued. This explains why we prefer the term of favouring packet more than prioritizing packet. Furthermore, FavourQueue does not introduced packet re-ordering inside a flow which obviously badly impacts TCP performance [7]. Finally, in the specific case where all the traffic becomes

Algorithm 1 FavourQueue algorithm

```
function enqueue(p)
# A new packet p of flow F is received
if less than 1 packet of F are present in the queue then
    # p is a favoured packet
    if the queue is full then
        if only favoured packets in the queue then
            p is drop
            return
        end if
    else
        # Push out
        the last standard packet is dropped
    end if
    p inserted in position pos_
    pos_ ← pos_ +1
else
    # p is a standard packet
    if the queue is not full then
        p is put at the end of the queue
    else
        p is dropped
    end if
end if
```

favoured, the behaviour of FavourQueue will be identical than DropTail.

III. Experimental methodology

We use ns-2 to evaluate the performance of FavourQueue. Our simulation model allows to apply different levels of load to efficiently compare FavourQueue with DropTail. The evaluations are done over a simple dumbbell topology. The network traffic is modeled in terms of flows where each flow corresponds to a TCP file transfer. We consider an isolated bottleneck link of capacity C in bit per second. The traffic demand, expressed as a bit rate, is the product of the flow arrival rate λ and the average flow size $E[\sigma]$. The load offered to the link is then defined by the following ratio:

$$\rho = \frac{\lambda E[\sigma]}{C}. \quad (1)$$

The load is changed by varying the arrival flow rate [8]. Thus, the congestion level increases as a function of the load. As all flows are independent, the flow arrivals are modeled by a Poisson process. A reasonable fit to the heavy-tail distribution of the flow size observed in practice is provided by the Pareto distribution. The shape parameter is set to 1.3 and the mean size to 30 packets. Left side in Figure 2 gives the flows' size distribution used in the simulation model.

At the TCP flow level, the ns-2 TCP connection establishment phase is enabled and the initial congestion window size is set to two packets. As a result, the TCP SYN packet is taken into account in all dataset. The load introduced in the network consists in several flows with different RTT according to the recommendation given in the "Common TCP evaluation suite" paper [8]. The load is ranging from 0.05 to 0.95 with a step of 0.1. The simulation is bounded to 500 seconds for each given load. To remove both TCP feedback synchronization and phase effect, a traffic load of 10% is generated in the opposite direction. The flows in the transient phase are removed from the analysis. More precisely, only flows starting after the first forty seconds are used in the analysis. The bottleneck link capacity is set to $10Mbps$. All other links have a capacity of $100Mbps$. According to the small buffers rule [9], buffers can be reduced by a factor of ten. The rule of thumb says the buffer size B can be set to $T \times C$ with T the round-trip propagation delay and C the link capacity. We choose $T = 100ms$ as it corresponds to the averaged RTT of the flows in the experiment. The buffer size at the two routers is set to a bandwidth-delay product with a delay of $10ms$. The packet length is fixed to 1500 bytes and the buffer size has a length of 8 packets.

To improve the confidence of these statistical results, each experiment for a given load is done ten times using different sequences of pseudo-random numbers (in the following we talk about *ten replications experiment*). Some figures also average the ten replications, meaning that we aggregate and average all flows from all ten replications and for all load conditions. In this case, we talk about *ten averaged experiment* results which represents a dataset of nearly 17 million of packets. The rationale is to consider these data as a real measurement capture where the load is varying as a function of time (as in [2]) since each load condition has the same duration. In other words, this represents a global network behaviour.

The purpose of these experiments is to weight up the benefits brought by our scheme in the context of TCP best-effort flows. To do this, we first experiment a given scenario with DropTail then, we compare with the results obtained with FavourQueue. We enable FavourQueue only on the uplink (data path) while DropTail always remains on the downlink (ACK path). We only compare all identical terminating flows for both experiments (i.e. DropTail and FavourQueue) in order to assess the performance obtained in terms of service for a same set of flows.

We assume our model follows Internet short TCP flows characteristics as we find the same general distribution latency form than Figure 1 which is as a function of the measurements obtained in Figure 2. This comparison provides a correct validation model in terms of latency. As explained above, Figure 2 corresponds and illustrates a *ten averaged experiment*.

IV. Performance evaluation of TCP flows with FavourQueue

We present in this section global performance obtained by FavourQueue then we deeper analyze its performance and investigate the case of persistent flows. We compare a same set of flows to assess the performance obtained with DropTail and FavourQueue.

A. Overall performance

We are interested in assessing the performance of each TCP flows in terms of latency and goodput. We recall from Section

I that we defined the latency as the time to complete a data download (i.e the transmission time) and the goodput is the average pace of the download. In order to assess the overall performance of FavourQueue compared to DropTail, Figure 3 gives the mean and standard deviation of the latency as a function of the traffic load of FavourQueue. We both study FavourQueue with and without the push-out mechanism in order to distinguish the supplementary gain provided by the drop precedence.

The results are unequivocal. FavourQueue version without push-out as presented in [5] provides a gain when the load increases compared to DropTail (i.e. when the queue has a significant probability of having a non-zero length) while the drop precedence (with push-out) clearly brings out a significant gain in terms of latency. Basically, Figure 4 shows that both queues (with and without push-out) globally drop the same amount of packets. However, the push-out version better protects short TCP flow (and more generally: all flows entering a slow-start phase) as when the queue is congested, it always enqueues a packet from a new flow. As a result, initial lost of packets further decreases. Indeed, as already emphasized in Figure 2 from the introduction, losses do not occur at the beginning of a connection and as a result, the flow is not impacted anymore by the retransmission overhead resulting from an RTO. Thus, our favouring scheme allows to prevent lost packets during the startup phase. As a matter of fact, this is explained by a different distribution of these losses.

Following this, we have computed the resulting normalized goodput for all flows size for all experiments and obtained is 2.4% with DropTail and 3.5% with FavourQueue (i.e. around 1% of difference). This value is not weak as it corresponds to an increase of 45%.

Figure 5 gives the average latency as a function of the flow length. The cumulative distribution function of the flow length is also represented. On average, we observe that FavourQueue obtains a lower latency than DropTail whatever the flow length. This difference is also larger for the short TCP flows which are also numerous (we recall that the distribution of the flows' size follows a Pareto distribution and as a result the number of short TCP flow is higher). This demonstrates that FavourQueue particularly favors the slow-start of every flow and as a matter of fact: short TCP flows. The cloud pattern obtained for a flow size higher than hundred is due to the decrease of the statistical sample (following the Pareto distribution used for the experiment) that result in a greater dispersion of the results obtained. As a result, we cannot drive a consistent latency analysis for sizes higher than hundred.

To complete these results, Figure 6 gives the latency obtained when we increase the size of both queues. We observe that whatever the queue size, FavourQueue always obtains a lower latency. Behind a given queuesize (in Figure 6 at $x = 60$), the increase of the queue does not have an impact on the latency. This enforces the consistency of the solution as Internet routers prevent the use of large queue size.

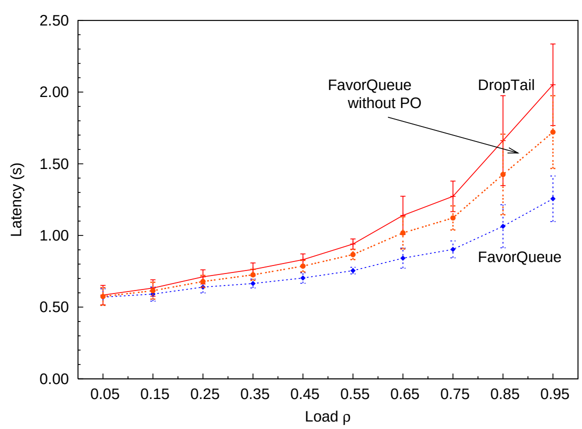

Fig. 3. Overall latency according to traffic load.

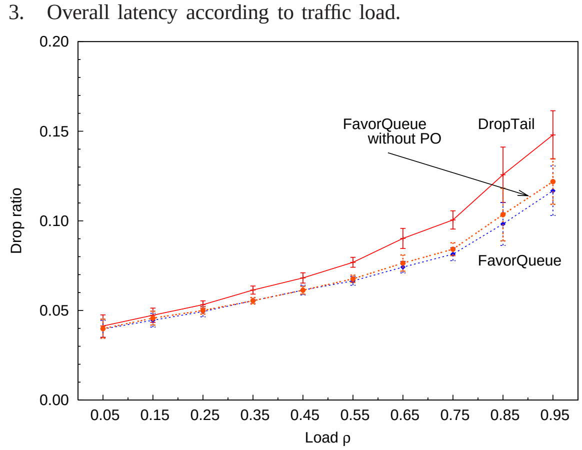

Fig. 4. Overall drop ratio according to traffic load.

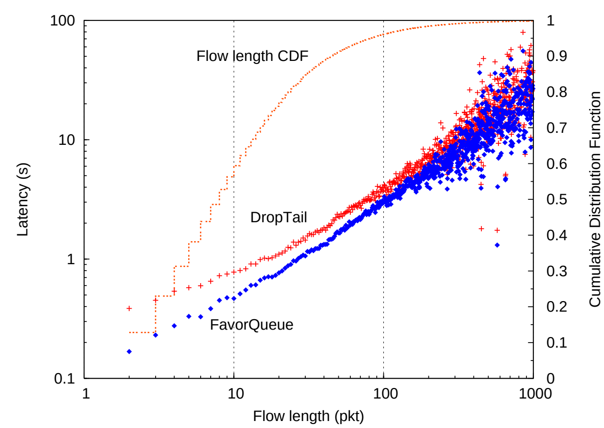

Fig. 5. Flow length CDF and mean latency as a function of the flow length.

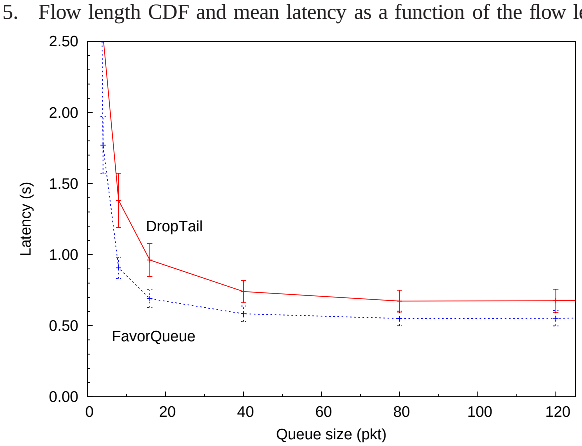

Fig. 6. Overall latency according to queuesize.

B. Performance analysis

To refine our analysis of the latency, we propose to evaluate the difference of latencies per flows for both queues. We denote $\Delta i = Td_i - Tf_i$ with Td and Tf the latency observed respectively by DropTail and FavourQueue for a given flow i. Figure 7 gives the distribution of the latencies difference. This figure illustrates that there is more decrease of the latency for each flow than increase. Furthermore for 16% of flows, there is no impact on the latency i.e. $\Delta = 0$. In other words, 84% of flows observe a change of latency; 55% of flows observe a decrease ($\Delta > 0$) and 10% of flows observe a significant change ($\Delta > 1$ second). However, 30% of the flows observe an increase of their latency ($\Delta < 0$). In summary, FavourQueue has a positive impact on certain flows that are penalised with DropTail.

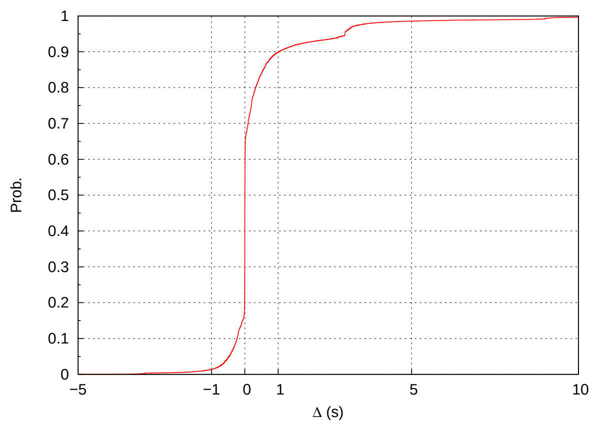

Fig. 7. Cumulative distribution function of latency difference Δ.

In order to assess the flows that gain in terms of latency, Figure 8 gives the probability of latency improvement. For the whole set of short TCP flows, (i.e. with a size lower than 10 packets), the probability to improve the latency reaches 58% while the probability to decrease is 25%. For long TCP flows (i.e. above 100 packets), the probability to improve and to decrease the latency is respectively 80% and 20%. The flows with a size around 30 packets are the ones with the highest probability to be penalised. For long TCP flows, the large variation of the probability indicates a uncertainty which mainly depends on the experimental conditions of the flows. We have to remark that long TCP flows are less present in this experimental model (approximately 2% of the flows have a size higher or equal to 100 packets). As this curve corresponds to a ten averaged experiment, each long TCP flows have experienced various load conditions and this explains these large oscillations.

Medium sized flows are characterized by a predominance of the slow-start phase. During this phase, each flow opportunistically occupies the queue and as a results less packets are favoured due to the growth of the TCP window. The increase of the latency observed for medium sized flows (ranging from 10 to 100) is investigated later in subsection V-A. We will also see in the next subsection IV-C that FavourQueue acts like a shaper for these particular flows.

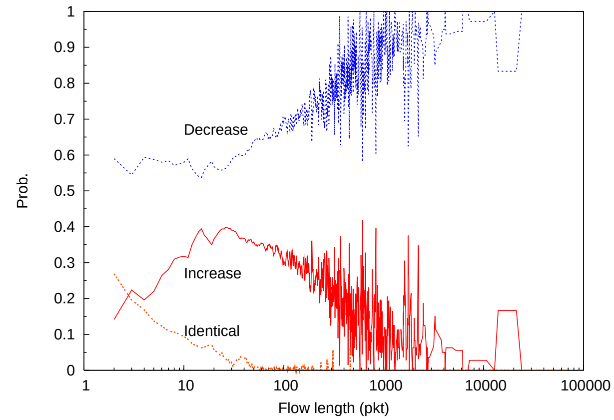

Fig. 8. Probability to change the latency.

To estimate the latency variation, we define $G(x)$ the latency gain for the flows of length x as follows:

$$G(x) = \frac{\sum_{i=1}^{N} \Delta_{x_i}}{\sum_{i=1}^{N} Td_{x_i}}. \quad (2)$$

with N, the number of flows of length x. A positive gain indicates a decrease of the latency with FavourQueue. Figure 9 provides the positive, negative and total gains as a function of the flows size. We observe an important total gain for the short TCP flows. The flows with an average size obtain the highest negative gain and this gain also decreases when the size of the flows increases. Although some short flows observe an increase of their latency, in a general manner, the positive gain is always higher. This preliminary analysis illustrates that FavourQueue improves by 30% on average the best-effort service in terms of latency. The flows that take the biggest advantage of this scheme are the short flows with a gain up to 55%.

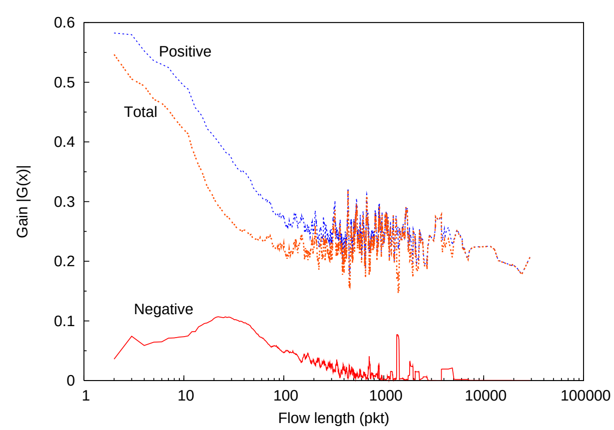

Fig. 9. Average Latency gain per flow length.

Finally and to conclude with this section, we plot in Figure 10 the number of flows in the system under both AQM as a function of time to assess the change in the stability of the network. We observe that FavourQueue considerably reduces both the average number of flows in the network as well as the variability.

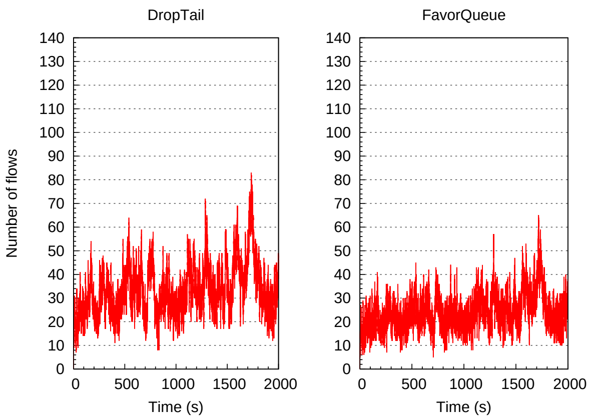

Fig. 10. Number of simultaneous flows in the network.

C. *The case of persistent flows*

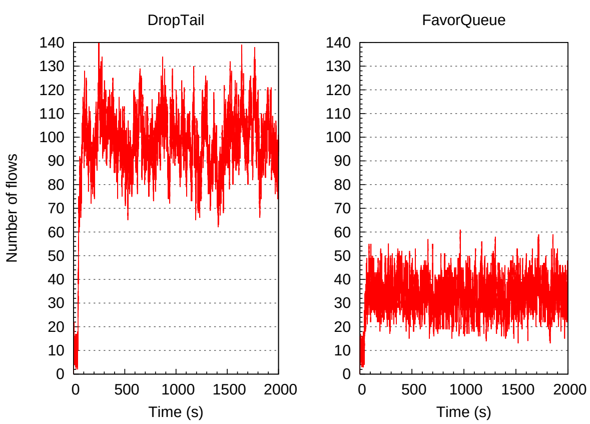

Fig. 11. Number of short flows in the network when persistent flows are actives.

Following [10], we evaluate how the proposed scheme affects persistent flows with randomly arriving short TCP flows. We now change the network conditions with 20% of short TCP flows with exponentially distributed flow sizes with a mean of 6 packets. Fourty seconds later, 50 persistent flows are sent. Figure 11 gives the number of simultaneous short flows in the network. When the 50 persistent flows start, the number of short flows increases and oscillates around 100 with DropTail. By using FavourQueue, the number increases to 30 short flows. The short flows still take advantage of the favour scheme and Figure 12 confirms this point. However we observe in Figure 13 that the persistent flows are not penalized. The mean throughput is nearly the same (1.81% for DropTail versus 1.86% for FavourQueue) and the variance is smaller with FavourQueue. Basically, FavourQueue acts as a shaper by slowing down opportunistic flows while decreasing the drop ratio of non opportunistic flows (those which less occupy the queue).

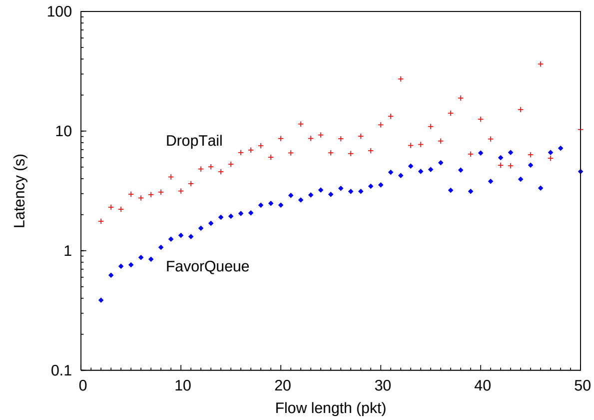

Fig. 12. Mean latency as a function of flow size for short flows in presence of persistent flows.

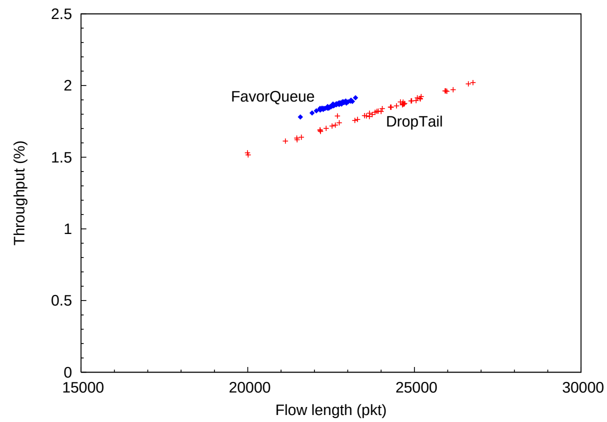

Fig. 13. Mean throughput as a function of flow length for 50 persistent flows.

V. Understanding FavourQueue

The previous section has shown the benefits obtained with FavourQueue in terms of service. In this section, we analyse the reasons of the improvements brought by FavourQueue by looking at the AQM performance. We study the drop ratio and the queueing delay obtained by both queues in order to assess the reasons of the gain obtained by FavourQueue. We recall that for all experiments, FavourQueue is only set on the upstream. The reverse path uses a DropTail queue. In a first part, we look at the impact of the AQM on the network then on the end-host.

A. *Impact on the network*

Figure 14 shows the evolution of the average queueing delay depending on the size of the flow. This figure corresponds to the 10 averaged replications experiment (as defined Section III). Basically, the results obtained by FavourQueue and DropTail are similar. Indeed, the average queueing delay is 2.8ms for FavourQueue versus 2.9ms for DropTail and both curves similarly behave. We can notice that the queueing delay for the medium sized flows slightly increases with FavourQueue. These flows are characterized by a predominance of the slow-start phase as most of the packets that belong to these flows are emitted during the slow-start. Since during this phase each flow opportunistically occupies the queue, less packets are favoured due to the growth of the TCP window. As a result, their queuing delay increases. When the size of the flow increases (above hundred packets length), the slow-start is not pervasive anymore and the average queueing delay of each packet of these flows tends to be either higher or lower as suggested by the cloud Figure 14 depending on the number of favoured packets during their congestion avoidance phase.

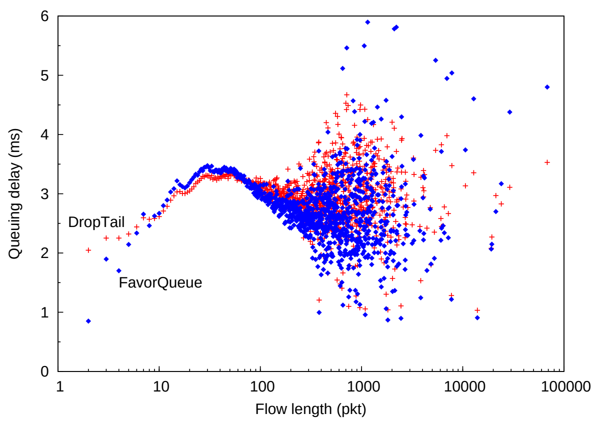

Fig. 14. Average queuing delay according to flow length.

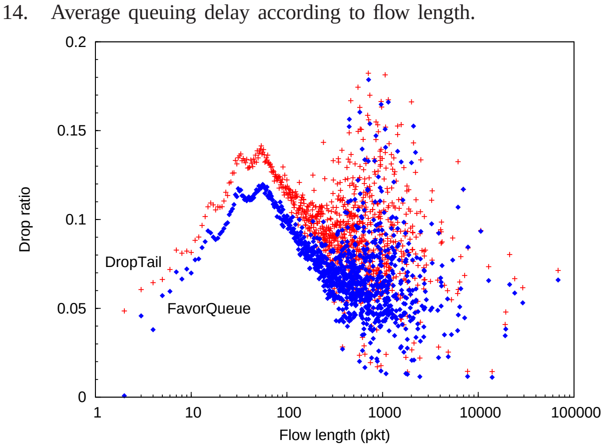

Fig. 15. Average drop ratio according to flow length.

However and as suggests Figure 15, the good performance in terms of latency obtained by FavourQueue previously shown Figure 3 in Section IV are mostly due to a significant decrease of the drop ratio. If we look at the average drop ratio of both queues in Figure 15, still as a function of the flow length, we clearly observe that the number of packets dropped is lower for FavourQueue. Furthermore, the loss ratio for the flow size of 2 packets is about 10^{-3} meaning that the flows of this size obtain a benefit compared to DropTail. The slow-start phase is known to send burst of data [11]. Thus, most of the packets sent during the slow-start phase have a high probability to be not favoured. This explains the increase of the drop ratio according to the flow size until 60 packets. Indeed, in the slow-start phase, packets are sent by burst of two packets. As a result, the first packet is favoured and the second one will be favoured only if the first is already served. Otherwise, the second packet might be therefore delayed. Then, when their respective acknowledgements are back to the source, the next sending will be more spaced. As FavourQueue might decrease the burstiness of the slow-start, we might decrease the packet loss rate and thus improve short TCP flow performance.

If we conjointly consider both figures 14 and 15, we observe that FavourQueue enables a kind of traffic shaping that decreases TCP aggressivity during the slow-start phase which results in a decrease of the number of dropped packets. As the TCP goodput is proportional to $1/(RTT.\sqrt{p})$ [12], the decrease of the drop ratio leads to an increase of the goodput which explains the good performance obtained by FavourQueue in terms of latency.

The loss ratio of SYN segments is on average 1.8% with DropTail. However for a load higher than 0.75, this loss ratio value reaches 2.09% while with FavourQueue, this ratio is 0.06%. Finally on average for all load conditions, this value is 0.04% with FavourQueue. These results demonstrate the positive effect to protect SYN segments from the loss. Obviously, by using FavourQueue in duplex mode (we recall that we have tested FavourQueue only on the upstream), this would further improve the results as SYN/ACK packets would have also been protected.

B. Impact on the end-host performance

The good performance obtained with FavourQueue in terms of latency are linked to the decrease of the losses at the beginning of the flow. In the following, we propose to estimate the benefits of our scheme by estimating the RTO ratio as a function of the network load. We define the RTO ratio $T(\rho)$ for a given load ρ as follows:

$$T(\rho) = \frac{\sum_{i=1}^{N} RTO_i}{\sum_{i=1}^{N}(L_i + R_i)}, \tag{3}$$

with RTO_i, the number of RTO for the i flow; R_i its number of retransmission and L_i its size. The ten replications experiment in Figure 16 presents the evolution of the RTO ratio for FavourQueue and DropTail and shows that the decrease of the loss ratio results in a decrease of the RTO ratio for FavourQueue. This also shows the advantage to use FavourQueue when the network is heavily loaded.

We now evaluate the RTO recovery ratio as a function of the flow length. We define this RTO recovery ratio as follows:

$$\tau(x) = \frac{\sum_{i=1}^{N} RTO_i}{\sum_{i=1}^{N}(RTO_i + FR_i)}, \tag{4}$$

with FR_i, the number of TCP Fast Retransmit for the i flow. In terms of RTO recovery, Figure 17 shows a significant decrease of the number of recovery with an RTO. Concerning the ratio of Fast Retransmit for this experiment, we observe an increase of 14% with FavourQueue. As a fast recovery packet is placed at the beginning of a window, FavourQueue prevents the loss of a retransmission. Then, the number of recovery with Fast Retransmit is higher with FavourQueue and the latency observed is better since the retransmission are faster.

For the flow with a size strictly below six packets, the recovery is exclusively done by a RTO. Indeed, in this case there is not enough duplicate acknowledgement to trigger a Fast Retransmit. For the flow above six packets, we observe

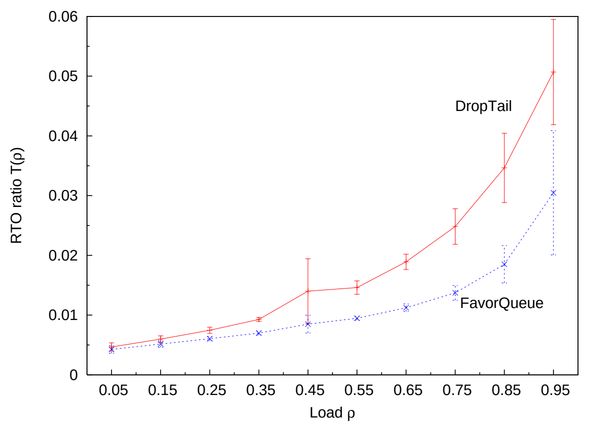

Fig. 16. RTO ratio as a function of the network load.

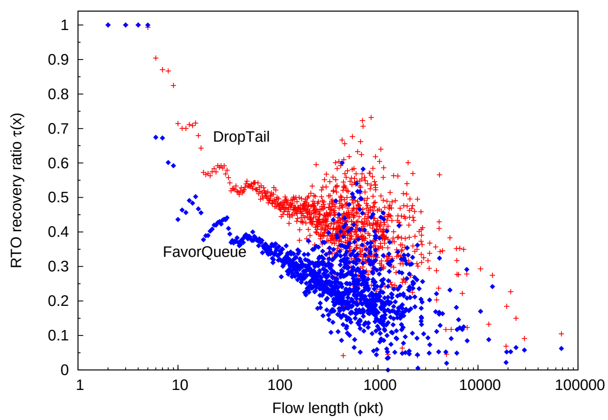

Fig. 17. RTO recovery ratio according to flow length.

a noticeable decrease of the RTO ratio due to the decrease of the packet lost rate on the first packets of the flow. Thus, the number of duplicate acknowledgement is higher, allowing to trigger a Fast Retransmit recovery phase. The trend shows a global decrease of the RTO ratio when the flow length increases. On the overall, the RTO recovery ratio reaches 56% for DropTail and 38% for FavourQueue. The decrease of the gain obtained follows the increase of the flow size. This means that FavourQueue helps the connection establishment phase.

VI. Stochastic model of FavourQueue

We analyze in this part the impacts of the temporal and drop priorities previously defined in Section II and propose a stochastic model of the mechanism.

A. Preliminary statistical analysis

We first estimate the probability to favour a flow as a function of its length by a statistical analysis. We define $P(Favor|S = x)$, the probability to favour a flow of size s, as follows:

$$P(Favor|S = s) = \frac{\sum_{i=1}^{N} fa_i}{\sum_{i=1}^{N} s + R_i}. \quad (5)$$

with fa_i, the number of packets which have been favoured and R_i the number of retransmitted packets of a given i flow. The number of favoured packets corresponds to the number of packets selected to be favoured at the router queue. Figure 18 gives the results obtained and shows that:

- the flows with a size of two packets are always favoured;
- the middle sized flows that mainly remain in a slow-start phase are less favoured compared to short flows. The ratio reaches 50% meaning that one packet among two is favoured;
- long TCP flows get a favouring ratio around 70%.

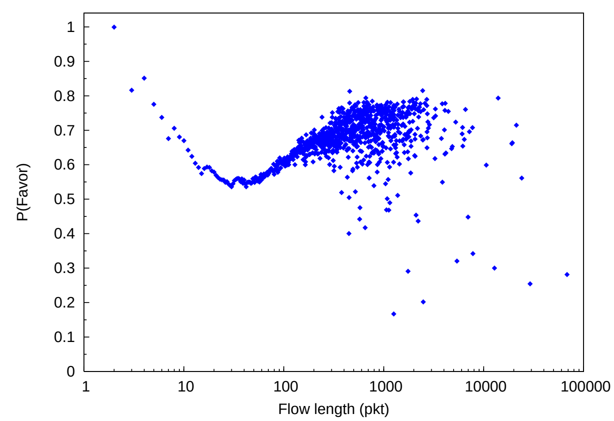

Fig. 18. Probability of packet favouring according to flow length.

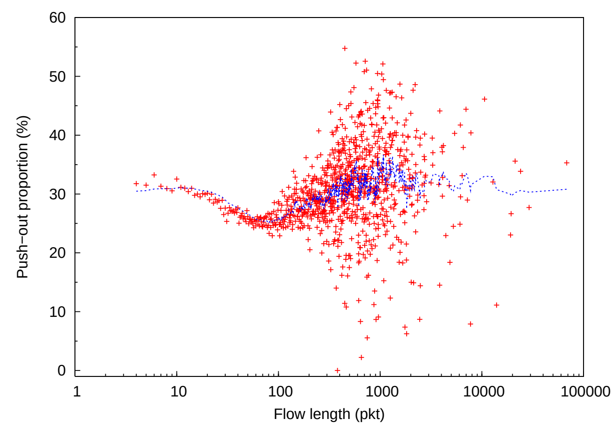

Fig. 19. Push-out proportion of drop as a function of flow length.

We also investigate the ratio of packets dropped resulting from the push-out algorithm as a function of the flow length in order to assess whether some flows are more penalised by push-out. As shown, Figure 19, the mean is about 30% for all flows, meaning that the push-out algorithm does not impact more short than long TCP flows.

We now propose to build a stochastic model of Figure 18 in the following.

B. Stochastic model

We denote S: the random variable of the size of the flow and Z: the Bernoulli random variable which is equal to 0 if no favoured packets are present in the queue and 1 otherwise. We then distinguish three different phases:

- phase #1 : each flows have a size lower than s_1. In this phase, the flows are in slow-start mode. This size is a parameter of the model which depends of the load. ;

- phase #2 : each flows have a size higher than s_1 and lower than s_2. In this phase, flows progressively leave the slow-start mode (corresponding to the bowl between $[10:100]$ in Figure 18). This is the most complex phase to model as all flows are either in the congestion avoidance phase or at the end of their slow-start. s_2 is also a parameter of the model which depends of the load;
- phase #3 : each flows have a size higher than s_2. All flows are in congestion avoidance phase. Note that the statistical sample which represents this cloud is not large enough to correctly model this part (as already pointed out in Section IV-A). However, one other important result given by Figure 18 is that 70% of packets of flows in congestion avoidance mode are favoured. We will use this information to infer the model. This also confirms that the spacing between each packet in the congestion avoidance phase increases the probability of an arriving packet to be favoured.

First phase: We consider a bursty arrival and assume that all packets belonging to the previous RTT have left the queue. Then, the burst size (BS) can take the following values: $BS = 1, 2, 4, 8, 16, 32, \ldots$. If $Z = 0$, we assume that a maximum of 3 packets can be favoured in a row[3], the packets number that are favoured are $1, 2, 3, 4, 5, 6, 8, 9, 10, 16, 17, 18, \ldots$ and $1, 2, 4, 8, 16, 32, \ldots$ if $Z = 1$. Thus, if $Z = 0$, the probability to favour a packet of a flow of size s is:

$$P(Favor|(Z=0,S=s)) = \begin{cases} s, s \leqslant 6 \\ \frac{s-1}{s}, 7 \leqslant s \leqslant 10 \\ \frac{9}{s}, 11 \leqslant s \leqslant 15 \\ \frac{s-6}{s}, 16 \leqslant s \leqslant 18 \\ \frac{12}{s}, 19 \leqslant s \leqslant 31 \\ (\ldots) \end{cases} \tag{6}$$

and with $Z = 1$:

$$P(Favor|(Z=1,S=s)) = \begin{cases} 1, s = 1 \\ \frac{2}{s}, 2 \leqslant s \leqslant 3 \\ \frac{3}{s}, 4 \leqslant s \leqslant 7 \\ \frac{4}{s}, 8 \leqslant s \leqslant 15 \\ \frac{5}{s}, 16 \leqslant s \leqslant 31 \\ (\ldots) \end{cases} \tag{7}$$

The probability to favour a packet of a flow of size s is thus:

$$\begin{aligned} P(Favor|S=s) = \quad & P(Z=0).P(Favor|(Z=0,S=s)) + \\ & P(Z=1).P(Favor|(Z=1,S=s)) \end{aligned} \tag{8}$$

Once again, $P(Z = 0)$ and $P(Z = 1)$ depends on the load of the experiment and must be given.

Second phase: In this phase, each flow progressively leaves the slow-start phase. First, when a flow finishes its slow-start phase, each following packets have a probability to be favoured of 70% (as shown in in Figure 18). So, we now need to compute an average value of the probabilty to favour a packet for a given flow. We also have to take into account that, for a given size of flow s, only a proportion of these flows have effectively left the slow-start phase. The other ones remain in slow-start and the analysis of their probabilty to favour a packet follows the first phase. To correctly describe this phase, we need to assess which part of flows of size s, $s_1 \leqslant s \leqslant s_2$, has left the slow start phase at packet s_1, $s_1 + 1$, ... s. As a first approximation, we use a uniform distribution between s_1 and s_2. This means that for flows of size s, the proportion of flows which have left the slow-start phase at s_1, $s_1 + 1$, ... $s - 1$ is $\frac{1}{s_2 - s_1}$ and the proportion of flows of size s which have not yet left the slow-start phase is thus $\frac{s_2 - s}{s_2 - s_1}$.

If we denote p_k the proportion of flows of size $s \geqslant s_1$ that have left the slow start-phase at k we have:

$$P(Favor|(S=s,Z=0)) = \sum_{i=0}^{s-s_1-1} p_k.P(Favor|k = s_1 + i, Z = 0, S = s)$$

and

$$P(Favor|(S=s,Z=1)) = \sum_{i=0}^{s-s_1-1} p_k.P(Favor|k = s_1 + i, Z = 1, S = s)$$

and as in (8) we obtain:

$$\begin{aligned} & P(Favor|S=s) = \\ & P(Z=0). \sum_{i=0}^{s-s_1-1} p_k.P(Favor|k = s_1 + i, Z = 0, S = s) + \\ & P(Z=1). \sum_{i=0}^{s-s_1-1} p_k.P(Favor|k = s_1 + i, Z = 1, S = s) \end{aligned}$$

Third phase: The model of this phase is quite simple. In fact, each packet of a flow which have left the slow-start phase have a probability to be favoured of 70%. We compute the probability for a packet to be favoured by taking into account the time at which a flow has left the slow-start phase and the proportion of flows as in the second phase.

Model fitting: To verify our model, among the ten loads that are averaged in Figure 18, we choose two verify our model for two loads: $\rho = 0.25$ and $\rho = 0.85$. For the first one we have estimated $P(Z = 1) = 0.25$ and $P(Z = 1) = 0.7$ for the second. Figures 20 and 21 show that our model correctly fits both experiments.

This model allows to understand the peaks in Figure 20 when the flow size is lower than hundred packets. These peaks are explained by the modelling of the first phase. Indeed, the traffic during the slow-start is bursty, then, each burst has either one or two packets favoured as a function of Z (i.e. up to three packets are favoured when $Z = 0$ and only one when $Z = 1$ as given by (6) and (7)).

VII. Related work

Several improvements have been proposed in the literature and at the IETF to attempt to solve the problem of short

[3]The rationale is the following, if $Z = 0$ a single packet (such as the SYN packet) is favoured and one RTT later, the burst of two packets (or larger) will be favoured if we consider that the first packet of this burst is directly served.

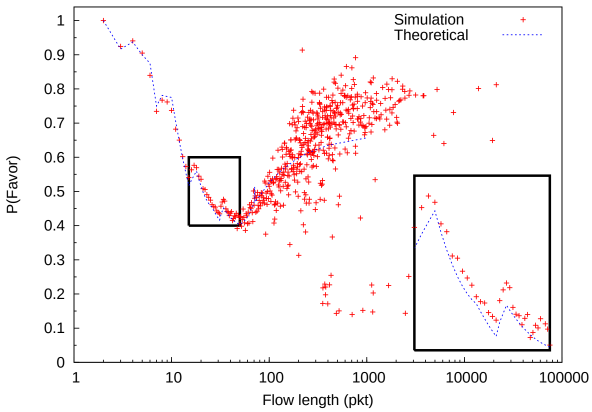

Fig. 20. Model fitting for $\rho = 0.25$ with $P(Z = 1) = 0.25$.

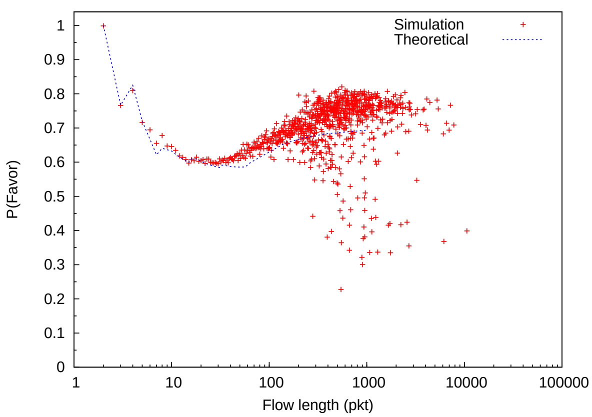

Fig. 21. Model fitting for $\rho = 0.85$ with $P(Z = 1) = 0.7$.

TCP flows performance. Existing solutions can be classified into three different action types: (1) to enable a scheduling algorithm at the router queue level; (2) to give a priority to certain TCP packets or (3) to act at the TCP level in order to decrease the number of RTO or the loss probability. Concerning the two first items, the solution involves the core network while the third one involves modifications at the end-host. In this related work, we first situate FaQ among several core network solutions and then explain how FaQ might complete end-hosts' solutions.

A. Enhancing short TCP flows performance inside the core network

1) The case of short and long TCP flows differentiation: Several studies [13][10][14] have proposed to serve first short TCP traffic to improve the overall system performance. These studies follow one queueing theory result which stands that the overall mean latency is reduced when the shortest job is served first [15]. One of the precursor in the area is [14], where the authors proposed to adapt the Least Attained Service (LAS) [15], which is a scheduling mechanism that favors short jobs without prior knowledge of job sizes, for packets networks. As for FavourQueue, LAS is not only a scheduling discipline but a buffer management mechanism. This mechanism follows FavourQueue principle since the priority given to the packet is done without knowlegde of the size of the flow and that the classification is closely related to the buffer management scheme. However, the next packet serviced under LAS is the one that belongs to the flow that has received the least amount of service. By this definition, LAS will serve packets from a newly arriving flow until that flow has received an amount of service equal to the amount of least service received by a flow in the system before its arrival. Compared to LAS, FavourQueue has no notion of amount of service as we seek to favour short job by accelerating their connection establishement. Thus, there is no configuration and no complex settings.

In [13] and [10], the authors push further the same idea and attempt to differentiate short from long TCP flows according to a scheduling algorithm. The differences between these solutions are based on the number of queues used which are either flow stateless or stateful. Theses solutions uses an AQM which enables a push out algorithm to protect short TCP flow packets from loss. Short TCP flows identification is done inside the router by looking at the TCP sequence number [10]. However and in order to correctly distinguish short from long TCP flows, the authors modify the standard TCP sequence numbering which involves a major modification of the TCP/IP stack. In [13], the authors propose another solution with a per-flow state and deficit round robin (DRR) scheduling to provide fairness guarantee. The main drawback of [14][13] is the need of a per-flow state while [10] requires TCP senders modifications.

2) The case of giving a priority to certain TCP packets: Giving a priority to certain TCP packets is not a novel idea. Several studies have tackled the benefit of this concept to improve the performance of TCP connection. This approach was really popular during the QoS networks research epoch as many queueing disciplines was enabled over IntServ and DiffServ testbed allowing researchers to investigate such priority effects. Basically, the priority can be set intra-flow or inter-flow. Marco Mellia et Al. [16] have proposed to use intra-flow priority in order to protect from loss some key identified packets of a TCP connection in order to increase the TCP throughput of a flow over an AF DiffServ class. In this study, the authors observe that TCP performance suffers significantly in the presence of bursty, non-adaptive cross-traffic or when it operates in the small window regime, *i.e.*, when the congestion window is small. The main argument is that bursty losses, or losses during the small window regime, may cause retransmission timeouts (RTOs) which will result in TCP entering the slow-start phase. As a possible solution, the authors propose qualitative enhancements to protect against loss: the first several packets of the flow in order to allow TCP to safely exit the initial small window regime; several packets after an RTO occurs to make sure that the retransmitted packet is delivered with high probability and that TCP sender exits the small window regime; several packets after receiving three duplicate acknowledgement packets in order to protect the retransmission. This allows to protect against losses the packets that strongly impact on the average TCP throughput. In [3][17], the authors propose a solution on inter-flow priority. The short TCP flow are marked IN. Thus, packets from these flows are marked as a low drop priority. The differentiation in core routers is applied by an active queue management. When sender has sent a number of packets that exceeds the flow identification threshold, the packet are marked OUT and the drop probability increase. However, these approaches need the support of a DiffServ architecture to perform [18].

B. Acting at the TCP level

The last solution is to act at the TCP level. The first possibility is to improve the behavior of TCP when a packet is dropped during this start up phase (i.e. initial window size, limited transit). The second one is to prevent this drop by decreasing the probability of segments lost. For instance, in [19], the authors propose to apply an ECN mark to SYN/ACK segments in order to avoid to drop them. The main drawback of these solutions is that they require important TCP sender modifications that might involve heavy standardisation process.

We wish to point out that one of the current hot topic currently discussed within the Internet Congestion Control Research Group (ICCRG) deals with the TCP initial window size. In a recent survey, the authors of [20] highlight that the problem of short-lived flows is still not yet fully investigated and that the congestion control schemes developed so far do not really work if the connection lifetime is only one or two RTTs. Clearly, they argue for further investigation on the impact of initial value of the congestion window on the performance of short-lived flows. Some recent studies have also demonstrated that larger initial TCP window helps faster recovery of packet losses and as a result improves the latency in spite of increased packet losses [21], [22]. Several proposals have also proposed solutions to mitigate the impact of the slow start [23], [24], [25].

Although we do not act at the end-host side, we share the common goal to reduce latency during the slow start phase of a short TCP connection. However, we do not target the same objective. Indeed, end-host solutions, that propose to increase the number of packets of the initial window, seek to mitigate the impact of the RTT loop while we seek to favour short TCP traffic when the network is congested. At the early stage of the connection, the number of packets exchanged is low and a short TCP request is both constrained by the RTT loop and the small amount of data exchange. Thus, some studies propose to increase this initial window value [21], [22]; to change the pace at which the slow-start sends data packets by shrinking the timescale at which TCP operates [26]; even to completely suppress the slow-start [24]. Basically, all these proposals attempt to mitigate the impact of the slow-start loop that might be counterproductive over large bandwidth product networks. On the contrary, FavourQueue do not act on the number of data exchanged but prevents losses at the beginning of the connection. As a result, we believe that FavourQueue must not be seen as a competitor of these end-host proposals but as a complementary mechanism. We propose to illustrate this complementarity by looking at the performance obtained with an initial congestion window sets to ten packets. Figure 22 gives the complementary cumulative distribution function of the latency for DropTail and FavourQueue with flows with an initial slow-start set to two or ten packets. We do not have changed the experimental conditions (i.e. the router buffer is still set to eight packets) and this experiment corresponds to a ten averaged experiments (see section III). As explained in [21], if we focus on the results obtained with DropTail for both initial window size, the increase of the initial window improves the latency (with the price of an increase of the loss rate as also denoted in [21]). However, the use of FavourQueue enforces the performance obtained and complement the action of such end-host modifications making FavourQueue a generic solution to improve short TCP traffic whatever the slow-start variant used.

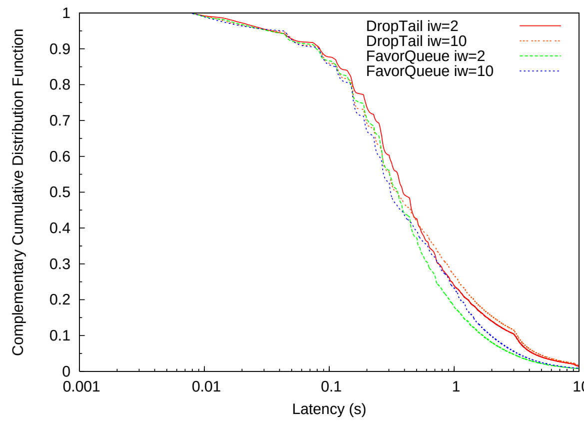

Fig. 22. Comparison of the benefit obtained in terms of latency with an initial TCP window size of ten packets.

VIII. Discussion

A. Security consideration

In the related work presented in Section VII, we present a similar solution to our proposal that gives priority to TCP packets with a SYN flag set. One of the main criticism that raises such kind of proposals usually deal with TCP SYN flood attack where TCP SYN packets may be used by malicious clients to improve this kind of threat [27]. However, this is a false problem as accelerating these packets do not introduce any novel security or stability side-effects as explained in [28]. Today, current kernel enables protection to mitigate such well-known denial of service attack[4] and current Intrusion Detection Systems (IDS) such as SNORT[5] combined with firewall rules allow network providers and companies to stop such attack. Indeed, the core network should not be involved in such end-host security issue that should remain under the reponsability of edge networks and end-hosts. Concerning the reverse path and as raised in [28], provoking web servers or hosts to send SYN/ACK packets to third parties in order to perform a SYN/ACK flood attack would be greatly inefficient. This is because the third parties would immediately drop such packets, since they would know that they did not generate the TCP SYN packets in the first place.

[4] See for instance http://www.symantec.com/connect/articles/hardening-tcpip-stack-syn-atta

[5] http://www.snort.org/

B. Deployment issue

Although there is no scalability issue anymore inside new Internet routers that can manage millions of per-flow state [6]. FavourQueue does not involve per-flow state management and the number of entries that need to handle a FavourQueue router is as a function of the number of packets that can be enqueued. Furthermore, as the size of a router buffer should be small [9], the number of states that need to be handle is thus bounded.

To sump up, the proposed scheme respects the following constraints:

- easily and quickly deployable; this means that FavourQueue has no tuning parameter and does not require any protocol modification at a transport or a network level;
- independently deployable: installation can be done without any coordination between network operators. Operation must be done without any signaling;
- scalable; no per-flow state is needed.

FavourQueue should be of interest for access networks; entreprise networks or universities where congestion might occur at their output Internet link.

IX. Conclusion

In this paper, we investigate a solution to accelerate short TCP flows. The main advantages of the proposed AQM is that FavourQueue is stateless; does not require any modification inside TCP; can be used over a best effort network; does not request to be completely deployed over an Internet path. Indeed, a partial deployment could only be done over routers from an Internet service provider or over a specific AS.

We drive several simulation scenarios showing that the drop ratio decreases for all flow length, thus decreasing their latency. FavourQueue significantly improves the performance of short TCP traffic in terms of transfer delay. The main reasons are that this mechanism strongly reduces the loss of a retransmitted packet triggered by an RTO and improves the connection establishment delay. Although FavourQueue targets short TCP performance, results show that by protecting retransmitted packets, the latency of the whole traffic and particularly non-opportunistic flows, is improved.

In a future work, we aim at investigating FavourQueue with rate-based transport protocols such as TFRC in order to verify whether we would benefit similar properties and with delay-based TCP protocol variants (such as TCP Vegas and TCP Compound) that should intuitively take large benefit of such AQM. We also expect to enable ECN support in FavourQueue.

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