# The persistent congestion problem of FAST-TCP: Analysis and solutions

M. Rodríguez-Pérez\*, S. Herrería-Alonso, M. Fernández-Veiga, C. López-García

Dept. Telematics Engineering, ETSE Telecomunicación, Campus Universitario Lagoas-Marcosende s/n, 36310 Vigo, Spain

### SUMMARY

FAST-TCP achieves better performance than traditional TCP-Reno schemes, but unfortunately it is inherently unfair to older connections due to wrong estimations of the round-trip propagation delay. This paper presents a model for this anomalous behavior of FAST flows, known as the persistent congestion problem. We first develop an elementary analysis for a scenario with just two flows, and then build up the general case with an arbitrary number of flows. The model correctly quantifies how much unfairness shows up among the different connections, confirming experimental observations made by several previous studies.

We built on this model to develop an algorithm to obtain a good estimate of the propagation delay for FAST-TCP that enables to achieve fairness between aged and new connections while preserving the high throughput and low buffer occupancy of the original protocol. Furthermore, our proposal only requires a modification of the sender host, avoiding the need to upgrade the intermediate routers in any way. Copyright © 2010 John Wiley & Sons, Ltd.

# 1. INTRODUCTION

Delay-based congestion avoidance (DCA) algorithms are a promising alternative to standard congestion avoidance algorithms which employ packet loss as an indicator of network congestion [1, 2, 3]. In fact, DCA algorithms outperform TCP-Reno [4] and its variants [5] in the aspects of overall network utilization, stability and low buffer occupancy [1, 3, 6].

FAST-TCP [1] is a good example of pure DCA methods. This algorithm reacts to increments in round-trip time (RTT) in an attempt to avoid network congestion before losses occur. Though it achieves higher throughput, lower transit delays and fewer packet losses than previous versions of TCP, the FAST congestion avoidance algorithm exhibits some anomalous behaviors that lead to an inefficient or unfair use of network resources. It may be regarded as its less harmful effect that the performance degrades when congestion arises in the return path, because several enhancements have been recently proposed that can overcome such problems [7, 8, 9, 10]. A second weakness is put in appearance when flows from different TCP implementations share a link and compete for bandwidth. Mixed with Reno or alike versions, in which packet drops are the unique congestion signals the sender reacts to, FAST flows are unable to achieve their fair share of bandwidth [11, 12] for the fundamental reason that their implicit utility functions are different [13, 14], thus making the network have multiple operating points [15]. Nevertheless, this phenomenon does not affect the stability of the network, neither does it affect intraprotocol fairness, nor does it prevent FAST from being useful in homogeneous high-throughput network domains. Furthermore, there are proposals that try to detect [16] or even react to this condition achieving a fair share of the bandwidth in the long run [17, 18].

The third idiosyncratic behavior of FAST is known as the *persistent congestion problem* [1, 19], and is a side effect of the procedure for detecting congestion. Recall that FAST interprets the increments in the RTT as a sign of incipient congestion. For tracking those changes, it keeps both an accurate measure of the current RTT and an estimate of the round trip propagation delay, which

<sup>\*</sup>Correspondence to: Email address: Miguel.Rodriguez@det.uvigo.es.

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is equated to the minimum RTT observed throughout a single connection. This sampling renders an almost exact value when there is actually a single FAST flow in the network, or when the bottleneck link is shared with TCP-Reno flows. In the second case because the buffer at the bottleneck is likely to empty frequently. However, when a set of FAST flows coexist in the network, the dynamics of the congestion control algorithm lead the buffer occupancy along their paths approximately to a constant level. At this point, if a new connection starts, it would mistake the current RTT for the propagation delay. This overestimation is the cause of serious unfairness among the contending flows [19, 20, 21].

Several techniques have been proposed to correct persistent congestion and ameliorate fairness between new and aged FAST connections [13, 22, 23, 24, 25]. Unfortunately, all these solutions rely on queue management mechanisms at the intermediate routers, thus hindering large scale deployment. A sound tentative solution to solve persistent congestion without assistance from routers has been presented in [26]. It relies on a new method to obtain a good estimate of the propagation delay that consists, basically, in throttling down briefly each newly started flow to allow router queues to empty. Nevertheless, as we demonstrate in this paper, this approach is not effective in all circumstances.

We provide a mathematical model that predicts the buffer occupancy and the individual throughput for unsynchronized FAST connections. The results of this analysis are used to clarify the network conditions that lead to the flawed performance of the rate reduction method. Moreover, this model is also used to build a solution able to remove the undesired effect of persistent congestion under more general conditions. As in [26], our proposal only requires the modification of the sender end host and, consequently, router queues can remain simple FIFO buffers without jeopardizing FAST performance. Further, our solution retains the useful properties of FAST-TCP attaining high throughput and low router buffer utilization [27].

The rest of this paper is organized as follows. Section 2 summarizes previous work on models for the FAST-TCP behavior and persistent congestion issues. Section 3 gives a brief overview of pure DCA algorithms in general and FAST-TCP in particular. In Section 4, we analyze the persistent congestion problem and its impact on the transmission rates of FAST flows on several scenarios. Section 5 illustrates why the rate reduction method proposed in [26] fails to solve this bias in networks with small propagation delays or when they are shared by many

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flows. In Section 6, we present a solution to the persistent congestion problem that lacks the limitations of the rate reduction method. Section 7 contains some simulation experiments that validate both the proposed analysis and our solution. Lastly, Section 8 summarizes this work.

### 2. RELATED WORK

Because of their ability to keep the network out of congestion, delay based congestion avoidance (DCA) proposals started to appear in the literature shortly after the appearance of Van Jacobson's seminal paper of TCP congestion control [28]. In fact, in [2] a proposal is presented to use delay for congestion avoidance in interconnected networks. However, the first approach to complement TCP congestion control with a delay avoidance algorithm did not appear until the TCP-Vegas proposal [3]. Models for the behavior of Vegas started appearing shortly after. Most of them are also applicable to other pure-DCA approaches, that is, protocols that only need packet delay measures to react to congestion, as, for instance, FAST-TCP [1].

These first works focused on the interactions between pure DCA approaches and the predominant TCP-Reno congestion control algorithm [11, 21, 19], and found both approaches not entirely compatible. In particular, Reno based algorithms were shown to be more aggressive occupying network buffers than DCA flows. This, coupled with a FIFO queueing discipline, made the latter get a throughput lower than expected. However, in the recent years some proposals have appeared that try to mitigate this problem by providing methods for pure DCA algorithms to self-tune themselves for being more aggressive when competing against Reno-like traffic [17, 18]. The problem with these proposals is that they either are very slow reacting or that they behave like Reno when the network is congested. In [16] a proposal is made to better detect the presence of Reno-like traffic that could lead to more rapid reaction to its presence, finally giving an incentive to users to employ FAST congestion control without having to sacrifice performance.

Another intrinsic limitation of some pure DCA methods, and the one that this paper deals with, is the *persistent congestion* problem. It was first described for Vegas flows in [19] and it is a direct consequence of an important property of both Vegas and FAST, namely, that once equilibrium is reached, they maintain a constant amount of traffic queued at network routers. Although this is usually a nice property, as it prevents jitter in packet arrivals and keeps latency to a minimum, it also hinders

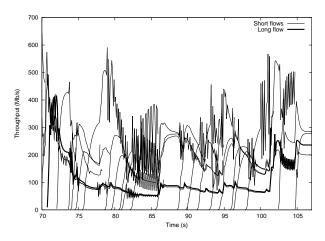


Figure 1. Example trajectory of long running FAST flows vs short flows showing the persistent congestion effect.

the measurement of the propagation delay. DCA protocols estimate the round trip propagation delay as the smallest measure of the round trip time during a whole connection. When the bottlenecks empty frequently, like it is the case when the network is shared with Reno flows, this procedure can produce accurate measurements. This is precisely the reason why the LEDBAT algorithm [29] does not encounter the persistent congestion problem in current networks. However, when the network is being used by pure DCA flows, bottlenecks always contain a certain amount of enqueued traffic and thus the propagation delay is overestimated. This overestimation, or more exactly, the different estimations done by the different flows depending on the minimum network load during their lifetime, is the cause of serious unfairness. Non pure DCA algorithms, like, for instance, Compound TCP [30] do not suffer from this problem, because the non DCA component of the protocol guaranties that the buffer occupation varies with time. Thus, the estimation of the round trip propagation delay improves as successive measures of the round trip time are more likely to encounter emptier buffer at bottlenecks after packet losses.

To illustrate it, we have run a short experiment with the help of the ns-2 [31] simulator. In a simple dumbbell topology we have set up two kind of FAST flows sharing a gigabit bottleneck. Flows of the first kind, *long flows*, send 500 megabytes of data, while the second kind of flows, *short flows*, only send 20 megabytes. Flows arrive at the network following two independent Poisson arrival processes. In Fig. 1 we can see in detail a small part of the simulation. The thicker line represents the throughput of a paradigmatic long flow. We see how at the start it

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takes more share than its fair rate, however, as time passes and shorter flows arrive at the network they consistently obtain more share than the long flow. Thus, as long flows suffer the arrival of more flows in their lifetime than short flows, they suffer more the consequences of the persistent congestion problem and are unable to get their fair share of network resources. Moreover, short flows enjoy for a significant percentage of their lifetime being the last coming flow to the bottleneck, and thus they benefit from the persistent congestion problem. In fact, we have measured in the above scenario that long flows take on average 40 % longer to finish than what it would take if they had enjoyed their fair share.

The first mathematical model of this problem appeared in [13], which provided a recursive set of equations to calculate the throughput of a set of flows reaching a bottleneck consecutively. However, this model employs a different interpretation of Vegas parameters than those found in actual implementations. This further biases the results against old flows.

In this paper we will provide a set of equations according to the actual interpretation of FAST (and Vegas) parameters for a scenario of sequential pure DCA arrivals to a single bottleneck and for the arrival of a single pure DCA flow to a bottleneck that is already being fairly shared. This latter case will help us to provide a better mechanism to estimate the propagation delay and thus fix the persistent congestion problem.

We build on equations for the expected throughput of a pure DCA connection in equilibrium provided also by these previous works. In fact, much of the work on this paper builds on the equilibrium formulas found in [32] for Vegas and [1] for FAST, that, in the absence of heavy congestion (packet losses) happen to be identical.

Different improvements have been proposed to correct persistent congestion and ameliorate fairness between new and aged pure DCA connections. [22] proposes the use of RED gateways at the routers to get a more even distribution of the bandwidth regardless of the starting time of competing flows. However, finding the appropriate threshold values for the RED gateways is not an easy problem and remains an open issue [33]. [13] suggests a way to eliminate persistent congestion using REM at the routers. REM [23] is an active queue management scheme that keeps buffer low while leading to the high utilization of the link at the same time. Certainly, with small queues, the minimum of all measured RTTs is an accurate approximation to propagation delay. In [24] a new IP option named AQT (Accumulate Queueing Time) is defined to collect the queueing time experienced by

FAST packets along the path. With this scheme, FAST sources must send some probing packets with the AQT option active while routers must compute the queueing time for each receiving probing packet and add it to the AQT field. As a result, each connection is able to obtain a good estimate of the propagation delay canceling out the queueing time from the RTT measurement. [25] solves the persistent congestion problem by marking the ToS field in the IP header with the highest priority for the first packet of each flow. With priority queueing at routers, highest priority packets will be dispatched immediately even if the router buffer is not empty and, therefore, FAST-TCP will obtain an accurate estimate of the propagation delay.

Unfortunately, all these solutions rely on queue management mechanisms at the intermediate routers, thus hindering large scale deployment. An interesting method to obtain a good estimate of the propagation delay without assistance from routers has been presented in [26]. Basically, it consists on throttling briefly each newly started flow in an attempt to empty router queues so that it can obtain a good estimate of its true propagation delay. Nevertheless, as we will demonstrate in Section 5, this solution is not effective in networks with short propagation delays or when shared by many flows.

# 3. FAST-TCP DESCRIPTION

DCA algorithms work on the assumption that it is possible to gain insight into network status by observing the variations in the RTT. The difference between the RTT and the propagation delay is directly related to the amount of data in transit, and, the more data in the network, the nearer it is to become congested. So, adjusting the window size based on these variations, DCA flows keep an appropriate transmission rate without causing congestion. In contrast, TCP-Reno and its variants need to drive the network to congestion to receive the feedback needed to adjust the window. They keep slowly incrementing the window size until it reaches a point that buffers overflow and packet losses occur. This leads sources to abruptly reduce their sending rates and the slow increment phase begins again, preventing Reno flows from fully using all the available bandwidth. Thus DCA algorithms are more suitable in long fat pipes where packet losses are too scarce to properly adjust the rate or for those applications negatively affected by sudden changes in the transmission rate.

Both TCP-Vegas and FAST-TCP employ a similar *modus operandi*, in fact, FAST-TCP can be treated as an improved (faster) version of the former [1]. Throughout the rest of this paper we will focus on FAST, although

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To modulate its transmission rate FAST-TCP employs a congestion window analogous to the one employed by Reno variants. The FAST congestion window can be characterized, at the flow level, by the following dynamic equation:

$$\dot{w}(t) = \gamma \alpha \left( 1 - \frac{q(t)x(t)}{\alpha} \right),$$
 (1)

where  $\gamma$  and  $\alpha$  are configuration parameters, q(t) is the instantaneous queueing delay and  $x(t) = \frac{w(t)}{d+q(t)}$  is the transmission rate, d being the round trip propagation delay. This equation has the property that the variation on the window size is directly proportional to the distance from equilibrium, yielding very *fast* convergence times.

The above dynamic flow level behavior is implemented at the packet level with the following rule. Every update interval, defined to be a constant time or some number of RTTs depending on the precise FAST version, the window size is updated as

$$w \leftarrow \gamma \left(\frac{\hat{d}w}{\hat{r}} + \alpha\right) + (1 - \gamma)w,$$
 (2)

where  $\hat{d}$  is the current estimation of the round trip propagation delay and  $\hat{r} = \hat{d} + \hat{q}$  is an estimation of the round trip time. The accurate estimation of d is a bit tricky as it can only be correctly measured in the absence of cross traffic. In practice  $\hat{d}$  is set to the minimum round trip time observed during the whole transmission. In the end, this is only a problem when different FAST flows have different overestimations. As long as all FAST flows make the same error, the fairness properties are not affected. Later in this paper, we will study this problem and provide solutions for it. Eq. (2) brings light to the meaning of  $\gamma$ . In fact it is just a *smoothing factor* or *gain* that controls the speed of convergence. Its value its taken from the semi-open range (0, 1], although  $\gamma = 0.5$  is its more common value.

The equilibrium properties of FAST are well established in the literature [1, 34] and coincide with those of Vegas [13, 32]. In fact, they are a direct consequence of its dynamic equation (1). It suffices to make  $\dot{w}(t) = 0$  to obtain that under equilibrium each connection attains a throughput

$$x^* = \frac{\alpha}{q^*} = \frac{\alpha}{\hat{r}^* - \hat{d}}.$$
(3)

This equilibrium formula merits some observations. Firstly, as long as all FAST flows share the same configuration (same  $\alpha$  value) and bottleneck, they

all obtain the same throughput, irrespective of their propagation delays. Secondly, it helps us to get insight into what is the effect of the  $\alpha$  parameter. In fact, if all the flows sharing a bottleneck, and thus observing the same queueing delay  $(\hat{r}^* - \hat{d})$ , are set up with the same  $\alpha$  parameter, they will get the same equilibrium transmission rate  $x_i^* = x^*$ . It then follows that

$$(\hat{r}^* - \hat{d}) \sum_{i=1}^n x_i^* = (\hat{r}^* - \hat{d})C = n\alpha,$$
(4)

for *n* flows and link capacity *C*. Finally, solving for  $\alpha$  in (4) it becomes apparent that each flow contributes  $\alpha$  packets to the bottleneck backlog.

There is a trade off for selecting an appropriate  $\alpha$  value. On the one hand we want to select a small value, to minimize overall latency and buffering needs in the network. On the other hand, big values provide faster convergence times. At the same time, too small  $\alpha$  values can produce too little queueing delays making their precise measure too difficult for end hosts.

Lastly, it must also be taken into consideration that not every combination of  $\alpha$  and  $\gamma$  produces stable configurations. Several papers deal with the conditions that both  $\alpha$  and  $\gamma$  must meet to reach equilibrium. More details can be found in [35, 36, 37, 38].

### 4. ANALYTICAL MODEL

Throughout this section we will build on the model for permanent congestion developed in [13] with the necessary adaptations for FAST and with a focus on reaching a closed-form formula that predicts persistent congesting effects. So, we will consider a stable all-FAST scenario where new flows arrive and modify the equilibrium throughput.

Under these conditions, with each flow keeping its amount of enqueued data in the network never below  $\alpha$ packets, new FAST flows are unable to obtain accurate measures of the path delays. Please note that scenarios with other types of TCP flows are of no interest to our study, as router queues get eventually empty, giving a chance to the FAST flows to accurately estimate their propagation delay.

### 4.1. Two flows scenario

We present firstly the most elementary case with just two FAST flows appearing consecutively on a network and sharing a single bottleneck. Without loss of generality, let

Copyright © 2010 John Wiley & Sons, Ltd. *Prepared using ettauth.cls*  us assume that  $d_i$  is the actual propagation delay of flow *i*, and that flow j > i starts after flow *i* reaches equilibrium.

Recall from Section 3 that in an all-FAST scenario without losses caused by congestion, each flow i achieves at equilibrium a throughput

$$x_i^* = \frac{\alpha_i}{\hat{r}_i^* - \hat{d}_i}.$$
(5)

where  $\hat{d}$  is an estimate of the round-trip propagation delay and r is the current RTT.<sup>†</sup> Let  $\alpha_i = \alpha$  for each flow i so as to achieve fairness [39]. Then, for the first flow where  $\hat{d}_1 = d_1 + \sum_l t_{\text{tx}_l}, t_{\text{tx}_l}$  is the transmission time of a packet on the  $l^{\text{th}}$  link, and  $r_1^* = d_1 + \sum_l t_{\text{tx}_l} + \alpha/C$ ,

$$x_1^* = \frac{\alpha}{(d_1 + \sum_l t_{tx_l} + \alpha/C) - (d_1 + \sum_l t_{tx_l})} = C,$$
(6)

where C is the capacity of the bottleneck link expressed in packets per second.

After the second flow starts,  $\alpha$  more packets should get enqueued at the bottleneck, and thus the RTT of the first flow would increase by  $\alpha/C$ , yielding hypothetical values of  $\hat{r}'_1 = \hat{d}_1 + 2 \cdot \alpha/C$  and  $x'_1 = C/2$ , as expected for a fair share of the bottleneck bandwidth. However, the second flow measures a wrong value for the propagation delay, because it encounters  $\alpha$  packets already enqueued at the bottleneck when it starts. Since the buffer occupancy never decreases, this causes an overestimation of the propagation delay and  $\hat{d}'_2 = d_2 + \sum_l t_{tx_l} + \alpha/C = \hat{r}_1^*$ . If this second flow were to enqueue just  $\alpha$  packets in the network, its RTT would then become  $\hat{r}'_2 = \alpha/C + \hat{d}'_2 = 2 \cdot \alpha/C + d_2 + \sum_l t_{tx_l}$ , for a (hypothetical) throughput of

$$x_2' = \frac{\alpha}{\alpha/C + \hat{d}_2' - \hat{d}_2'} = C.$$
 (7)

However, in the course of reaching  $x'_2$ , the aggregate throughput surpasses the bottleneck capacity, making the queue grow and leading to a bigger queueing delay. Let us call  $r_1^{\tau}(t)$  and  $r_2^{\tau}(t)$  the new round trip time measured by the first and the second flow after the second flow has enqueued at least  $\alpha$  packets in the network. Because both flows see the same increase in queueing delay, we can write  $r_1^{\tau}(t) = \hat{r}'_1 + \frac{\delta(t)}{C} = \hat{r}_1^* + \frac{\alpha + \delta(t)}{C}$  and  $r_2^{\tau}(t) = \hat{r}'_2 + \frac{\delta(t)}{C}$ , where  $\delta(t)$  accounts for the increase in queue length because of permanent congestion. For notational simplicity let us define a(t) such that  $\delta(t) = \alpha \cdot a(t)$ , and lets call

<sup>&</sup>lt;sup>†</sup>This equilibrium throughput is not exclusive of FAST-TCP, but is also obtained by at least by TCP-Vegas [32], in an all-Vegas scenario.

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 $r_i^{\tau}$ ,  $\delta$  and a the values of, respectively,  $r_i^{\tau}(t)$ ,  $\delta(t)$  and a(t) after equilibrium is reached again. With this notation, the throughput of each flow in the new equilibrium can be written as

$$x_1' = \frac{\alpha}{r_1^{\tau} - \hat{d}_1^*} = \frac{\alpha}{r_1^* + \frac{\alpha(1+a)}{C} - \hat{d}_1^*} = \frac{C}{2+a},$$
(8)

$$x'_{2} = \frac{\alpha}{r_{2}^{\tau} - \hat{d}'_{2}} = \frac{\alpha}{r'_{2} + \frac{\alpha a}{C} - \hat{d}'_{2}} = \frac{C}{1 + a},$$
(9)

taking into consideration that  $r_1^* - \hat{d}_1^* = r_2' - \hat{d}_2' = \alpha/C$ . The new equilibrium will be reached when the two

- following conditions are met:
  - 1. Both flows notice that they have enqueued  $\alpha$  packets in the network, and
  - 2. The aggregated throughput does not exceed the link capacity.

That is,  $x'_1 + x'_2 = C$ , and solving for a yields

$$a = \frac{\sqrt{5} - 1}{2}.$$
 (10)

For this particular a value, it holds that  $x'_1 = (1 - a) \cdot C$ and  $x'_2 = a \cdot C$ . That is, for the case with just two flows, the unfair bottleneck share achieved after both of them reach equilibrium is independent of  $\alpha$  and the respective propagation delays of the flows.

#### 4.2. Several flows arriving consecutively

We now extend the previous model to a scenario with an arbitrary number of flows n arriving sequentially. This is the same case studied in [13].

In our new scenario with n flows, flow i starts after the flow i-1 stabilizes. Extending the reasoning of the previous version, we can define  $\hat{r}_i^{\tau}(t) = \hat{d}_i^* + \frac{(n-i)\alpha + \delta(t)}{C}$ , as flow i only sees in its queueing delay the traffic enqueued by flows arriving later.

Under this condition, and without using any proposal to avoid persistent congestion, the normalized throughput of the  $i^{\text{th}}$  flow is

$$\frac{x'_i}{C} = \frac{1}{1 + n - i + \sum_{j=i}^n a_j},\tag{11}$$

where  $a_j$  accounts for the increase in the queue size required to make  $\sum_{i=1}^{j} x'_i/C \leq 1$  after the  $j^{\text{th}}$  flow joins. Obviously,  $a_0 = 0$ . Notice how, again, the values of  $a_i$  and,

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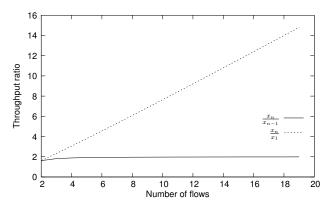


Figure 2. Throughput ratio achieved by the late coming flow relative to both the first flow and the previous one.

hence, the relative throughput obtained by each flow, do not depend on C or any other network property.<sup>‡</sup>

The vector  $\vec{a} = \{0, a_1, a_2, \dots, a_n\}$  can be calculated iteratively by solving the equation  $\sum_{i=1}^{n} x'_i/C = 1$  starting with n = 2. While there are simple algebraic formulae for this until n reaches 4, for greater values only numerical procedures are possible, but a few iterations of the Newton's method would suffice to give accurate results, for instance.

We have represented in Fig. 2 the expected throughput ratio of a newly arrived flow compared to both the oldest one and the previous arrival for up to twenty consecutive flows. It can be seen how even for a small number of consecutive arrivals the unfairness is severe. The newest flow always gets twice the bandwidth of the previous arrival and the oldest flow only gets one  $n^{\text{th}}$  of the newest flow throughput.

4.2.1. Consequences in Buffer Dimensioning and Queueing Delay The growth in the number of packets enqueued under persistent congestion can have a dramatic effect on buffer dimensioning. Under ideal circumstances it suffices to have a buffer size  $\alpha$  times the number of possible FAST flows to ensure that performance does not degrade. However, when taking into account the extra packets

<sup>&</sup>lt;sup>‡</sup>Note that eq. (11) predicts better fairness than [13]. This is because the latter uses an interpretation of Vegas behavior that further biases results towards flows with larger propagation delays. They consider  $\alpha$  and  $\beta$  parameters to represent a desired target transmission rate, whereas we interpret them as a target amount of data enqueued at the network buffers. This latter interpretation has the added benefit of making the actual values of the parameters independent of the physical characteristics of the network elements. Although it can be argued that their interpretation is more to the letter of the original Vegas paper [3], our model is in greater accordance with actual implementations [31, 40], FAST description [1] and other Vegas models [32].

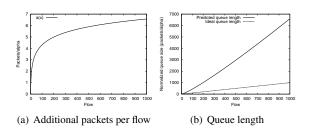


Figure 3. Additional packets enqueued under persistent congestion and predicted queue length.

enqueued under persistent congestion, the maximum buffer size increases substantially.

Fig. 3 shows the values of vector  $\vec{a}$  and the queue length at the bottleneck router when up to 1 000 FAST flows are created sequentially.<sup>§</sup> The number of extra packets per flow needed to reach equilibrium  $(a_i)$  grows with the logarithm of the number of flows. As a direct consequence, from around ten flows onwards, each new flow enqueues more than  $2\alpha$  packets. In fact, it is clearly seen that for one thousand flows the buffer size is almost seven times larger than the calculated value for ideal behavior.

#### 4.3. One flow arriving to a *n*-flows stable scenario

Another scenario of particular interest is the one in which a new flow arrives when n previous flows are fairly sharing the bottleneck bandwidth. This is what would happen if the flows had a method to counter-measure the persistent congestion problem, and thus we will employ this model in the following sections to analyze the rate reduction approach presented in [26] and to develop a new algorithm that reacts to persistent congestion in a distributed fashion.

So, in our new scenario, there are *n* flows with a correct estimation of the propagation delay that, hence, have enqueued exactly  $\alpha$  packets in the network each. As a result, they all experience the same RTT of  $\hat{r}_i^* = n \cdot \alpha/C + \hat{d}_i^*, \forall i \leq n$ . When the new flow arrives it will estimate its propagation delay as  $\hat{d}'_{n+1} = r_n = \ldots = r_1$ , for a target throughput equal to the bottleneck bandwidth *C*, leading to increased queueing delay. When equilibrium is restored the flows measure  $\hat{r}'_i = \hat{r}_i^* + \frac{\alpha+\delta}{C}, i \leq n$  and  $\hat{r}'_{n+1} = \hat{d}'_{n+1} + \frac{\alpha+\delta}{C}$ . Defining *a* again so as that  $\delta = \alpha \cdot a$ , the throughput in equilibrium of all the flows can be

expressed as

$$x'_{i} = \frac{\alpha}{\frac{n \cdot \alpha + \alpha \cdot a}{C}} = \frac{C}{n+1+a}, \ i \le n, \quad (12)$$

$$x'_{n+1} = \frac{\alpha \cdot C}{\alpha \cdot (1+a)} = \frac{C}{1+a},$$
(13)

where  $x'_i$  and  $x'_{n+1}$  must hold

$$\sum_{j=1}^{n} x'_{i} + x'_{n+1} = n \frac{C}{n+1+a} + \frac{C}{1+a} = C.$$
 (14)

Solving for a in eq. (14) yields

$$a = \frac{\sqrt{1+4n}-1}{2}.$$
 (15)

We can see again how the unfairness in the bottleneck sharing is independent of C and  $\alpha$ , and only depends on the number of flows sharing the resources. Moreover, this unfairness increases with the number of flows as  $\frac{x_{n+1}}{x_i} = O(\sqrt{n}), \forall i \leq n$ .

# 5. THE RATE REDUCTION APPROACH

In this section we will analyze the rate reduction approach presented in [26] to solve the persistent congestion problem. We will consider a scenario similar to that analyzed in Section 4.3 in which a late-coming flow arrives to a bottleneck being fairly shared by n FAST flows. In a dynamic environment, when connections depart, the reduction in throughput causes the occupancy of router buffers to drop, giving a chance to remaining FAST connections to obtain a better estimate of their propagation delays. Thus, it is reasonable to assume that the new connections must deal with the presence of a number of existing flows aware of their true propagation delays.

The proposed solution consists in restraining transiently the transmission rate of a new flow by a given factor to allow router queues to get eventually empty, thus giving new FAST connections a chance to measure the true roundtrip propagation delay.<sup>¶</sup> Unfortunately, and despite of the reduction on its rate, the new connection is not always able to observe the empty queues. Note that, as the new flow drains queues by reducing its own rate, competing flows respond by increasing their rates. Hence, the new flow will only obtain the true propagation delay if queues empty

 $<sup>{}^{\</sup>S}$  Although one thousand sequentially started flows is unlikely, it helps to show the asymptotic behavior of  $\vec{a}$ .

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The authors argue that the rate scaling factor should be set to a value similar to the  $\alpha$  threshold.

before existing flows are aware of this event, that is, if the time required to empty the queues is less than the RTT of the existing flows.

The total backlog buffered at the core of the network in equilibrium,  $B^*$ , is the sum of the backlog buffered by all active flows:

$$B^* = \sum_{i=1}^{n+1} b_i^* = n\alpha + b_{n+1}^*, \qquad (16)$$

where  $b_i^*$  is the backlog buffered by flow *i* in equilibrium. Assuming that each flow  $i, \forall i \leq n$ , knows its true propagation delay, then these connections will each maintain  $\alpha$  packets in the router queues  $(b_i^* = \alpha)$ . On the other hand, the backlog buffered by the newly arrived flow satisfies

$$b_{n+1}^* = C(r_{n+1}^* - \hat{d}_{n+1}) = C \frac{\alpha}{x_{n+1}^*}.$$
 (17)

Substituting (13) and (15) into (17)

$$b_{n+1}^* = \alpha(1+a) = \frac{\alpha(1+\sqrt{1+4n})}{2}.$$
 (18)

This backlog will be drained from the queue at a rate equal to the bottleneck link capacity minus the sum of the transmission rates of all active flows. In the most favorable case, the new connection will completely pause its transmission  $(x_{n+1} = 0)$ . Considering the case in which all flows  $i, \forall i \leq n$ , share a similar propagation delay  $(d_i \approx d)$  and hence experience a similar RTT  $(r_i^* \approx r^*)$ , the fairness condition becomes

$$\frac{B^*}{C - \sum_{i=1}^n x_i^*} < r^* = d + \frac{B^*}{C}.$$
 (19)

Finally, substituting (12), (16) and (18) into eq. (19), it follows that

$$d > \frac{n\alpha \left(1 + \sqrt{1 + 4n}\right)}{2C} = \frac{nb_{n+1}^*}{C}.$$
 (20)

Thus, the rate reduction method is only effective when the round-trip propagation delay of competing flows exceeds the lower bound calculated in (20). Note that the lower bound scales as  $O(n^{3/2})$  with the number of active flows.

### 6. OUR SOLUTION

In this section we will present a solution to the persistent congestion problem that lacks the rate reduction method

Copyright © 2010 John Wiley & Sons, Ltd. Prepared using ettauth.cls limitations. We noticed that, when the newly arriving flow stabilizes, it can indirectly obtain a good estimation of its actual round-trip propagation delay.

As we pointed in Section 4.3, the new flow overestimates its propagation delay as  $\hat{d}_{n+1} = d_{n+1} + \sum_l t_{tx_l} + n\alpha/C$ . That is, there is an error  $e = n \alpha/C$ . Therefore, provided the new flow knows (or accurately guesses) n and the bottleneck link capacity, C, a more precise estimate of the propagation delay could be calculated as

$$\hat{d}'_{n+1} = \hat{d}_{n+1} - \hat{e}, \text{ with } \hat{e} = \hat{n} \frac{\alpha}{\hat{C}},$$
 (21)

where  $\hat{n}$  and  $\hat{C}$  are the inferred values of n and C, respectively.

In order to obtain good values for  $\hat{n}$  and  $\hat{C}$  it suffices to induce short variations in the throughput of the late coming flow and measure the changes it produces in queueing delay. Let  $r_{n+1}^*$  be the RTT of the newest flow once it reaches a stable throughput. If this connection modifies its transmission rate, for instance by changing the value of the window size,  $w_{n+1}^{\epsilon} = (1 - \theta)w_{n+1}^*$  with  $\theta < 1$  for a brief time  $t_{\epsilon}$  it will measure a new round trip time  $r_{n+1}^{\epsilon}$ after this time.<sup>||</sup> Let  $\Delta r_{n+1} = r_{n+1}^{\epsilon} - r_{n+1}^*$ . Under such circumstances

$$C\Delta r_{n+1} = \left(C - \sum_{i=1}^{n} \frac{w_i^*}{r_i^*} - (1-\theta) \frac{w_{n+1}^*}{r_{n+1}^*}\right) t_{\epsilon}, \quad (22)$$

as long as  $t_{\epsilon}$  is short enough so that the transmission rate of the first *n* sources remains constant. For this it is enough to make  $t_{\epsilon}$  of the same order as  $r_{i}^{*}$ .

A estimation of  $\hat{n}$  can be directly obtained substituting (12), (13) and (15) into (22) and using the fact that  $x = \frac{w}{r}$ . Solving for  $\hat{n}$ , we reach

$$\hat{n} = \frac{\theta t_{\epsilon}}{\Delta r_{n+1}} \left( \frac{\theta t_{\epsilon}}{\Delta r_{n+1}} - 1 \right).$$
(23)

Once we have  $\hat{n}$  it is trivial to obtain  $\hat{C}$  using (13) and (15), obtaining

$$\hat{C} = \frac{\left(1 + \sqrt{1 + 4n}\right)w_{n+1}}{2r_{n+1}}.$$
(24)

The proposed adjustment of the round-trip propagation delay suffices to solve the persistent congestion problem

Note that using positive values for  $\Theta$  can cause the queues to deplete before the time  $t_{\epsilon}$  is over, thus rendering the following analysis inaccurate. This can be avoided using small negative values for  $\Theta$  causing the queueing delay to increase. Although this can lead to packet drops in insufficiently dimensioned routers, this situation is easily detected and avoided using smaller values for  $\Theta$  in posterior measures.



Figure 4. Network topology used in the simulation experiments.

since transmission rates of competing flows will eventually converge to their expected values. As the former method, our proposal can be applied without the need of any network support but, in this case, the propagation delay of competing flows does not affect their behavior.

Additionally, the estimates obtained with our proposal can be used to reduce convergence time by computing an optimal value for the congestion window of the newly arrived flow. Recall that, when these estimates are obtained, the new flow is enjoying more bandwidth than its fair share, so it should reduce its congestion window to a more suitable value as well. Ideally, all the flows must maintain  $\alpha$  packets buffered into the core of the network. In such circumstances, the RTT experienced by the new flow should be  $r'_{n+1} = d'_{n+1} + (n+1)\alpha/C$ . Therefore, since w = rx, the value that should be assigned to the congestion window is

$$w'_{n+1} = \frac{\alpha r'_{n+1}}{r'_{n+1} - d'_{n+1}} = \alpha + \frac{d'_{n+1}\hat{C}}{\hat{n}+1}.$$
 (25)

### 7. EXPERIMENTAL VALIDATION

In this section we present the results of two series of experiments. The first group tests the validity of the model of the persistent congestion problem presented in Section 4. The second group verifies the appropriateness of our solution to the persistent congestion problem, available for download at [41], when compared with the rate reduction approach [26] and with the original FAST protocol.

All the experiments were simulated with version 2.31 of ns-2 [31] and, unless otherwise noted are based on the scenario depicted in Fig. 4 with little variations explained in each experiment. We establish a FAST-TCP connection between each source  $S_i$  and its corresponding destination  $D_i$ . Unless otherwise noted, each connection is configured with  $\alpha = 50$  packets and all packets carry a payload of 1 000 bytes. Because in this paper we are only concerned with the congestion avoidance characteristics of FAST, buffer sizes are big enough to hold all the packets enqueued by the flows, as predicted by eq. (11).

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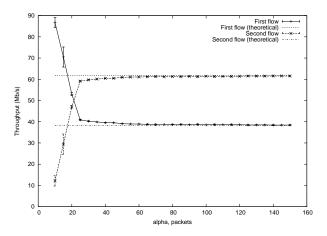


Figure 5. Average throughput obtained by both flows for different values of  $\alpha$ . Error bars show a 95 % confidence interval.

#### 7.1. Analytical Model

The following experiments are all designed to validate our modeling of the persistent congestion problem.

7.1.1. Two flows scenario We start with the simple twoflows scenario and validate the claims presented in Section 4.1, namely that the unfairness is independent on both FAST configuration and network characteristics. To this end we simulate the previously described test network both under ideal conditions, i.e., with no background traffic, and in a noisy environment.

Fig. 5 plots the throughput obtained by two FAST flows started sequentially when we vary the value of  $\alpha$ .\*\* Flows were started with a 2 seconds gap to ensure the first flow have had plenty of time to achieve its steady state and an uniformly distributed random time (between 0 and 1 second) to add some randomness. In this experiment we have run both flows for 10s and plotted the averaged throughput of both flows after 25 simulations with slightly different starting times. It can clearly be seen that the theoretical results hold for almost any value of  $\alpha$ . The deviations when  $\alpha < 30$  are caused by inherent instabilities in FAST when  $\alpha$  is too small for the network. With such small  $\alpha$  values FAST is unable to converge. The stability characteristics of FAST have been extensively studied in the literature, in fact [35, 36, 37] give sufficient conditions  $\alpha$  must meet to avoid this problem.

In order to test how the modeling behaves in more stringent scenarios, with non-100% FAST traffic, we have

<sup>\*\*</sup>In this and following figures, error bars correspond with a 95% confidence interval.

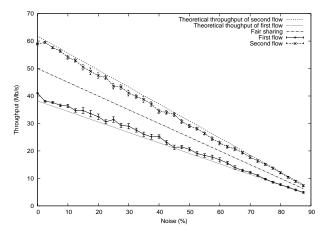


Figure 6. Average throughput of two sequentially started FAST flows in the presence of background traffic.

repeated the above experiment adding some background traffic to the bottleneck link. This background traffic was simulated with several Pareto traffic sources on top of UDP in a similar fashion as in [1]. Each Pareto flow was set up with a shape factor of 1.5, average burst and idle time of 100 ms and a peak rate of 1 Mb/s, thus consuming, on average, 0.5% of the bottleneck bandwidth.

We have run simulations for a different number of background flows (from none to two hundred flows) and represented the results in Fig. 6. It can be observed how, despite the high amount of noise, that reaches the full bandwidth of the bottleneck link, results match those predicted. That is, both flows share in an unfair manner the bandwidth not used by the noise.

7.1.2. FAST flows arriving sequentially This second set of experiments measures the impact of persistent congestion in a worst case scenario: flows arriving sequentially at a bottleneck link.

The first experiment compares the relative throughput obtained by each sequentially started flow under an all-FAST scenario and for a different number of total flows, from just two flows up to nine.

Fig. 7 shows both the predicted and the measured throughput. The measured throughput is the result of averaging the values obtained for different values of  $\alpha$  (between 40 and 60). For easier observation, the results corresponding to the same number of total flows are joined by a continuous line. That is, there are 8 lines, one for the two flows experiment, a second one joining the throughput of the flows in the three flows experiment, and so on. Each line has as many points as flows, each point representing

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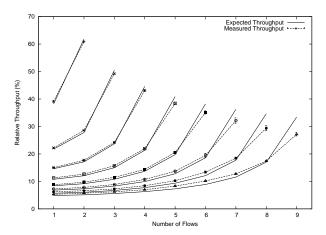


Figure 7. Measured throughput for different number of flows started sequentially. Each line joins the final relative throughput of each flow for each experiment (same number of total flows).

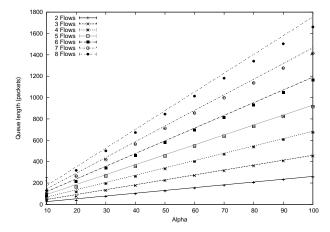


Figure 8. Measured queue length for different number of flows started sequentially. Continuous lines show the values as predicted by the model in Section 4.2.1.

the averaged throughput obtained by the i-th coming flow. The model produces very accurate predictions that match the values obtained by simulation. These results agree with those observed in other works where the asymmetry in throughput among Vegas flows was first pointed out [19, 21].

Persistent congestion does not only have adverse effects on fairness, but transmission delay worsens as well, as buffer occupancy grows larger that expected. In fact, the results in Fig. 8 confirm that queue sizes grow much larger than  $\alpha$  times the number of flows. These results clearly show that the model predictions are quite accurate. The small differences in the plot are somewhat misleading: they

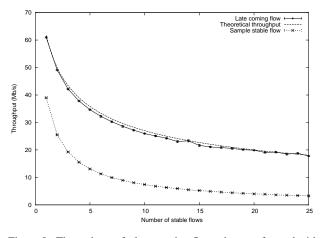


Figure 9. Throughput of a late coming flow when confronted with several FAST flows already getting their fair share.

are due to the fact that the model computes real values for the queue length, while the length obtained in the simulations comes expressed in integer units.

7.1.3. One flow arriving to a n flows stable scenario Finally we test the model for the scenario we are most interested in. This is the situation that happens in a network shared by FAST flows enhanced with some mechanism to correct persistent congestion. The experiment thus test the validity of eq. (15), that forms the basis for our solution to the persistent congestion problem presented in Section 6.

In order to obtain n flows with the proper estimation propagation delay without using yet our modification we let each one to run for some time in isolation to later restart the n flows simultaneously. After all n flows reach equilibrium, we start the late coming flow and measure the throughput of the latter and a representative flow from the initial set. We have employed a fixed packet size of 1 000 bytes and different values of  $\alpha$  for each simulation. The results, for different values of n, are plotted in Fig. 9.

### 7.2. Solutions

We have also conducted several simulation experiments to verify our claims regarding the rate reduction approach and validate our proposal. We have implemented both methods in the ns-2 simulator. We have employed the same network topology (Fig. 4) and configuration parameters than in the previous experiments.

7.2.1. Impact of the value of  $\theta$  A precise estimation of the number of flows is essential to remove the error introduced by persistent congestion on the measured propagation

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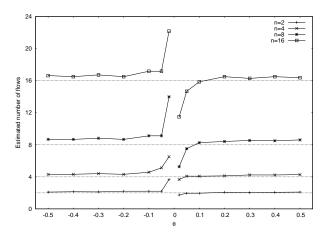


Figure 10. Impact of  $\theta$  on the estimation of the number of flows (n).

delay. The following experiment measures the accurateness of this estimation for different values of  $\theta$ .

The simulation is as follows. A set of FAST flows aware of their true propagation delay share a single bottleneck link. Once their (equal) rates stabilize, a new flow using our proposed measurement method starts. The simulation is repeated ten times varying slightly the starting time of the flows. Fig. 10 shows the number of existing flows estimated by the late coming flow using different values of  $\theta$  for different values of n, the number of existing flows.<sup>††</sup> The proposed method is unable to obtain reliable estimates when  $\theta$  gets too close to 0. However, variations from  $|\theta| = 0.1$  onwards avoid undesired deviations. In the following experiments, we have employed  $\theta = -0.5$  to acquire good estimates while preventing the bottleneck from getting empty at the same time.

7.2.2. Effectiveness of our proposal Firstly, we have simulated a scenario where five FAST connections are sharing the bottleneck link. The sources are started at intervals of 20 s each.<sup>‡‡</sup> Fig. 11(a) shows the instantaneous throughputs of the FAST connections when the original congestion avoidance mechanism is used. As expected, FAST strongly favors new sources and recent connections enjoy larger throughputs compared to old connections.

 $<sup>^{\</sup>dagger\dagger}Although 95\%$  confidence intervals have been calculated, they are not represented since they were consistently lower than  $\pm1\%$  and just cluttered the figure.

<sup>&</sup>lt;sup>‡‡</sup>Only consecutive arrivals were considered. Departures are a trivial case if we assume that the system converges to a fair share. When a flow leaves the network, the queue occupation eventually just diminishes in  $\alpha$  packets, and the situation is no different than that of n- flows already sharing fairly a bottleneck link, with n being the number or previous flows.

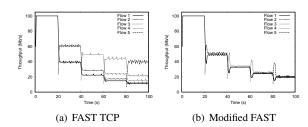


Figure 11. Throughput comparison.

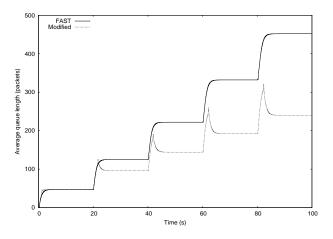


Figure 12. Core queue length comparison.

When our method is applied, this bias disappears and the network bandwidth is shared among competing FAST connections in a fair manner (Fig. 11(b)).

The average queue length at the bottleneck is also shown in Fig. 12. Due to persistent congestion, the amount of extra data introduced by FAST is larger than the targeted amount ( $\alpha$  packets per connection). However, our proposal keeps the proper amount of extra data into the network and thus the average queue size is smaller.

7.2.3. Impact of different round-trip propagation delays In the second experiment, we have examined the impact of different round-trip propagation delays on bandwidth distribution. We consider a number of existing FAST flows knowing their true propagation delays and, therefore, sharing the available bandwidth uniformly. Once their transmission rates have stabilized, a new flow starts its transmission. In order to study the effect of different propagation delays, the delay of the link between nodes  $R_1$  and  $R_2$  has been changed from 3 to 53 ms. To evaluate the fairness among the new and the existing connections,

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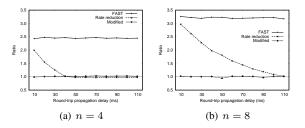


Figure 13. Impact of round trip propagation delay.

we used the following ratio:

Fairness Ratio = 
$$\frac{n\bar{x}_{n+1}}{\sum_{i=1}^{n}\bar{x}_i}$$
, (26)

where n is the number of existing flows,  $\bar{x}_{n+1}$  is the average transmission rate of the new flow and  $\bar{x}_i$  is the average transmission rate of existing flow  $i = 1, \ldots, n$ . Clearly, if the new connection obtains the same throughput as its competitors, the ratio will be 1. Fig. 13 compares the performance of original FAST-TCP, the rate reduction approach and our proposal for two different values of n. As expected, with FAST-TCP, the new connection obtains a higher throughput. With the rate reduction method, the bandwidth sharing depends on the experienced propagation delay. The minimum round-trip propagation delays required for the rate reduction approach to work properly as calculated using eq. (20) are 40.9 ms for n = 4 and 107.9 ms for n = 8. Graphs show how, as the propagation delay of existing flows falls below these thresholds, the bandwidth distribution becomes less fair. In contrast, with our proposal, fairness is preserved in all simulated scenarios.

7.2.4. Impact of the number of flows We have also compared our proposal to the rate reduction approach when the number of flow increases, while maintaining the rest of the simulation parameters fixed.

Fig. 14 shows that, while our solution manages to stay fair irrespectively of the number of flows, the rate reduction approach deviates from fairness and approximates original FAST behavior as the number of flows increases.

7.2.5. Impact of background traffic We end the validation section presenting a non-ideal scenario. For this we use a more realistic and stringent topology and noisy background traffic that interferes with our estimation method. Fig. 15 shows the topology employed, a variant of the classic parking-lot topology. The various bottlenecks are traversed by five flows running from nodes  $S_1, ..., S_5$  towards node

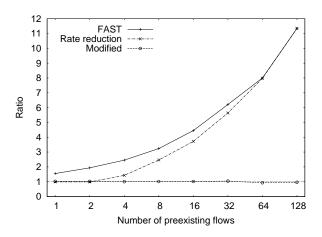


Figure 14. Impact of the number of preexisting flows.

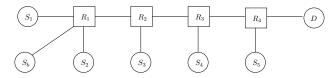


Figure 15. Parking lot topology with multiple bottlenecks. Every link has a 100 Mb/s capacity and a 5 ms propagation delay.

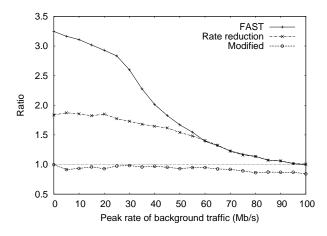


Figure 16. Impact of background traffic.

D. The flow from node  $S_1$  starts its transmission last, giving time to the previous flows to stabilize. Additionally, in a similar way as in [1], some background traffic was simulated via a Pareto flow  $(S_b, D)$  with a shape factor of 1.25, average burst and idle time of 100 ms and a peak rate ranging from 5 to 100 Mb/s. Fig. 16 shows the results. Not surprisingly, with both FAST and the rate reduction method, fairness improves as the peak rate of background traffic increases. The reason is that during active periods

Copyright © 2010 John Wiley & Sons, Ltd. *Prepared using ettauth.cls*  FAST flows reduce their rates as router queues fill up due to background traffic. When the background noise diminishes (idle periods) queues drain as the total FAST traffic is smaller than the link capacity and flows can seize a better estimate of their respective propagation delays before queues start to fill up again. In contrast, our solution deviates from absolute fairness when the background noise gets too high, because it interferes with our estimation method. However, it reaches a fairness index of 0.84 even with a peak noise level equal to the bottleneck bandwidth and obtains significantly better results than both original FAST and the Rate reduction method when there is less background traffic. This is a very good result if we keep in mind that our method was designed for the case when there is just FAST traffic in the network.

# 8. CONCLUSIONS

Taking as a starting point the FAST model in [1] we have established explicit formulæ that predict the throughput of a given DCA flow under persistent congestion conditions. We have found that assuming all flows have the same configuration parameters, the bandwidth share is independent of both their actual values and network configuration.

We have employed the aforementioned model to analyze the rate reduction approach, one of the most promising end-to-end proposals that try to deal with the persistent congestion problem. We have found that it does not work in every network configuration. In fact, we have provided necessary conditions a network must meet for the rate reduction approach to be useful.

Finally, we have presented an amendment to FAST that makes it immune to the persistent congestion problem. We used our analysis to give FAST senders the ability to discern persistent congestion and react accordingly. Our proposed solution outperforms previous approaches and does not need network modification to work. In fact, it is insensible to the values of propagation delay and bottleneck capacity.

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