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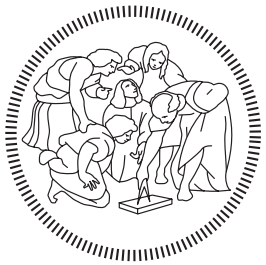

Design and experimental validation of a new bandwidth sharing scheme based on dynamic queue assignment

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Abstract

Even if the fair sharing of available resources in the network has been one of the basic principles since the Internet beginnings, it is still today not fully implemented. The fairness among traffic flows is mainly managed by the transport layer and the network does not have an active role but it just provides a Best Effort type of service without quality guarantees.

The objective of the work is to present a practical solution to the bandwidth sharing problem in a network node. Therefore, the current approaches were analyzed in order to propose an original and ready for use scheme. The new bandwidth sharing allocation engine (named UGUALE) works by assigning packets to the queues of a Strict Priority scheduler based on the measured rate offered by users.

The main idea is to prioritize well-behaving users in order to collaborate with the end-to-end congestion control mechanisms they implement at the transport or application layer. The allocation engine was implemented with the abstractions made available by the OpenFlow switch model and APIs, with the aim of proposing a solution amenable to be deployed in existing Software Defined Networking (SDN) architectures. Moreover, UGUALE is easy to configure since the only requested parameters are the guaranteed rates that users should obtain.

To validate our approach, a real testbed composed of five PCs was set up and several automation and monitoring scripts were written. When tested, the proposed allocation engine proved to be very effective in guaranteeing minimum rates and in the fair allocation of the free capacity, even when users have a different number of TCP connections or different Round Trip Times (RTTs). In conclusion, the promising results obtained by UGUALE makes us believe that such an approach is worth of further analysis.

Sommario

Sebbene l'equa condivisione delle risorse di rete sia stata uno dei principi fondamentali sin dagli inizi di Internet, ad oggi non è ancora stata completamente implementata. La ripartizione della banda tra i flussi di traffico è gestita principalmente dal livello di trasporto, pertanto la rete non ha un ruolo attivo ma fornisce semplicemente un tipo di servizio Best Effort senza garanzie di qualità.

L'obiettivo del lavoro è di presentare una soluzione pratica al problema della suddivisione della banda in un nodo della rete. Per questo motivo abbiamo analizzato le soluzioni attuali così da poter proporre uno schema innovativo e di facile adozione. Il nuovo schema di allocazione di banda (chiamato UGUALE) è basato sull'assegnamento dei pacchetti alle code di uno scheduler Strict Priority in base al tasso d'arrivo offerto dagli utenti.

L'idea si basa sulla prioritizzazione degli utenti che rispettano i propri limiti così da collaborare con i meccanismi di controllo di congestione implementati al livello di trasporto o applicativo. Lo schema di allocazione è stato implementato con le astrazioni rese disponibili dal modello di switch OpenFlow e dalle sue API, con lo scopo di proporre una soluzione facilmente utilizzabile nelle architetture Software Defined Networking (SDN) già esistenti. UGUALE è facile da configurare siccome l'unico parametro richiesto è la banda da garantire a ciascun utente.

Al fine di verificare la bontà della nostra soluzione, abbiamo configurato una rete composta da cinque computer, inoltre abbiamo scritto diversi programmi per automatizzare i test e per controllare in tempo reale la suddivisione della banda. Dai test emerge che UGUALE è efficace nel garantire la banda minima e nell'allocare equamente la banda eccedente, anche quando gli utenti hanno un numero diverso di connessioni TCP o dei Round Trip Time (RTT) differenti. In conclusione, visti i risultati promettenti ottenuti, siamo convinti che il funzionamento dello schema di allocazione proposto meriti ulteriori approfondimenti.

Chapter 1

Introduction

1.1 Context

Today's Internet is still facing the problem of congestion control, due to the increasing number of users and the convergence of many services on the global network. The architecture of the Internet is still too much Best Effort oriented, in the sense that routers rarely implement technologies to guarantee fairness between users and to maximize the network utilization at the same time. In the most common case users' flows share the same link served with a First In First Out (FIFO) queue, loosing any chance to apply fairness schemes able to guarantee a certain rate to every one. In fact connections that do not implement any end-to-end congestion control mechanism might end up monopolizing queues, finally preventing other users to access the link capacity. Most Internet traffic is TCP-compatible [2] but the end-to-end congestion control mechanisms apply only to separate flows. This means that a user can obtain more bandwidth by instantiating many connections to the same destination, thus violating the intrinsic TCP fairness at the user level. Hence end-to-end mechanisms are not enough to guarantee per-user fairness because they are not collaborative. Nevertheless telecommunication operators need to sell services on a per-user basis, for example charging more users who desire high bandwidth and low delay. Operators tend to solve the bandwidth sharing problem implementing different solutions that perform good enough only

if combined with over-provisioned networks. Summarizing, the standard schemes produce a coarse per-user fairness, oblige operators to stipulate contracts based on the maximum rate achievable by users and result in a low network efficiency from the point of view of the link utilization.

1.2 Objective

For the reason sustained in [5], we agree that fairness should be guaranteed and enforced between users and not between connection layer flows. A user is defined as an arbitrary aggregate of connection layer flows.

The objective of this work is to propose, implement and analyze a new forwarding engine able to guarantee a weighted fairness between users in a network node. The system, to be applied on real network switches, should produce some features like:

Scalability The ability to operate with a large number of users.

Flexibility The parameters of the system must be easily modifiable.

Stability The system should produce a stable output, so that rate oscillations are contained.

Performance The system should be able to manage an high total transmission rate.

Efficiency In presence of elastic traffic the aggregate throughput must be always the maximum allowed by the link.

1.3 Problem statement

The problem we want to solve is formally stated as follows. We have a set of users U routed on the same egress interface of a switch, whose link has capacity c. Every user $u \in U$ has a guaranteed rate g_u [bit/s] subject to the constraint

$$\sum_{u \in U} g_u \leq c$$

Let a_u and o_u be respectively the offered arrival rate and departure rate of the user u aggregated flows to the interested port of the switch. At a given instant we measure a_u for each user, then we can define N as the subset of U containing the users for which $a_u \leq g_u$. Analogously we can define E as the subset of U containing those users for which $a_u > g_u$. Finally, we define

$$f = c - \sum_{u \in N} a_u - \sum_{u \in E} g_u$$

as the amount of free capacity to be fairly shared. Thus, we want the system to enforce the following conditions for the evaluation of o_u:

$$o_u = \begin{cases} a_u & \forall u \in N \\ g_u + \frac{f}{|E|} & \forall u \in E \end{cases}$$

We will refer to the output rate of a user that satisfy the previous expression as the Optimal Fair Rate of a user (OFR_u). To highlight the efficiency condition we remark that

$$\sum_{u \in U} o_u = c$$

if there is at least a user with elastic traffic.

1.4 Summary

The bandwidth sharing problem defined in section 1.3 is faced in the rest of the document that is organized as follows. In chapter 2 the current standard solutions to bandwidth sharing are analyzed to understand their main drawbacks. In chapter 3 our approach to the problem is proposed and described in detail. In chapter 4 the implementation of the proposed scheme on a real testbed is illustrated, taking into considerations all the practical problems encountered and the solutions found. The results obtained from the emulations are analyzed and commented in chapter 5. Finally in chapter 6 the open issues are presented and solutions to solve them are outlined.

Chapter 2

State of the art

For the Internet Engineering Task Force (IETF), Quality of Service (QoS) is a term used to express a set of parameters and describes the overall performance offered by a network. These parameters are measurable and include bit rate, bit error rate, throughput, latency, delay, jitter and service availability [12]. Even though the term QoS appeared in the last years of the twentieth-century, today's Internet is still very similar to the classic one: all packets are treated in the same way, with the Best Effort (BE) paradigm, without providing many guarantees of QoS. However, the convergence of various services on the Internet and the consequent possibility to bill users in different ways, pushed the proposal of QoS strategies to differentiate traffic and serve it based on its requirements. Despite the variety of QoS parameters, the most used is the maximum rate that a user can get, just like in a standard DSL contract. So the first problem to face is the allocation of a certain bandwidth to every user. Today this allocation is coarsely obtained with a combination of end-to-end mechanism and technologies implemented in network nodes. Nevertheless these standard techniques are not enough to guarantee per-user fairness, at least if they are not used in a collaborative way inside a joint infrastructure. In sections 2.1,2.2 and 2.3 we analyze respectively end-to-end, scheduling and AQM solutions applied to bandwidth management. In section 2.4 we cite some architectures strictly related to our work that is finally sustained in section 2.5.

2.1 End-to-end congestion control

In IP networks, the most used transport protocols are TCP and UDP. User Datagram Protocol (UDP) is a connectionless protocol: the packets, called datagrams, are sent as soon as the application generates them. No congestion control is done at the connection layer, but it will be hopefully enforced at the application layer with a bigger time granularity.

On the other hand Transmission Control Protocol (TCP) is connection-oriented because it assures the correct transmission of data between two ports. TCP executes a congestion control mechanism: the transmission rate is automatically adjusted based on the acknowledgements packets (ACKs) received from the destination. Since packets are mostly lost in congested nodes, to avoid this situation and so further retransmissions, TCP tries to estimate the congestion status of the network and to send only as much as sustainable at a time. This mechanism provides a coarse per-flow fairness dependent on the characteristics of the flow, such as the Round Trip Time (RTT).

When many TCP flows competing for the same link are queued in the same First In Firt Out (FIFO) queue, there will be fairness between flows. This means that each flow on the long period will converge to the same sending rate. This is a well known phenomenon that however causes unfairness between users. In fact, a user that opens many parallel TCP connections will get a higher share of the link capacity with respect to a user with only one connection [30]. This observation is at the base of applications like download enhancers and torrent clients. Another factor that impairs the TCP fairness mechanism is the different Round Trip Time (RTT) between flows. In fact when TCP flows with different RTTs are enqueued in the same FIFO queue, they will experience throughputs inversely proportional to their RTT [16]. This effect becomes less evident if there are many users or if an AQM strategy is adopted, but it needs to be accounted in future solutions.

2.2 Scheduling

Scheduling is the act of choosing the transmission order of packets. Each switch's port is equipped with a certain scheduler that manages the transmission on the outgoing link. The simplest scheduling policy is First In First Out (FIFO): packets are enqueued in a single buffer and served in the same order of the arrivals. In a Best Effort (BE) architecture FIFO is the used policy, so users cannot be distinguished because all packets share the same queue. If the total incoming rate is greater than the serving rate, the FIFO policy becomes unfair. In fact, a user that offers too much traffic can monopolize the entire buffer thus preventing other users to enqueue packets. Moreover FIFO does not allow to prioritize some packets with respect to others, making impossible to implement QoS policies.

Indeed to guarantee QoS a more complex scheduler must be used, in particular a scheduler that utilizes various FIFO queues as base elements. The complexity of schedulers increases with the number of queues, because the queuing and dequeuing processes must consider a lot of information.

The queuing process is in charge of deciding to which queue assign the packet. This task is called classification and it is usually based on the data written in the IP packet header.

The dequeuing process is way more complicate because it takes into account the actual and the past occupation of queues. The serving policies of queues can be subdivided in two main categories: fair queuing and strict priority.

Fair Queuing (FQ) and its Weighted variant (WFQ) [18][3] serve queues cyclically. Queues are allowed to transmit a certain number of packets in each round. Since packets can have different sizes, FQ must take into account even this parameter, becoming very complex. The parameters for WFQ schedulers are the weights assigned to each queue. A weight represents the percentage of the total capacity that a queue should obtain. Said that, FQ can be viewed as a version of WFQ where all queues have the same weight.

In Strict priority schedulers each queue has a fixed priority. A packet of a queue with priority i will be served if and only if the queue of priority $i + 1$ is empty. Even though strict priority clearly risks to starve low-priority queues, this policy is fundamental to guarantee low delays to packets assigned to high priority queues.

However the classical approach in QoS networks is the mixed policy depicted in figure 2.1 [4]. In the mixed policy, an external strict priority scheduler serves a few queues and the output of a WFQ scheduler. Usually flows with low-rate and low-delay requirements such as VoIP are served with the maximum priority, then the capacity that they do not use is shared with a FQ policy.

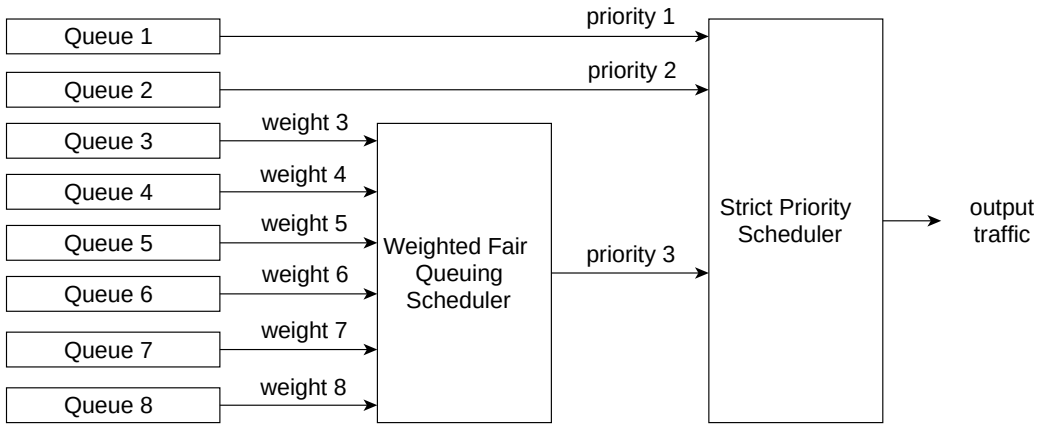

Figure 2.1: *Example of hierarchical scheduler.*

FQ schedulers constitute the text-book approach to bandwidth sharing because they are theoretically able to achieve the perfect short-time fairness between queues. To obtain per-user fairness with schedulers, each user must be sent on a different queue and served with a weight that reflects its requirements. Even if FQ is normally implemented with the Deficit Round Robin (DRR) mechanism [29], the complexity of FQ schedulers remains high and limits the number of available queues. Some schemes such as Stochastic Fair Queuing (SFQ) [17] try to speed-up the queuing operation and to reduce the number of queues by using hash functions. However, to reach a fairness comparable to FQ, SFQ still need from 1000

to 2000 queues [1].

We analyzed the QoS configuration manuals of commercial switches that some vendors make freely available for consultation. The models studied are:

- Arista - 7500 Series
- Brocade - MLX Series
- Cisco - CRS Router
- Extreme Networks - BlackDiamond Series
- Fujitsu - XG2600
- HP - FlexFabric 7900 Switch Series
- Huawei - Quidway S300 Terabit Routing Switch
- Juniper - T-series
- Mellanox
- Pica8 - PicaOS

It clearly emerged that every switch has a scheduler for every output port. These schedulers are configurable and they can always implement priority, round robin and mixed policies. Priority policies are the pure Strict Priority and Rate Limited versions. Rate Limited is a Strict Priority scheduler where each queue has a maximum rate parameter. If a queue exceeds its maximum rate, the lower priority queue is served. Round robin policies such as Weighted Round Robin, Weighted Fair Queuing and Deficit Round Robin are different only for the implementation and for the configuration parameters. Finally mixed policies allow to subdivide queues in subgroups and to serve them in a nested way, as illustrated in figure 2.1.

Nevertheless, the most important observation is that the number of queues is always fixed to 8. The classic solution is to group flows and to let them share queues, obtaining only a coarse graded fairness. For example architectures like Differentiated Services (DiffServ) [28] aggregate traffic in classes based on the value of the Type of Service (ToS) and serves queues with arbitrary weights. The ToS is a field in the IP header and its value depends from the the application that generates packets. The weights used in DiffServ are based on the operator experience and they do not guarantee for sure the fairness between users. In fact, in order to provide enough bandwidth to each class, operators tend to rate-limit or delay packets at the ingress nodes and to over-provision the networks. These operations produce a coarse fairness, a low network efficiency and oblige operators to stipulate contracts based on the maximum rate reachable by users.

The complexity of schedulers makes impossible to have a queue for each user, so the bandwidth sharing problem cannot be solved only with schedulers. This observation triggered the development of Active Queue Management techniques, nevertheless recently proposed architectures like Flow Aware Networking (FAN) [21] are still based on variants of per-flow FQ. FAN's authors motivate their work saying that Internet needs a global rethinking that should consider congestion control and QoS as built-in features. This opinion is supported also in [14], but since the Internet architecture will hardly change in the next future, a solution that do not involve more than 8 queues is still needed.

2.3 Active Queue Management

Another class of router algorithms for congestion control is Active Queue Management (AQM). The primary goal of AQM techniques is to maintain the queue occupancy low while allowing bursts of packets. Since AQM techniques were thought as alternatives to complicated scheduling algorithms, they are normally applied on single FIFO queues. AQM algorithms have been well known for two

decades and their effectiveness have been proved, but their adoption is still limited because of the difficulty of carefully tuning their parameters [19]

The reference AQM scheme used in routers is Random Early Drop (RED) that simply discards packets with a probability dependent on the estimated queue occupancy. RED has only a few parameters for tuning the dropping reactivity and the target number of packets in the controlled queue. Since these parameters do not map directly to flow rates, they are still considered a complicated setting. Even if a self-configuring RED that automatically reacts to the average queue occupancy was proposed [33], its adoption is still very limited.

RED adapts well to the AQM's main objective since queues are maintained short and so latency, jitter and flows synchronization are reduced. Nevertheless RED does not constitute a solution to fairly allocate bandwidth. In fact RED is based on two wrong assumption [6]:

1. All flows adapt their sending rate.

2. Dropping packets randomly will affect more high rate flows.

The first assumption is clearly unrealistic because some protocols are non-adaptive. These protocols do not slow down when they encounter drops since they do not include congestion avoidance mechanisms. Furthermore there are protocols that implement congestion avoidance algorithms too much robust to dropping events.

The second assumption is wrong because it is based on the first one. In fact, non-adaptive connections will keep the dropping probability of the system high. As a consequence, on a long time scale every flow will see the same dropping probability and even low-bandwidth flows will be prevented to grow.

To solve this unfairness problem, many variants implementing the differential dropping idea were proposed. Differential dropping consists in recognizing high-rate flows in order to penalize them. The penalization is made by dropping more high-rate flows with respect to low-rate flows. Differential dropping schemes vary

basically for how high-rate flows are recognized. These algorithms obtain a different degree of fairness based on the amount of per-flow information maintained: they show a trade-off between fairness and complexity.

The first well-known proposal was Fair Random Early Drop (FRED) [6] that calculates different dropping probabilities for flows based on their queue occupancy and on their history in terms of fairness violations. In other words, FRED estimates a flow rate by counting the number of enqueued packets belonging to that flow. Since FRED executes per-flow operations, it requires too many states. For this reason the authors of [11] proposed Core Stateless Fair Queuing (CSFQ). In CSFQ edge routers estimate flow rates and write this information in an extra-label of packets' headers. Then core routers apply a probabilistic dropping based on extra-labels and on the routers' measurement of the aggregated traffic. While performing good and maintaining the core-routers' complexity low, CSFQ is not practical because it requires to change all the network's routers. In fact, CSFQ involves non standard routers' operations and a new IP header field.

CHOose and Keep for responsive flows, CHOose and Keep for unresponsive flows (CHOKe) [26] constitutes a sort of stateless pre-filter for RED. When a packet arrives, another packet is drawn in a random position from the buffer. If the two packets belong to the same flow, a strike is declared and both packets are dropped. Statistically, packets of high-rate flows are more likely to be picked or to trigger a comparison, so they should be dropped more frequently. CHOKe is very simple but it does not perform well if there are many flows and it achieves limited performance with high bandwidth unresponsive flows [9]. Moreover CHOKe implies non-standard operations such as randomly choosing packets from a FIFO and eliminating packets of a queue.

The first CHOKe's variant XCHOKe [22] maintains a list of flows that incurred in strikes, while RECHOKe [31] eliminates the FIFO random draw by adding another data structure. These CHOKe variants obtain good performance

in detecting and controlling aggressive flows. Unfortunately they are stateful: lookup tables are maintained to categorize active flows.

Other technologies such as Approximate Fair Dropping (AFD)[25], RED with Preferential Dropping (RED-PD) [24] and BLACK (BLACKlisting unresponsive flows) [9] proposed different solutions to estimate more accurately arrival rates and fair rates. Moreover they keep the amount of data required low by maintaining only information for flows suspected to be high-rate. AFD, RED-PD and BLACK perform better than the simple RED, but they need extra features and they add extra configuration problems to network administrators. For these reasons they are still rarely utilized in todays networks.

At the end, tail-drop is still the most widely deployed (non) queue management scheme.

2.4 Dynamic assignment of queues

We have seen that per-user scheduling and AQM techniques, if used alone, are not precise and scalable solutions to per-user fairness. Hence we looked for QoS architectures different from the classical DiffServ and IntServ that can solve our problem.

The authors of [27] propose a feedback distribution architecture that provides per-flow QoS in a scalable way. In this architecture, the rate of the traffic received by a user is measured in the access network and communicated to markers placed in proximity of the sending server. Subsequent packets of the same flow are marked by the server into a few categories based on the feedback measurement. Core-routers act as droppers by sending packets on different strict priority queues based on packets' mark and protocol. This proposal is very interesting because it does not drop out-of-profile packets, but it simply puts them on different priority queues. Therefore, once in the network, low-priority packets will be served only if there is spare capacity. Unfortunately the involved elements — meters,

droppers and markers — are spread in different points of the network and so this architecture does not constitute a one-link solution to our problem.

Approximate Flow-Aware Networking (AFAN) is a network architecture based on FAN [13], so it implements admission control and router cross-protect mechanisms. AFAN recognizes the drawbacks of per-flow scheduling and tries to solve them using only two queues served by a strict priority scheduler. One queue is used for priority flows and the other one for elastic flows. More precisely, the first queue is reserved to packets belonging to well-behaving flows. Well-behaving flows are defined as flows having less than Q bytes in the buffer. The other queue is subject to a mechanism very similar to CHOke: the arriving packet and a random packet from the elastic queue are compared. If the two packets belong to the same flow, the picked packet is dropped while the one arriving is dropped with a certain probability. AFAN is more compact than the architecture presented in [27] because the assignment to queues is done based on measurements performed by the switches. AFAN confirmed the idea of doing queue management and rate control with strict priority schedulers.

2.5 Analysis

As showed in previous sections, per-flow FQ has complexity issues while active queue management requires extra router features or non-intuitive configurations. We tried to overcome these problems by taking inspiration from the schemes presented in section 2.4. Nevertheless our proposal aims to an immediate acceptance in todays networks, so we tackled one by one the problems of the macro-categories previously exposed.

The proposed solution uses only features already available in switches and considers their actual limits. More specifically it involves only rate metering processes, strict priority schedulers and a small number of queues. The used elements are present in the OpenFlow's abstractions, so we are very confident that they are

widely diffused in todays switches. In fact, since the main objective of SDN and OpenFlow is to provide commands able to manage different data planes, the abstractions given should conform to the vast majority of switches.

Our solution is easy to configure because the only settings required are the minimum rates that should be guaranteed to users. Indeed there is a mapping one to one between the desired effect of the system and its parameters, so the configuration is really intuitive and agnostic to its implementation.

Chapter 3

Approach

This chapter describes our proposal to solve the problem of bandwidth sharing formulated in chapter 1. The proposed forwarding engine is called UGUALE, that stands for User GUaranteed ALlocation Engine. The main idea of UGUALE is described in section 3.1 while section 3.2 shows in detail the pipeline that packets encounter in a switch. Section 3.3 proposes some rules to set up the system parameters. If more information about users is available, the system can be enhanced with the methods presented in section 3.4.

3.1 Main idea

To understand the main idea of UGUALE, it is important to keep in mind three key facts:

1. A Strict Priority (SP) scheduler serves packets of a queue only if all the queues with higher priority are empty. We assume that the queue with the highest priority is denoted as queue 1 or first queue.

2. A user is defined as an arbitrary aggregate of connection layer flows. E.g. a user is identified by all the TCP flows between two IP addresses.

3. Elastic users adapt their sending rate to the rate permitted by the network based on the end-to-end congestion control algorithm they implement.

The notation used in this chapter is based on the notation introduced in chapter 1. N is the subset of users whose offered rate is lower or equal to their guaranteed rate, while E is the subset of users whose offered rate is greater than their offered rate. We will indicate users $\in N$ as low-rate users and users $\in E$ as high-rate users.

The idea behind UGUALE is simple: to send packets on different priority queues based on the measured offered rate. For the sake of simplicity, the system is firstly explained when the available SP scheduler has only 2 queues and only later extended to more queues. The set of scheduler's queues is denoted by $Q = 1..|Q|$.

3.1.1 UGUALE with two queues

With $|Q| = 2$ queues, the proposed mechanism has only two rules:

- Packets of users whose measured input rate a_u is less or equal to g_u will be enqueued in the first queue.
- Packets of users whose measured input rate a_u is greater than g_u will be enqueued in the second queue.

This dynamic assignment of packets to queues is enough to collaborate with the congestion control mechanisms implemented by elastic users and to stabilize the transmission rates around the optimal fair rates. In fact, when some users are requesting less than their g_u, there will be some unused bandwidth. In this situation, elastic users will try to increase their sending rate to use all the available bandwidth. When elastic users' offered rate becomes bigger than their guaranteed rates, their packets begin to be enqueued in the second queue. In queue 2, elastic users can compete for bandwidth without stealing resources from the well-behaving users of the first queue. Indeed the bandwidth available in the second queue is the bandwidth not used by the first queue. So if low-rate users begin to offer more traffic, high-rate users will have less bandwidth. If the service rates of users in the second queue begin to decrease, they will reduce their offered rates.

Finally, when the offered rates of elastic users return within their guaranteed rates, these users will be re-admitted in the first queue. An example of the forwarding engine with two queues and three users is shown in figure 3.1

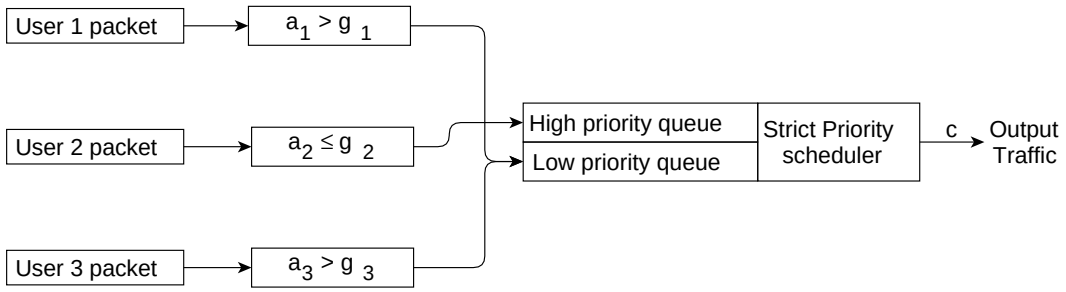

Figure 3.1: *Example of the UGUALE scheme with $|Q| = 2$ queues and $|U| = 3$ users.*

With this mechanism, every user will always be able to reach its guaranteed rate and the throughput of the link is maximized. UGUALE with two queues is enough to assure guaranteed rates to an arbitrary number of users, but it does not enforce fairness in the distribution of excess bandwidth. To reach better levels of fairness and stability, UGUALE can operate with more than two queues.

3.1.2 UGUALE with more queues

As previously mentioned, the UGUALE scheme can be extended to more queues, so that the more a user increases its rate, the less priority it will get.

For every user u, up to $|Q|$ thresholds can be defined. Thresholds are indicated by $t_{u,q}$, where $u \in U$ and $q \in Q$. For every user, the first threshold corresponds to the guaranteed rate and the last threshold corresponds to the link capacity. The other thresholds are spaced between the first and the last one, as explained in section 3.3.

The new enqueueing rule is the following: *when user u offers a rate a_u between thresholds $t_{u,q-1}$ and $t_{u,q}$, its packets are enqueued in queue q.*

Since there are many thresholds, from now on it is more practical to talk about bands. Bands are interval of rates between two consecutive thresholds. Band 1

includes rates lower than the first threshold, band 2 contains rates between the first and the second threshold and so on. Also bands are denoted by $b_{u,q}$, where $u \in U$ and $q \in Q$. Now the enqueueing rule can be expressed as follows: *when user u offers a rate a_u falling in band $b_{u,q}$, its packets will be enqueued in queue q.*

Intuitively the subdivision in more than two bands allows a fairer distribution of the excess bandwidth. An example of the scheme with many queues is illustrated in figure 3.2.

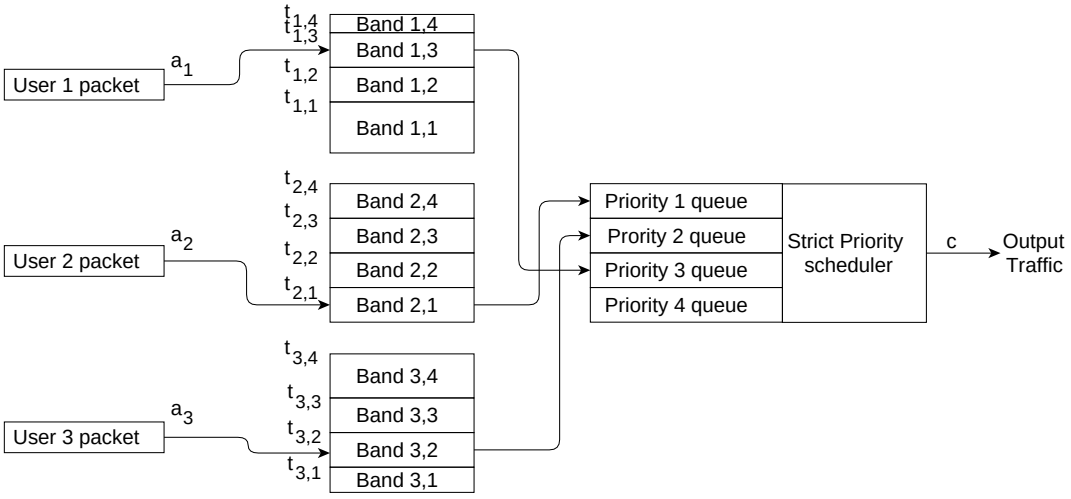

Figure 3.2: *Example of the UGUALE scheme with $|Q| = 4$ queues and $|U| = 3$ users.*

Since the objective of the project is to guarantee fairness between users, connection-layer flows belonging to the same user are aggregated before being measured and enqueued. Nevertheless each connection-layer flow implements its own end-to-end congestion control algorithm. Hence single flows will compete in queues, re-enabling the TCP per-flow fairness in the subdivision of buffers' capacity. Fortunately the per-flow contention of bandwidth will take place only in the last used priority queue, therefore the per-flow effect is limited to the amount of bandwidth available in the last used queue. As result, the per-flow fairness can be mitigated by reducing the bandwidth available to queues and so by increasing the number of queues.

3.2 Pipeline

A packet arriving to a switch implementing UGUALE will first follow the normal routing procedure, then it will encounter the UGUALE's processing pipeline represented in figure 3.3.

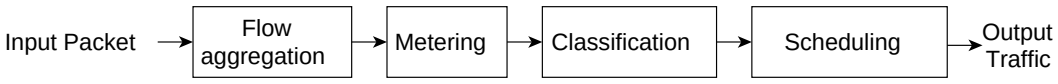

Figure 3.3: *UGUALE's pipeline*

The stages depicted in the figure are:

- Flow Aggregation: flows belonging to the same user are aggregated and treated as a single macro-flow. For example, the most likely aggregation will be based on the source IP address.
- Metering: the rate of the aggregated flow is updated considering the new packet. Then the meter compares the measured rate with a set of thresholds and decides in which band the rate falls. Finally the meter communicates its decision to the next element.
- Classification: the classifier receives the packet and the information to classify it. Then it sends the packet to the priority queue corresponding to the band communicated by the meter.
- Scheduling: the packet is served by a Strict Priority scheduler.

The pipeline enriched with more details is depicted in figure 3.4, whose steps are thoroughly explained and motivated in the following subsections.

3.2.1 Flow Aggregation

To implement per-user operations, all packets belonging to the same user must be treated in the same way. For this reason, packets of different connection-layer

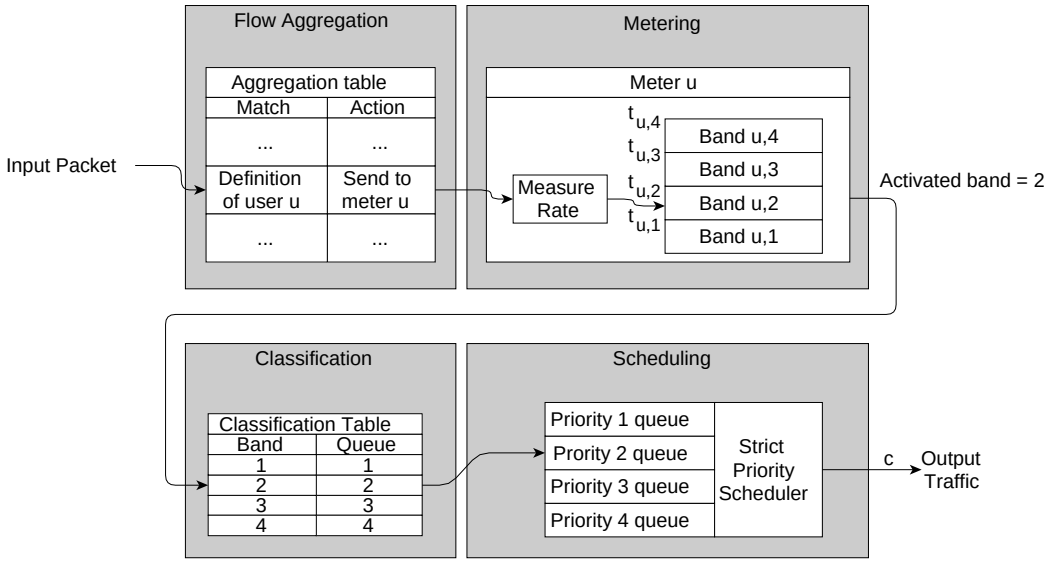

Figure 3.4: *UGUALE's detailed pipeline*

flows must be aggregated as a single user. Practically, a user must be defined by a set of rules related to the IP packet header: for instance a user can be identified by a particular source IP address or by the couple source IP address and destination TCP port. Therefore the matching fields of the aggregation rules define users and to each rule corresponds an action to be applied on matching packets. With UGUALE, the action is sending the packet to a user-dedicated meter. For example, an aggregation table such as table 3.1 identifies users by the source IP address and applies a user-specific action.

Match Rule	Action
SRC IP = 192.168.1.1	Send to meter 1
SRC IP = 192.168.1.2	Send to meter 2
...	...
SRC IP = 192.168.1.N	Send to meter N

Table 3.1: *Example of aggregation table for a system in which users are identified by the source IP address. All packets belonging to the same user are sent to a user-dedicated meter.*

3.2.2 Metering

A meter is a switch element that can take actions based on the measured rate of packets passing through it. In our scheme, a meter has a set of $|Q|$ bands defined by $|Q|$ thresholds and each band has an associated action. These actions are applied when the corresponding bands are activated, i.e. when the measured rate falls in that band. Since bands are disjoint, a packet can activate only a band.

The meter receives a packet and updates the measured rate. Then the meter decides to which band the rate belongs to and applies the corresponding action on the packet: in our case the action is to communicate to the classifier the band activated by the packet. This communication can be done by associating some meta-data to the packet.

In our system there is a univocal correspondence between users and meters. In fact a meter maintains information on the corresponding user, such as the offered rate, the thresholds and the action. For this reason, UGUALE can manage a number of users corresponding to the number of meters available in the switch.

3.2.3 Classification

Classification is the act of deciding to which queue a packet should be sent. Generally, the classification is done based on a packet's header field, but in our system it is based on the meta-data written by the meter. So at the end of the pipeline, the packet will be sent on a certain port of the switch. Every port will have a multi-queue SP scheduler and a classifier to decide in which queue to send the packets. The information of the meter band is retrieved by the classifier that finally puts the packet on a certain queue. To keep the system simple, the classifier executes a one to one mapping between bands and queues, so that band q will correspond to queue q.

3.2.4 Scheduling

Packets are served by a Strict Priority (SP) scheduler. Since no Active Queue Management (AQM) is implemented, packets can experience tail drop: a packet sent to a queue temporarily full will be dropped.

3.3 Thresholds setting

Thresholds must be set in a way that enforces guaranteed rates and fairness. The rules to distribute bands are elucidated in this section.

3.3.1 First threshold

Since a SP scheduler is used, the only queue that is always able to transmit is the first one. Moreover, rates lower than the guaranteed ones must always be assured. To be enqueued in the first queue a user's rate must remain within its first threshold, so the first threshold corresponds to the guaranteed rate.

3.3.2 Other thresholds

The other thresholds must be properly set to obtain a fair sharing of the excess bandwidth. Following the line of thinking of subsection 3.1.2, the dimension of the bands must be minimized to enhance the control of rates. Since the subset of bands effectively used in a certain moment is unknown, all bands should be kept as short as possible. To achieve this principle we chose a uniform distribution: all thresholds must be equally spaced.

The easiest solution is to assign the first threshold and then to divide the non-guaranteed bandwidth in $|Q| - 1$ equal slices, as depicted in figure 3.5. The non-guaranteed bandwidth of user u is defined as

$$f_u = c - g_u \quad \forall u \in U$$

The figure highlights the fact that users will have bands of different width based

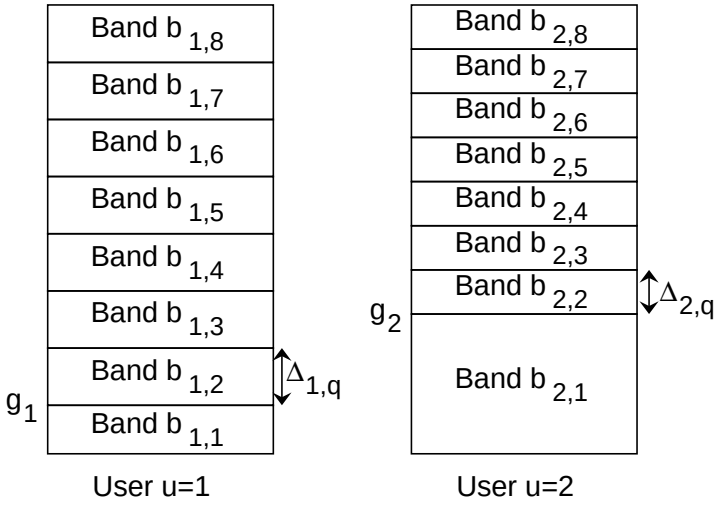

Figure 3.5: *A possible assignment of bands with $|Q| = 8$ available queues. Bands are obtained by dividing the non-guaranteed bandwidth in $|Q| - 1$ regular intervals.*

on their guaranteed rates. More precisely,

$$g_1 < g_2 \implies \Delta_{1,q} > \Delta_{2,q} \quad 2 \leq q \leq |Q|$$

Since the bandwidth obtained in a queue depends on the band's width, this difference in width causes a different subdivision of the excess bandwidth. In the example of figure 3.5, user $u = 1$ will obtain a bigger share of the excess bandwidth with respect to user $u = 2$. Nevertheless thanks to this observation, different policies of division of the excess bandwidth can be implemented by adjusting thresholds.

Hence, in order to obtain a fair division of the excess bandwidth, all bands $b_{u,q} \quad \forall u \in U, 2 \leq q \leq |Q|$ should have the same width: another bands assignment policy is needed. If $|Q|$ queues are available, a common width can be

$$\Delta_{\text{ref}} = \frac{c}{|Q|}$$

Thresholds are then assigned by stacking many Δ_{ref} on top of the first threshold. By doing that, the sum of bands may be different from c, so the last band is left flexible as depicted in figure 3.6.

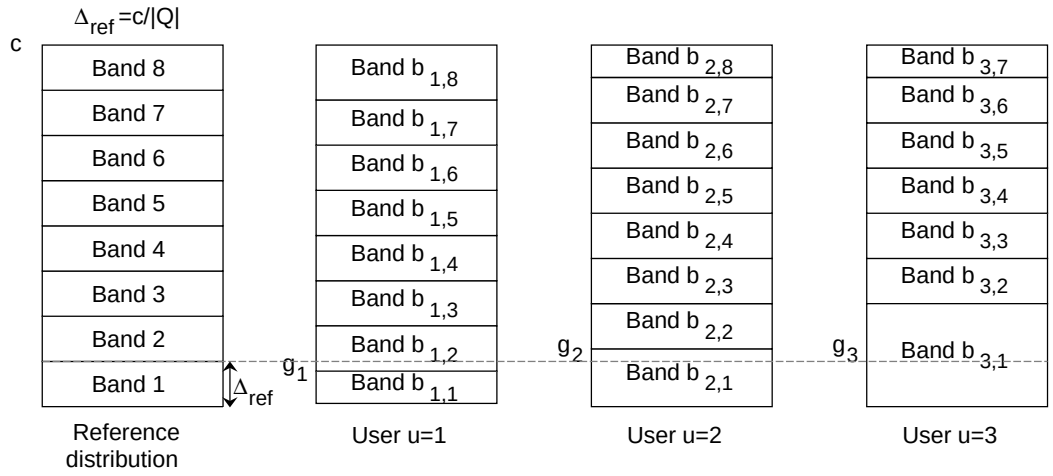

Figure 3.6: *Assignment of bands with $|Q| = 8$ available queues. Many Δ_{ref} are stacked on top of the guaranteed rate and the last fitting band is adapted to the capacity.*

If $g_u < \Delta_{\text{ref}}$ the sum of bands will be smaller than c:

$$g_u < \Delta_{\text{ref}} \implies g_u + (|Q| - 1) \cdot \Delta_{\text{ref}} < c$$

In this case — user $u = 1$ in figure 3.6 — it will be necessary to enlarge the last band. On the contrary, if $g_u > \Delta_{\text{ref}}$ there will be a band that does not fit in the capacity:

$$g_u > \Delta_{\text{ref}} \implies g_u + (|Q| - 1) \cdot \Delta_{\text{ref}} > c$$

In this case — users $u = 2, 3$ in figure 3.6 — the last stacked band will be squeezed to fit c. If the last fitting band's index is smaller than $|Q|$, the following bands will not be used nor assigned for that user. The index of the last band assigned to user u is denoted as $q_{u,\max} \leq |Q|$.

To resume, bands of users are assigned such as

$$b_{u,q} = \begin{cases} g_u & q = 1 \\ \Delta_{\text{ref}} & 2 \leq q < q_{u,\max} \\ c - \sum_{1 \leq q < q_{u,\max}} b_{u,q} & q = q_{u,\max} \end{cases} \quad \forall u \in U$$

3.4 Enhancements

The previously exposed policy to set thresholds assumes only the knowledge of users' guaranteed rates. If more information is available, the control on users' rates can be enhanced using the methods presented in this section.

3.4.1 Ad hoc thresholds

If offered rates are known, the Optimal Fair Rate of users can be calculated as done in subsection 1.2:

$$\mathrm{OFR}_u = \begin{cases} a_u & \forall u \in N \\ g_u + \frac{f}{|E|} & \forall u \in E \end{cases}$$

where

$$f = c - \sum_{u \in N} a_u - \sum_{u \in E} g_u$$

To obtain fairness in this situation it is enough to let the first threshold of user u correspond to its Optimal Fair Rate OFR_u as depicted in figure 3.7.

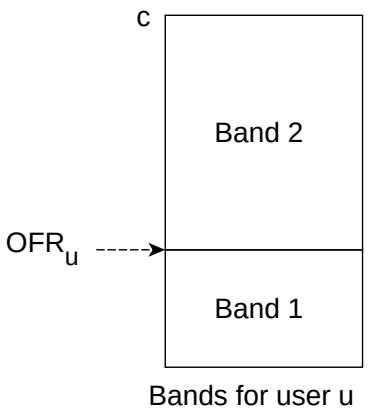

Figure 3.7: *When the OFR of a user is known, a single threshold in the OFR is enough to obtain fairness.*

To improve the stability of the system, other bands can be placed around the OFR threshold, as depicted in figure 3.8. Since bands are placed only in the used

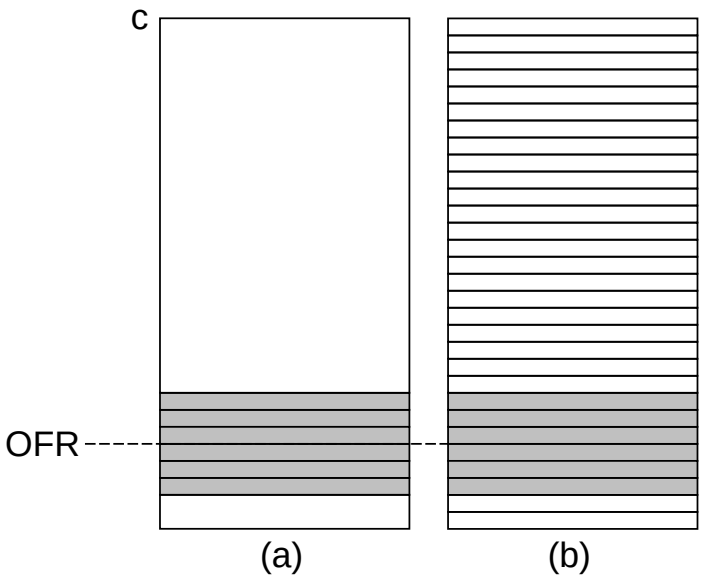

Figure 3.8: *When the OFR is known, the assignment of bands (a) can emulate the presence of much more bands (b) because the rates will oscillate only in a small interval.*

rates interval, this assignment emulates the presence of much more than $|Q|$ bands as pointed out in figure 3.8.

Even if this policy might assure the perfect fairness, the system would require a mechanism that continuously change thresholds based on users' offered rates: this proposal seems unpractical and not scalable, so it will be tested only for theoretical reasons.

3.4.2 Maximum Marking Rate

The UGUALE's enhancement presented in this subsection assumes the knowledge of the maximum Optimal Fair Rate. The maximum OFR of the system is:

$$\mathrm{OFR}_{\max} = \max_{u \in U} \mathrm{OFR}_u$$

To obtain its value, offered rates must be known, but the solution proposed does not require continuous adjustments of thresholds. If the $\mathrm{OFR}_{\max}$ is very low with respect to the link capacity, why bands should be distributed to cover all

the link capacity? If the $(|Q|-1)^{th}$ reference threshold is reduced to a common value slightly greater than all users' rates, the Δ_{ref} is reduced, too. That means enhancing the control on every user's rate while maintaining the fairness in the division of the free bandwidth. The new value of the $(|Q|-1)^{th}$ reference threshold is indicated as the Maximum Marking Rate (MMR) of the system. The MMR should be large enough to contain the rate of the user having $\text{OFR}_u = \text{OFR}_{\max}$ and is defined as:

$$\text{MMR} = \text{OFR}_{\max} \cdot \frac{|Q|-1}{|Q|-w-1} \tag{3.1}$$

where w is an integer number representing the number of guard bands. Guard bands are bands over the $\text{OFR}_{\max}$, without counting the last band. Guard bands are necessary to contain the oscillations of the highest-rate user and to permit some flexibility. Without guard bands, if a user reduces its offered rate, the output rates of other users may grow and users might be regulated by a single large last band. The optimal number of guard bands will be found experimentally.

From the MMR the new reference width of bands is obtained with the rule

$$\Delta_{\text{ref,MMR}} = \frac{\text{MMR}}{|Q|-1} = \frac{\text{OFR}_{\max}}{|Q|-w-1}$$

as illustrated in figure 3.9

As before, many $\Delta_{\text{ref,MMR}}$ are stacked on top of the guaranteed rate and the last used band is adapted to fit the link capacity. Formally,

$$b_{u,q} = \begin{cases} g_u & q = 1 \\ \Delta_{\text{ref,MMR}} & 2 \le q < q_{u,\max} \\ c - \sum_{1 \le q < q_{u,\max}} b_{u,q} & q = q_{u,\max} \end{cases} \quad \forall u \in U$$

3.4.3 Round Trip Time compensation

The UGUALE's enhancement presented in this subsection assumes the knowledge of the RTTs of users.

Flows with different RTT competing for the same capacity will obtain different rates. This aspect should be taken into account because it can notably impact

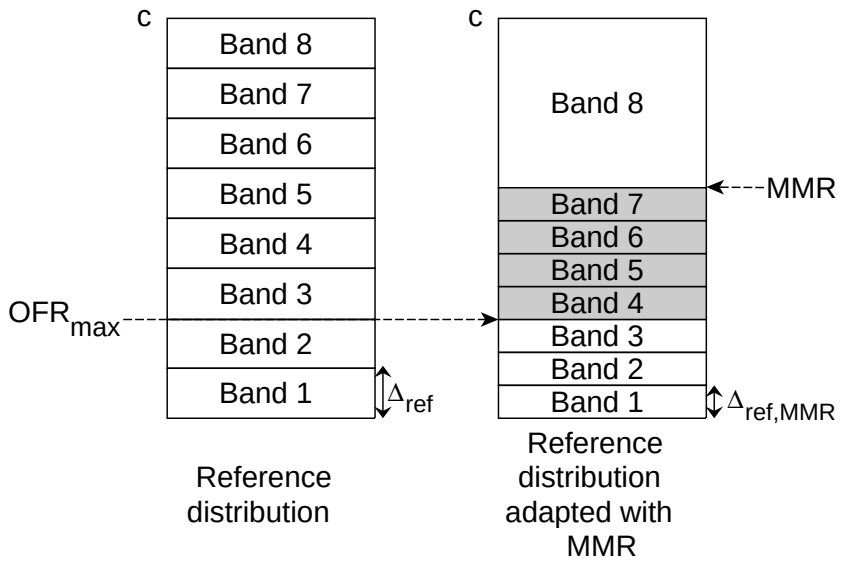

Figure 3.9: *Example of bands assignment with the Maximum Marking Rate strategy using* $w = 4$ *guard bands (in grey).*

the fairness of the system. In fact, a difference of RTT is translated in a different goodput following the relation

$$\frac{\text{goodput}_1}{\text{goodput}_i} \approx \frac{\text{RTT}_i}{\text{RTT}_1} \tag{3.2}$$

as reported in [16]. With UGUALE it is possible to mitigate this effect by properly enlarging or shrinking bands by a factor depending from the user's RTT.

The goodput is defined as the difference between the throughput and the rate lost for retransmission. For our scope the goodput can be approximated with the throughput: from now on the goodput will be referred as throughput or rate and it will be indicated by $r_u \quad \forall u \in U$.

Since a user is an arbitrary aggregate of flows, each user's flow can experience a different RTT. We define the RTT of a user as the average RTT of its flows.

For each user $u \in U$, we want to find a coefficient m_u defined as the factor that multiplies the estimated rate r_u to get the OFR_u:

$$r_u \cdot m_u \equiv \text{OFR}_u \quad \forall u \in U$$

We assume that there are N users sharing the same FIFO queue served by a link of capacity c. We also assume that the users' RTTs are uniformly distributed between a minimum and a maximum RTT. Following the relation 3.2 users will obtain

$$r_u = \frac{\mathrm{RTT}_1}{\mathrm{RTT}_u} \cdot r_1 \quad \forall u \in U$$

Fixing $r_1 = 1$, the relation becomes:

$$r_u = \frac{\mathrm{RTT}_1}{\mathrm{RTT}_u} \quad \forall u \in U$$

The distribution of rates with $r_1 = 1, \mathrm{RTT}_1 = 1$ is depicted in figure 3.10. Assuming that all users deserve the same rate, the optimal fair rate is a con-

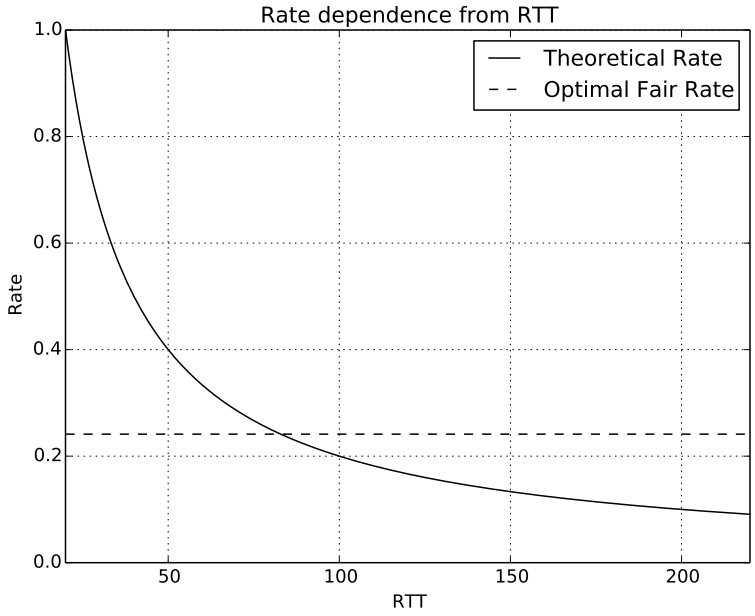

Figure 3.10: *Flows with different RTTs sharing the same FIFO queue obtain different throughputs. There are 200 flows, each one with a different RTT in the range [20,220]*

stant:

$$\mathrm{OFR}_u = \frac{c}{N} = \mathrm{OFR} \quad \forall u \in U$$

The coefficient m_u is then

$$m_u = \frac{\mathrm{OFR}}{r_u} \quad \forall u \in U$$

By substituting the expressions of the optimal fair rate and of the estimated rate, we obtain

$$m_u = \frac{c \cdot \mathrm{RTT}_u}{N \cdot \mathrm{RTT}_1 \cdot r_1} \quad \forall u \in U$$

Finally c, N, RTT_1 and r_1 are constant and can be grouped under a common factor k, so that the linear dependence of m_u from RTT_u is highlighted:

$$m_u = \frac{c}{N \cdot \mathrm{RTT}_1 \cdot r_1} \cdot \mathrm{RTT}_u = k \cdot \mathrm{RTT}_u \quad \forall u \in U$$

The multiplier is represented in figure 3.11.

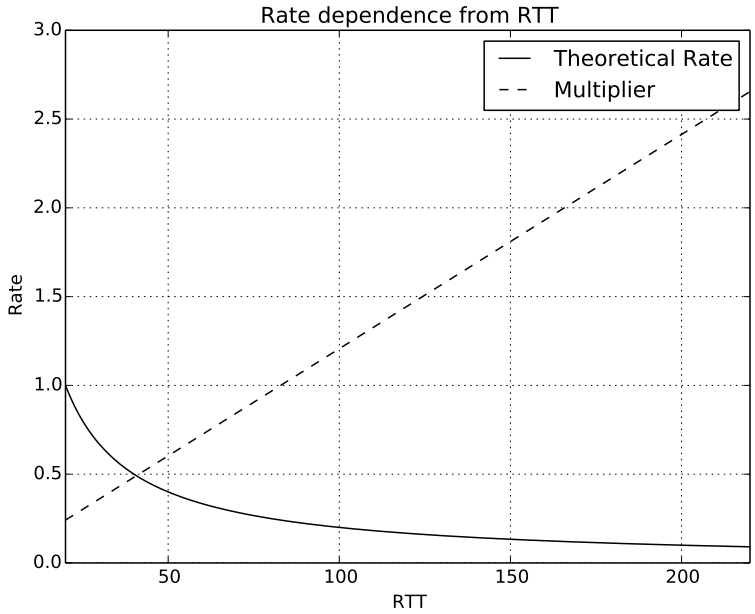

Figure 3.11: *The multiplication of the estimated rate by the multiplier produces a straight line corresponding to the OFR.*

The knowledge of users' RTTs is enough to calculate the coefficient m_u of each user. We suppose that with UGUALE, by multiplying thresholds by these coefficients, rates of users having different RTTs can be equalized. More precisely, bands are assigned such that:

$$b_{u,q} = \begin{cases} g_u & q = 1 \\ \Delta_{\mathrm{ref}} \cdot m_u & 2 \leq q < q_{u,\max} \\ c - \sum_{1 \leq q < q_{u,\max}} b_{u,q} & q = q_{u,\max} \end{cases} \quad \forall u \in U$$

Calibration of the multiplier

Multipliers obtained with the method previously proposed may have a too strong effect when using UGUALE. In fact users are not sharing a simple FIFO queue and their rates might be already quite similar to their optimal fair rates. For this reason, the effect of the multiplier can be lightened as depicted in figure 3.12. To

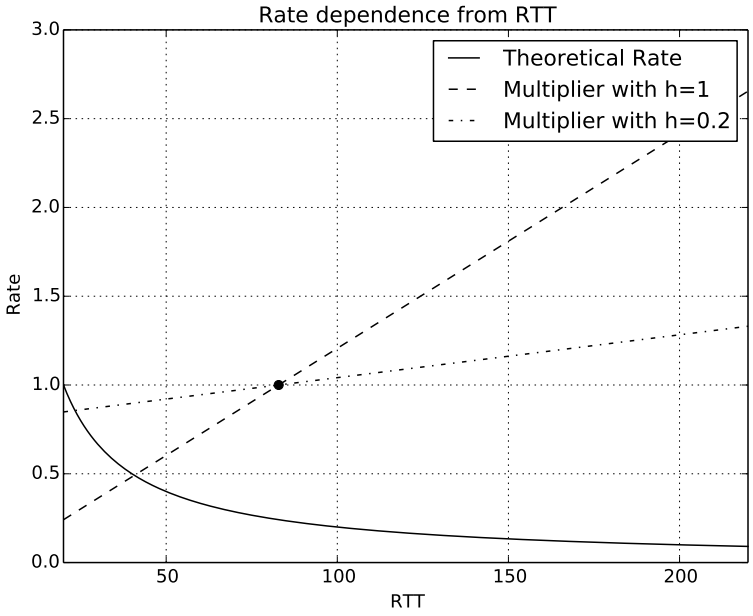

Figure 3.12: *The new multiplier line has a smaller angular coefficient and passes from the point where the old multiplier line is unitary.*

reduce the effect of the multiplier, the angular coefficient is multiplied by a factor $0 < h < 1$. Moreover, the new multiplier line should pass from the point where the old multiplier is unitary. If only the angular coefficient k were reduced, all users will uselessly get a smaller multiplier. If the line were rotated with respect to another point, all users might get a multiplier greater or bigger than one. The optimal value of h will be searched experimentally.

Chapter 4

Implementation

The approach proposed in the previous chapter was implemented on the real hardware testbed described in section 4.1. The chosen software switch was picked following the reasoning presented in section 4.2. The pipeline of UGUALE was adapted to the testbed and implemented as described in section 4.3. Section 4.4 shows the procedure necessary to obtain a Strict Priority scheduler with Open vSwitch, while in section 4.5 the implementation of meters is discussed. Section 4.6 shows the softwares used to generate and collect data. Finally, to demonstrate the effectiveness of UGUALE, an arbitrary number of users with different characteristics is emulated as described in section 4.7.

4.1 Testbed

UGUALE has been implemented on a physical network in order to have a better understanding of the available technologies and to discover all the real problems that can affect computer network communications. This should also demonstrate the feasibility of our solution.

The testbed was named *Redfox* and it is composed of five PCs. There are four Dell Studio XPS 8100, with Intel i7 processor and named Redfox1, Redfox2, Redfox3 and Redfox4. These PCs are connected through Ethernet CAT6 cables to a Ciaratech PC, with processor Intel Qual Core 2 running at 3GHz, named

Redfox0. Redfox0 can act as a software switch thanks to an extra 4-NIC network interface card, whose ports run at 1Gbps. All PCs run a standard distribution of the operative system Ubuntu 14.04 LTS. Figure 4.1 shows a picture of the testbed.

Figure 4.1: *Photo of the testbed.*

The testbed is used with the scheme presented in figure 4.2: Redfox1, Redfox2 and Redfox3 send data to Redfox4. Hence the link *(Redfox0, Redfox4)* becomes the bottleneck of the network on which fairness is evaluated. Given this working scheme, from now on Redfox1, Redfox2 and Redfox3 will be indicated as the *client PCs*, Redfox4 as the *server* and Redfox0 as the *switch*.

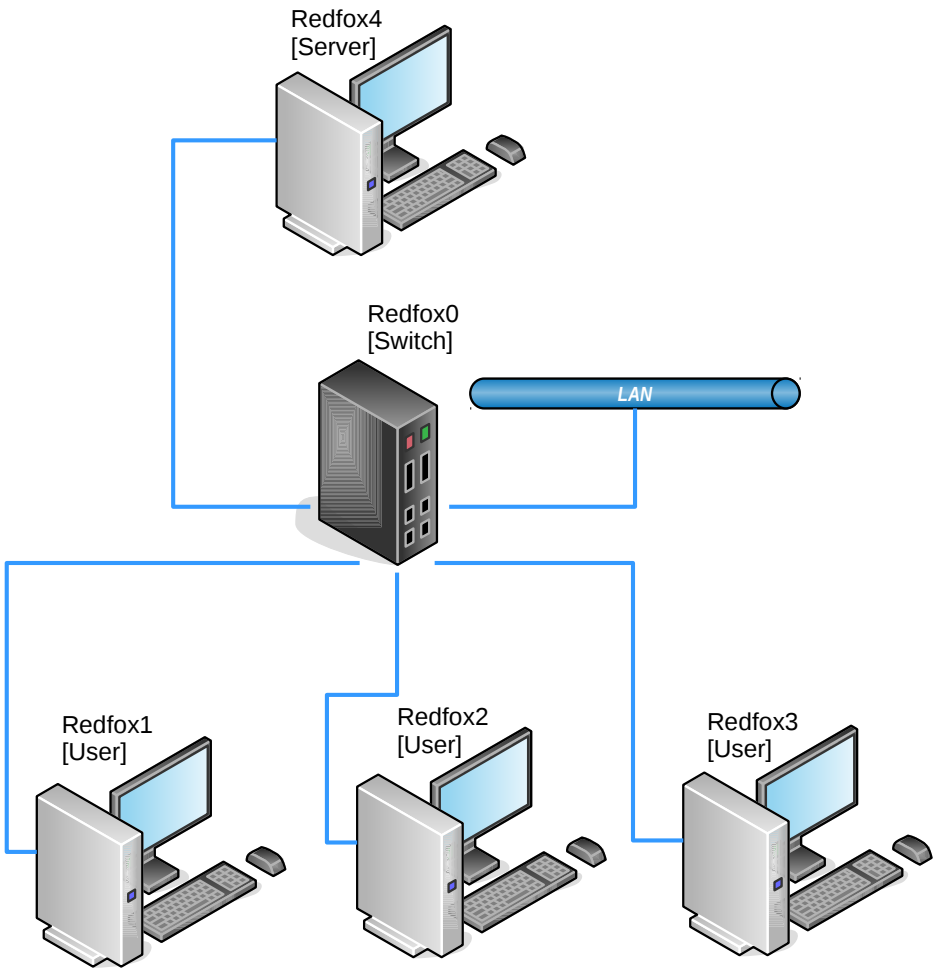

Figure 4.2: *Network scheme of the testbed.*

4.2 Switch

To implement UGUALE, a switch that allows to configure the pipeline presented in section 3.2 is needed. Moreover, the switch should assure good performance in terms of stability and throughput.

The searched ease of programmability was found in the Software Defined Networking (SDN) domain and in particular in the abstractions made available by OpenFlow. SDN is a networking paradigm that enables the separation of the

control logic from the data plane, while Openflow is the most diffused controller to data plane interface. OpenFlow compliant switches can be programmed with a set of instructions that, acting on some abstractions of the physical elements of the data plane, allow the configuration of the packet pipeline with good flexibility. Hence, the ideal solution would have been to use a physical OpenFlow compliant switch to have high-performance and compatibility with all the QoS features. Unfortunately this kind of switches have a prohibitory cost.

We initially thought of implementing the whole testbed in a virtual environment such as Mininet, but the first simple tests, even with a standard switch not implementing UGUALE, made clear the instability of the virtual solution in particular for the evaluation of QoS parameters.

Then we decided to implement the whole system on real machines and to equip a PC with an extra 4-NICs network card to use it as a software switch. This solution is notably cheaper than an OpenFlow switch, in fact this kind of network card costs around 600 CAD. When we got the network card, we replaced it with the graphic card of a PC and we obtained our switch.

The last step was to find a switching software that could provide all the necessary QoS OpenFlow abstractions and that could guarantee high performance in terms of total throughput and stability. Although the only OpenFlow abstractions necessary to implement the proposed scheme are meters and queues, we discovered that the QoS support in software switches is still limited.

The most popular software switch is Open vSwitch (OVS) [32]. Today it is widely deployed in data centers because it guarantees high performance and stability while supporting all QoS features. Unfortunately we discovered that OVS does not support meters.

Another software switch that we tried is Lagopus [15]. Lagopus implements OpenFlow 1.3 and is based on the high-performance packet processing library DPDK (Data Plane Development Kit) [7]. At the time we discovered it, in June

2015, Lagopus was still at version 0.1.2, supporting meters but not queues.

Another solution that we tried is ofsoftswitch13 [20] that is considered the reference implementation of OpenFlow: all functionalities are implemented as described in the standard. The main drawback is that ofsoftswitch13 is not optimized for performance, imposing a maximum throughput of only tens on Mbps.

We thought to a workaround: implementing meters directly on client PCs and using OVS as switching software. In this way we were able to use OVS that was the only switch guaranteeing high performance and queue programmability.

4.3 Pipeline

The pipeline described in section 3.2 was modified to be executed on our testbed. In particular the pipeline was adapted to the absence of meters on the switch.

In OpenFlow, a meter is an entity that measures the rate of packets assigned to it and that applies actions subsequently. Each meter maintains a list of meter bands, where each band specifies a rate and an action executed only if the packet is processed with that band. The rate of a band can be specified in Kbit/s or in packets/s and a meter can also take into account bursts. The action applied by a band can be *drop* or *remark*. The drop action discards the packet while the remark action marks the DSCP field of the IP packet header. Packets are always processed by a single band: the meter chooses only the band with the highest configured rate that is lower than the measured rate. Since the available actions are only the two just presented, the only way for the meter to communicate to the classifier which band is activated is to mark the packet.

The pipeline implemented with an OpenFlow compliant switch would look like the one depicted in figure 4.3.

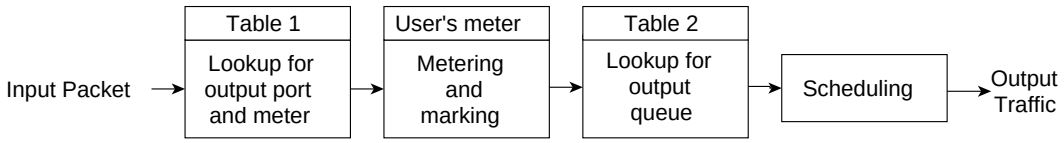

Figure 4.3: *UGUALE's pipeline with an OpenFlow switch.*

The arriving packet is first matched against a table whose fields determine the output port and define the users. The matching rule appends the chosen output port to the pipeline of the packet and sends it to the meter dedicated to the user. The meter chooses a band depending on the measured rate and applies the correspondent action, that is marking the DSCP field of the packet with a number corresponding to the band activated. Then the packet is passed to another table that acts as a classifier for the scheduler: the only matching field is the DSCP value that corresponds to the queue on which to send the packet. The enqueue action is appended to the packet's pipeline that is finally executed.

With the workaround involving meter and marker in the client PC, the pipeline is quite different and it is represented in figure 4.4.

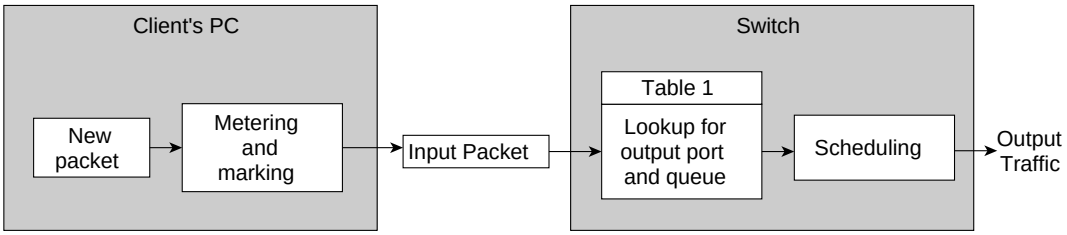

Figure 4.4: *UGUALE's pipeline when meter and marker are placed in the client PC.*

Packets generated by clients are marked just before exiting the network interface. The switch receives the packets and match them against a table whose fields are the source and destination IP addresses and the DSCP field. The IP address determines the output port while the DSCP value determines the output queue. Then the packet's pipeline is immediately executed.

The meters that we managed to implement on client PCs are not the same

used in OpenFlow, but the pipeline is still coherent to the proposed approach. In fact positioning meters at the output interface of client PCs is equivalent to have them at the input interface of the switch.

4.4 Strict Priority scheduler

OVS supports and implements queues, but the only scheduling policy configurable with the furnished utility *ovs-vsctl* is Weighted Round Robin (WRR): we found a solution also for this problem, as described below.

First of all, using ovs-vsctl, we created a WRR scheduler with $|Q|$ queues of arbitrary weight. Then, inspecting the kernel configuration with the Traffic Control (*tc*) command suite, we noticed that the scheduler was always created with a fixed reference name "1:" and its queues were named "1:1", "1:2", and so on until "1:$|Q|$". To be precise, scheduler and queues in Linux Kernel are called respectively Queuing Discipline (*qdisc*) and *classes*. The WRR configuration was useful because it stored references of qdisc and classes in the OVS QoS database.

The next step was to substitute the WRR scheduler with a Strict Priority one. Indeed thanks to tc commands, we could delete the old *htb* qdisc whose classes were automatically destroyed. Then we created a new *prio* qdisc having the same reference name "1:" and $|Q|$ classes. When instantiating a prio qdisc, it is possible to specify only the number of priority queues $|Q|$ that it should have. These classes are automatically created with names ranging from "1:1" to "1:$|Q|$". By default, the queue with the highest priority is "1:1" and the lowest one is "1:$|Q|$".

Finally, inspecting the OVS database with the ovs-vsctl utility, we checked that the references to the qdisc and its classes had not been lost. We executed some tests specifically designed to verify the correct behavior of the Strict Priority scheduler and it worked as expected.

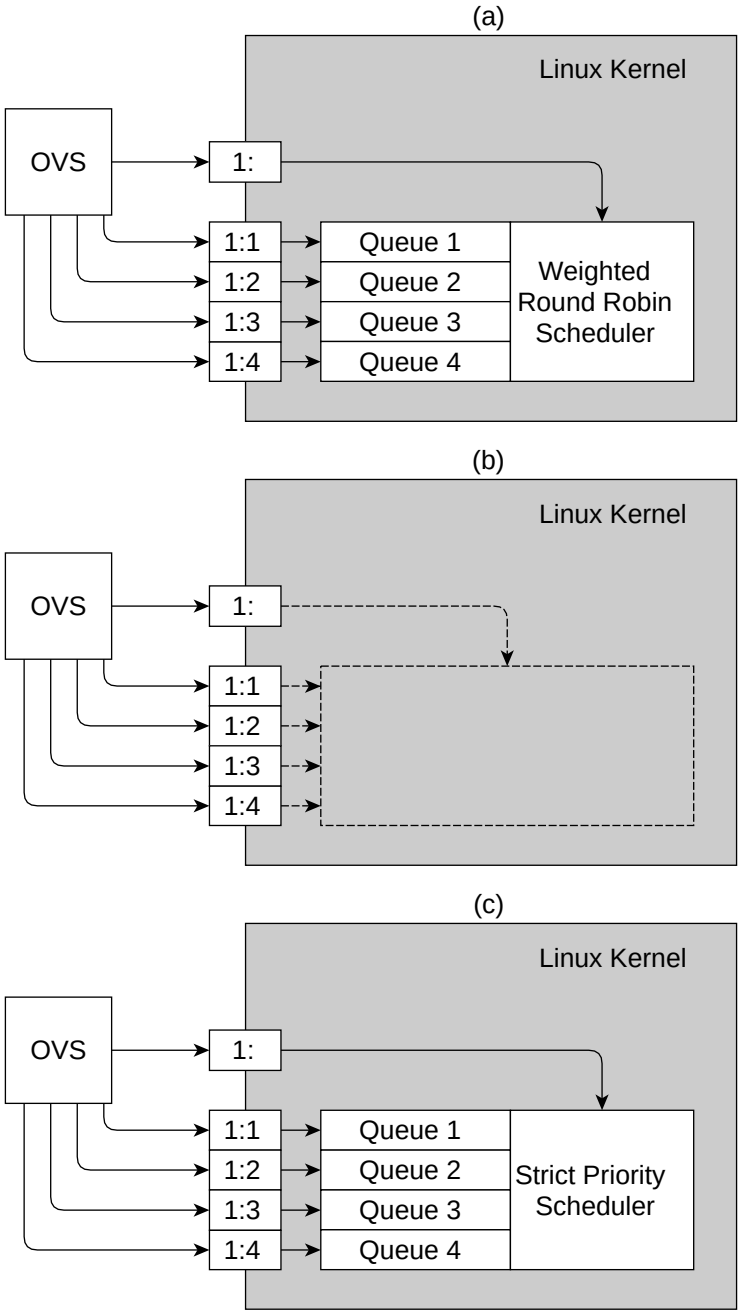

Figure 4.5: *Process to obtain a SP scheduler. A WRR scheduler is created with ovs-vsctl (a). Then the scheduler and the queues are deleted with tc commands (b) so that ovs pointers remain valid. Finally the new scheduler and the new queues are created with the same references using tc commands (c).*

4.5 Meters

The reference implementation of OpenFlow meters is described in ofsoftswitch13's code and it is based on token buckets.

A token bucket is a data structure that enables the comparison of the packet rate with a given threshold. The bucket collects tokens that arrive with a rate equivalent to the reference threshold: each token is a grant to transmit a Byte. When a packet arrives, its size B [Bytes] is compared to the number of tokens T in the bucket.

- If $T - B \geq 0$, B tokens are consumed and the packet rate is considered below the threshold. The packet is said to be in-profile.
- If $T - B < 0$, the bucket is voided and the packet rate is considered above the threshold. The packet is said to be out-of-profile.

A token bucket allows a certain burstiness of the packet rate based on the burst size parameter. This parameter represents the maximum number of tokens that the bucket can accumulate.

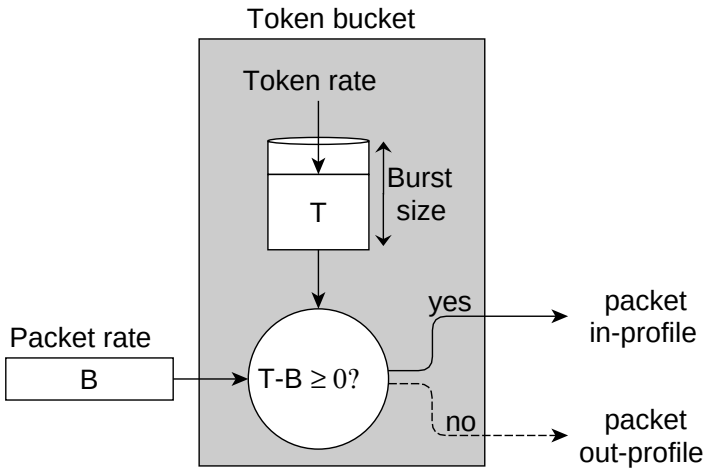

Figure 4.6: *Structure of a token bucket that enables the comparison of the packet rate with a threshold corresponding to the token arrival rate.*

In the reference implementation, a meter is implemented like in figure 4.7. For

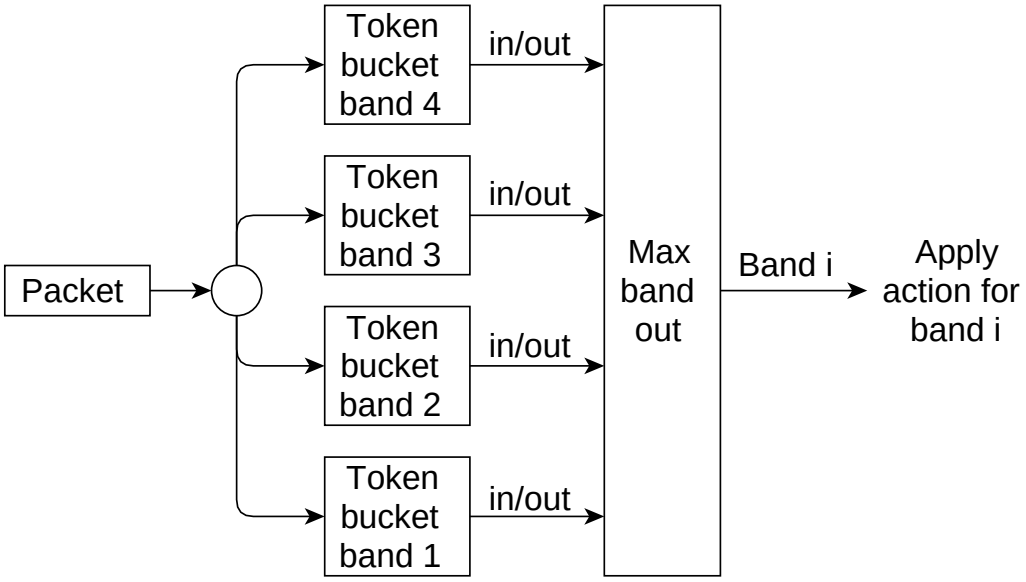

Figure 4.7: *Scheme of an OpenFlow Meter. The meter has 4 bands.*

every band, a token bucket is instantiated. All buckets are periodically refilled at the same time with a number of tokens based on band rates. In other words, the token rate corresponds to the band rate. A packet arriving to the meter consumes tokens from every bucket, then the biggest band with an empty bucket is applied.

While this logic seems easy to implement, we did not find a way to obtain it in Linux Kernel and with token buckets. Indeed, we proposed two different implementations of meters: the first uses a cascade of token buckets and the second uses the rate estimators made available by the Linux Kernel Firewall. The first solution produces an output marking that is quite different from the OpenFlow standard while the output of the second solution is more OpenFlow-like, yet not compliant.

4.5.1 Series of token bucket filters meter

The first solution is illustrated in figure 4.8 and it is totally implemented with the Linux Kernel suite tc. First of all a DSMARK Queuing Discipline (qdisc) is instantiated. Even though DSMARK is a qdisc, it does not schedule, reorder or

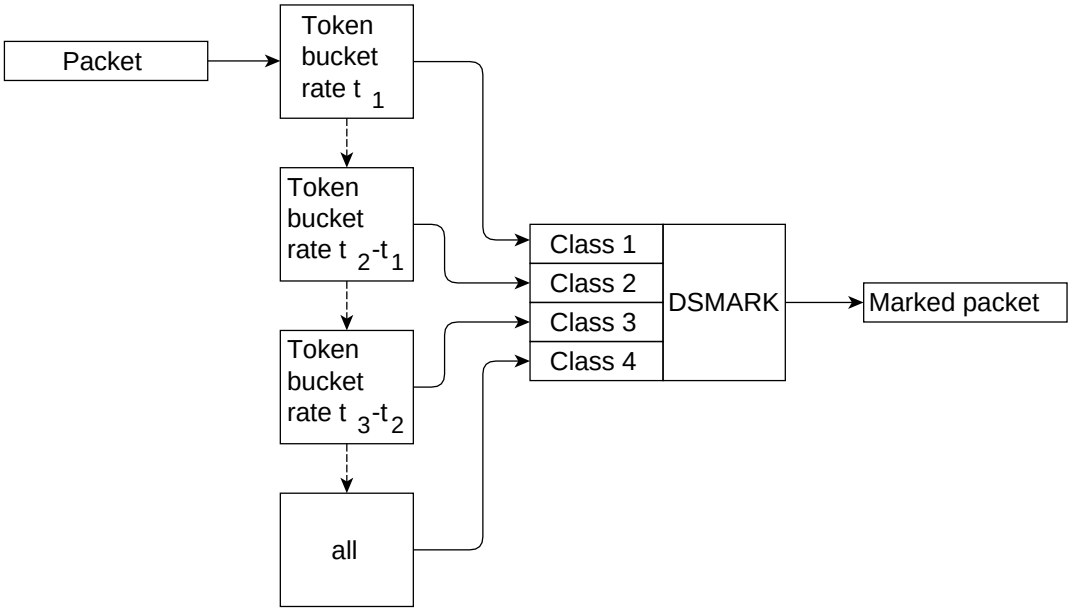

Figure 4.8: *Meter implemented with a series of token bucket filters. The meter has 4 bands.*

delay packets but it simply marks the DSCP field of packets passing through it. The DSMARK is created with $|Q|$ classes and each class is in charge of marking packets with a certain value. In our scheme, the first class marks DSCP=1, the second one DSCP=2 and so on. A qdisc having many classes needs a classifier to send packets to the correct class. The classifier is implemented with a series of tc filters and each filter has a priority and a token bucket. A filter matches if its token bucket declares the packet in-profile: in this case the matching packet is sent to a certain class. Filters have priorities to determine in which order they should be checked. The first filter has a token rate equivalent to the first threshold $t_{u,1}$ and sends matching packets to DSMARK class 1. If the first filter does not match, the second filter is tried. The second filter has a token rate equivalent to the second band width, so equal to $t_{u,2} - t_{u,1}$. The second filter sends packets to DSMARK class 2. This mechanism is repeated until the filter with priority $|Q| - 1$ is reached, because the $|Q|^{th}$ filter sends everything left to the last class. This version of meters produces an output marking subdivided in bands, so totally

different from the OpenFlow standard.

As an example, a user transmits at a constant bit rate of 400Mbps and has thresholds at 100Mbps, 200Mbps, 500Mbps and 800Mbps. With this kind of meters the 25% of packets are marked as 1, another 25% are marked as 2 and the remaining 50% are marked as 3. According to the OpenFlow standard these packets should be all marked as 3. The example is depicted in figure 4.9.

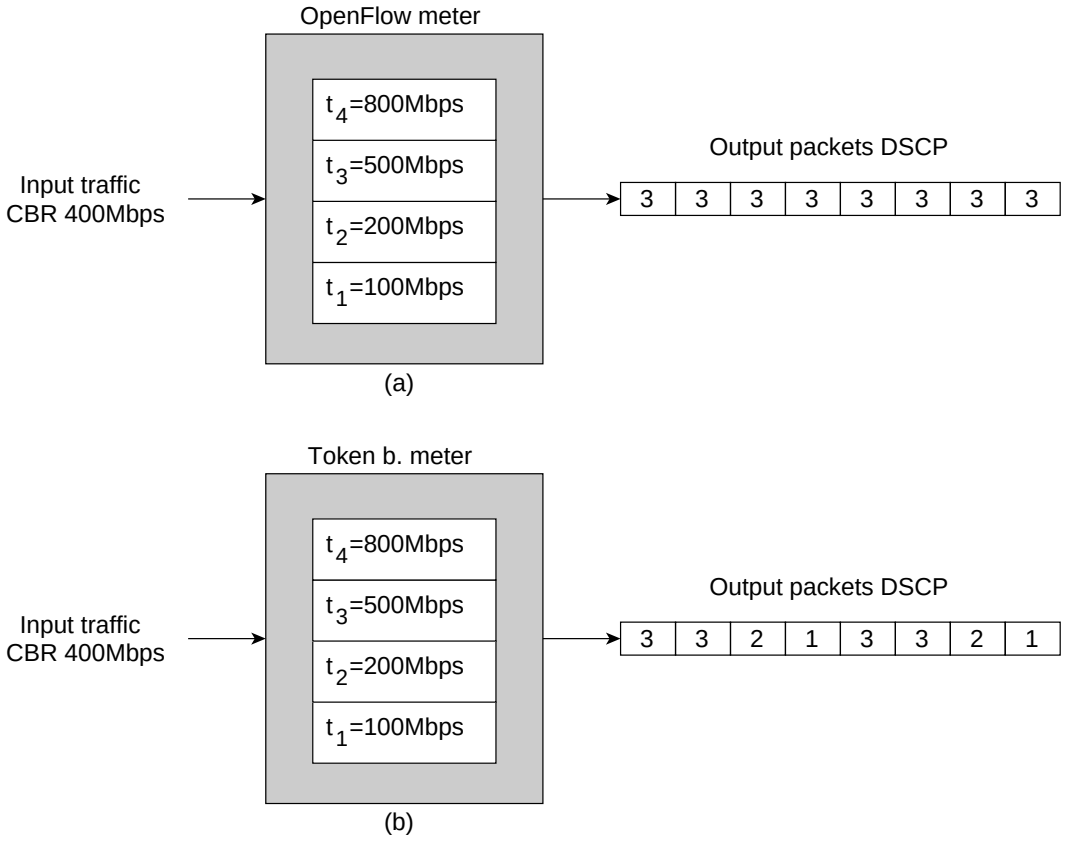

Figure 4.9: *Example showing the output of different meter implementations.*

4.5.2 Iptables estimator meter

To obtain an OpenFlow-like behavior, we tried the implementation of meters illustrated in figure 4.10. This version uses the Linux Kernel Firewall suite *iptables* and the tc suite. First of all, a DSMARK qdisc and its classes are configured as

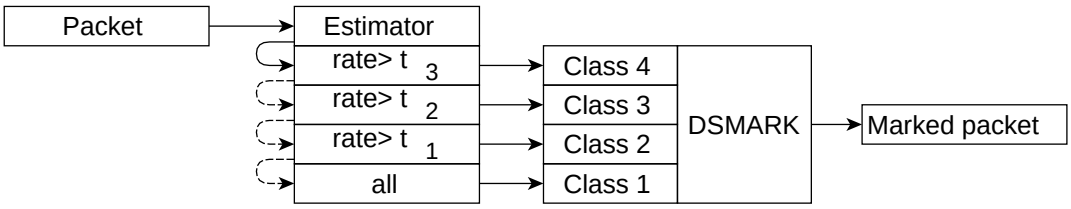

Figure 4.10: *Meter implemented with an iptables estimator. The meter has 4 bands.*

before. Then a firewall table is populated with an estimator and two rules for each class.

Every table of iptables is composed of rules, and every rule has a matching criterion and an action called target. The target is applied only if the packet matches the rule. A packet is compared with all rules from the top to the bottom until it encounters the target ACCEPT. This target lets the packet pass to an eventual next table or to the Linux Kernel.

The first rule matches every packet and sends them to an estimator. Then, for each class two rules are written:

1. The first rule compares the rate of the estimator with a threshold. If the estimated rate is greater than the threshold, the packet matches and the CLASSIFY target is applied. This action classifies the packet: it decides on which DSMARK class the packet should be sent.

2. The second rule has the same matching criterion of the previous one, but it applies the ACCEPT target.

Since we check if a rate is greater than a threshold, the first couple of rules has thresholds corresponding to the $(|Q| - 1)^{th}$ band and sends packets to the $|Q|^{th}$ class. All rules until the second band are specified in the same way. Also in this case, the last rule is different because it will match everything left. The estimator has a minimum measurement interval of 250ms with a Logarithmic Estimated Weighted Moving Average (ewma-log) of 500ms. For this reason this meter does not produce an output completely equivalent to the reference implementation.

4.6 Collecting data

To generate data and collect statistics we used two softwares: *iPerf* and Bandwidth Monitor Next Generation (*bwm-ng*).

IPerf allows to establish TCP and UDP connections between two sockets and to send data for the desired amount of time. The process receiving data is indicated as the iPerf server, while the sender is called client. A client can open an arbitrary number of parallel connections to the server. If TCP connections are used, the transmitting rate is the maximum permitted by the network: for this reason iPerf is commonly used to test links capacity. When using UDP connections, a sending rate must be specified. This rate, if supported by the network, is maintained for all the time specified. Both the server and the client processes produce reports about the actual goodput of flows. These reports are produced with an arbitrary time period. If a client opens parallel connections to a server, iPerf produces client-aggregated reports.

Bwm-ng is a tool that reliably monitors the current bandwidth of network interfaces in a specified direction. Bwm-ng produces reports with an arbitrary time period and was used to check the total rate received by the server.

To have an immediate idea about the rates obtained by users, we needed a tool to gather and visualize in real-time the reports generated by the mentioned softwares. For this reason we wrote a Python script named *plotServer* that instantiates many iPerf server processes and a bwm-ng process and then starts to plot the reports received. A screenshot of plotServer is presented in figure 4.11. The figure is composed of two subplots:

- The first one shows raw samples produced by iPerf aggregated by source IP address and protocol. Connections from the same IP address of different protocols are plotted with the same color but with different line styles. In particular, continuous lines are used for TCP connections, while UDP connections are represented by dashed lines. This plot shows also the raw

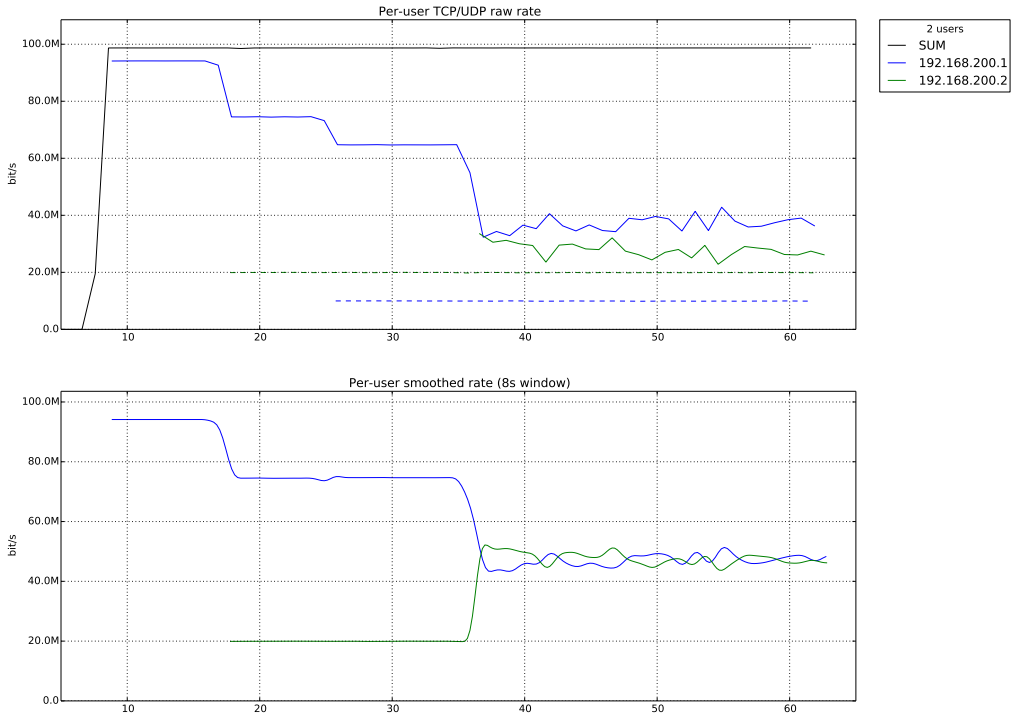

Figure 4.11: *Screenshot of plotServer with two users having both TCP and UDP connections. Users are defined only by the source IP address. In the first subplot, continuous lines represents TCP flows while dashed lines UDP flows. In the time interval [8sec,15sec], the difference between the goodput obtained by a TCP flow and the interface rate is observable.*

bwm-ng reports labeled as "SUM".

- The second subplot shows the total rate obtained by users, where a user is only defined by the source IP address. Lines are smoothed by a weighted moving average algorithm.

To execute many test cases with different users' configurations, we wrote another Python script called *test* that:

1. Configures the UGUALE's pipeline on the switch.
2. Configures client PCs, e.g. by instantiating meters and markers.
3. Starts the plotServer script.

4. Starts iPerf client processes.

5. Saves the reports and the testbed configuration for future analysis.

The latter script permits to easily instantiate many users, as depicted in figure 4.12.

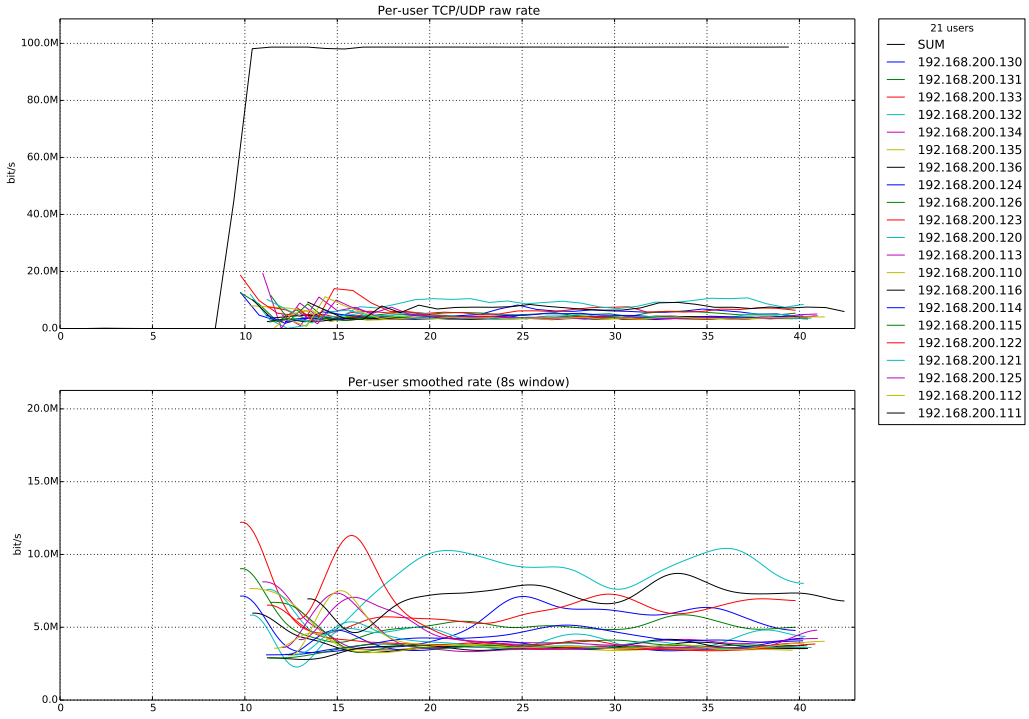

Figure 4.12: *Screenshot of plotServer during a test with 21 users using only TCP connections. Users are defined only by the source IP address.*

4.7 Emulating more users

To show that our forwarding engine can guarantee Optimal Fair Rates to many users with only a few queues, more users than queues are needed.

If users are defined by the source IP address, the easiest solution to emulate users is to use a PC for every user. Since our switch has only four ports, other PCs cannot be attached. The other solution is emulating more users from a PC and redefining users. In this case, a user is defined as the couple source IP address

and destination TCP port. The server instantiates many iPerf server processes, each one listening on a different port. The client PCs start different iPerf client processes, each one connecting to a different server port, in order to obtain the situation depicted in figure 4.13.

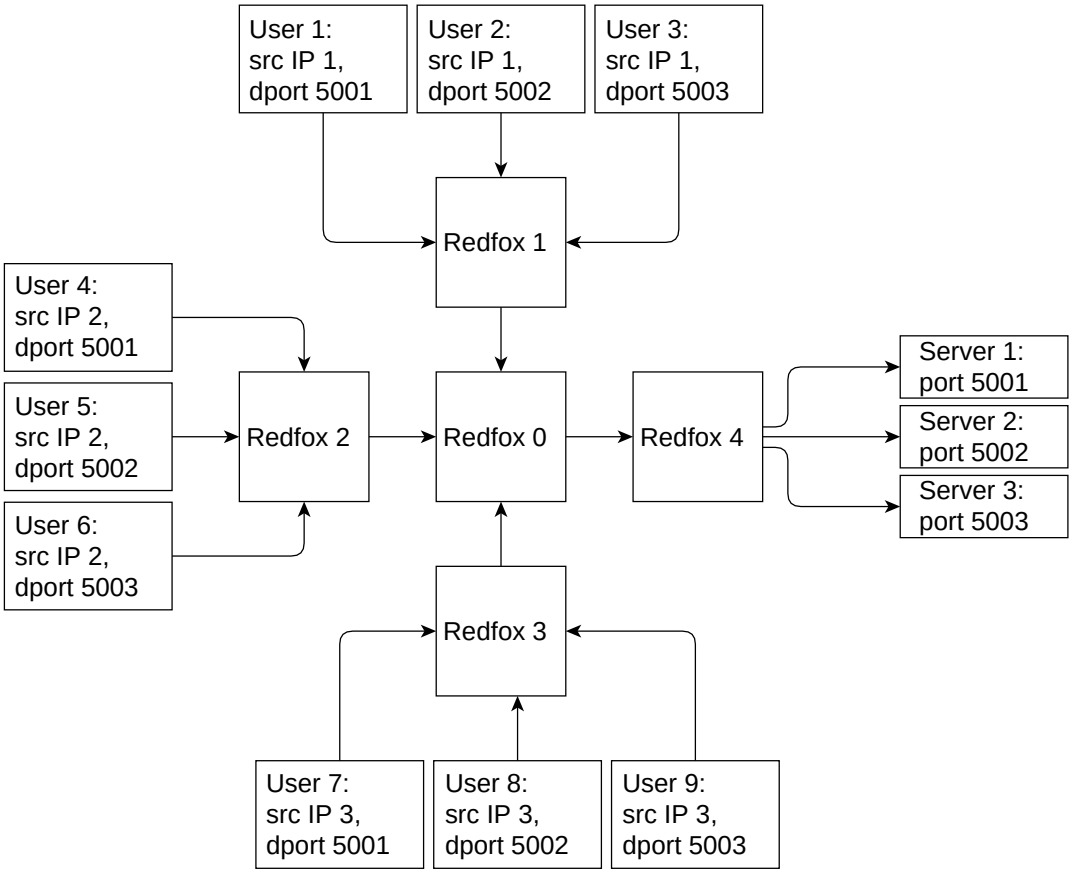

Figure 4.13: *Testbed configured to emulate many users. Each user is identified by the couple source IP address, destination port.*

The only bottleneck must be the link *(switch, server)* whose bandwidth is regulated by UGUALE. In other words, users should not contend other links' capacity. Unfortunately now users do not have dedicated links to the switch, so the first bandwidth contention could happen on each link *(client PC, switch)*

To avoid this contention, each user is limited to a rate such that the sum of users' limits exiting a PC remains under the capacity of the link exiting that PC.

Then also the link *(switch, server)* should be limited to the same value, otherwise it would not be the bottleneck: each user would obtain a rate corresponding to its first limit. Formally, the emulated users and the bottleneck must be limited link such that

$$\sum_{u \in U_i} c_u \leq l_i \quad \forall i \in I$$
$$c_u = c \quad \forall u \in U$$

where

- I is the set of PCs
- U_i is the set of users emulated on PC $i \in I$
- c_u is the link bottleneck capacity for the emulated user u.
- l_i is the capacity of the link *(i,switch)*
- c is the bottleneck link capacity

The first constraint says that the total capacity available for users of the same PC cannot be greater than the capacity of the link exiting the PC itself. The second constraint puts the obtained capacity to a common value corresponding to the bottleneck link capacity. To avoid perturbations, the bottleneck negotiates a capacity of c=100Mbps. Since $l_i = 1Gbps \quad \forall i \in I$, the maximum number of users per PC is $l_i/c = 10 \quad \forall i \in I$.

If all users would start to transmit at the same time, the capacity of the bottleneck link would be the first to be contended, so it would not be strictly necessary to limit users' processes.

4.7.1 Emulating users with different Round Trip Times

To reproduce the situation of users with different Round Trip Times (RTTs) described in subsection 3.4.3, a tool to delay processes is needed.

It possible to delay processes with softwares but the output is not realistic at all. E.g. using Trickle, processes simply get throttled for regular time intervals, modifying completely the TCP behavior.

A better solution is to use NetEm, that is able to delay and rate limit packets. As reported in [10], the TCP sequence and windows obtained using NetEm are very similar to the ones obtained with real links. NetEm is a Queueing Discipline (qdisc) available in the tc suite, so it must be attached to an egress interface or to another qdisc. The latter option is easy but unsuitable for our tests: the only types of qdisc allowing to attach more than one qdisc are classful qdiscs. Classful qdiscs are schedulers with more than one queue. In this case, each NetEm can be attached to a different queue. This solution modifies the fairness between emulated users because packets get scheduled for transmission on the real link *(client PC, switch)* with an order different from FIFO. In other words, a classful qdisc would begin to schedule its queues and all the emulated users would get perfectly the same rate.

A solution that does not alter the bandwidth sharing between users is to attach a NetEm to virtual interfaces (veth). Virtual interfaces must be linked to the real Ethernet interface using an internal bridge. The internal bridge can be the default Linux bridge or an instance of Open vSwitch (OVS). We chose OVS because it seems more stable and because it does not apply any implicit fairness mechanism between its ports. In this way, users are served as if they were flowing into a single FIFO queue.

Another problem was to make packets generated by different processes pass through different veths based on the definition of user. This problem was solved using *iptables*, *ip rules* and routing tables together, as depicted in figure 4.14 and described below:

1. The packet is generated in the main user space, with a certain destination TCP port.

2. The packet is managed by the firewall (iptables), where it receives a firewall mark based on the destination port.

3. Thanks to *ip rule* the packet is sent to a certain routing table based on the carried firewall mark.

4. Each routing table contains a single entry that directs all packets to a certain veth, to which a NetEm is attached.

Passing through a NetEm, the packet is delayed. If the PC is implementing also meters and markers, the packet is marked when passing through the DSMARK qdisc attached to the physical interface eth0.

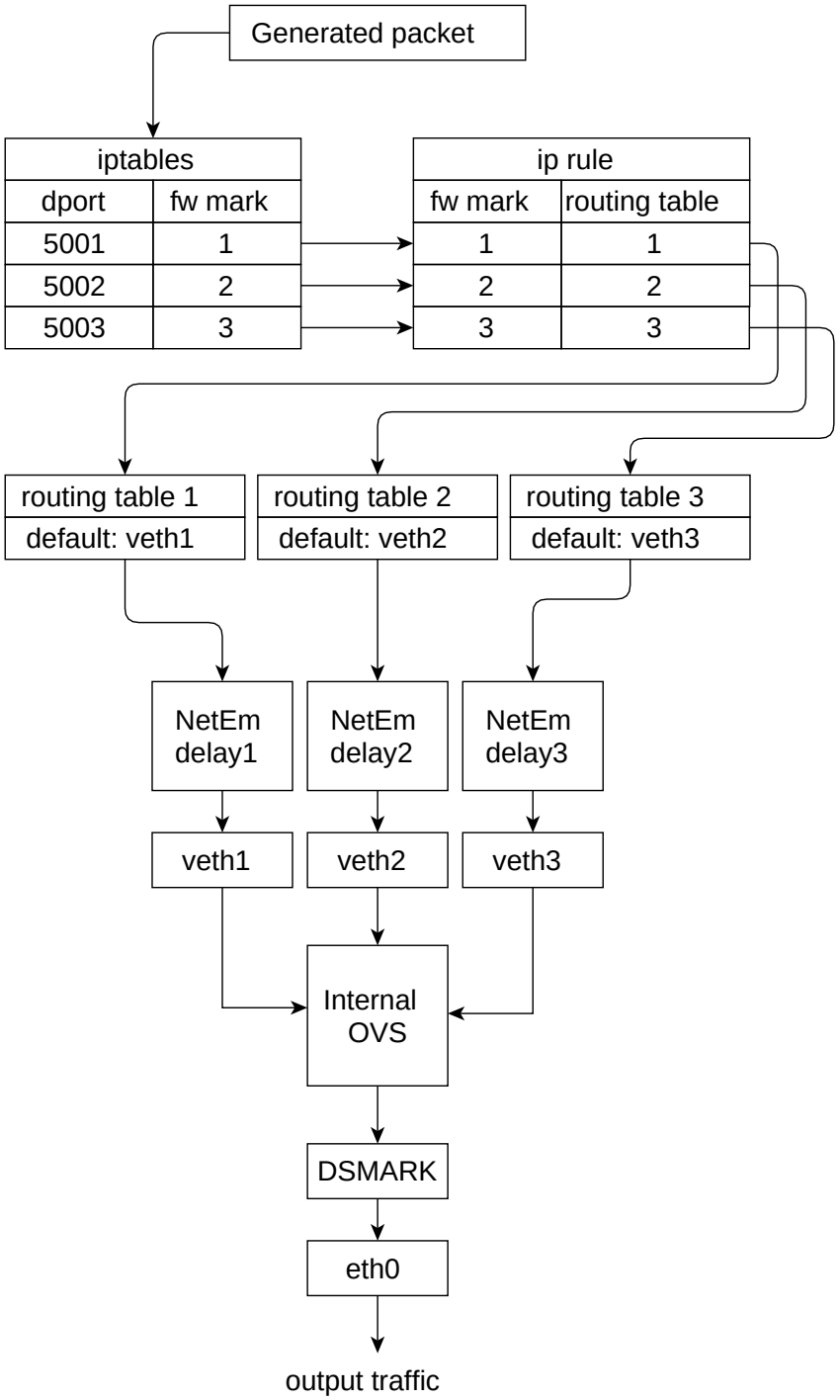

Figure 4.14: *PC configuration to emulate many users with different delays.*

Chapter 5

Experimental results

Section 5.1 describes how tests were executed and how data was validated, analyzed and presented. In section 5.2 the results obtained are showed and commented to clarify the functioning of UGUALE.

5.1 Test

A test is defined by the configuration of the testbed and by the data collected by the server using that configuration. The data consists in a set of rate samples for every user. Each sample is identified by the time stamp (ts) in which the sample was produced and by the value of the user's rate (val) in that instant. The structure of a test file is depicted in figure 5.1

The testbed is configured as shown in figure 4.13: Redfox0 is the switch, Redfox4 executes several iPerf server processes and the other PCs act as iPerf clients. Since the client PCs send data to the server, the link *(Redfox0, Redfox4)* becomes the bottleneck on which fairness is evaluated. UGUALE will be used only on the switch port of the bottleneck link toward the server. In the opposite direction (from the server to the client PCs) there will be only flows of TCP ACKs that for certain do not consume enough bandwidth to saturate the links' capacity.

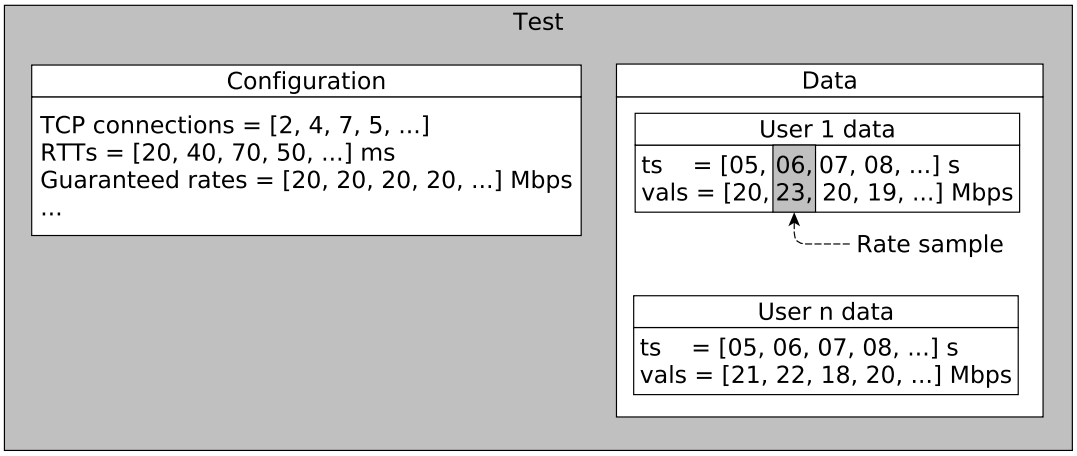

Figure 5.1: *Structure of a test file*

5.1.1 Test validation

The data collected during the tests must be validated before proceeding with the analysis. To evaluate fairness, all rate samples must belong to the same time interval in which all users are in a steady state. In each test, the server processes start at time t_0 while the client processes start after a fixed time interval necessary to transmit and apply the particular test configuration to the client PCs. All the timestamps of a test are relative to its t_0.

To declare a test valid:

- The number of recorded users must be equal to the number of instantiated users.
- All users must begin to transmit within an instant indicated by t_{start}.
- All users must be still active at the end of the test, when the server processes are stopped. This instant is indicated by t_{end}.
- All users must produce approximately the same number of rate samples in the interval $[t_{start}, t_{end}]$.

- Every user can have just a few low rate samples, i.e. periods in which transmission was interrupted.

The first instant t_{start} was set to 35 seconds. This interval is quite large since rates become stable within a couple of seconds, but it is ensures that all connections started and reached a stable rate. The final instant t_{end} was set to 3 seconds before the end of the test, so that all samples are taken strictly before the end of transmissions. If the test is valid, data can be recorded and analyzed.

5.1.2 Data analysis

First of all, data is trimmed in order to consider only the rate samples falling in the time interval $[t_{start}, t_{end}]$. Then many statistics are calculated.

Jain's Fairness Index To evaluate the fairness between users, the Jain's Fairness Index (JFI) can be used [23]. The Jain's Fairness Index is defined as:

$$f(r_1, r_2, ...r_{|U|}) = \frac{(\sum_{u \in U} x_u)^2}{|U| \cdot \sum_{u \in U} x_u^2}$$

where x_u is the normalized throughput of a user u. The normalized throughput is defined as the ratio between the measured throughput r_u and the fair throughput OFR_u of a user u:

$$x_u = \frac{r_u}{\mathrm{OFR}_u}$$

This index is bounded between 0 and 1, where 1 is a fair distribution and 0 is a discriminating one. The Jain's Fairness Index can be calculated with the rate allocation vector taken in different instants or with average rates. The first option allows to evaluate also the variance of the JFI, but since iPerf does not produce synchronized samples, the allocation vectors can be obtained only by interpolating and re-sampling data at fixed time instants.

Throughput and goodput Another aspect to evaluate is the total link utilization. For each instant of the previous re-sampling, the goodput is obtained as the

sum of the allocation vector elements while the throughput samples are obtained from the bwm-ng reports. Both the goodput and the throughput samples are normalized with respect to the bottleneck link capacity c. Finally, to calculate the ratio goodput/throughput in various time instants, also the throughput samples should be interpolated and re-sampled. For all of the presented link utilization indexes, the mean and the variance are calculated.

Data transformation To further analyze data, each raw sample $r_{u,t}$ of user u at time t is transformed with the function

$$g(r_{u,t}, \mathrm{OFR}_u, c) = \frac{r_{u,t} - \mathrm{OFR}_u}{c}$$

The sample is related to the OFR, enabling the comparison between users with different guaranteed rates and thus different OFRs. The normalization with respect to the bottleneck capacity allows to express rates in the bounded interval $[-1, +1]$. For instance, a transformed rate equal to 0 means that the raw rate is equal to the OFR, while a transformed rate equal to -0.1 means that the raw rate is the 10% lower than the OFR with respect to the bottleneck capacity.

To be precise, the assignment of guaranteed rates and the estimation of the OFRs consider the maximum goodput achievable on the link instead of the nominal link capacity. If it were not so, when all users offer their guaranteed rates no one will be able to reach it. The practical link capacity is the maximum rate that iPerf reports when only an elastic user is transmitting on the link. For instance on 100 Mbps links the practical capacity was $c = 94.1$ Mbps.

5.1.3 Data presentation

Each test is presented in a figure composed of four subplots and a text box, e.g. figure 5.2. The statistics exhibited in the subplots are calculated on transformed rates. The figure is organized as follows:

- Subplot (a) shows the discrete distribution of samples. The transformed samples of all users are joined in a single list on which histograms and other statistics are computed. Histograms count the occurrences of values grouped by intervals called bins. In our case, the space of rates $[-1, +1]$ was divided in 500 symmetric bins.

- Subplot (b) shows the average and the standard deviation of users' rates. In this graph, users are ordered by ID (e.g. the IP address) in order to show details about single users.

- Subplot (c) shows average and standard deviation of transformed rates for users aggregated by the number of TCP connections. This subplot allows to evaluate the dependence of rates from the number of TCP connections.

- Subplot (d) shows average and standard deviation of transformed rates for users aggregated by RTT. This subplot allows to evaluate the dependence of rates from the RTT.

- The text-box is divided in three sections. The first section lists the ID, the number of TCP connections and the RTT of each user. The second and third sections report respectively the configuration and other statistics of the test.

To resume, the presented statistics are:

- The mean and the variance of the Jain's Fairness Index.

- The mean and the variance of the throughput.

- The mean and the variance of the goodput.

- The mean and the variance of the ratio between the goodput and the throughput.

- The mean value, the variance, the standard deviation and the mean squared error (MSE) of transformed samples.
- The mean and the standard deviation of transformed samples for each user.
- The mean and the standard deviation of transformed samples grouped by their user's number of TCP connections.
- The mean and the standard deviation of transformed samples grouped by their user's RTT.

5.1.4 Test configuration

To study and validate several concepts, many test configuration were tried. The tunable test parameters are:

- the activation of UGUALE on the switch
- the number of users
- the bands assignment policy
- the number of TCP connections of users
- the RTT of users
- the number of queues used by UGUALE
- the activation of the Maximum Marking Rate strategy
- the activation of the RTT compensation strategy
- the type of meters
- the length of switch's queues

5.1.5 Outline of tests

The first step is to run tests without UGUALE to evaluate the basic behavior of the testbed. For this purpose OVS in standalone fail-mode is used: the switch executes the standard MAC learning and does not enforce QoS policies. In this configuration, when all users are in the same conditions (that is when all users have the same number of TCP connections and the same RTT) they should obtain the same rate. This case can be considered as a baseline, in other words as the target fairness that the testbed can achieve. By contrast, executing tests with users in different conditions is useful to show the TCP per-flow fairness and the impairments caused by different RTTs.

The second step is to run tests with UGUALE to prove that the conditions presented in section 1.2 are enforced. To verify that the guaranteed rates are assured, $\sum_u g_u$ should be set equal to c (free bandwidth equal to zero) and users should be set in different conditions. Then, by increasing the free bandwidth, the fairness in the allocation of the excess capacity can be evaluated. Finally other tests can be executed to verify the effectiveness of the enhancements proposed in section 3.4 and to calibrate their parameters.

5.2 Results

This section presents the obtained results and observations about the texts executed.

5.2.1 Switch in standalone fail-mode

When the switch is in standalone fail-mode, QoS is not enforced and fairness depends on users' conditions. In particular, three different situations can occur:

- If all users have the same number of TCP connections and the same RTT, every user obtains the same throughput and the JFI is very high. This situation, represented in figure 5.2, constitutes a baseline and demonstrates the stability of the testbed.
- If users have a different number of TCP connections, the TCP per-flow fairness shows up and the per-user fairness is impaired. The relation between the number of TCP connections and the rate obtained is linear, as highlighted in figure 5.3(c).
- If users have different Round Trip Times, the fairness is impaired as discussed in subsection 3.4.3: rates are inversely proportional to the RTT, as shown in figure 5.4(d).

The latter two situations, in which the JFI drops to approximately 0.8, explain why a QoS enforcing mechanism like UGUALE is necessary.

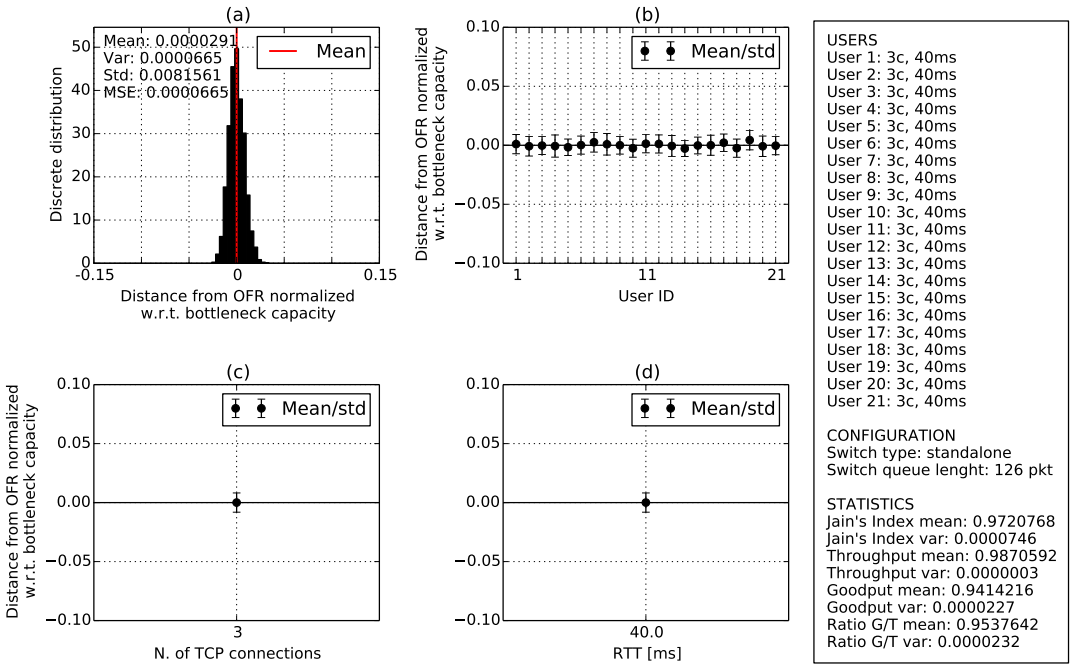

Figure 5.2: *When all users have the same number of TCP connections and the same RTT, QoS enforcing is not necessary to obtain fairness.*

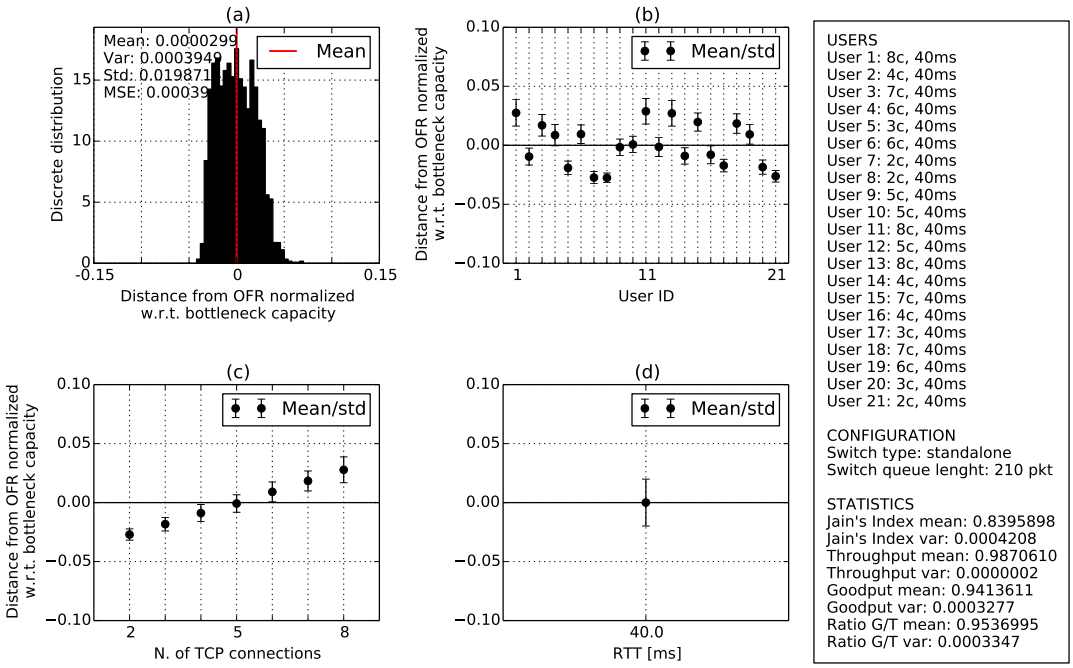

Figure 5.3: *The points in subplot (c) are disposed on a tilted line, highlighting the linear dependence of users' rates from the number of TCP connections.*

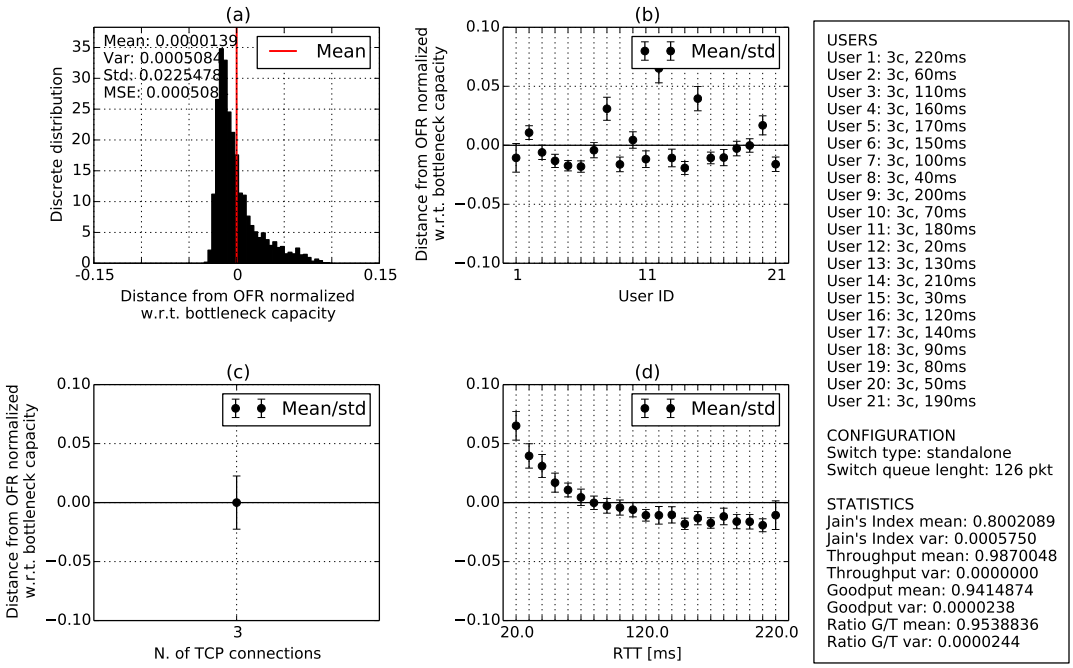

Figure 5.4: *Users obtain rates inversely proportional to their RTT, as shown by the curve in subplot (d).*

5.2.2 UGUALE

The results presented in figures 5.5 and 5.6 show that the proposed allocation engine is able to assure guaranteed rates to a big number of users in different conditions.

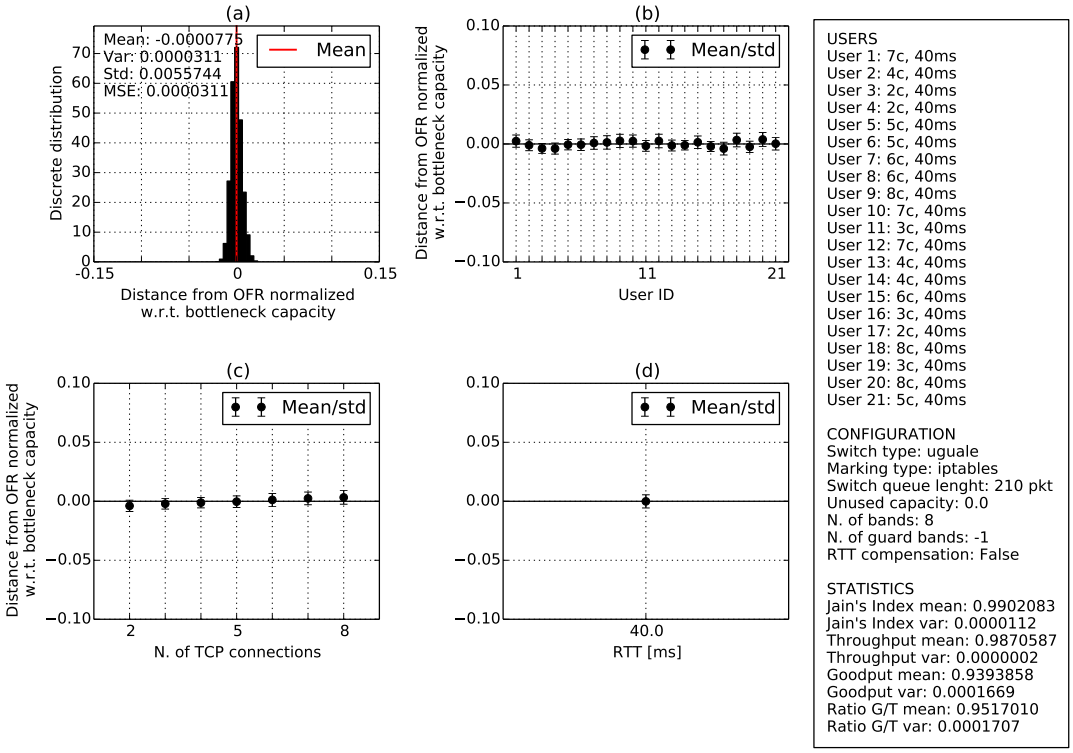

Figure 5.5: *UGUALE is able to assure guaranteed rates to users having a different number of TCP connections: the points in subplot (c) remain on the horizontal axis.*

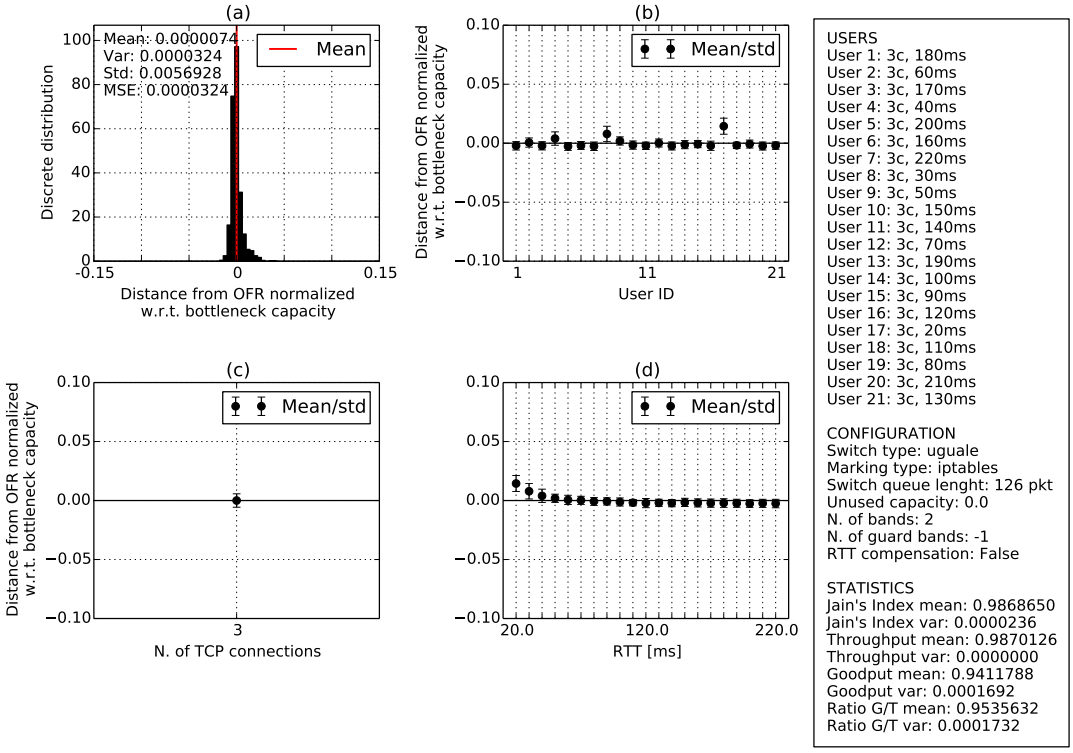

Figure 5.6: *UGUALE is able to assure guaranteed rates to users having different RTTs: the points in subplot (d) remain on the horizontal axis.*

When there is no free bandwidth, users oscillate around their first threshold and rates are almost perfectly controlled. To test how the free bandwidth is shared, the OFRs of users should be bigger than their guaranteed rates. To emulate this situation, the sum of guaranteed rates of active users must be lower than c. Since thresholds are assigned using the rules presented in section 3.3, bands are $c/|Q|$ wide and rate oscillations can be that big. In the unlucky case where the OFR is far from thresholds, all packets are regulated by the same band and they end up in the same queue, so fairness cannot improve significantly with respect to the standalone case.

To confirm this idea and to clarify how the width of bands influences the amplitude of rate oscillations, the bands assignment illustrated in figure 5.7 was tried with several central band's widths. Since the OFR should fall exactly in the

middle of a band, rates should be contained between thresholds $t_{u,1}$ and $t_{u,2}$. By squeezing the central band, fairness and stability should improve because rates get bounded around the optimal one.

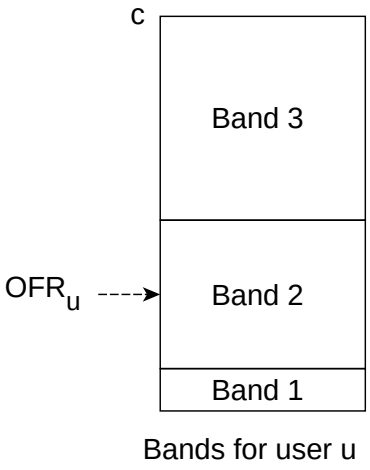

Figure 5.7: *Bands assignment used to clarify the dependence of fairness from the width of bands. The OFR of a user falls exactly in the middle of the central band.*

From the results presented in figure 5.8, it turns out that there are improvements up to a certain width $\Delta_{\max}$ corresponding to 1 Mbit/s. When the central band is larger than $\Delta_{\max}$, flows are not strictly controlled: they rarely cross thresholds and performance is bad. In our opinion, performance stops to improve because when the band becomes too small, just a few packets end up in that queue, so the band behaves as a single threshold.

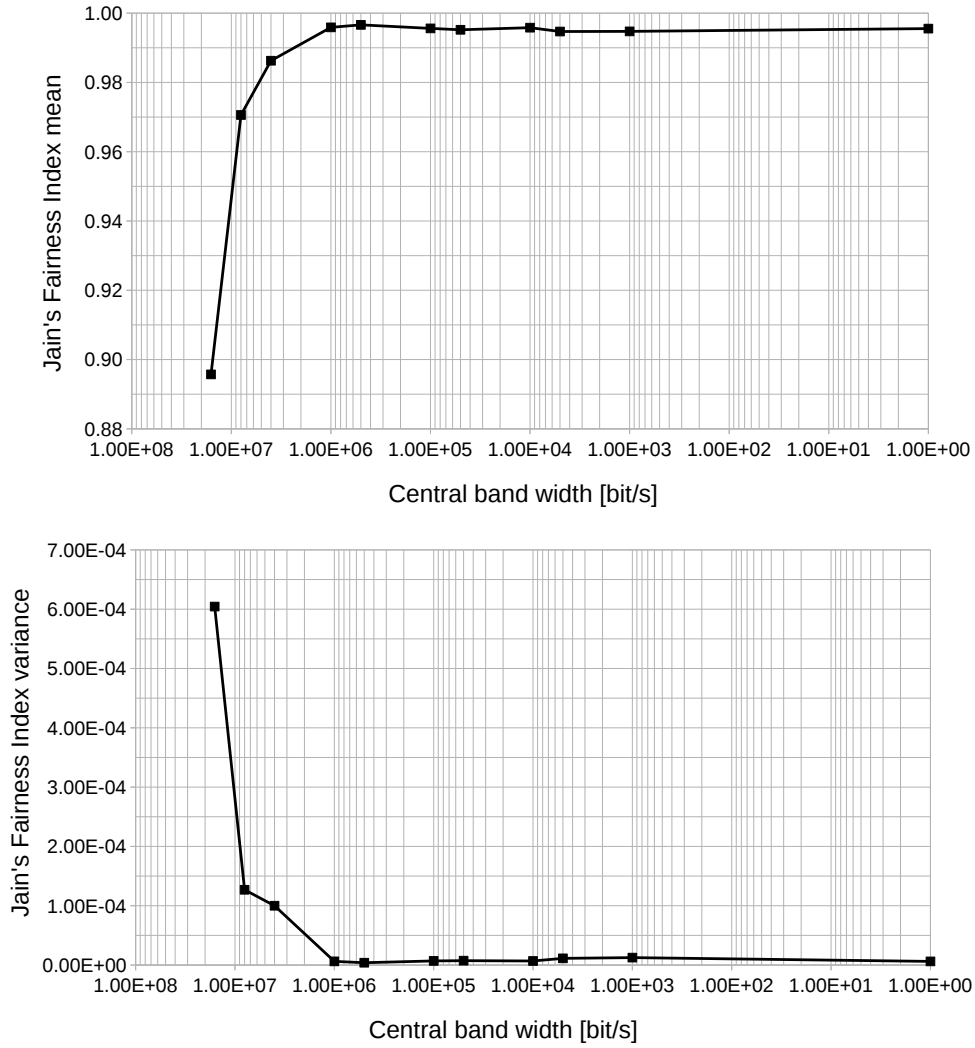

Figure 5.8: *The graphs shows the mean and the variance of the Jain's Fairness Index depending on the width of the central band. Statistics were obtained using the band assignment depicted in figure 5.7 and 9 users having a different number of TCP connections. A test was executed for each central width. The unitary width corresponds to a single threshold in the OFR.*

5.2.3 UGUALE with ad hoc thresholds

To exploit the band assignment proposed in subsection 3.4.1, the offered rates of users must be known. Hence, this scheme were tried to test:

- if a single threshold in the OFR is enough to assure fairness
- if other bands can be concentrated around the OFR threshold to increase fairness and stability

The results, presented in figure 5.9, show that generally performance does not improve significantly by adding bands around the OFR threshold. When the packets of a user are spread on too many queues, performance drops, e.g. when small bands (125 Kbit/s, 250Kbit/s and 500Kbit/s) are used. On the contrary, adding bands that will not be used is useless, as in the case of large bands (2 Mbit/s and 4 Mbit/s): at a certain point, performance remains stationary. In conclusion, each flow should be regulated by not more than 4 bands at a time, since this is the point that obtains the best fairness and stability. This rule of thumb determines the optimal band's width Δ_{opt}, that in our case is still equal to 1 Mbit/s.

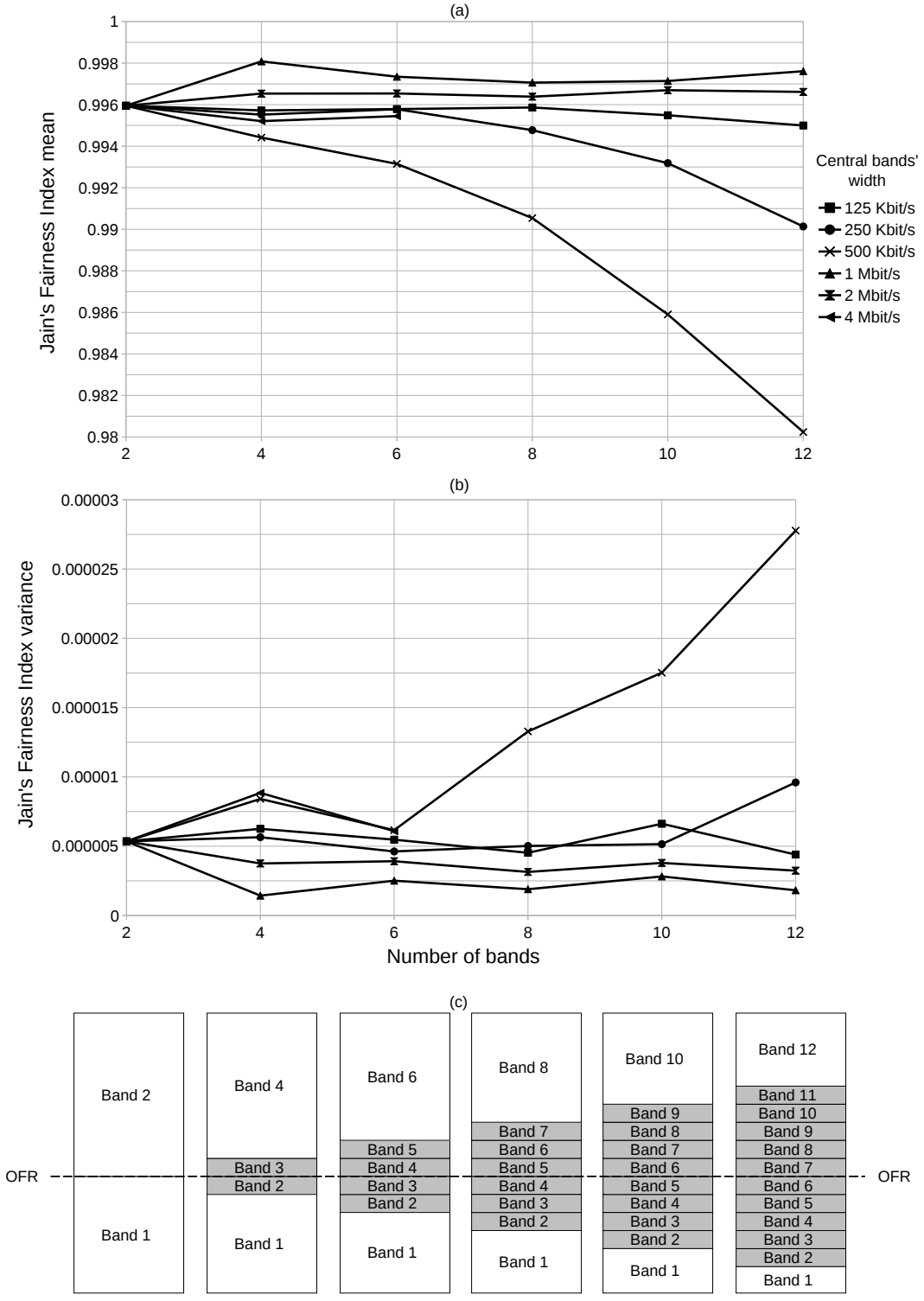

Figure 5.9: *The graphs shows the mean and the variance of the Jain's Fairness Index depending on the width of the central bands. Statistics were obtained using the band assignment depicted in sub-figure (c) and 9 users having a different number of TCP connections.*

5.2.4 UGUALE with MMR

When the maximum Optimal Fair Rate is known, the Maximum Marking Rate strategy presented in subsection 3.4.2 can be applied. Since bands in the used range are small with respect to the basic case, UGUALE assures an excellent per-user fairness even in the free bandwidth case. As shown in figures 5.10 and 5.11, fairness is guaranteed to users having a different number of TCP connections or a different RTT.

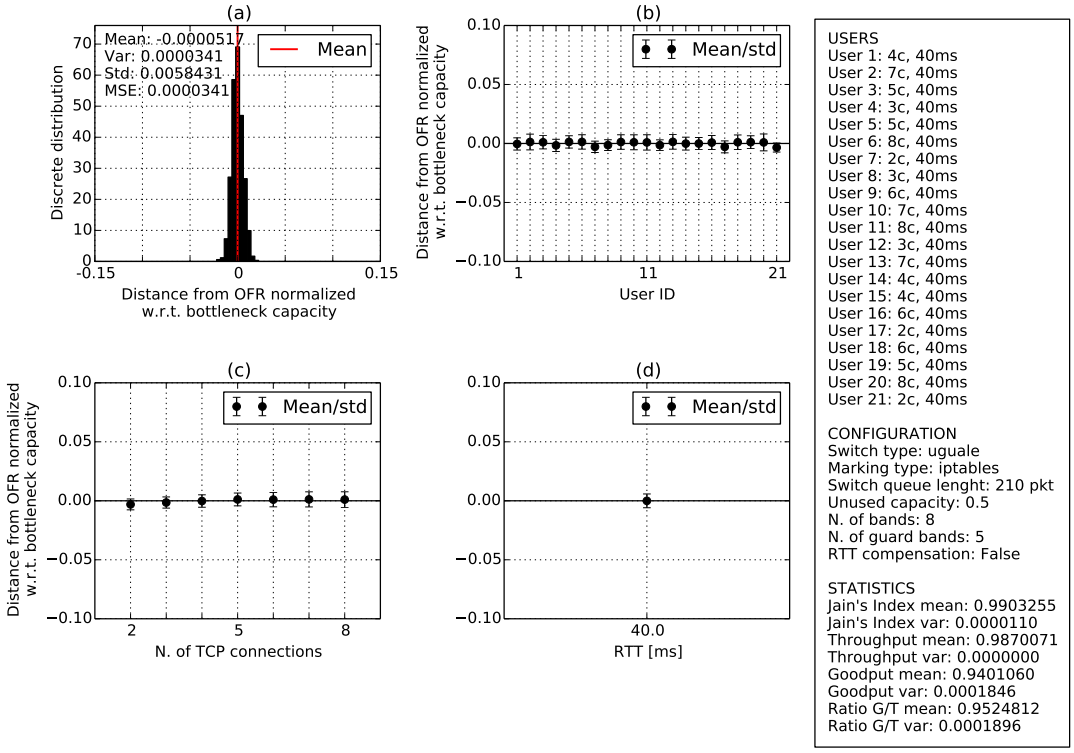

Figure 5.10: *UGUALE with the MMR strategy improves the fairness of users having a different number of TCP connections even in presence of free bandwidth: the points in subplot (c) remain on the horizontal axis.*

The only parameter to optimize was the number of guard bands. In all the test configurations tried, the best results were obtained using $w = 4$ or $w = 5$ guard bands out of $|Q| = 8$ total bands: these values make one of the first thresholds fall closer to the OFR. If less guard bands are used, the bands' width results to small and the rates above the OFR will not be regulated. If more guard bands are used,

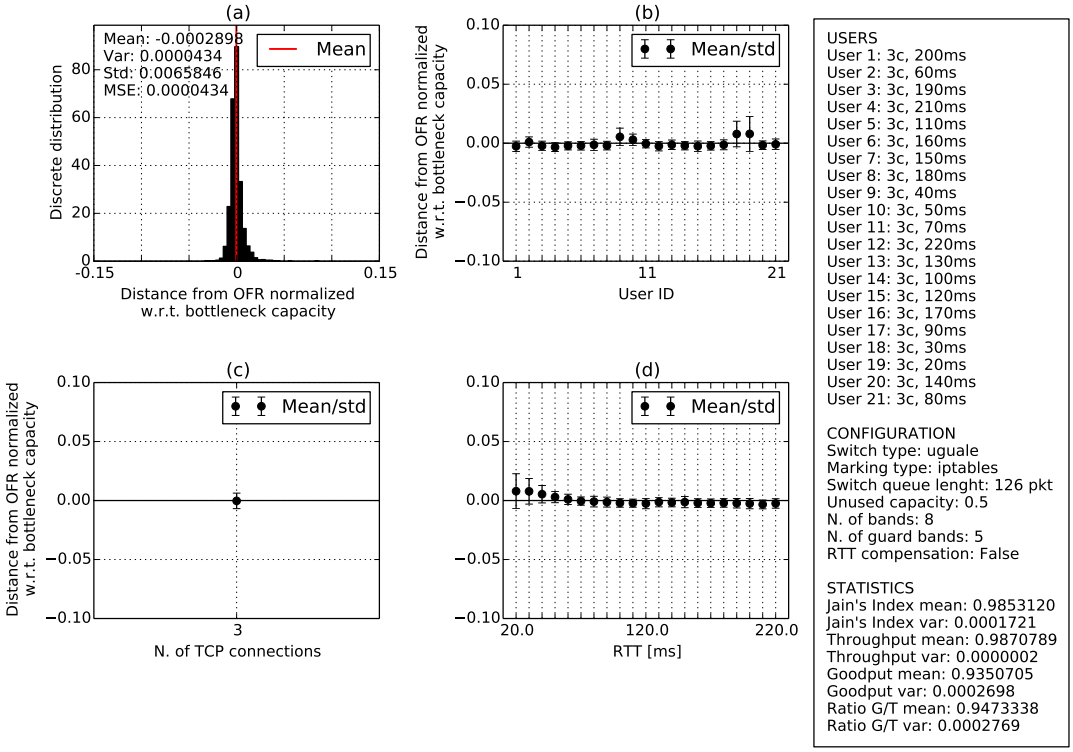

Figure 5.11: *UGUALE with the MMR strategy improves the fairness of users having different RTTs even in presence of free bandwidth: the points in subplot (d) remain on the horizontal axis.*

bands remain large and performance does not improve significantly (with respect to the no-MMR case) because wide oscillations are still permitted.

5.2.5 UGUALE with RTT compensation

When users' RTTs are known, the RTT compensation strategy presented in subsection 3.4.3 can be applied. With this mechanism, UGUALE can slightly enhance the fairness between users having different RTTs, as shown in figure 5.12,

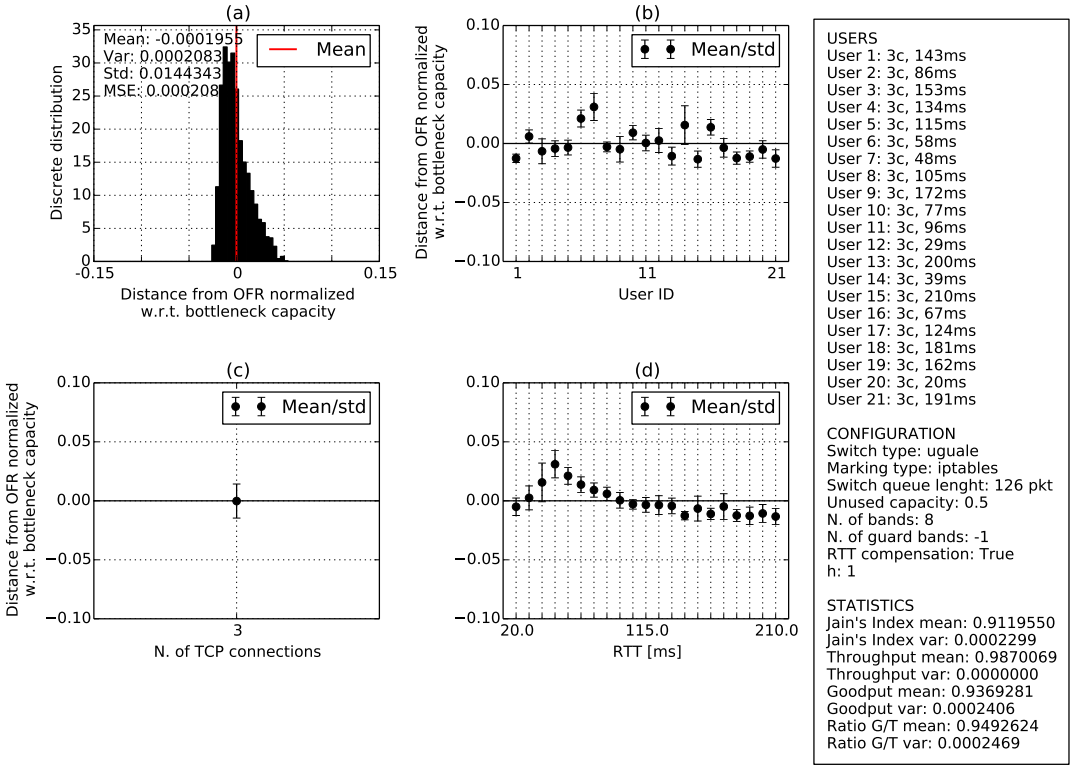

Figure 5.12: *UGUALE with the RTT compensation method improves slightly the fairness of users having different RTTs in presence of free bandwidth.*

If there is no free capacity (so if the there is a threshold in the OFR) this mechanism is useless since the Jain's Fairness Index obtained by UGUALE is already greater than 0.98. In other cases, the RTT compensation strategy can raise fairness of 10% as shown by the graph in figure 5.13.

Finally, from figure 5.13 we observe that the RTT compensation method should not be calibrated since it performs better with $h = 1$.

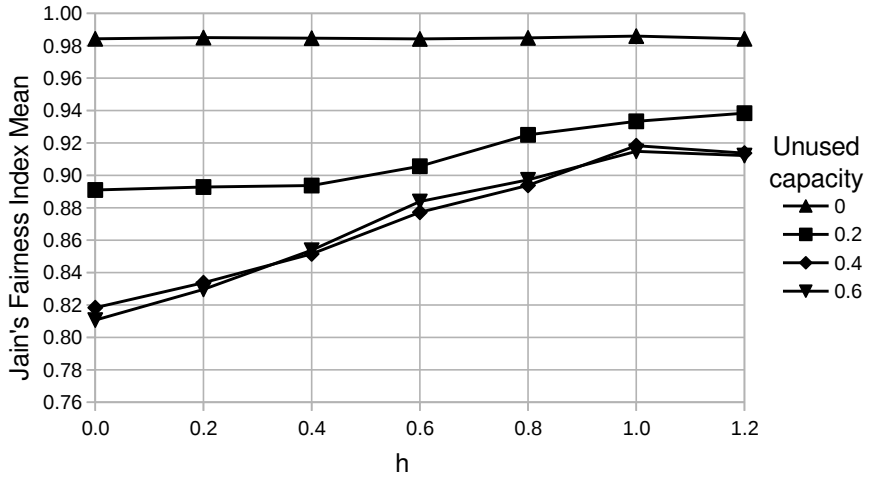

Figure 5.13: *Variation of the Jain's Fairness Index based on the multiplier h applied to the RTT compensation strategy. There are 21 users, each one with 3 TCP connections, with RTT uniformly distributed between 20ms and 220ms. Note that with $h = 0$, the RTT compensation strategy is not applied.*

5.2.6 Type of meter

In section 4.5 two different implementations of meters were discussed. UGUALE enforces fairness with both implementations, but statistics are different.

The first meter type, implemented with a series of token bucket filters, produces an output subdivided in bands even for constant rate flows: consecutive packets can be marked differently and thus end up in different queues. This might cause packet reordering and so many packet retransmissions.

The second type of meter, implemented with a rate estimator, ensures that as long as the rate remains constant, packets are marked in the same way. This meter might cause less packet reordering, but when the offered rate oscillates around a threshold, consecutive bursts of packets receive different priorities and the reactiveness of the control might decrease.

These considerations are confirmed by test results: the token bucket implementation causes a small degradation of the goodput with respect to the estimator case. The goodput reduction affects all users in the same way, so the fairness remains high but the mean rates are lower than expected. Nevertheless, the to-

ken bucket meter implementation reduces oscillations of rates with respect to the estimator case: all the variance indexes are remarkably lower.

For example the configuration of the test in figure 5.5, when tested with token bucket meters, obtained the results presented in figure 5.14. This difference is

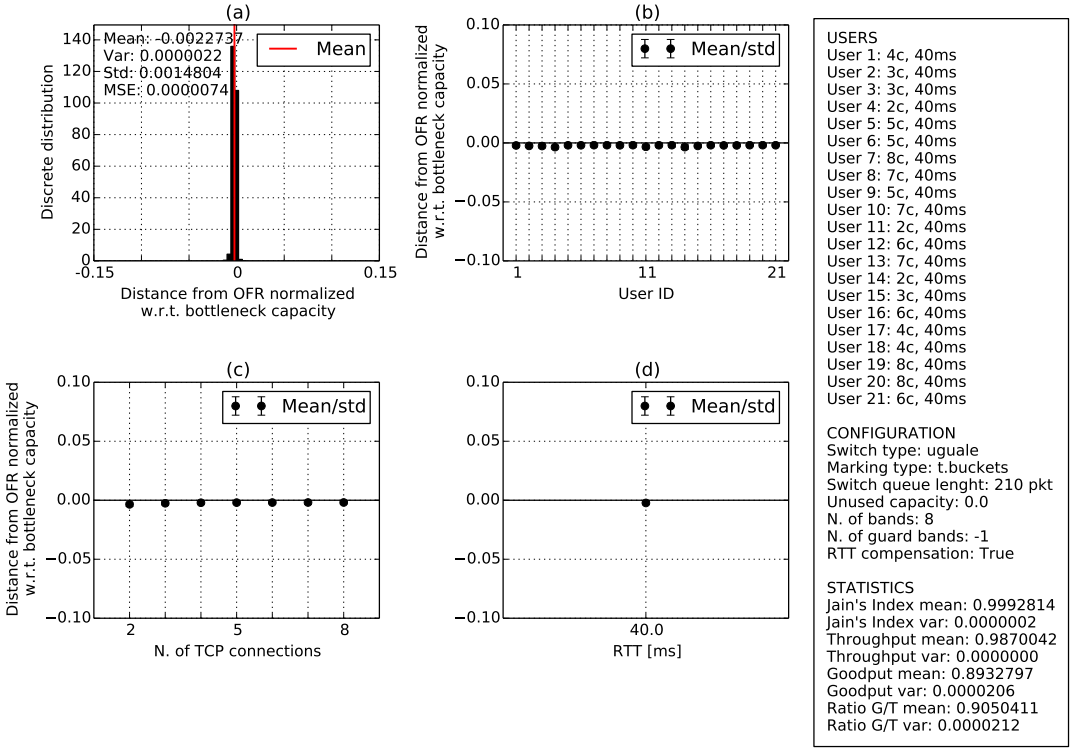

Figure 5.14: *The token bucket implementation of meters improves remarkably the variance but causes a reduction of goodput, so all rates are a bit lower than expected.*

highlighted by the results presented in table 5.1. The table shows the average statistics obtained in a set of tests executed with both meter types.

Meter	JFI mean	Goodput mean	Distr. mean	Distr. var
Token B.	0.985	0.908	-0.00153	$3.27 \cdot 10^{-5}$
Estimator	0.969	0.937	-0.00015	$8.42 \cdot 10^{-5}$

Table 5.1: *Average statistics of different meter implementations.*

5.3 Length of switch queues

A parameter that impact fairness and stability is the length of switch queues. If queues are long, packets have a big latency and the end-to-end congestion control mechanisms cannot react in short times. Generally, this causes unfairness and wide rate oscillations. By contrast, short queues keep the variance and the latency low. The default Linux queue length (1000 packets for each NIC) is too big for our testbed configuration. In fact, the best results were obtained when the queue lengths were set with the rule presented in [8]:

$$L = \frac{\overline{\mathrm{RTT}} \cdot c}{\sqrt{n}} \quad [\mathrm{bit}]$$

where:

- $\overline{\mathrm{RTT}}$ is the average RTT of users
- c is the link capacity
- n is the number of TCP flows on the link

Since even UGUALE improved using this rule, it was used in all tests presented.

Only when users have different RTTs, long queues enhance the fairness. In fact, the different reactiveness of users' end-to-end congestion control is smoothed and so the JFI is higher with respect to the equivalent case having short queues. An example is presented in figure 5.15: the test in figure 5.4 was repeated with long queues and fairness improved significantly.

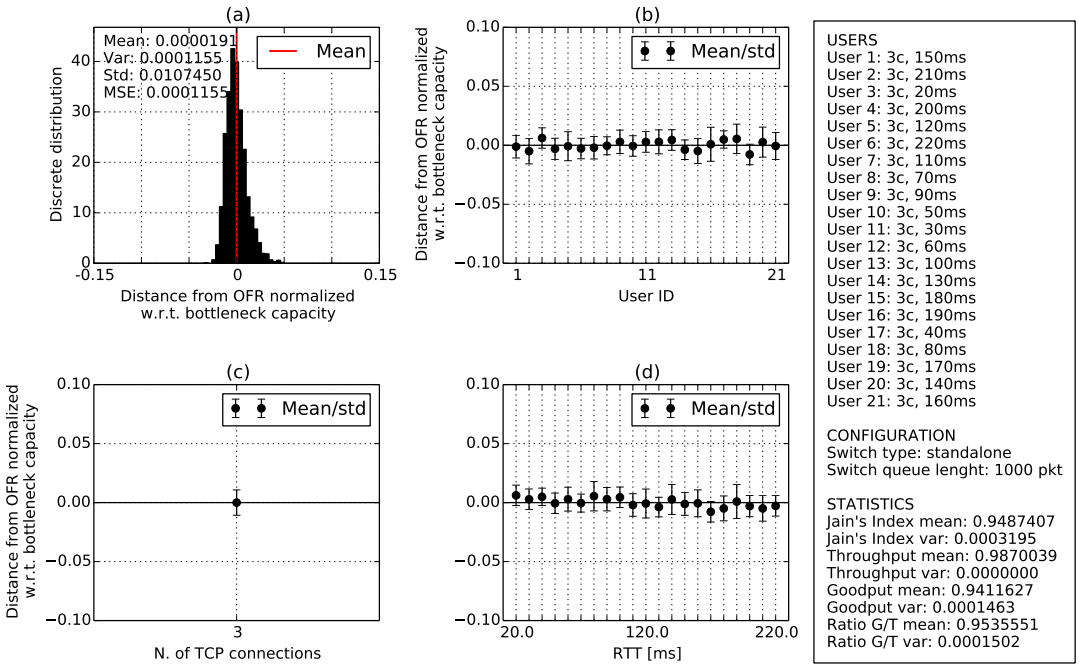

Figure 5.15: *Test without QoS enforcing, where users have different RTTs and the switch has long queues.*

Chapter 6

Conclusion

Bandwidth sharing is still an issue in todays networks, since it is difficult to provide fairness to a large number of users while fully exploiting the links' capacity. As discussed in the document, there are several solutions but they are either complicate to set up or not scalable, so practically the problem remains unsolved and Internet is still Best Effort oriented.

The bandwidth sharing problem was formally defined and the current approaches were analyzed in order to propose an original solution. The new bandwidth allocation scheme (UGUALE) works by dynamically assigning users to the queues of a Strict Priority scheduler, based on the measured input rate. In particular, the more traffic is offered, the less priority is obtained. This assignment interacts with the end-to-end congestion control mechanisms implemented by users and stabilizes the output rates around the desired ones. Since there are no theoretical models involving dynamic queue assignment, we decided to first assess the approach using experiments. To do so, a real testbed composed of five PCs was set up and one of those PCs was adapted to operate as the switch hosting UGUALE. The bandwidth sharing scheme was implemented using the abstractions made available by the OpenFlow switch model and APIs. Finally, in order to perform tests and collect data, several automation and monitoring scripts were written.

The results show that when bandwidth sharing is not enforced by the net-

work, users obtain a rate proportional to their number of TCP connections and inversely proportional to their RTT. By contrast, with UGUALE, users obtain their guaranteed rates, no matter the number of TCP connections instantiated or the RTT of their flows. We noticed that the fairness in the assignment of the spare capacity depends on the number and width of used bands. Hence, with a simple enhancement that concentrate thresholds close to the used rate interval, even the free bandwidth is optimally redistributed to elastic users. Moreover we observed that the goodput and the stability of rates are influenced by the type of meters and that performance depends on the length of switch queues.

In summary, the proposed allocation engine proved to be very effective in guaranteeing minimum rates and in the fair repartition of the excess bandwidth among users. In fact, UGUALE allows an easy assignment of bandwidth shares to users and the maximum link utilization using only the instruments already available in modern switches. In conclusion, the promising results obtained by UGUALE makes us believe that such an approach is worth of further analysis.

6.1 Future works

The proposed forwarding engine performs well in tested situations, but further cases and aspects need to be evaluated.

In static cases, users do not change their behavior during the test and the system is evaluated in a steady state. Until now, only a set of static cases have been examined but many others are necessary:

- A scalability test to discover the maximum number of users supported. The number of users should be increased until the performance drops significantly.

- Tests where users have different types of congestion control mechanisms (including inelastic users).

- Tests that clarify the effect of meters' parameters on performance. In particular, the update frequency of the estimator (for iptables meters) and the burst size (for token bucket meters) should be investigated.

- Tests using a real OpenFlow switch to verify the behavior of UGUALE with OpenFlow meters.

Moreover, other QoS parameters can be evaluated, such as delay and jitter. Then the dynamic performance of UGUALE should be studied by executing:

- Tests where the number of active users varies in time to evaluate the convergence time of rates to the OFR. Users should implement different congestion control mechanisms.

- Tests with realistic TCP traffic, i.e. composed of a combination of elephant and mice flows. Elephant flows emulate bulk transfers, as the long-lived connections already tested wit iPerf. By contrast, mice flows emulate HTTP traffic that is usually composed of many short bursts. Mice flows are very short-lived and end up transmissions before reaching their Slow Start Threshold (SSTHRESH). For this reason, other parameter such as the Flow Completion Time (FCT) can be considered.

Leaving the single node problem, the utilization of UGUALE in a network must be considered. More precisely, the network architecture to embrace it should be defined.

The first architectural option is to run UGUALE on every network node, so that fairness is guaranteed on each link. This solution might present some scalability and management issues, since all routers should know users' guaranteed rates and meter all packets.

Another option is to run UGUALE only on edge routers thus leaving a stateless version in core routers. Core routers should send packets on different queues based on the band previously chosen by edge routers. The band should be communicated

to the core routers by writing some data in the packet header, e.g. by marking the DSCP field. Nevertheless, to obtain fairness all packets sharing a link should be marked consistently. This solution seems more practical since it resembles the well-known DiffServ architecture.

In both the sketched architectures, a centralized control of the network is necessary to set thresholds in a coherent way. This control can be enforced using the standard network engineering protocols or more simply in the SDN domain. Moreover, if a reactive control mechanism were available, thresholds could be periodically tuned based on the network traffic so to enhance the performance of the system.

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