# Cross-Layer Flow and Congestion Control for Datacenter Networks (Invited Paper)

Andreea S. Anghel, Robert Birke, Daniel Crisan and Mitchell Gusat
IBM Research, Zürich Research Laboratory
Säumerstrasse 4, CH-8803 Rüschlikon, Switzerland
{aan,bir,dcr,mig}@zurich.ibm.com

***Abstract*—A key feature of the upcoming datacenter networks is their losslessness, achieved by the means of Priority Flow Control (PFC). Inherited from the cluster and HPC networks that traditionally use link level flow control to prevent packet loss across multiple virtual lanes, channels and/or priorities, this feature is now also becoming widely available in the next generation 10, 40 and 100Gbps Ethernet switches and adapters. Nevertheless, excepting storage protocols such as Fibre Channel over Ethernet, PFC is new and unfamiliar to the majority of datacenter applications and protocols. That is, despite PFC's key role in the datacenter and its increasing availability – supported by virtually all future Converged Enhanced Ethernet (CEE) products – its impact on the higher layer routing and transport protocols has yet to be investigated. Hence our motivation to assess the performance exposure of three widespread TCP versions to PFC, as well as to the potentially conflicting Quantized Congestion Notification (QCN) congestion management mechanism, which apparently replicates on Layer 2 some more advanced TCP functionality.**

**As workloads of interest we have selected a few revealing commercial and scientific applications. For quantitative performance evaluation we use two distinct methodologies: (a) Our reference is an accurate Layer 2 CEE 10Gbps network simulator intercoupled with TCP implementations extracted from FreeBSD v9; (b) A hardware setup scaled down in speed and size. The main outcome of our work is that PFC can notably improve the TCP performance across all tested configurations and workloads. This result was validated in both environments. Hence our recommendation to enable PFC whenever this is possible. By contrast, QCN can either harm or help depending on its parameter settings, and essentially, on the co-existence of competing UDP or other non-congestion-managed traffic.**

## I. Introduction and Motivation

Ethernet was originally designed to be a simple and inexpensive Plug-and-Play LAN solution. Nonetheless, it was not optimized for performance i.e. low latency, high throughput, traffic differentiation, low packet loss ratio. In order to fill the gap, HPC and datacenter architects have built specialized networks such as, Infiniband, Myrinet, Fibre Channel etc. While outperforming the low cost Ethernet, this specialized multiple network infrastructure is expensive to build and difficult to maintain. As a consequence, in order to save overhead, power and cost, a better solution was to converge all the datacenter traffic in one single consolidated network, i.e. Converged Enhanced Ethernet (CEE) datacenter network. One of the CEE key features is Layer 2 (L2) losslessness. This is achieved via per priority link-level flow control as defined by 802.1Qbb PFC [1]. However, the benefits of losslessness via PFC are not free of side effects. Besides the potential for multiple types of deadlocks in various common topologies and switch architectures, any stiff link level flow control may lead to saturation tree congestion, oscillations and critical instabilities. To counteract the potentially severe performance degradation due to such undesirable phenomena, IEEE has recently standardized a new L2 congestion control scheme capable to deal with sustainable network congestion, Quantized Congestion Notification (QCN, 802.1Qau) [2].

Future CEE datacenter networks will have to manage up to one million physical – and 20-300x more virtual – nodes connected in a single L2 domain via multipathing across 10-100 *Gbps* links of at most a few 10s of meters. The round-trip times (RTT) range from 0.5 $\mu s$ up to a few tens of $\mu s$, except under hotspot congestion, when the end-to-end delays can grow by several orders of magnitude, reaching into 10s of *ms* or more[3]. However, even if a congested DCN may have RTTs similar in magnitude with a WAN, there is a key difference. Unlike the wide area networks where the RTT is mostly due to flight delays, in datacenter environments the network RTT is dominated by queuing delays, which under bursty workloads [4], [5], [6], [7], [8], [9], [10], [11], lead to a difficult traffic engineering and control problem. Hence the recent surge in research and standardization efforts addressing the new challenges in datacenter virtualization, flow and congestion control, routing and high performance transports.

Apart from enabling the convergence of legacy and future datacenter applications, CEE datacenter networks must also meet the QoS requirements of a wide variety of applications such as business analytics, algorithmic trading, large scale Web services, cloud services (storage, computation), multimedia streaming, MPI workloads etc. These applications generate traffic with specific communication patterns. The bulk of the datacenter communications remains based on Layer 4 (L4) transport protocols, mostly TCP - with some notable exceptions such as UDP, SCTP or RDMA. In this paper we study the impact of the new CEE features on the typical datacenter applications performance. We focus on analyzing TCP commercial workloads that generate *scatter/gather* com-

978-0-9836283-2-3 



This paper was peer reviewed by subject matter experts for publication in the Proceedings of DC–CaVES 2011

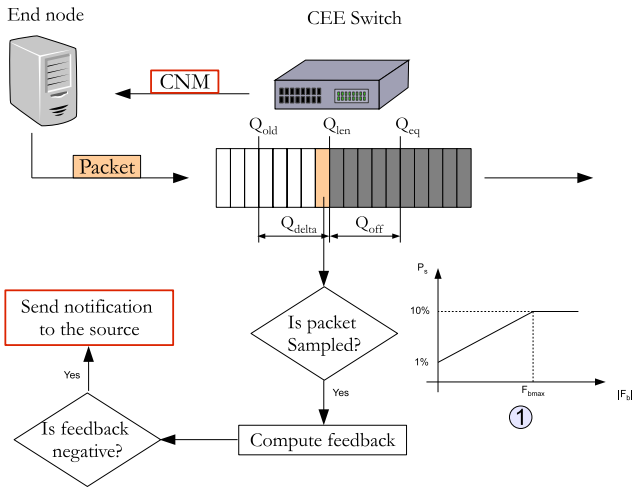



Figure 1. *QCN Congestion Detection – Load Sensor at Congestion Point*: It is implemented at the switch buffer level by monitoring the output queues. Its main objective is to keep the buffer occupancy at a certain level $Q_{eq}$. *Algorithm:* The QCN load sensor is sampling the input frames based on the probability function in 1. For each sampled frame, the load sensor measures the current queue occupancy level ($Q_{len}$) and computes the queue occupancy excess ($Q_{offset} = Q_{eq} - Q_{len}$), the queue rate excess ($Q_{delta} = Q_{old} - Q_{len}$) and the feedback value ($F_b = Q_{offset} + w \cdot Q_{delta}$). $Q_{old}$ is the occupancy level at the previous sample and $Q_{eq}$ is the target queue equilibrium (setpoint). If the computed feedback is negative a 64 B congestion notification message (CNM) frame, including a 6-bit hotspot severity value (feedback), is sent back directly to the source of the sampled frame.

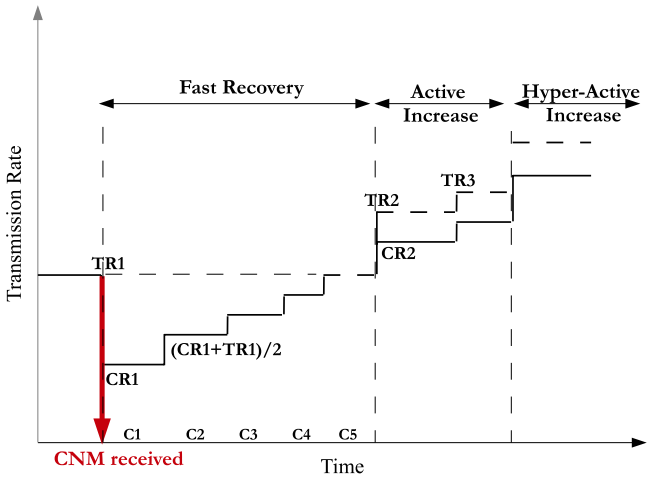



Figure 2. *QCN Rate Limiter (RL) – Rate Control at Reaction Point*: Upon receiving a CNM, a QCN-compliant source reacts by instantiating a rate limiter (RL) and setting the Target Rate (TR) to the Current Rate (CR). CR is the RL TX rate at any instant of time while TR is the RL TX rate just before the reception of a CNM. After that it adjusts its TX rate based on an AIMD-like strategy. *Algorithm*: Upon reception of a CNM, the RL reduces CR to $CR = CR * (1 - G * |F_b|)$. The higher the feedback value, the higher the rate reduction according to the rate decrease control law. The rate increase is performed in 3 phases: Fast Recovery (FR), Active Increase (AI) and HyperActive Increase (HAI). For each CNM received, the source resets a byte counter and enters the Fast Recovery phase to autonomously recover the lost TX rate. FR has 5 cycles (from C1 to C5), each of them consisting of 150 *KB* transmitted by the RL. At the end of a cycle, the RL CR is updated to $CR = 1/2 * (CR + TR)$. After 5 cycles in FR, the RL enters the AI phase. In this phase, the RL counts cycles of 75 *KB* each. At the end of a cycle, the target rate is updated to $TR = TR + R$ where R is 5 *Mpbs* and the current rate to $CR = 1/2 * (CR + TR)$. The byte counter is used simultaneously with a timer. They are executed independently, but they jointly determine the rate increase of the RL. The timer was introduced to speed-up the rate recovery especially when the RL current rates at the moment of a CNM reception, are small. Upon reception of a feedback message, the timer is reset and the RL enters the FR phase. It counts 5 cycles of $T = 10$ *ms* duration, and then enters the AI phase where each cycle is $T/2 = 5$ *ms* long. In each of the two phases, FR and AI, a cycle ends when either a byte counter or a timer completes its cycle. When both the timer and the byte counter are in the Active Increase phase, the RL is in the HAI phase. If the RL succeded to enter this phase, the network condition is considered stable and the probability for the RL to cause network congestion is small. Thus, CR and its corresponding RT have a higher growth rate: TR grows in steps of 50 *Mbps* and CR accordingly to the same increase law as in the AI phase.

munication patterns –aka, Partition/Aggregate– as described in [4], [6].

Assuming that TCP will continue as the paramount transport for the foreseable future, we ask three TCP-related questions: (**Q1**) How does TCP perform over CEE networks - does it work 'out of the box', or is tuning/adaption necessary? (**Q2**) Is PFC, with its correlated multiple hotspots and saturation trees, beneficial, neutral or detrimental to TCP, which traditionally has relied on loss as congestion feedback from *uncorrelated single* bottlenecks? (**Q3**) Is QCN's TCP-like L2 rate control beneficial, neutral or detrimental to TCP? By addressing these questions we hope to provide some useful insight to datacenter architects, network vendors, as well as operating system, hypervisor and application designers.

Our contributions are twofold, structured as follows.

**Methodology**: We use two evaluation methodologies presented in SECTION III and SECTION IV corresponding to: (i). simulation environment; (ii). hardware setup.

In the *simulation environment*: (i). In SECTION III-B we explain the network models including the necessary parameter changes for TCP over *10 Gbps* CEE; (ii). As benchmark applications, we have adopted three datacenter workloads, including commercial workloads (see SECTION III-C); (ii). At Layer 4, we extracted three TCP versions from FreeBSD v9, each representing a different class of TCP implementations, and ported them to our simulation environment: New Reno, Vegas and Cubic; (iii). At Layer 2, we instrumented our simulator with PFC and QCN. For performance evaluation we have mixed hardware and software experiments with detailed L2 simulation models of CEE switches and adapters.

In the *hardware environment*: (i). In SECTION IV we present the network models including the necessary parameter changes for TCP over FastEthernet networks; (ii). As benchmark applications, we implemented a version of commercial workloads; (ii). At Layer 4, we use three TCP modules already present in Linux Kernel 2.6.32 for New Reno, Vegas and Cubic; (iii). At Layer 2, we use PFC with one priority only (i.e. 802.3x PAUSE).

**Results**: In both SECTION V and SECTION VI we present evidence of PFC's benefits to TCP performance, leading to our recommendation to enable PFC also for TCP applications. The hardware testbed validates the PFC results obtained in the simulation environment. With respect to QCN, we identify cases in which it is beneficial, repectively detrimental to

| L4 TCP (Reno) | L2 QCN |
|---|---|
| **DETECTION MECHANISM** | |
| - Implemented (1) at the destination (duplicate ACK); (2) in the network at the congestion point (AQM/ECN); (3). at the source (RTO)<br>(1) For each packet loss the destination will ask the sender for retransmission by sending it a duplicate ACK. Upon reception of 3 duplicate ACKs, the source enters the fast recovery phase.<br>(2) When an ECN packet arrives at a RED-enabled congested queue, the router may mark the packet instead of dropping it.<br>(3) The RTO value is set larger than the maximum RTT value of the network. A packet loss can imply an RTO and then TCP enters the slow start setting the congestion window at 1 MSS. | - Implemented at the congestion point in the network.<br>- It monitors the L2 queues and in the event of congestion it sends a CNM to the culprit source node.<br>- The congestion point sensor randomly samples with a variable sampling probability the queue characterized by two state variables, position and velocity, defined with respect to a pre-established set-point $Q_{eq}$. Based on the queue size and velocity values, the CP evaluates the level of congestion by computing a congestion feedback. If negative, a CNM is sent back to the culprit of the sampled frame. |
| **FEEDBACK TYPE** | |
| Duplicate ACK<br>ECN/RED single bit feedback<br>Retransmission Time-out | If the feedback computed is negative, it is quantized on 6 bits (multibit). |
| **BURST TOLERANCE** | |
| Tolerant to bursts | The value of the setpoint $Q_{eq}$ determines how tolerant QCN is to bursty traffic, but overall QCN is less tolerant than established L4 CM schemes. |
| **TIMESCALE** | |
| 100s of *ms*. Depends on network RTT and default RTO. | 10s to 100s $\mu s$ |

Table I
CONGESTION DETECTION

L4 transports. The QCN performance proves to be highly dependent on the type of workload and its generated communication pattern. The same remark applies for ECN-RED: the simulation results show either benefit or detriment to TCP performance.

The rest of the paper is organized as follows. The CEE datacenter network stack is described in SECTION II. In SECTION III we present the simulation environment, the network models, the workloads used in our evaluation and the performance metrics. In SECTION IV we explain the hardware experimental setup, the corresponding network topology and the applications used for the hardware evaluation. In SECTION V we present simulation results and discuss their implications. The results obtained in the hardware testbed are shown in SECTION VI. Related work is presented in SECTION VII, after which we conclude in SECTION VIII.

## II. DATACENTER NETWORK STACK

### A. Layer 2 - Converged Enhanced Ethernet (CEE)

There is a growing interest in consolidated network solutions that meet the requirements of datacenter applications, i.e., low latency, no loss, burst tolerance, energy efficiency. One possible answer to the universal datacenter fabric is CEE with the following key features: (i) per-priority link-level flow-control and traffic differentiation, i.e., Priority Flow Control (PFC; IEEE 802.1Qbb) [1]; (ii) congestion management, i.e., Quantized Congestion Notification (QCN, optional in CEE; IEEE 802.1Qau) [2]; (iii) transmission scheduling: Enhanced Transmission Selection (ETS; IEEE 802.1Qaz).

*1) PFC:* Conventional IEEE 802.3 Ethernet does not provide reliability. Indeed, whenever a network device buffer reaches its maximum capacity, packets are dropped. Upper layer protocols such as TCP rely on these events and interpret them as congestion feedback, thus triggering window or rate adjustments. This goes against the strict QoS requirements of many of the nowadays datacenter applications such as FCoE, MPI or Business Analytics.

A mechanism that can avoid packet losses is IEEE 802.3x PAUSE [1]. This protocol pauses the transmitter side of an Ethernet link whenever the receiver side buffer occupancy exceeds a maximum threshold by means of explicit PAUSE XOFF control frames. The transmission resumes once a time-out expires or a PAUSE XON control frame is received. The PAUSE XON frames are produced when the occupancy falls below a given minimum threshold.

Two main drawbacks affect this solution: (i) no knowledge of service classes: pausing a link due to one application will affect all applications using that link; (ii) possible throughput collapses: if a sender is blocked, its buffer may get filled up and recursively other network devices will possibly get paused and will fill up as well, creating congestion spreading and thus a congestion tree which will drastically decrease the network throughput.

PFC addresses both by dividing the traffic into 8 different priorities based on the IEEE 802.1p class of service field. Inside each priority PFC acts exactly as 802.3x PAUSE, but one priority does not affect the others. This addresses directly drawback (i) and reduces the probability of drawback (ii).

However, incorrectly implemented, PFC can still generate switch memory-to-memory (M2M) circular dependency and routing deadlocks. Routing deadlocks can be avoided using a loop-free topology such as fat-tree topologies [12]. M2Ms are a more general issue and happen due to un-ordered access to mutually blocking resources. For instance if two shared-

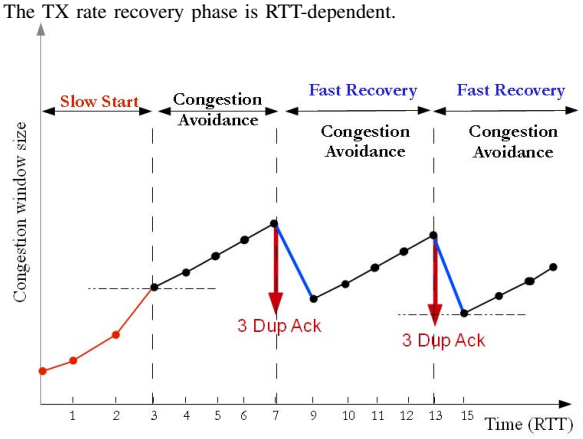
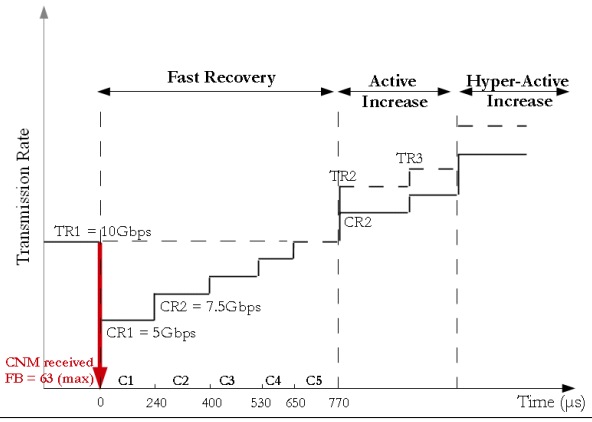
| L4 TCP (Reno) | L2 QCN |
|---|---|
| **PRINCIPLE OF OPERATION** | |
| - Implemented at the source - Window Controller.<br>- It uses a congestion window for controlling the injection rate. | - Implemented at the source - Rate-based Controller.<br>- It uses a Finite State Machine (based on a byte counter and a timer) and a Cubic like algorithm. |
| **INCREASE/DECREASE RATE CONTROL LAW** | |
| Additive Increase (AI) - Multiplicative Decrease (MD)<br>- A time-out reduces the current congestion window at 1 MSS and TCP enters the Slow Start phase. Three duplicate ACKs halve the TX congestion window, which is the standard TCP decrease law according to MD.<br>- The TX rate is recovered slowly (the duration is in the range of multipliers of RTT).<br>- The TX rate recovery phase is RTT-dependent.<br>  | Fast Recovery - Active Increase - Hyper Active Increase (Cyclic like)<br>- The reception of a CNM reduces the TX rate by a variable amount proportional to the feedback magnitude, up to maximum 50 percent of the CR.<br>- For a reference, 10 *Gbps* CEE with a singular maximum rate reduction (downto CR/2), the CR/2 rate 'drop' can be recovered in min. 770 $\mu s$.<br>- The TX rate recovery control phase is RTT-independent (unlike CUBIC, QCN does not exhibit RTT-based self-clocking).<br>  |

Table II
CONGESTION CONTROL

memory switches A and B exceed their thresholds and if all the traffic from A is destined to B and vice-versa, neither can send. This deadlock can be overcome by partitioning the switch memory and enforcing the thresholds on a per input basis.

*2) QCN:* QCN is an L2 congestion management technique that provides a set of mechanisms for controlling the network congestion generated by long-lived data flows. QCN detects congestion, reports it via explicit congestion feedback and adapts the source node injection rate to the available network capacity. QCN accomplishes these two steps by using two distinct algorithms.

One is the QCN load sensor mechanism which is implemented at the Congestion Point (CP) in the network. Its objective is to monitor the queues and, in the event of congestion, send a notification to the culprit source node. The congestion point sensor randomly samples with a variable sampling probability $P_s$ the queue characterized by two state variables, position and velocity $\{q, q'\}$, defined with respect to a pre-established setpoint $Q_{eq}$. The algorithm is explained in FIGURE 1.

The second algorithm is implemented at the Reaction Point (RP) (i.e. source node). It is called the *QCN Rate Limiter* and reacts to the congestion signals by reducing/increasing the transmission rate according to the feedback received from the CP. This algorithm uses a byte counter and a timer. FIGURE 2 shows how the RP algorithm adjusts the injection rate.

### B. Layer 3 - ECN - RED

Random Early Detection (RED) is an L3 Active Queue Management (AQM) congestion avoidance technique for packet-switched networks. Unlike QCN whose congestion estimation is based on instantaneous queue size measurements, RED detects network congestion by computing the average queue length and comparing it with a given threshold. The RED-enabled queue has a minimum and a maximum threshold. If the average queue length is below the minimum threshold, all incoming packets are forwarded unchanged. If the average queue length is above the maximum threshold, all the incoming packets are dropped. Finally, if the average queue length is between the threshold values, some of the incoming packets are dropped according to a liniar probability which is function of the average queue length. Out of the remaining packets, the ones that are ECN capable are *marked* [13]. An important advantage of using this mechanism is that by monitoring the queue and by keeping its length small, RED allows the network to absorb a limited amount of bursty traffic with little performance degradation. Unlike the QCN load sensor mechanism, RED is more tolerant to bursts as confirmed later in the paper.

Explicit Congestion Notification (ECN) is a L3-L4 end-to-end congestion management protocol defined in RFC 3168. The principle of this protocol is for the transmission node to receive network congestion notifications without the need for any packets to be dropped. In order to be operational and efficiently used, ECN has to be supported by both endpoints and also by the intermediate network devices.

ECN uses the two least significant bits of the Differential Services (DiffServ) field in the IP header. Once the communicating nodes have negociated ECN, the transport layer of the source node sets the ECN codepoint in the IP header of the packet and sends the packet to the destination. When the ECN-capable packet arrives at a RED-enabled queue that is experiencing a congestion, the router may decide to mark the packet instead of dropping it. Upon receiving the marked packet, the destination sends back to the transmission node an ACK packet with the ECN-Echo bit set. This way the transmission node receives a signal to reduce its transmission rate. The destination repeats sending ACK packets until the transmission node acknowledges having received the congestion indication.

### C. Layer 4 - TCP Versions

We have selected three representative TCP versions:

1) TCP New Reno [14] since it is both the best known and the most implemented one;
2) TCP Cubic [15] since it is the standard implementation in today's Linux kernels;
3) TCP Vegas [16] since it uses a distinct congestion control scheme than the others two i.e. delay-probing instead of loss-based congestion window adjustment.

TCP New Reno, like Reno, includes the slow-start, congestion avoidance, fast retransmit and fast recovery algorithms. Its congestion feedback is either packet loss and/or ECN-marked packets. TCP New Reno outperforms TCP Reno in the presence of multiple holes in the sequence space, but performs worse in case of reordering due to useless retransmissions. It was the standard TCP implementation in Linux kernels till version 2.6.8.

TCP Cubic is RTT-independent. It has been optimized for high speed networks with high latency (due to flight delays) and is a less aggressive derivative of TCP BIC (Binary Increase Congestion control). TCP BIC uses a binary search to probe the maximum congestion window size. TCP Cubic replaces the binary search with a cubic function. The concave region of the function is used to quickly recover bandwidth after a congestion event happened, while the convex part is used to probe for more bandwidth, slowly at first and then very rapidly. The time spent in the plateau between the concave and convex regions allows the network to stabilize before TCP Cubic starts looking for more bandwidth. TCP Cubic replaced TCP BIC as the standard TCP implementation in Linux kernels from version 2.6.19 onwards.

While both TCP New Reno and TCP Cubic rely on losses to detect congestions and react accordingly, Vegas avoids congestion by comparing the expected throughput in the absence of congestion with the actual achieved throughput and then it adjusts the transmitter window size accordingly. TCP Vegas is representative for the delay-probing class of TCPs similar to Adaptive Reno and Compund TCP.

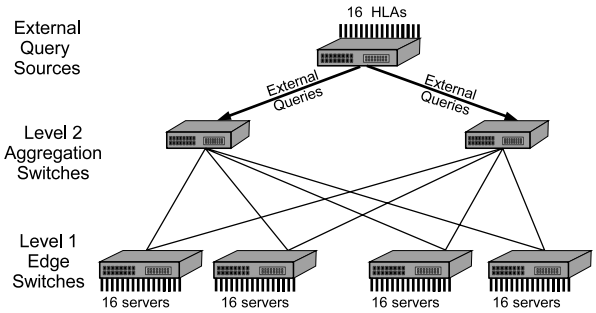



Figure 3. Two-tiered datacenter network topology with edge and aggregation switches and 64 end-nodes distributed in 4 racks. This topology is an XGFT(2;16,4;1,2). The 16 external query sources are acting as the HLAs for the *scatter/gather* communication pattern generated by the application. These external sources inject TCP queries in the datacenter network through the Level 2 aggregation switches.

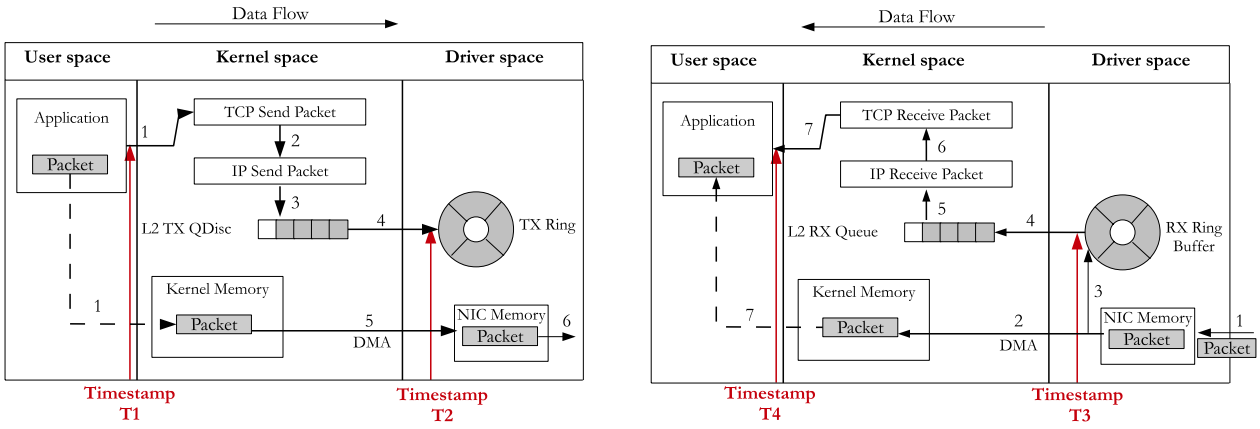
Table I shows a qualitative comparison between TCP and QCN with respect to the congestion detection mechanism. Table II shows a similar comparison but with regard to the congestion control scheme.

## III. Evaluation Methodology I: Simulation Environment, Models, Workloads

### A. Simulation Environment

A network simulation environment brings a set of clear advantages against the hardware experiments: (1). It gives access to a detailed overview of the behavior and interaction of different network protocols which may be unavailable on hardware equipments; (2). The scalability of the protocols can be tested with large-scale scenarios and configurations; (3). The protocols and the system design parameters can be easily modified in order to find the optimal system configuration which delivers the best performance results; (4). The experiments results can be easily replicated. On the other hand, there are deficiencies that have to be corrected in the case of a network simulation environment. The most relevant drawback is that it is difficult to integrate all the details from real environments and create realistic models for all the L2-L4 protocols and datacenter network components. Hence, for our simulations we chose to calibrate the simulation models by using hardware experiments. However, apart from the simulation environment, we also propose and test a hardware setup described in IV.

Even though there are already well established network simulation tools in the community such as NS-3 which has a large library of supported L3/L4 protocols, we chose not to use NS-3, but our own network simulation enviroment based on Omnet++ for the following reasons: (1). It has support for CEE standards including PFC, QCN and ETS; (2). It contains management and monitoring for the L2 queues and buffers, L2 scheduling, link-level flow control and memory management




(a) TX Stack Delay.

(b) RX Stack Delay.

Figure 4. *(a)*: The TX delay is measured as follows: the UDP traffic generator application adds a timestamp (T1) to the payload of the packet right before writing the packet to the L4 socket buffer (step 1); the driver adds a timestamp (T2) just before queuing the packet onto the TX ring buffer (a shared buffer between the driver and the NIC hardware) (step 4); the TX delay is computed as the difference between these two timestamps (T2-T1). *(b)*: The RX delay is measured as follows: the driver adds a timestamp (T3) to the payload of the packet that it receives from the RX ring buffer (again the same shared buffer between the driver and the NIC hardware) (step 4); the application adds a timestamp (T4) to the payload of the packet just after reading the packet from the L4 socket buffer (step 7); the RX delay is computed as the difference between these two timestamps (T3-T4).

at both switch and adapter micro-architecture level; (3). L4 protocols are ported from the BSD, AIX and Linux kernels and they are more complex than the TCP libraries in NS-3 which trade accuracy for simulation efficiency. Based on these features, our simulation environment is able to analyze the interaction between the L4 and L2 protocols. Our platform entails two simulators coupled in an end-to-end framework, Dimemas and Venus [17] and the TCP simulations are realistically calibrated against the actual OS stacks running on hardware.

### B. Network Models

*1) Datacenter Topologies:* A widely used topology for data center networks is the fat tree topology. A fat tree can be described as a multi-stage tree topology where the bandwidth of the connections increases towards the root of the tree [18]. The network topologies used in our simulations belong to the family of extended generalized fat trees (XGFT) [12] which is a more general formalization of fat trees. This family also includes other popular data center interconnection networks, such as k-ary n-trees and slimmed k-ary n-trees. An XGFT $(h; m_1, ...m_h; w_1, ...w_h)$ has $h+1$ levels of nodes divided into leaf nodes and inner nodes. There are $\prod_{i=1}^{h} m_i$ leaf nodes occupying level 0 and they serve as end nodes/servers. The inner nodes occupy levels 1 to $h$ and serve as switches. Each non-leaf node on level $i$ has $m_i$ child nodes and each non-root node on level $j$ has $w_{j+1}$ parent nodes.

For commercial workload evaluations we use two practical, albeit scaled down in size, XGFT topologies: *XGFT(2;32,4;1,2)* and *XGFT(2;16,4;1,2)*. The latter is shown in FIGURE 3. In the first scenario, described in SECTION V-B, we use the *XGFT(2;16,4;1,2)* topology and we inject only TCP traffic in the network. In the second scenario, mentioned in SECTION V-C, we inject both TCP and UDP traffic, but we use the *XGFT(2;32,4;1,2)* network topology.

For the MPI workloads using scientific application traffic we use two slightly different topologies: *XGFT(2;16,7;1,2)* and *XGFT(2;32,7;1,2)*.

*2) TCP Transport Model:* According to [19] TCP computes the RTO value based on two variables: a smoothed round-trip time average (*SRTT*) and an average RTT variance ($RTT_{var}$). The RTT of a packet is measured as the interval between sending a segment and receiving its corresponding ACK. Initially, when a first RTT measurement $RTT^1$ is made, $SRTT^1 = RTT^1$, $RTT^1_{var} = RTT^1/2$ and $RTO^1 = SRTT^1 + max(G, K * RTT^1_{var})$, where $K$ was originally set to 4, and G is the clock granularity in seconds. For the next RTT measurements, $SRTT^N$ and $RTT^N_{var}$ are calculated as follows: $RTT^N_{var} = (1-a) * RTT^{N-1}_{var} + a * |SRTT^{N-1} - RTT^N|$ and $SRTT^N = (1-b) * SRTT^{N-1} + b * RTT^N$ where $0 \leq a < 1$ and $0 \leq b < 1$. Once the source has computed these values, it will update the RTO according to $RTO^N = SRTT^N + max(G, K * RTT^N_{var})$ *(Eq. 1)*.

In order to realistically model the TCP transport protocol, we ported the TCP stack from the FreeBSD v9 kernel in our simulation environment. Further calibration was needed to adapt the TCP parameters to the actual network characteristics. Thus we changed three parameters: (1) the CPU clock tick rate of the system; (2) the RTO default values and, (3) the RTO variance. Similar changes will be considered for the hardware testbed in IV.

(1) *CPU clock tick rate*: In our simulations we use a network with 10 *Gbps* links. Given this and the topology, the RTT of the network is small, in the range of tens of microseconds. The kernel timer quanta is set by default at 1 *ms*. By using this default coarse-grain kernel timer, the RTT measurements cannot be accurate. The accuracy of the RTT estimation is critical especially for delay-probing TCP protocols such as

Vegas and Compound TCP [20]. Compared to Cubic and New Reno which react to congestion upon packet loss, TCP Vegas detects congestion by comparing the measured throughput with an expected one, whose computation is RTT-dependent. Thus, TCP Vegas relies on fine resolution timers for obtaining accurate timing information to compute the actual and the expected throughputs, and to accordingly adjust its congestion window. As a consequence we increased the timer granularity from 1 *ms* to 1 $\mu s$. A higher gramularity time also contributes to correctly compute the RTO value (see *Eq.* 1).

(2) *Default RTO values*: In addition to the CPU clock tick rate calibration, we also had to adapt the default RTO value to the actual RTT value of the network. Based on our network measurements and [10], [7], we reduced the value of the minimum RTO to 2 $ms$. The kernel default value for the RTO value was 3 $s$. We reduced the default RTO to 10 $ms$, larger than the maximum RTT of our network, so that we could avoid the Flow Completion Time (FCT) to be drastically penalized by a SYN packet loss when PFC is disabled.

(3) *RTO variance*: The retransmission time-out is computed by using the Van Jacobson's timer estimator which takes into account the variance measured in the RTT values. A constant term is added to the estimation, accounting for the variance in segment processing at the end-point kernels. Hence, in order to match the current fast processors, we also modified the term *K* in *Eq.* 1, from the default FreeBSD value of 200 $ms$ to 20 $ms$.

Nevertheless, by porting TCP to our simulation environment, we had to perform some other optimizations. Some of them are related to the allocation and deallocation of segments. Others are: (1). *Connection TCP parameter cache*: the congestion window and the RTT estimation are inherited from the previous connection; (2). *Adaptive TCP buffers*: the TCP buffers sizes are dynamically modified in response to an RTT change.

*3) TCP Stack Delay Evaluation:* When a packet is sent from an application, it is first written to a TCP send buffer and copied to the kernel memory. Then it is pushed to the IP layer and, if the transmission queue *tx_qdisc* is not full, it is succesfully enqueued. After that the driver removes the packet from the *tx_qdisc* and maps it into the transmission descriptor ring buffer called *tx_ring*. Finally, if resources are available, the NIC consumes the packet from the *tx_ring,* copies it to its memory using DMA and sends it out on the wire.

When a packet is received, the NIC transfers the packet via DMA into a free descriptor taken from the reception descriptor ring buffer called *rx_ring*. Once fully transfered an interrupt is raised to signal the new packet to the device driver. The device driver takes the packet out of the *rx_ring* and sends it to the network stack. In order to reduce the system overhead, while being processed by the upper layers, the packet remains in kernel memory. The packet is first consumed by the IP layer and then, if it is a TCP packet, it is forwarded to the TCP receive process. Finally the packet is eventually consumed by the application.

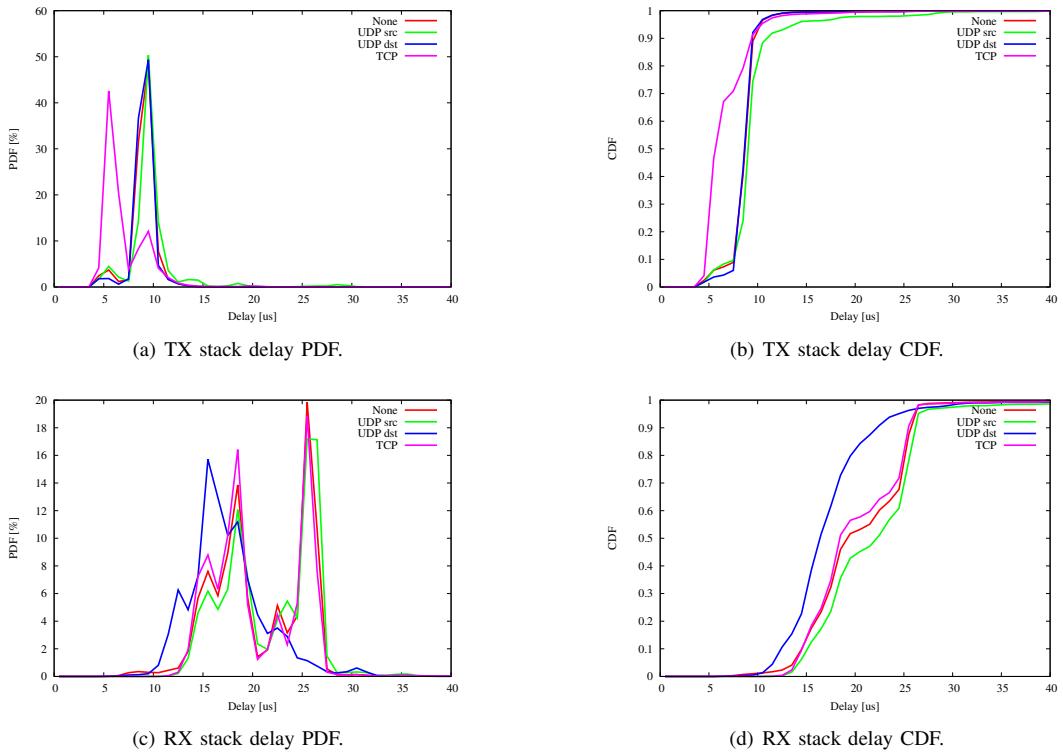
We determined the delay of the TCP stack by modifying a Linux 2.6.32.24 kernel running on an Intel i5 @ 3.2GHz machine with 4 GB of memory. We instrumented the E1000e Ethernet device driver of an Intel 82578DM Gigabit Ethernet controller. The stack delay was measured both in transmission and reception. The transmission delay was measured as the time spent between the application sending the packet and the moment the packet being enqueued in the *tx_ring*. Similarly, the reception delay was measured as the time spent from the moment a frame being ataken out of the *rx_ring* and the application receiving the data. Unfortunately the delay introduced by the NIC hardware operations is not measurable from software. For details see FIGURE 4. The average values are reported in TABLE III and the probability distributions are shown in FIGURE 5.

Table III
SIMULATION NETWORK PARAMETERS

| Parameter | Value | Unit | Parameter | Value | Unit |
|---|---|---|---|---|---|
| ***TCP*** | | | ***Other*** | | |
| buffer size | 128 | KB | TX delay | 9.5 | $\mu$s |
| max buffer size | 256 | KB | RX delay | 24 | $\mu$s |
| default RTO | 10 | ms | timer quanta | 1 | $\mu$s |
| min RTO | 2 | ms | reassembly queue | 200 | seg. |
| RTO variance | 20 | ms | | | |
| ***ECN-RED*** | | | | | |
| min thresh. | 25.6 | KB | $W_q$ | 0.002 | |
| max thresh. | 76.8 | KB | $P_{max}$ | 0.02 | |
| ***QCN*** | | | | | |
| $Q_{eq}$ | 20 or 66 | KB | fast recovery thresh. | 5 | |
| $W_d$ | 2 | | min. rate | 100 | Kb/s |
| $G_d$ | 0.5 | | active incr. | 5 | Mb/s |
| CM timer | 15 | ms | hyperactive incr. | 50 | Mb/s |
| sample interval | 150 | KB | min decr. factor | 0.5 | |
| byte count limit | 150 | KB | extra fast recovery | enabled | |
| ***PFC*** | | | | | |
| min thresh. | 80 | KB | max thresh. | 97 | KB |
| ***Network hardware*** | | | | | |
| link speed | 10 | Gb/s | adapter delay | 500 | ns |
| frame size | 1500 | B | switch buffer size/port | 100 | KB |
| adapter buffer size | 512 | KB | switch delay | 100 | ns |

*4) Simulation Network Parameters:* The network link speed is 10 Gbps. In accordance with the 802 Data Center Bridging reference switch model, we have assumed a canonical ideal input buffered (IB), output queued (OQ) switch with 100 *KB* per input buffer. While both of the PFC thresholds are defined with regard to this input buffer, the QCN setpoint is defined with respect to the output queues. Practically, however, such an OQ is emulated by the means of input buffered (IB) and crossbar hardware switches, which may not provide a dedicated hardware OQ - which was the key element for L2 QCN (the QCN algorithm implemented is based on version 2.4 [2]). In addition, our switch has *N* input buffers and the write bandwidth into an OQ is equal to $N$ times the line rate. The size of an OQ is bounded only by the sum of all the input buffers for all ports. However this is not true in reality, because a single output queue can not monopolize the full memory.

In addition, each input adapter provides one virtual output queue (VOQ) for each destination. This avoids most of the head-of-line blocking, which can be further exacerbated by the QCN rate limiters. TABLE III contains the key parameters




(a) TX stack delay PDF.

(b) TX stack delay CDF.

(c) RX stack delay PDF.

(d) RX stack delay CDF.

Figure 5. Linux 2.6.32.24 Kernel Stack Delay Results

of the network.

### C. Applications, Workloads and Traffic

As stated at the onset, we aim to evaluate the impact of different L2, L3 and L4 protocols on performance at the application level, centered on revealing the TCP sensitivities to the two new CEE features: PFC and QCN. Therefore we have selected a few key datacenter applications, divided in two groups: commercial and scientific workloads.

*1) Commercial Applications:* For evaluation we focus on commercial workloads that generate the *scatter/gather* (*partition/aggregate*) traffic communication pattern. This pattern is currently used by many large scale web applications such as web search or social networking applications. The traffic study from [4] confirms this finding. In addition, *scatter/gather* is a good candidate to analyze the challenges that TCP has to deal with in current datacenter networks.

The principle of operation of this communication pattern is as follows: (1). A central process receives a query/request (i.e. a web search request); (2). It divides the query into subqueries (i.e. parts from the web search index) and sends them to multiple workers (*scatter phase*); (3). The responses from the workers are collected by an aggregator (*gather phase*); (4). The aggregator merges the responses into a single response message which is sent back to the source of the query. This process might repeat on the workers in the sense that a certain worker might further segment the subquery and dispatch the segments to a second layer of workers. We obtain thus several layers of scatter of queries and gather of responses. In order to meet the latency requirements of the user application, the workers are assigned tight deadlines to provide the answer to the query. However, TCP can prevent the workers from meeting the deadlines thus reducing the relevance of the final response. Moreover, the *gather phase* of the pattern is intrisically prone to generating bursty traffic thus increasing the probability to produce network congestion and lose packets. Depending on what L4 congestion control scheme it is used at the end nodes, packet loss can increase dramatically the flow completion time.

In our simulations we use a three level *scatter/gather* pattern. The High Level Aggregators (HLA) are placed as in 3. The HLAs generate queries triggered from HTTP requests. The queries follow the inter-arrival time distribution in [4]. In the scatter phase, the HLA contacts multiple Mid-Level Aggregators (MLA) – one in each rack – and sends them a subquery. Upon reception of a subquery, the MLA will distribute it to all the other servers in the same rack. Later, the MLA will collect all the responses from the workers and send the aggregated response to the HLA.

We built a traffic generator that injects *foreground traffic* composed of latency critical queries (*mice*) on top of background traffic of long-lived flows (*elephants*). As previously mentioned, the foreground traffic is generated by the HLA. The queries have a fixed size i.e. 20 $KB$ and they follow the inter-arrival time distribution in FIGURE 6. The background traffic is composed of both short and long-lived flows. The background flows follow the inter-arrival time and size distributions in

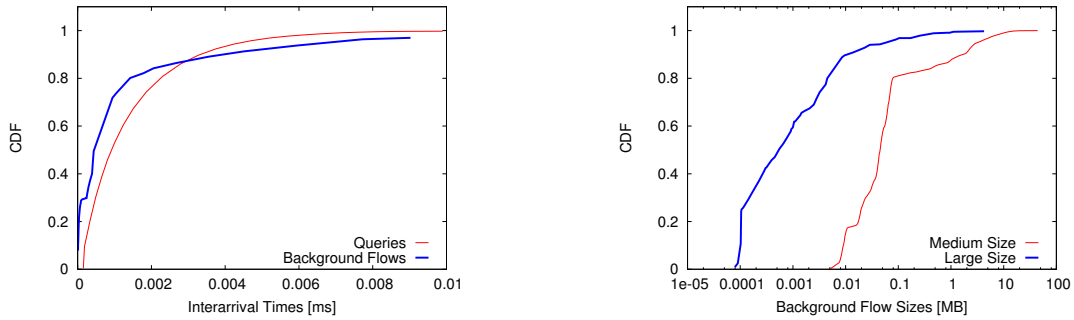



Figure 6. Flow inter-arrival and size distributions. For background flows we use the inter-arrival time and flow size distributions given in [4], [5]. The queries (foreground traffic) follow the inter-arrival time distribution from [4] accelerated 100×.

FIGURE 6. Each source randomly chooses a destination in order to match the ratio between the intra-rack and inter-rack of 30. For both foreground and background flows we measure the flow completion time (FCT) as application level metric [21]. SECTION III-D explains why we used FCT as the main performance metric for our experiments.

The traffic generator that we designed is based on findings from a few recent papers. In [6] the authors instrumented 19 datacenters to find evidence of *ON/OFF* traffic pattern behavior. They collected two datasets: (i). coarse-grained data such as number of bytes sent and received, errors and number of discarded packets; (ii). fine-grained data by using traces from a packet sniffer. The results showed: (i) higher utilisation of the links in the core than in the aggregation/edge levels; (ii) higher packet loss towards the edge links than in the core; (iii) the edge switches generate ON/OFF traffic with the duration of the ON/OFF periods and with the packet inter-arrival time during the ON period following lognormal distributions. In-depth studies of spatial and temporal distributions of the flows are performed in [4], [5].

*2) Scientific Applications:* We selected nine different MPI applications that were run on the MareNostrum cluster at the Barcelona Supercomputing Center. The MPI calls of the applications were recorded into trace files that fed our end-to-end simulation environment [17]. We assume that the MPI library on each processing node uses TCP sockets as underlying transport. The collected traces are in the order of a few seconds. Therefore, the TCP socket between a source and a destination of an MPI communication is opened only once, when the first transfer occurs and it is kept open during the entire run of the trace. Five of these applications belong to the NAS Parallel Benchmark [22]: BT, CG, FT, IS and MG. This benchmark aims to measure the performance of highly parallel supercomputers. In addition we used another 4 applications for weather prediction (WRF), parallel molecular simulations (NAMD) and fluid dynamics simulations (LISO and Airbus).

### D. Performance Metrics

There are various metrics that we could use to evaluate the TCP performance over the CEE protocols: throughput, delay, packet loss rates, fairness, response time to congestion, flow completion time etc. In this paper we will evaluate the TCP performance from the user perspective, thus we will use as performance metric the flow completion time (FCT). FCT was proposed in [21] which shows why the flow completion time is the best metric to evaluate congestion control algorithms. However, defining this metric can be challenging: (i). How is a flow defined, a problem still open for L2? (ii). How is completion defined? (iii). What benchmarking measurement methods are used? Generically, FCT is defined as the time from when the first packet of a flow is sent (the SYN packet in TCP) until the last packet is received [21]. In our simulations, FCT will refer to the amount of time from when a query is received in the datacenter network (for instance a Web search query) until the query is resolved.

For measuring the variability of the FCT results, we use the Coefficient of Variation (CoV), defined as the ratio between the standard deviation and the mean, in percentage. It is used to measure variability in relation to the mean of the distribution. We choose to use it because: (i). It is a traditional measure in engineering which is fairly easy to understand and which does not imply complex mathematical computations; (ii). It is scale free; (iii). It summarizes variability. However, it is sensitive to outliers.

## IV. EVALUATION METHODOLOGY II: HARDWARE ENVIRONMENT, MODELS AND WORKLOADS

### A. Hardware Environment

For completeness of the results and in order to validate those obtained in the simulation environment, we built a hardware testbed whose structure is shown in FIGURE 9. However, even if the hardware experiments are more accurate and two-three orders of magnitude faster than the simulations, the simulation environment remains our main instrument. Indeed, the hardware experiments are not: (i) Observable: the queues in the switches and the adapters in the network are not measurable; (ii) Available: the switches and the end nodes are partially CEE-enabled (they have support for PAUSE only); (iii) Flexible: the PAUSE settings are not configurable and the architectural changes on the end nodes are difficult, if possible at all [9]; (iv) Repeatable: scheduling of multi-threaded CPUs and asynchronous events, such as interrupts and I/O, is not

deterministic for our purpose. Nevertheless, we configured the Linux systems of the end nodes and we analyzed the impact of the PAUSE-enabled network on their TCP performance.

### B. Testbed

The testbed comprises 10 laptops, 1 desktop and 3 switches. The laptops are equipped with Pentium M @ 1.3 *GHz*, 1 *GB* of memory and an Intel Fast-Ethernet adapter. Instead, the desktop comprises a Pentium 4 @ 3 *GHz*, 3 *GB* of memory and an Intel Gigabit Ethernet Adapter. All PCs run Ubuntu 10.04 with a customized v2.6.32 kernel (more details are given in Section IV-C4). The switches are unmanaged 8-port consumer Fast-Ethernet switches. They conform to IEEE 802.3, 802.3u and 802.3x standards and they have support for link level flow control (i.e. 802.3x PAUSE).

### C. Hardware Models

*1) Topology:* The network is built in a tree topology as shown in FIGURE 9. The ten laptops are placed at the leaves of the tree, while the desktop is placed at the root of the tree. The 3 switches constitute the inner nodes of the tree. All links in the network have the same speed: 100 *Mbps* full duplex.

At a larger scale (more in an XGFT topology), this type of topology is typical for datacenters. It allows us to easily generate congestion spreading and it maps perfectly on the *scatter/gather* communication pattern. Given the simplicity of the topology and capabilities of the switches, no special routing algorithm is being used.

*2) Layer 2 :* Given the available resource capabilities, we emulate PFC with one priority through the use of PAUSE. This is consistent with our simulations of a single priority PFC network.

The tested switches have built-in support for 802.3x PAUSE which is always active (the switches are unmanaged), so no modifications were required at the switch level. However, PAUSE support at the host nodes is disabled by default. Therefore modifications to the network adapter driver were necessary to add the option to enable/disable flow control while loading the driver module (i.e. e100).

We tested the correct functionality of PAUSE both of the switches and of the network adapters with the modified driver using the following methodology:

1) We send ping tests from the controller towards a PAUSE enabled end node.
2) In parallel with the ping test we generate PAUSE frames in software with the PAUSE timer set to its maximum value (i.e. 65535).
3) We observe the increase in the measured RTTs.

If PAUSE is active we expect an increase in the measured RTTs due to the added PAUSE timer delay $D_P$. The PAUSE timer is expressed in quanta of 512 bit times on the wire, hence at 100Mbps this delay is $D_P = \frac{512}{100x10^6} * 65535 = 0.33\ s$ and therefore easily observable compared to normal RTTs in the order of *ms*. To test the network adapter we directly connected an end note and the controller, while to test the switch we

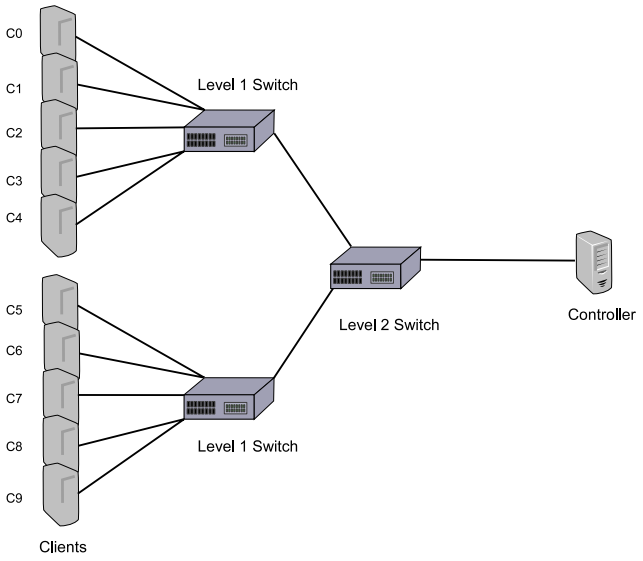



Figure 9. *Hardware Testbed Topology*: Two-tiered network topology with edge and aggregation switches and 10 end nodes. The controller at the root of the tree is acting as the HLA from the *scatter/gather* communication pattern, while the workers are represented by the 10 end nodes. The TCP queries are injected in the network through the Level 2 switch.

inserted a switch between them and disabled PAUSE frame generation on the end node network adapters.

QCN can not be tested as it is not supported neither in the switches, nor in the end nodes.

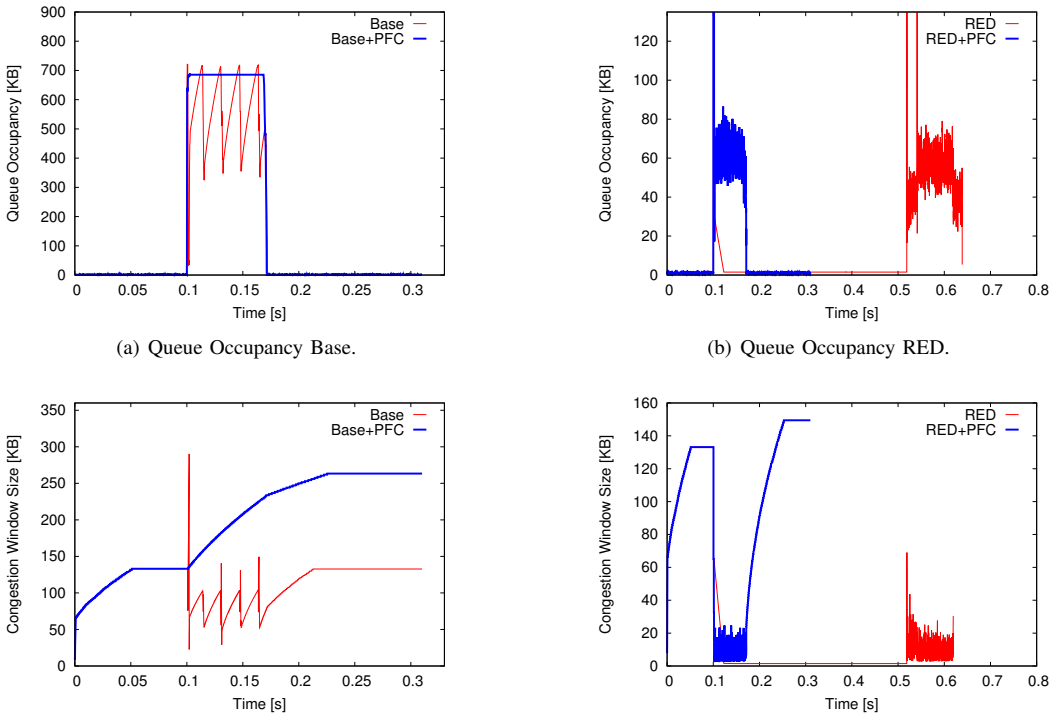
*3) Layer 3:* RED is available as a kernel module at the end nodes, but the switches do not support it. Hence, we do not recreate the L3 protocols tests presented in the simulation sections.

*4) Layer 4:* The Linux kernel 2.6.32 has built-in support for TCP Reno, TCP Vegas and TCP Cubic. TCP Cubic is the default TCP congestion control scheme at boot time. However, TCP Cubic can be easily changed at runtime by setting the /proc/sys/net/ipv4/tcp_congestion_control file to the appropriate TCP version. Thus, we have access to the same three TCP versions presented in the simulation environment.

Before running the experiments we performed some TCP tuning similar to those presented in III-B2. We changed the default RTO values from the maximum $120s$, minimum 200 $ms$ and initial 3 $s$ respectively to 10 $s$, 10 $ms$ and 30 $ms$. Furthermore, to increase the reactiveness of the operating system, we increased the CPU clock tick rate to 1000 $Hz$. Initially the clock interrupt rate was set at 250 $Hz$. [23] shows the impact of a proper CPU clock rate calibration on the scheduling mechanisms performed by an operating system.

*5) Layer 7 and Workload:* In the hardware environment we consider only the commercial applications described in III-C1. To re-create those workloads, we implemented a prototype application that simulates the *scatter/gather* communication pattern. The application is developed in JAVA and uses the JAVA multi-threading support.

In our hardware testbed we assume that the HLA coincides




(a) Queue Occupancy Base.

(b) Queue Occupancy RED.

(c) Congestion Window Base.

(d) Congestion Window RED.

Figure 7. Congestive synthetic traffic: many to one. 7 TCP sources send to the same destination. From $t_0 = 0$ *ms* to $t_1 = 100$ *ms* admissible offered load. $t_1$ to $t_2 = 110$ *ms* burst: all 7 sources inject a 4 times overload of the destination sink capacity. After $t_2$ admissible load. The congestive event extends past $t_2$ due to backlog draining.

with the MLA located at the root of the tree topology. A query is generated by the MLA and sent in parallel to all the leaf nodes. All queries have the same size (i.e. 20 *KB*) and follow the inter-arrival time distribution as described in III-C1. The leaf nodes are passive listeners. Once they receive a query from the controller, they run a four-step algorithm:

1) They randomly draw an inter-arrival duration $D$ from the inter-arrival times distribution in FIGURE 6;
2) They simulate the duration of searching the answer to the query by waiting for $D$ *ms*;
3) They randomly draw a size $S$ for the answer from the flow size distribution in FIGURE 6;
4) They send back the answer of $S$ *KB* to the controller.

All communications use TCP sockets as the underlying transport.

Preliminary tests of the application showed that the inter-arrival time distribution cannot be guaranteed. Indeed, we noticed that on the available resources, the high-granularity inter-arrival times used by the hosts cannot be insured by the regular JAVA multi-threading schedulers. One possible way to mitigate this problem would be to use the real-time multi-threading support offered by JAVA, but we did not consider it for this work. Instead, we simplified the commercial workload by synchronizing the queries among the clients inserting barriers between queries: i.e. the MLA waits for all answers before sending out a new query to all leaf nodes. Even if this workload is slightly different in terms of inter-arrival time distribution from the commercial workloads presented in III-C1, it remains representative and we will be referred to as *commercial*.

We use a constant number of 2000 transactions per run. We do not run any background traffic in the network.

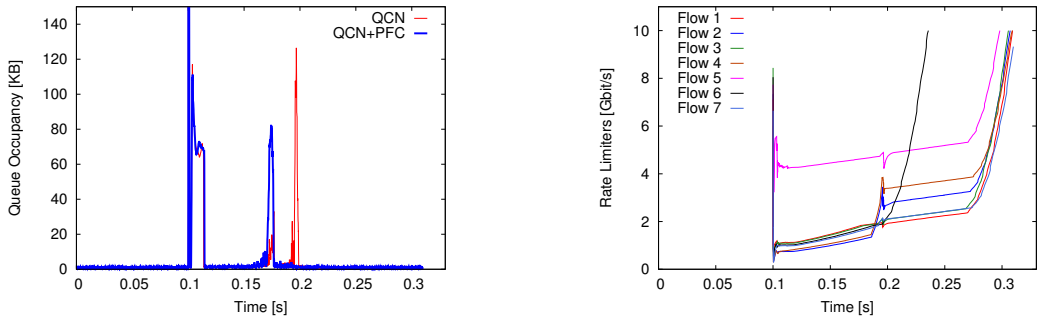
## V. SIMULATION RESULTS AND DISCUSSION

In this section we use four notations: *Base* – no congestion management (no CM) scheme; *QCN 20/66* – Quantized Congestion Notifications (QCN) enabled with $Q_{eq}$ = 20 *KB* or 66 *KB* respectively; *RED* – Random Early Detection with Explicit Congestion Notifications (ECN-RED) enabled. We run each of these congestion management schemes with or without PFC and with different TCP congestion control schemes: New Reno, Vegas and Cubic.

### A. Congestive Synthetic Traffic

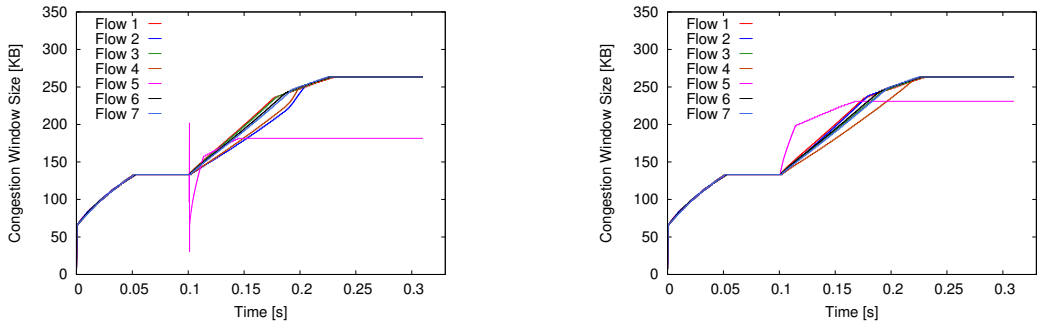
To debug the simulation model and calibrate our expectations, we run preliminary tests with TCP New Reno in a congestive synthetic traffic scenario, described in FIGURE 7 and derived from the 802.1Qau input-generated hotspot benchmark.

*1) Base:* We first analyze the network behavior in the absence of any congestion management scheme other than those used by standard TCP. On the switches, the evolution of the queues, when congestion is present, takes the shapes outlined in FIGURE 7(a). On the end nodes, we monitor the




(a) Queue Occupancy QCN 66.

(b) Rate Limiters QCN 66.




(c) Congestion window QCN 66.

(d) Congestion window QCN 66 + PFC.

Figure 8. Congestive synthetic traffic: many to one. Same traffic pattern as in FIGURE 7. The rate limiters for the QCN+PFC configuration are not shown because they exhibit the same unfairness as those without PFC i.e. flow 5 gets more than 40% of the bandwidth.

evolution of the TCP congestion window showed in FIGURE 7(c). When no flow control is enabled (red curve), the size of the congestion window has a periodic, sawtooth like, evolution that is a consequence of the TCP fast retransmit approach to window size management. On the contrary, when PFC is enabled (blue curve) there are no more dropped packets. This comes at the cost of an abrupt augmentation of the RTT (as can be seen at $t_1 = 100$ *ms*) which in turn leads to the buffer management mechanism increasing the reception buffer. This will lead to the increase of the source transmission window size.

*2) ECN-RED:* FIGURE 7(b,d) show the queue occupancy and one TCP source congestion window evolution, respectively, when using ECN-RED.

By enabling PFC (blue curve) we observe the same behavior as in the base scenario except for the queue occupancy which is much lower during the congestive event. Due to the ECN feedback, we also notice that the TCP source reduces its transmission rate by adjusting its congestion window. On the other hand, by disabling PFC we observe that additional network feedback i.e. ECN can have a negative impact on the network performance. Indeed, disabling PFC leads to lower ECN-RED performance and the throughput decreases between $0.17s$ and $0.53s$. In congestion phase, the queues fill up and ECN feedback is sent back to the TCP culprits. Upon reception of ECN packets, the TCP sources enter Congestion Recovery, but it is too late to avoid losses as the traffic has been already injected into the network. Since the sources are in recovery phase, the received duplicate ACKs are ignored and no Fast Retransmit happens before the retransmission timeout expires.

*3) QCN 66:* FIGURE 8(a,b) show the congested queue and the evolution of the QCN rate limiters, respectively. FIGURE 8(b) shows clearly one of the main QCN drawbacks i.e. unfairness [24]. Out of the seven flows, there is only one flow that monopolizes more than $40\%$ of the link capacity and succeds to finish its transmission first. However, the other flows still cannot increase their injection rate because they are in recovery phase. FIGURE 8(c,d) show the congestion window with and without PFC, respectively. QCN per se is capable of avoiding all the losses but one (of the 'winner').

### *B. Commercial Workload with and without TCP Background Traffic*

The communication traffic pattern is described in SECTION III-C1. FIGURE 10 shows the average flow completion time and its corresponding coeficient of variation (CoV) for query traffic (*mice*) without background traffic. FIGURE 11 and FIGURE 12 show the average flow completion time in two background traffic flow sizes scenarios respectively: FIGURE 11(a) and FIGURE 11(b) illustrate the average query completion time and the average background FCT for medium sized background traffic, while FIGURE 12(a) and FIGURE 12(b) show the average query FCT and the average background FCT for large sized *elephants* traffic.

*1) TCP Vegas:* TCP Vegas adjusts its congestion window based on RTT estimation and not on packet losses. PFC would be effective only when flows experience drops which Vegas avoids. Hence, PFC does not play any role in this L4 congestion control scheme. The same behavior is observed for QCN 20/66 and ECN-RED. Vegas does not reveal any impact in the query completion time. The coefficient of variation of the query completion times is relatively constant and small showing stable network conditions and high level of confidence intervals for the obtained results.

*2) TCP Cubic and TCP New Reno:* Cubic [15] performs worse than New Reno and Vegas in this environment. Overall we observed that the aggressive increases in the congestion window generate more losses than New Reno, therefore drastically penalizing the query completion times. Another factor which influences this result is the Cubic RTT independence which leads to increased losses and poor performance in dynamic networks like datacenter environments. Indeed, by default, Cubic was designed for network with high RTT values with long in-flight delays, whereas in our datacenter environment the RTT is small and mostly related to queuing delays and less to flight delays. In contrast to TCP Vegas, in some scenarios, the coefficient of variation for the average query completion time results can be rateher high, up to even *3.8x*, which significantly reduces the confidence interval.

*3) PFC:* In all the tests, PFC reduces the FCT, with the exception of the QCN 20 configuration. TCP Vegas does not benefit from the advantages of a lossless environment. It is a delay-probing TCP representative. However, TCP Cubic and TCP New Reno do react positively to PFC. On average, FCT is improved by 27% and up to 91%.

We attribute the PFC gains to avoiding TCP waiting for retransmissions. In the datacenter environment, the RTT is dominated by queuing delays which are extremely dynamic, instead of flight delays [3]. Whereas link delays are constant, the queuing delays are extremely dynamic. The original RTO estimator, however, reacts slowly. This is compounded with its kernel calculation: a constant term is added that accounts for the cumulated variances in the segment processing at the two end-point kernels. This constant is orders of magnitude higher than the typical datacenter RTT.

*4) QCN:* In order to compare the results with [4] we choose to test two different $Q_{eq}$ thresholds: (1). 20% of the queue size i.e. 20 *KB* – the value recommended in the IEEE standard; (2). 66% of the queue size i.e. 66 *KB* – an experimental value. Overall QCN 66 shows better performance than QCN 20 – see FIGURE 11 and FIGURE 12. This result shows that an L2 congestion management technique that has a low tolerance to bursts can affect the TCP performance. Intrinsically the TCP workloads are bursty. TCP sends bursts of segments until either the congestion window or the receiver window is exhausted. The first burst of segments will trigger a burst of ACKs, which in turn will produce another burst of segments to be sent and so on [6]. In comparison with QCN 20, QCN 66 has a higher tolerance to the intrinsic burstiness of the transport layer.

Currently we cannot argue whether QCN is beneficial or

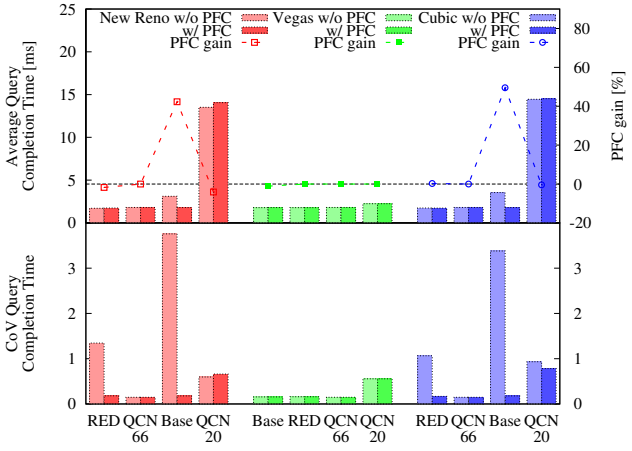
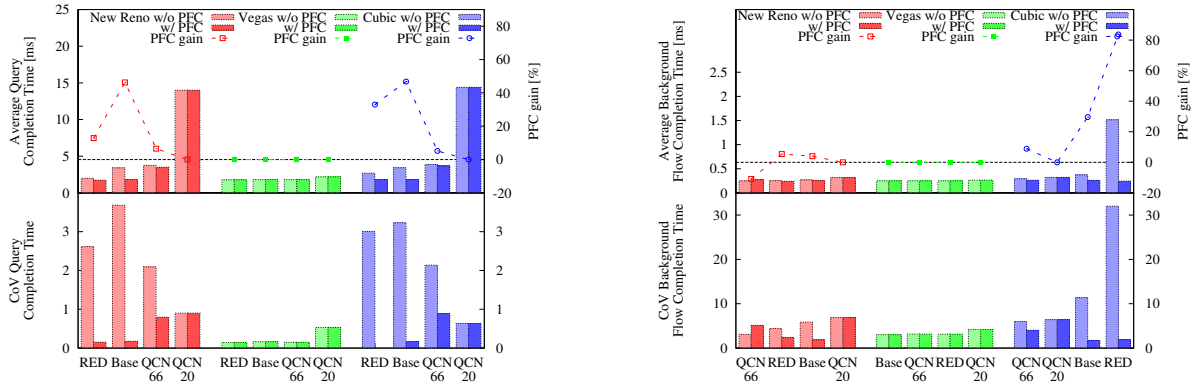



Figure 10. Commercial Workload without Background Flows. The upper part of the graph shows the average query completion time, while the lower part shows the corresponding coefficient of variation (CoV). The bars are grouped in three categories, based on the TCP version. In the upper part of the graphs, within a category, bars are sorted increasing with average query completion time without PFC.

detrimental for the TCP workloads FCT performance, because different scenarios and configurations provide different results. In all tests, QCN 20 without PFC degrades performance significantly: on average by 131% (by 44.3% for the medium sized background flows scenario FIGURE 11(a) and by 118% for the large sized background flows scenario FIGURE 12(a)) and up to 311% for New Reno and 321% for Cubic. On the other hand, a more burst tolerant QCN i.e. QCN 66 (without PFC) proves that it can have a positive impact in an environment with long-lived flows. Indeed, QCN 66 without PFC improves performance on average by 20.5% for the long sized background flows scenario FIGURE 11(a), but, for the medium sized background flows, it slightly decreases the performance by 3.67% FIGURE 12(a). Setting the QCN set-point proves to be a non-trivial task, because it requires first an in-depth analysis of the workloads and their generated communication traffic patterns in the network.

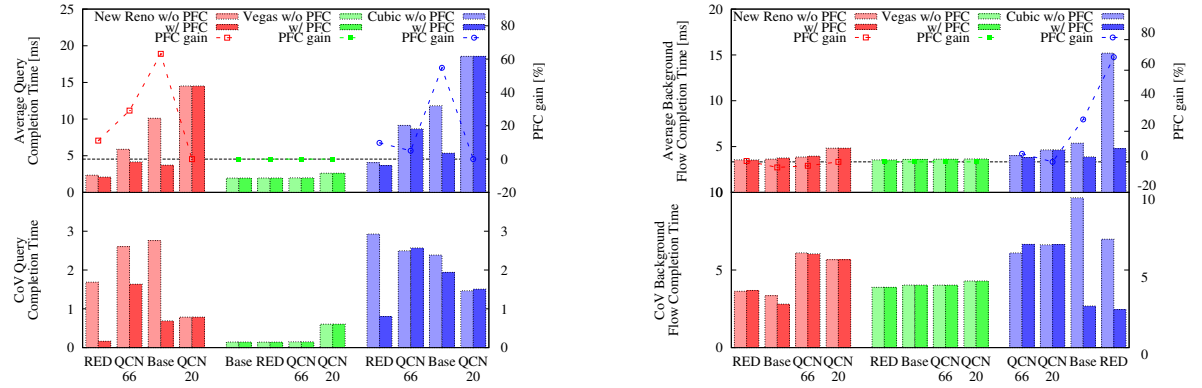
*5) ECN-RED:* ECN-RED delivers the best performance, which can be further improved by enabling PFC. ECN-RED outperforms QCN first because ECN-RED is more tolerant to bursty traffic than QCN. Indeed, ECN-RED congestion feedback is based on a smoothed averaged, whereas QCN is based on instantaneous queue length. Therefore, with RED enabled, a transient burst will not trigger a reduction of the injection rate. Secondly, the congestion feedback sent by ECN-RED is processed directly by the L4 transport protocol. Based on the feedback, the L4 transport adjusts accordingly the congestion window thus controlling the transmission rate. By contrast, TCP remains oblivious of L2 congestion feedback. Finally, RED can distinguish between data traffic and control traffic thus it is able to generate congestion notification only for data traffic. On the other hand, QCN's rate limiter can not differentiate between control and data traffic. Thus, it is possible to delay a query/transaction by throttling the initial




(a) Query Completion Time.

(b) Background Flow Completion Time.

Figure 11. Commercial Workload with TCP Background Traffic - Medium sized background flows. The upper part of the graphs shows the average query completion time in FIGURE 11(a) and the average background flow completion time in FIGURE 11(b). The lower part shows their corresponding coefficient of variation (CoV). The bars are grouped in three categories, based on the TCP version. In the upper part of the graphs, within a category, bars are sorted increasing with average query completion time without PFC.




(a) Query Completion Time.

(b) Background Flow Completion Time.

Figure 12. Commercial Workload with TCP Background Traffic - Large sized background flows. The upper part of the graphs shows the average query completion time in FIGURE 12(a) and the average background flow completion time in FIGURE 12(b). The lower part shows their corresponding coefficient of variation (CoV). The bars are grouped in three categories, based on the TCP version. In the upper part of the graphs, within a category, bars are sorted increasing with average query completion time without PFC.

SYN packet of the TCP connection.

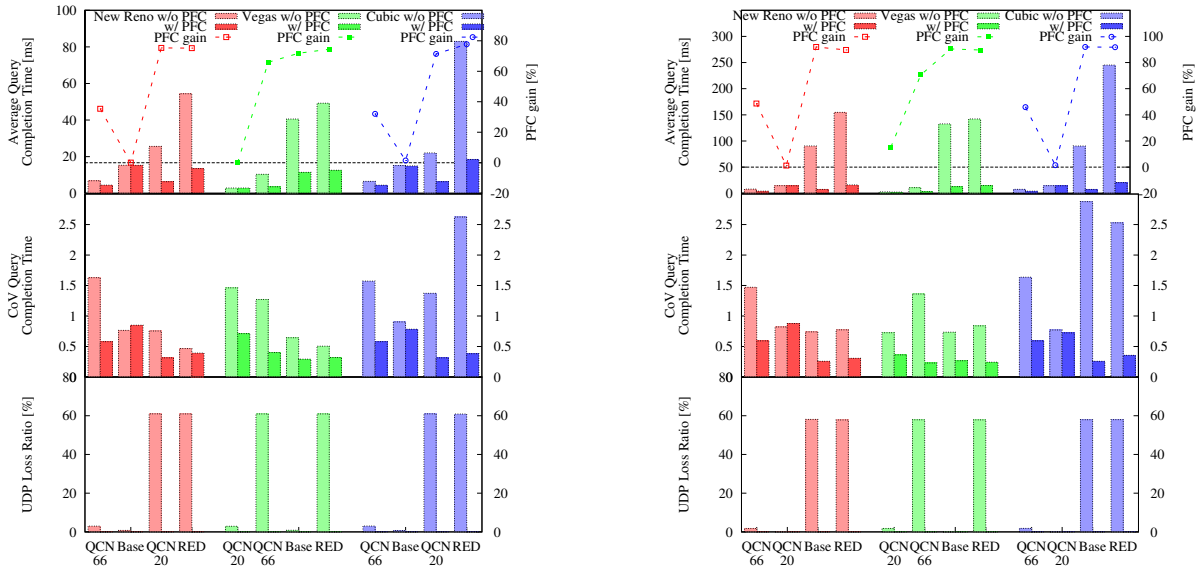
RED brings a FCT improvement by 46.6% and by 21.33% for long-sized and medium-sized background flows scenarios respectively and up to 76% for New Reno and 65% for Cubic.

### C. Commercial Workload with UDP Background Traffic

We also tested the performance of the commercial workloads in a mixed-environment: TCP queries injected in a network with UDP background flows. In contrast to the previous scenario presented in V-B, the TCP flows have to compete against aggresive UDP background flows ('elephants'). Thus, we double the number of end-nodes in the network: half of them are TCP, while the others are UDP flows sources. The average burst size for the UDP flows is determined according to the background flow size distributions from FIGURE III-C1. Hence, the UDP sources inject traffic with average burst size of 28 *KB* and 583 *KB*.

The results for the average flow completion time for the TCP queries are shown in FIGURE 13 together with their corresponding coefficient of variation values. In addition, in the lower part of the figure we represent the loss ratio for the UDP flows – the ratio is computed as the percentage of dropped bytes vs. the total amount of injected bytes. We observe that most of the dropped bytes are originating in UDP flows. We consider that this is due to the lack of any congestion control mechanism in the UDP protocol. Indeed, TCP is able to reduce its congestion window and reduce its transmission rate whenever losses are detected in the network.

*1) TCP Vegas, TCP New Reno, TCP Cubic:* In contrast with the previous section, in this scenario all the TCP versions are sensitive to ECN-RED and QCN, including TCP Vegas. This can be due to the effect of the UDP being throttled strongly by ECN or QCN, thereby leaving more network capacity available for the TCP flows. In addition, the CoV for the query FCT has high values up 2.8 for TCP Cubic. This shows unstable network conditions and it reduces the confidence interval for




(a) Medium sized background flows.

(b) Large sized background flows.

Figure 13. Commercial Workload with UDP Background Traffic. The upper part of the graphs shows the average query completion time, while the lower part shows the their corresponding coefficient of variation (CoV) and the loss ratios of the background UDP flows. Bars are grouped in three categories based on the TCP version. Within a category bars are sorted increasing with average query completion time without PFC.

the query FCT.

*2) PFC:* Again enabling PFC improves performance over all the tested configurations, including those with TCP Vegas. Two antagonist effects on performance are simultaneously at work. On one hand, UDP will cause more frequent PFCs than the well-behaved TCP; thus a throughput and FCT penalty on TCP. On the other hand, this could be compensated by TCP's inherent performance benefits derived from PFC - which may dominate the results. The figures, however, are similar to QCN66 below - which has a clearly positive causality on TCP by penalizing UDP. On average the query FCT is improved by 81.6% and up to 91.94% for TCP Cubic.

*3) QCN:* In this mixed environment QCN brings a significant improvement. The best performer over all the configurations is QCN 66. Thus, again, QCN 66 outperforms QCN 20. QCN proves to be the best option in this scenario because when we introduce non-cooperative UDP sources, only QCN's rate limiters can restore some of the fairness lost by TCP in competing against UDP. On average QCN 66 improves the FCT by 80.25% and up to 91.95% for TCP Vegas. Also QCN 20 results show an FCT improvement by 71.66% on average and up to 97.99% for TCP Vegas.

### D. Scientific Workload

The simulated MPI traces are described in SECTION III-C2. Initially we run each benchmark on a reference system where we assume we have a perfect hardware accelerated transport and lossless network. We run every benchmark on each configuration while measuring the execution times. Then we compute the relative slowdown of each benchmark vs. the ideal reference. Finally we average all the slowdowns across the nine benchmarks, plotted in FIGURE 14.

Enabling PFC improves performance across all the configurations by 45% on average and up tp 92%. The previous observations from SECTION V-B apply also to this workload. In contrast to commercial workloads with TCP background flows only, by enabling QCN the network provides better FCT results. Again an analysis of the type of communication pattern generated in the network by the workload is needed in order to justify these results.

Commercial workloads exhibit only sparse transient congestive events, whereas in the scientific workload the congestive events are sustained and involve all the end-nodes. The MPI applications use barriers to synchronize between execution phases. All the nodes start communicating quasi-simultaneously, which generates heavy congestion. The aggressive $Q_{eq}$ setpoint of QCN 20 effectively mitigates these congestive high degree hotspots. Thus, when PFC is disabled, the best perfomer for the scientific workload is QCN 20: this configuration improves performance on average by 31% and up to 59%. Nonetheless, by enabling PFC (the most common course of action in HPC networks), QCN is no longer the best option : QCN 20 degrades performance on average by 5.4% and up to 8.2% and the best performer in this configuration is ECN-RED.

## VI. HARDWARE RESULTS AND DISCUSSION

Out of all the configurations tested in the simulation environment, on the hardware testbed we run experiments only for the following scenario: commercial workloads with no background traffic in the network, with different congestion control schemes (TCP Cubic, TCP Vegas, TCP New Reno) and with and without PAUSE enabled. We run each of the 6

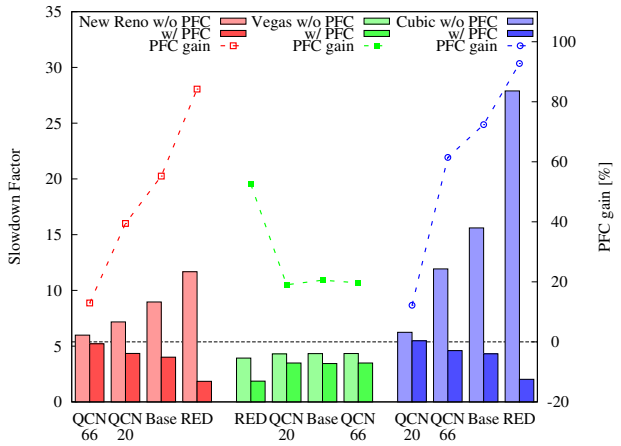



Figure 14. Scientific Workload: MPI Traces relative slowdowns. Bars are grouped in three categories based on the TCP version. Within a category bars are in increasing order of the relative slowdown factors with PFC disabled.

configurations 30 times and each time the controller is sending 2000 queries to the end nodes. The flow completion time is computed as the average over all the query completion times measured at the controller.

The the switches are actively using the 802.3x PAUSE flow control. In addition, the end nodes run an e100 network driver capable of handling and originating PAUSE frames. Thus, by enabling PAUSE the network becomes lossless. However, the network can still drop packets even when enabling PAUSE as above. These losses can occur at the edge of the network, in the end node's kernel stack. In order to detect such occurences, we have run the commercial application while monitoring the TCP statistics with netstat to determine the amount of packet loss in the kernel between TCP and the IP layers. We did not register any segments lost at this level.

### A. Hardware Tests Results

FIGURE 15 shows the results for the hardware tests together with their corresponding results obtained in the simulation environment.

*1) TCP Vegas:* Based on its congestion control scheme, TCP Vegas validates the results obtained in the simulation environment. Enabling PAUSE does not influence the flow completion time results. Over all the runs, the flow completion time is relatively constant with small variation showing stable network conditions.

*2) TCP Cubic and TCP New Reno:* TCP Cubic and TCP New Reno show similar results. However, in the context of a larger network, 100x faster and with more complex traffic communication patterns, TCP Cubic would have performed worse than New Reno. Indeed, as an RTT-independent TCP representative, its aggressive increases in the congestion window would have generated more losses than New Reno, therefore penalizing drastically the query completion time.

*3) PAUSE:* The main insight that we obtained in the simulaton environment about the influence of PFC over all the tested configurarions is validated with our hardware testbed (all our PFC simulations are single priority, hence congruent with 802.3x PAUSE). In all the tests, PAUSE reduces the query flow completion time. We attribute this result to avoiding TCP having to wait for retransmissions. TCP Cubic shows an FCT improvement on average of 7.85%, up to 13%, while TCP New Reno has an average improvement of 7.43%, up to 10.9%. In the worst case, by enabling PAUSE, TCP Cubic shows an improvement of only 4.09% while TCP New Reno only 2.89%.

### B. Hardware testbed investigation

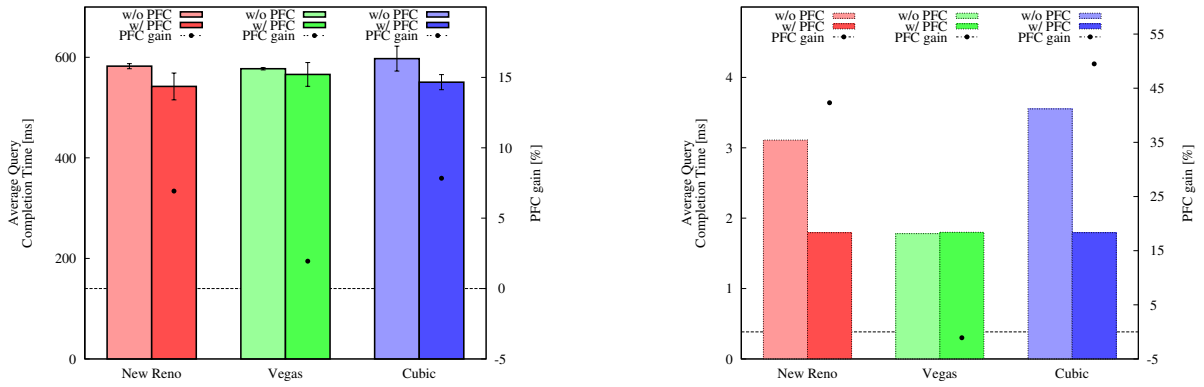
Our hardware testbed results show the same trend/reaction of the L4 protocols to PFC/PAUSE as in the simulation environment. However, the PFC/PAUSE gains are different from those in the simulations, because: (1). The hardware network speed (100 *Mbps*) is $100x$ smaller than the simulated one (10 *Gbps*); (2). The hardware network does not have CEE switches and those that we used are just three 10/100 *Mbps* desktop switches (not datacenter switches); (3). We tested only with 10 nodes against the 80 clients that we had in the simulation environment; (4). In the hardware environment we did not have access to setting some of the L2 parameters that are tunable in the simulation environment, such as the PFC thresholds. Since we cannot map the simulation environment to the hardware testbed, to cross-check our results we can adapt the simulation parameters to the hardware settings. We consider this option for further investigation as future work.

While our results suggest that PAUSE/PFC is beneficial to TCP workloads in CEE datacenter networks (and are consistent w/ the simulation results), given the limitations of the hardware testbed we can not exclude that the achievable gains are masked by the bottlenecks inherent to the OS and the hardware in use.

## VII. Selected Related Work

Our work is at the confluence of established, e.g. TCP, and emerging research areas, such as datacenter workload analysis and new L2 networking protocols. Our commercial workload traffic generator is based on the datacenter traffic characteristics discovered in [6], [5], [4], [8]. The main drawbacks of [6] are the following: (i). even though all the TCP/IP fields of the sampled packets are recorded, only part of them are actually used to obtain information about the traffic pattern; (ii). the ON/OFF traffic pattern observed at the monitored devices are generalized to all the devices. Nevertheless, these studies provide valuable information about the flows spatial and temporal distributions in real datacenters.

QCN is defined in [2] and further analyzed in [25]. Its unfairness and lack of RTT adaptivity [4] have been addressed by $E^2CM$ [26]. $E^2CM$ (Extended Ethernet Congestion Management) is a congestion management technique that inherits the best features from BCN (Backward Congestion Notification), ECN (Explicit Congestion Notification), delay-based probing and QCN. $E^2CM$ introduces a per-flow probing sensor in the edge node for insuring a fast max-min convergence between




(a) Hardware Testbed Results.

(b) Simulation Results.

Figure 15. Hardware vs. Simulation Results: (a) shows the query completion times obtained in the hardware environment, their corresponding 95% confidence intervals and the PFC gains (the three black dots); (b) shows the query completion times obtained in the simulation environment and their corresponding PFC gains (the three black dots).

the flows, even without fair share and rate-based computations in the bridge. This scheme manages to eliminate the drawbacks of BCN (slow convergence, unfairness and oscillations) and increases convergence and scalability with network size. An alternative solution is *AF-QCN*, proposed in [27]. *AF-QCN* keeps the QCN simplicity, responsiveness and stability features and improves the QCN fairness by providing a programmable bandwidth partitioning per-flow or per-class of traffic.

The TCP Incast problem has been analyzed in [7], [10], [4]. TCP Incast happens in the context of barrier synchronized many-to-one traffic in high bandwidth and low latency networks like datacenter networks. This type of traffic, as in scatter/gather, generates traffic with a high degree of burstiness. Even if it is temporarily, bursty traffic can cause packet loss and retransmissions which would increase the application response time. One solution is suggested in [10] where a $10 - 1000\times$ RTO reduction and high resolution timers have been proposed. Another TCP Incast solution is DCTCP [4], using a modified RED/ECN, a new multibit feedback estimator that filters the incoming single-bit ECN stream and a modified TCP at both sender and receiver sides. By contrast, [7] brings a solution based on TCP changes at the receiver side only. Closely related is [24] which analyzes the TCP Incast problem in a QCN-enabled *lossy* network – arguably in conflict with default assumption of lossless CEE. The authors propose a new version of QCN with adaptive sampling at the congestion point and adaptive rate increase at the reaction point. The main drawbacks are the use of NS-2 simulations and the overly aggressive sampling proposal.

Finally, most recent and relevant for the experimental part of our assessment, TCP Incast is studied in [9] using SCTP and ECN with hardware experiments. The performance improvements range from 5.5% to 7.9%, most likely limited by the experimental platform. Although unrelated and complementary to our experimental hardware and software platform, the notably similar performance improvement ranges between these two independent studies prompt for further investigation. While experimental results in the field traditionally constitute the ultimate validation, the recent improvements in performance, accuracy and capacity make simulations a key modeling tool for datacenter research.

## VIII. Conclusions and Final Remarks

In this paper we have performed an extended evaluation of the TCP performance over CEE datacenter networks as initially introduced in [28]: The main difference here is in employing two modeling methods, i.e., the Omnet++ Venus ZRL simulation platform, and, an adhoc hardware testbed. In simulation, we used as traffic generators three types of workloads: congestive synthetic traffic, commercial and scientific applications. In hardware, we emulated and tested commercial workloads only. The hardware testbed – albeit reduced in speed and scale – still validates the main results obtained from simulations. We summarize the results obtained by answering the original questions from Section I.

(**Q1**) How does TCP perform over CEE networks?: In both simulation and hardware environments, the delay-probing TCP Vegas remains the best performer as before in [28], requiring the same minimal changes, e.g., high resolution timers. By contrast, as an RTT-independent TCP representative, Cubic entails the most adaptation effort for datacenter environments, eliciting parameter tuning and potentially core algorithmic changes - which we have not performed. Indeed, because of its aggressivity and slow convergence of congestion windows, Cubic as it stands today does not seem the most suitable transport protocol for datacenter networks. Even though it was only partly CEE-enabled, also the hardware enviroment confirms that Cubic yields worse FCT results. Whether the RTT independence, as in BIC and Cubic, is harmful in CEE networks with high variations of queueing delays, from 0.5 $\mu s$ up to tens of $ms$, remains an open problem. TCP New Reno lies in between the two extremes above, requiring more parametrical retuning than Vegas, but no invasive changes such as Cubic - while producing promising performance results.

New Reno's key features as a DCN transport candidate are in its relative simplicity of operation, ubuiquity and compatibility with AQM methods such as RED and REM. We argue that **TCP Vegas** currently is our best transport protocol candidate for CEE datacenter networks. To achieve its full performance potential some changes are suggested.

1) Increased time granularity: a finer granularity allows for (much) more precise RTT measurements. To obtain a finer time granularity in Linux, one can increase the HZ kernel parameter - directly changing the 'jiffy' time reference of the TCP code. A more complex method would be to modify the TCP kernel code to implement a finer time reference for the time sensitive modules. Time sources with finer granularity than jiffies do exist, e.g.. Linux *doGetTimeOfDay* directly queries the hardware and it is not updated when the timer ticks – though they are complex and architecture-dependent, possibly eliciting an increased CPU overhead. A trade-off between TCP performance and CPU overhead study is left as an interesting future work.
2) Default RTO: The current Linux kernels have a default initial RTO value of 3s. If the first TCP SYN packet is lost, the source will have to wait 3s before retransmitting the packet and re-initiating the connection. This affects drastically the flow completion time in a DCN environment and, the performance of delay sensitive applications such as, business analytics, algorithmic trading and P/A workloads. The problem can be mitigated by setting the default initial RTO to a small multiplier of the maximum RTT of the datacenter network when fully congested. The multiplier should be conservative enough, since a smaller RTO (then $\text{RTT}_{\text{max}}$) can trigger false positives: unnecessary retransmissions, thus wasting bandwidth and again degrading the network performance.

(**Q2**) Is PFC beneficial or detrimental to TCP?: Despite our initially different expectations, PFC has systematically improved performance across all the tested configurations and benchmarks in both the simulation *and* the hardware environments. In simulation, the commercial workload completion time improves by 27% on average, and up to 91%. Scientific workloads show higher gains by enabling PFC: 45% on average, and up to 92%. In hardware, over all the configurations, the results show the same positive trend by enabling PFC (our case, single priority PAUSE). On average, PFC increases performance by 6.45% and up to 13%. Note on PFC tuning: PFC implementation in input buffer and distributed crossbar switches is relatively straightforward, except the optimal settings of the two thresholds.

The upper limit (STOP) should be set as high as possible, but low enough to ensure that once it is reached and the PFC=ON frame is sent out, the buffer has enough remaining space to accomodate all the still in-flight packets that the upstream source might have sent before receiving the notification. The amount of in-flight traffic can be calculated as maximum between the local link level RTT plus the logical delays times the maximum injection rate (BDP), and, the worst-case MTU, i.e. 2x the largest MTU supported (2 Jumbo frames). This value may or not coincide with the PFC timing recommendations made by 802 DCB in bit time units.

The lower limit (GO) must be set to (i) avoid oscillations (PFC=ON/OFF/ON...) through a sufficiently wide hysteresis, (ii) reduce the 'dead time' to restart (prevent underflow), and, (iii) prevent the synchronized restarts of multiple traffic sources.

The PFC gains recorded from our hardware testbed are notably inferior to those obtained in the simulation. We attribute this discrepancy to the practical limitations of our testbed: the consumer-level hardware is not CEE equipped, works at $100x$ slower rate than the 10 *Gbps* simulation platform, and the hardware 'datacenter' network contains 5x less end nodes . We plan to redo the experiments in a more realistic datacenter testbed, with direct access to the hardware CEE configuration of the network. Currently we are not aware of the availability of DCN switches and adapters implementing user-accessible CEE and OpenFlow interfaces, as required to conduct our investigation.

(**Q3**). Is QCN beneficial or detrimental to TCP?: The answer to this question must be more carefully qualified, depending on (a) whether TCP is alone, or in competition with (e.g.) UDP traffic; (b) the proper parameter settings per type of application. Given the available hardware capabilities, we tested the TCP performance over QCN only in the simulation environment.

On the *positive* side, properly tuned for mixed environments of commercial TCP with UDP applications, QCN 66 with PFC improves performance on average by 49%, up to 70%. Regardless of the upper layer protocols, QCN 66 succeds to keep congestion under control even when we introduce in the network non-cooperative, greedy UDP flows. For scientific workloads QCN 20 without PFC improves performance on average by 31%, up to 59%. HPC applications typically exhibit alternating phases of computation and communication. During the latter, typically all nodes start communicating almost at the same time, which can generate heavy congestion, especially in slim networks as simulated here. The aggressive $Q_{eq}$ setpoint of QCN 20 effectively mitigates such congestive events.

On the *negative* side, mistuned QCN can severely degrade performance. For example, for the commercial workload without UDP, QCN 20 without PFC degrades performance on average 131%, up to 311% for New Reno and 321% for Cubic. For scientific workloads QCN 66 with PFC degrades performance on average by 5.4%, up to 8.2%. QCN suffers also from inherent unfairness as it tends to randomly favor only some of the flows over the others harming the average completion time.

As a conclusion for **Q3**, we argue that QCN needs further investigation and improvement with respect to adaptivity and fairness. Even if QCN is customizable also by the means of the $Q_{eq}$ parameter, setting its value is more challenging than that of the PFC thresholds. As one example, is improvement of

QCN's burst tolerance: Can be trivially achieved by increasing the setpoint values, such as $Q_{eq} = 66$ (much higher than the .1Qau recommendation). Another –albeit complex and non-standard– option to improve the QCN burst tolerance would be to set a low $Q_{eq}$= [8-17], and to filter the (multibit) congestion feedback notifications stream at the source, somewhat similar to single-bit DCTCP feedback in [4]. When in doubt about the nature of their DC workloads and QCN's impact on the performance, the datacenter managers may consider **QCN** for conservative deployment: Hence, originally the equillibrium point should be set starting from high values ($Q_{eq} = 80$ or $Q_{eq} = 90$) that will minimize the amount of false positivies and the QCN activity (relax its control loop). Subsequently, the QCN loop can be progressively tightened by reducing $Q_{eq}$toward the .1Qau recommended values. It is essential that this manual QCN tuning is performed while carefully monitoring the DCN performance.

Finally we have also shown results of TCP with RED. Our simulation results show that RED can handle the transient congestion episodes generated by commercial applications better than QCN. Properly configured, RED is less sensitive to bursty traffic than QCN. Indeed, RED relies on smoothed queue length and not on instantaneous queue size as QCN does. By contrast, QCN has an intrinsic higher burst-sensitivity than RED. However, this can be prevented by properly configuring its setpoint $Q_{eq}$. The simulation results showed that RED can reduce the query completion time by up to 76%, which suggests that future CEE switches should implement RED, and other such L3 primitives directly in L2.

*Future Work:* We intend to cross-validate the results obtained for the hardware setup through an accurate simulation of the actual platform used. Foremost, our experimental testbed must be upgraded with new CEE-compatible switches and adapters. Currently this is our main limitation. We aim to build a CEE virtualized network environment (already partially instrumented) and test it under the same workloads and configurations.

## ACKNOWLEDGMENTS

We are deeply indebted to C. Minkenberg for his contributions.

## REFERENCES

[1] *P802.1Qbb/D1.3 Virtual Bridged Local Area Networks - Amendment: Priority-based Flow Control*, IEEE Draft Standard, 2010. [Online]. Available: http://www.ieee802.org/1/pages/802.1bb.html

[2] *P802.1Qau/D2.4 Virtual Bridged Local Area Networks - Amendment: Congestion Notification*, IEEE Draft Standard, 2009. [Online]. Available: http://www.ieee802.org/1/pages/802.1au.html

[3] M. Gusat *et al.*, "Delay-based Cloud Congestion Control," in *Proc. IEEE GLOBECOM 2009 Global Communications Conference*, Honolulu, HI, USA, December 2009.

[4] M. Alizadeh *et al.*, "DCTCP: Efficient Packet Transport for the Commoditized Data Center," in *Proc. ACM SIGCOMM 2010 Conference on Data Communication*, New Delhi, India, August 2010.

[5] T. Benson *et al.*, "Network Traffic Characteristics of Data Centers in the Wild," in *Proc. Internet Measurement Conference (IMC 2010)*, Melbourne, Australia, November 2010.

[6] ——, "Understanding Data Center Traffic Characteristics," in *Proc. ACM SIGCOMM Workshop for Research on Enterprise Networks (WREN 2009)*, Barcelona, Spain, August 2009.

[7] Y. Chen *et al.*, "Understanding TCP Incast Throughput Collapse in Datacenter Networks," in *Proc. 1st ACM Workshop on Research on Enterprise Networking (WREN 2009)*, Barcelona, Spain, August 2009.

[8] S. Kandula *et al.*, "The Nature of Datacenter Traffic: Measurements & Analysis," in *Proc. Internet Measurement Conference (IMC 2009)*, Chicago, IL, USA, November 2009.

[9] R. R. Stewart *et al.*, "An Investigation into Data Center Congestion with ECN," in *Proc. 2011 Technical BSD Conference (BSDCan 2011)*, Ottawa, Canada, May 2011.

[10] V. Vasudevan *et al.*, "Safe and Effective Fine-grained TCP Retransmissions for Datacenter Communication," in *Proc. ACM SIGCOMM 2009 Conference on Data Communication*, Barcelona, Spain, August 2009.

[11] J. Dean and S. Ghemawat, "MapReduce: Simplified Data Processing on Large Clusters," in *Proc. 6th Symposium on Operating System Design and Implementation (OSDI 2004)*, San Francisco, CA, USA, December 2004.

[12] S. R. Öhring *et al.*, "On Generalized Fat Trees," in *Proc. 9th International Parallel Processing Symposium (IPPS 1995)*, Santa Barbara, CA, USA, April 1995.

[13] S. Floyd and V. Jacobson, "Random Early Detection Gateways for Congestion Avoidance," *IEEE/ACM Transactions on Networking*, vol. 1, no. 4, pp. 397–413, August 1993.

[14] *RFC 3782*, RFC, April 2004. [Online]. Available: http://www.faqs.org/rfcs/rfc3782.html

[15] I. Rhee and L. Xu, "CUBIC: A New TCP-Friendly High-Speed TCP Variant," in *Proc. PFLDnet*, Lyon, France, February 2005.

[16] L. Brakmo *et al.*, "TCP Vegas: New Techniques for Congestion Detection and Avoidance," in *Proc. ACM SIGCOMM 1994 Conference on Data Communication*, London, UK, August 1994.

[17] C. Minkenberg and G. Rodriguez, "Trace-driven Co-simulation of High-Performance Computing Systems using OMNeT++," in *Proc. SIMUTools 2nd International Workshop on OMNeT++*, Rome, Italy, March 2009.

[18] G. Rodriguez, "Understanding and Reducing Contention in Generalized Fat Tree Networks for High Performance Computing," Ph.D. dissertation, Technical University of Catalonia, Barcelona, Spain, 2011.

[19] *RFC 2988 - Computing TCP's Retransmission Timer*, RFC, November 2000. [Online]. Available: http://potaroo.net/ietf/idref/rfc2988/

[20] K. Tan and J. Song, "A compound tcp approach for high-speed and long distance networks," in *In Proc. IEEE INFOCOM*, 2006.

[21] N. Dukkipati and N. McKeown, "Why Flow-Completion Time is the Right Metric for Congestion Control," *ACM SIGCOMM Computer Communication Review*, vol. 36, no. 1, pp. 59–62, January 2006.

[22] D. Bailey *et al.*, "The NAS Parallel Benchmarks," NASA Ames Research Center, Moffett Field, CA, NASA Technical Report RNR-94-007, March 1994.

[23] Y. Etsion *et al.*, "Effects of Clock Resolution on the Scheduling of Real-Time and Interactive Processes," in *SIGMETRICS '03 Proceedings of the 2003 ACM SIGMETRICS international conference on Measurement and modeling of computer systems*.

[24] P. Devkota and A. L. N. Reddy, "Performance of Quantized Congestion Notification in TCP Incast Scenarios of Data Centers," in *Proc. 2010 IEEE International Symposium on Modeling, Analysis and Simulation of Computer and Telecommunication Systems (MASCOTS 2010)*, Miami Beach, FL, USA, August 2010.

[25] C. Minkenberg and M. Gusat, "Congestion Management for 10G Ethernet," in *Proc. Second Workshop on Interconnection Network Architectures: On-Chip, Multi-Chip (INA-OCMC 2008)*, Goteborg, Sweden, January 2008.

[26] M. Gusat *et al.* (2007, March) Extended Ethernet Congestion Management (E2CM): Per Path ECM - A Hybrid Proposal. [Online]. Available: http://ieee802.org/1/files/public/docs2007/au-sim-IBM-ZRL-E2CM-proposal-r1.09.pdf

[27] A. Kabbani *et al.*, "AF-QCN: Approximate Fairness with Quantized Congestion Notification for Multi-tenanted Data Centers," in *Proc. 18th Annual IEEE Symposium on High-Performance Interconnects (HOTI 2010)*, Mountain View, CA, USA, August 2010.

[28] D. Crisan *et al.*, "Short and Fat: TCP Performance in CEE Datacenter Networks," in *HOT Interconnects (HOTI 2011)*, Santa Barbara, California, US, August 2011.