# Accumulation-based Congestion Control

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*Abstract*— **This paper generalizes the TCP Vegas congestion avoidance mechanism and proposes a model to use *accumulation*, buffered packets of a flow inside network routers, as a congestion measure on which a *family* of congestion control schemes can be derived. We call this model accumulation-based congestion control (ACC). We use a fluid analysis to define accumulation and develop a general control algorithm which includes a *set* of control policies with the proportional fairness and global stability. The ACC model serves as a reference for packet network implementations. We show that TCP Vegas is one possible scheme which fits into the ACC model. It is well known that Vegas suffers from round trip propagation delay estimation error and reverse path queuing delay. We therefore design a new scheme called Monaco which, with two FIFO queues at each router, solves these problems by employing an *out-of-band receiver-based* accumulation estimation. Analysis and simulation comparisons between Vegas and Monaco demonstrate the effectiveness of the Monaco accumulation estimator. We use ns-2 simulation and Linux kernel v2.2.18 implementation experiments to show that the static and dynamic performance of Monaco matches the theoretic results. Several key issues regarding the ACC model in general, such as the scalability of router buffer requirement and its possible solutions, are discussed.**

## I. Introduction

Much research has been conducted toward achieving stable, efficient and fair operation of packet-switching networks. TCP congestion control [8] is an end-to-end mechanism which has been critical for the stability of the Internet. It detects network congestion by inferring packet loss assumed to be caused only by congestion. As an alternative TCP implementation, Vegas [4] uses another measure called backlog, the estimated number of buffered packets inside network routers along the path, to detect network congestion.

Unfortunately Vegas has technical problems inherent to its backlog estimator. There has been a substantial body of work on these issues. For instance, Mo et al [16] (and references therein) pointed out Vegas' drawbacks of estimating round trip propagation delay incorrectly and possible persistent congestion. Low and Peterson [14] developed an optimization model for Vegas and suggested to improve Vegas performance by using an active queue management mechanism [1]. But none of them provides a solution to estimate backlog unbiasedly in case of round trip propagation delay estimation error or reverse path congestion.

In this paper, we develop a systematic model to generalize the Vegas congestion avoidance mechanism and offer a solution to the above problems. Formally, we define in the fluid model the backlog (hereafter we call *accumulation*) as a time-shifted, distributed sum of the queue contributions of a flow at a sequence of FIFO routers. We show that flow rates can be controlled by controlling the accumulations in a distributed manner. We study a *family* of closed-loop congestion control schemes based upon accumulation estimation, instead of depending on packet loss for congestion detection.

In Section II we first develop the key concepts and propose a general control algorithm which is globally stable and steers the network to the equilibrium for this accumulation-based congestion control (ACC) model. A range of traditional algorithms including additive-increase/additive-decrease [5] [4] and other algorithms [15] can be used in the ACC model. Detailed proofs of the fairness and stability are given in the technical report [18].

Within the ACC model a family of different schemes make choices in each of the ACC components and put together the entire scheme. We describe two packet network example schemes in Section III. We demonstrate that the TCP Vegas [4] congestion avoidance mechanism attempts to estimate accumulation, and fits into the ACC family. But Vegas often fails because it cannot provide an unbiased accumulation estimation. Then we develop a new scheme called Monaco that emulates the ACC fluid model in a better way. Particularly, Monaco solves the above problems of Vegas by employing an *out-of-band receiver-based* accumulation estimation. In Section IV we use simulations to show the static and dynamic performance of the Monaco scheme, which is also validated by a set of experiments based on a Monaco implementation in Linux kernel v2.2.18. We conclude this paper by suggesting future research issues in Section V.

## II. Fluid Model

In this section we describe the ACC model. We define the accumulation concept using a bit-by-bit fluid model and use accumulation to control network congestion. We develop a general control algorithm of global stability for each flow to achieve its target accumulation.

### A. Accumulation

Consider an ordered sequence of FIFO nodes $\{R_1, \ldots, R_J\}$ along the path of a *unidirectional* flow $i$ in Figure 1(a). The flow comes into the ingress node $R_1$ and, after passing some intermediate nodes $R_2, \ldots, R_{J-1}$, goes out from the egress node $R_J$[1]. At time $t$ in any node $R_j$ $(1 \leq j \leq J)$, flow $i$'s input rate is $\lambda_{ij}(t)$, output rate $\mu_{ij}(t)$. The propagation delay from node $R_j$ to node $R_{j+1}$ is $d_j$.

We define the arrival curve $A_{ij}(t)$ of a flow $i$ at a node $R_j$ as the number of bits from that flow which have cumulatively

[1] In practice $R_1/R_J$ can be mapped as source/destination to form end-to-end control loop or ingress/egress edge router to form edge-to-edge control loop. Here we focus on the ACC model itself. We'll discuss architectural issues in a separate paper.

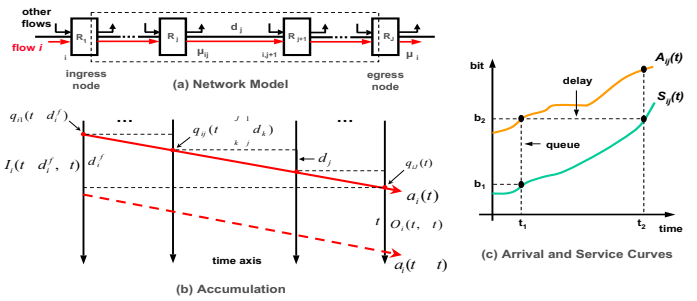



Fig. 1. Network Fluid Model

arrived at the node up to time $t$, and similarly the service curve $S_{ij}(t)$ as flow $i$'s bits cumulatively serviced at node $R_j$, drawn in Figure 1(c). For any FIFO node $R_j$, both $A_{ij}(t)$ and $S_{ij}(t)$ are continuous[2] and non-decreasing functions. If there is no packet loss, then at any time $t$, by definition, flow $i$'s buffered bits $q_{ij}(t)$ in node $R_j$ is the difference between $A_{ij}(t)$ and $S_{ij}(t)$, as shown in Figure 1(c):

$$q_{ij}(t) = A_{ij}(t) - S_{ij}(t). \tag{1}$$

We compute the change of flow $i$'s queued bits at node $R_j$

$$\begin{aligned} \Delta q_{ij}(t) &= q_{ij}(t+\Delta t) - q_{ij}(t) \\ &= [A_{ij}(t+\Delta t) - A_{ij}(t)] - [S_{ij}(t+\Delta t) - S_{ij}(t)] \\ &= [\overline{\lambda}_{ij}(t,\Delta t) - \overline{\mu}_{ij}(t,\Delta t)] \times \Delta t \\ &= I_{ij}(t,\Delta t) - O_{ij}(t,\Delta t) \end{aligned} \tag{2}$$

where $I_{ij}(t,\Delta t)$ and $O_{ij}(t,\Delta t)$ are incoming and outgoing bits of flow $i$ at node $R_j$ during the time interval $[t, t+\Delta t]$; $\overline{\lambda}_{ij}(t,\Delta t)$ and $\overline{\mu}_{ij}(t,\Delta t)$ are the correspondent average input and output rates, respectively.

Now consider the flow's queuing behavior at a *sequence* of FIFO nodes. Reasonably, suppose data link layer transmission could be modelled as a line, then flow $i$'s input rate $\lambda_{i,j+1}(t)$ at a node $R_{j+1}$ is a delayed version of its output rate $\mu_{ij}(t)$ at the upstream neighbor node $R_j$, namely

$$\mu_{ij}(t - d_j) = \lambda_{i,j+1}(t). \tag{3}$$

Define flow $i$'s *accumulation as a time-shifted, distributed sum of the queued bits* in all nodes along its path from the ingress node $R_1$ to the egress node $R_J$, i.e.,

$$a_i(t) = \sum_{j=1}^{J} q_{ij}(t - \sum_{k=j}^{J-1} d_k) \tag{4}$$

which is shown as the solid slant line in Figure 1(b). Note this definition includes only those bits backlogged inside the node buffers, not those stored on transmission links. Using it as a reference we provide an unbiased accumulation estimator in Section III-B.1. We define flow $i$'s ingress and egress rates as those at the ingress and egress nodes, respectively:

$$\begin{aligned} \lambda_i(t) &= \lambda_{i1}(t) \\ \mu_i(t) &= \mu_{iJ}(t). \end{aligned} \tag{5}$$

[2]Strictly this is true if we accept that a bit is infinitely small.

Using Equations (2)–(5), we calculate flow $i$'s accumulation change as follows:

$$\begin{aligned} \Delta a_i(t) &= a_i(t+\Delta t) - a_i(t) \\ &= \sum_{j=1}^{J} \Delta q_{ij}(t - \sum_{k=j}^{J-1} d_k) \\ &= [\overline{\lambda}_i(t - d_i^f, \Delta t) - \overline{\mu}_i(t,\Delta t)] \times \Delta t \\ &= I_i(t - d_i^f, \Delta t) - O_i(t,\Delta t) \end{aligned} \tag{6}$$

where $d_i^f = \sum_{j=1}^{J-1} d_j$ is the forward direction propagation delay of flow $i$ from node $R_1$ all the way down to node $R_J$. Similar to Equation (2), $I_i(t - d_i^f, \Delta t)$ and $O_i(t,\Delta t)$ are flow $i$'s bits coming into and going out of network during two *different* time intervals but both of length $\Delta t$; while $\overline{\lambda}_i(t - d_i^f, \Delta t)$ and $\overline{\mu}_i(t,\Delta t)$ are the correspondent average ingress and egress rates. The result, illustrated in Figure 1(b), shows the change of a flow's accumulation on its path is only related to its input and output at the ingress and egress nodes. Further, this means it is possible to control accumulation at only the ingress and egress nodes.

For one FIFO node, it's straight-forward to control flow rates by controlling the number of queued packets [6], since packets buffered decide service received if the scheduling discipline is FIFO. Due to the similarity of Equations (2) and (6), for a sequence of FIFO nodes, we aim to *control flow rates by controlling the accumulations, i.e., keeping a steady state accumulation for each flow* inside network. Note Equations (2) and (6) have a significant difference of the propagation delay $d_i^f$, which is a constant as long as flow $i$'s route is fixed.

### B. Queuing and Optimization Analysis

To facilitate better understanding of using accumulation as the steering parameter for congestion control, in [18] we provide physically a simple queuing analysis and mathematically an optimization theory to demonstrate the *steady state* picture by leveraging results from [9] [13] [15]. It turns out that, by using accumulation as a steering parameter to control flow rates, the network is actually doing a nonlinear optimization which steers the network to an equilibrium of proportionally fair bandwidth allocation.

### C. Control Algorithm

In the ACC model we use accumulation to measure network congestion as well as to probe available bandwidth. If accumulation is low, we increase congestion window; otherwise, we decrease it to drain accumulation. Specifically, we try to maintain a constant target accumulation $a_i^*$ for each flow $i$ by applying a general ACC control algorithm:

$$\dot{w}_i(t) = -\kappa \cdot f(a_i(t) - a_i^*) \tag{7}$$

where $w_i(t)$, $a_i(t)$ and $a_i^*$ are respectively the congestion window size, instantaneous accumulation and target accumulation value of flow $i$, $f(\cdot)$ is a strictly increasing function with a unique root 0 (i.e., only $f(0) = 0$) and $\kappa > 0$.

Obviously Equation (7) includes a *set* of algorithms. The reason we present a general algorithm here is that *all instance algorithms which fit into Equation (7) share a common steady state property of proportional fairness*, as shown in the next subsection.

By choosing different $f$ functions, we instantiate the above general algorithm into a set of control policies including the well-known additive-increase-additive-decrease (AIAD) policy [5] popularized by TCP Vegas [4], an algorithm proposed by Mo and Walrand [15], and another proportional control policy.

### D. Properties

For any flow control algorithm, major theoretic concerns are its stability, fairness and queue bound. Stability is to guarantee equilibrium operation of the algorithm. Fairness, e.g., max-min [3] and proportional fairness [9], determines the allocation of network bandwidth among competing flows. Queue bound provides an upper limit on the router buffer requirement, which is important for real deployment. We prove in Appendix that

*Proposition 1*: The flow control algorithm given by Equation (7) is globally stable and weighted proportionally fair.

Even we keep a finite accumulation inside network for every flow, the steady state queue at a node scales up to the number of flows passing that bottleneck. In practice, we need to provide more buffers to avoid packet loss and make the control algorithm robust to such loss (see Section III-B). Another way to alleviate this problem is to control aggregate flow, instead of individual source-destination flow. More details on buffer size scalability are discussed in Section V.

Interestingly, as Proposition 1 states, different ACC control policies can achieve the same fairness property, as long as they fit into Equation (7). Thus to achieve a particular steady state performance, we have the freedom to choose from a set of control policies. In this sense, we regard *ACC flow control as a two-step issue of setting a target steady state allocation (fairness) and then designing a control policy (stability and dynamics) to achieve that allocation.*

## III. ACC Schemes

In this section we instantiate the ACC fluid model into two example schemes for packet-switching networks. Firstly we show that TCP Vegas [4] tries to estimate accumulation and fits into the ACC model. Unfortunately Vegas often fails to provide an unbiased accumulation estimation. Then we design a new scheme called Monaco which solves the estimation problems of Vegas. Monaco also improves the congestion response by utilizing the value of estimated accumulation, unlike Vegas' AIAD policy which is possibly slow in reacting a sudden change in demands or network capacity. By comparing Monaco and Vegas via analysis and simulation we reach two observations: It is effective to employ 1) a *receiver-based* mechanism and, 2) the measurement of *forward path queuing delay*, instead of round trip queuing delay as in Vegas, to estimate accumulation unbiasedly. The scheme design is guided by the following goals:

*Goal 1:* **Stability:** The scheme should converge to an equilibrium in a reasonably dynamic environment with changing demands or capacity;
*Goal 2:* **Proportional Fairness:** Given enough buffers, the scheme must achieve proportional fairness and operate without packet loss at the steady state;
*Goal 3:* **High Utilization:** When a path is presented with sufficient demand, the scheme should converge around full utilization of the path's resources;
*Goal 4:* **Avoidance of Persistent Loss:** If the queue should grow to the point of loss due to underprovisioned buffers, the scheme must back off to avoid persistent loss.

### A. Vegas

Vegas [4] was proposed as an alternative TCP implementation. It includes several modifications over TCP Reno [8]. However, we focus only on its congestion avoidance mechanism, which fits well as an example ACC scheme.

The Vegas estimator for accumulation was originally called "backlog", a term we use interchangeably in this paper. For each flow, the Vegas estimator takes as input an estimate of its round trip propagation delay, hereafter called $rtt_p$ (or $basertt$ in [4] [16]). Vegas then estimates the backlog as

$$\hat{a}_V = (\frac{cwnd}{rtt_p} - \frac{cwnd}{rtt}) \times rtt_p \quad (8)$$

$$= \frac{cwnd}{rtt} \times rtt_q \quad (9)$$

where $cwnd/rtt$ is the average sending rate during that round trip time (RTT) and $rtt_q = rtt - rtt_p$ is the round trip queuing delay. According to Little's Law, $\hat{a}_V$ is the backlogged packets inside bottleneck routers. If $rtt_p$ is accurately available and there is no reverse path queuing delay, then $\hat{a}_V$ provides an unbiased estimation for accumulation.

Vegas estimates the $rtt_p$ as the minimum RTT measured so far. So, if the queues drain often, it is likely that each control loop will eventually obtain a sample that reflects the $rtt_p$. The Vegas estimator is used to adjust its congestion window size, $cwnd$, so that $\hat{a}_V$ approaches a target range of $\varepsilon_1$ to $\varepsilon_2$ packets. More accurately stated, the sender adjusts $cwnd$ using:

$$cwnd(n+1) = \begin{cases} cwnd(n)+1 & if \quad \hat{a}_V < \varepsilon_1 \\ cwnd(n)-1 & if \quad \hat{a}_V > \varepsilon_2 \end{cases} \quad (10)$$

where $\varepsilon_1$ and $\varepsilon_2$ are set to 1 and 3 packets, respectively. Vegas has several well-known problems when there exists $rtt_p$ estimation errors or reverse path congestion, violating goals listed above.

### B. Monaco

Monaco emulates the accumulation defined by Equation (4) and implements a receiver-based out-of-band measurement. It

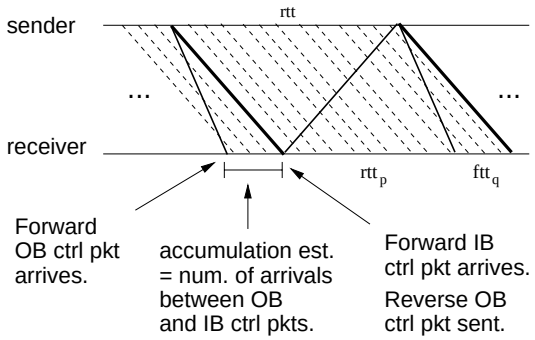



Fig. 2. Monaco Accumulation Estimator

is immune to issues such as $rtt_p$ sensitivities and reverse path congestion and robust to control and data packet losses. We describe firstly the Monaco accumulation estimator and then its congestion response policy.

### B.1 Monaco: Congestion Estimation Protocol

According to its definition, accumulation of a flow is the sum of its queued bits at a sequence of FIFO routers, including both ingress and egress nodes as well as intermediate routers. We aim to eliminate the computation at intermediate routers to keep them simple. Actually it is impossible for *all* nodes $R_j$ $(1 \leq j \leq J)$ to compute synchronously their queues $q_{ij}(t - \sum_{k=j}^{J-1} d_k)$ since no common clock is maintained.

To estimate accumulation without explicit computation at intermediate routers, Monaco generates a pair of back-to-back control packets once per RTT at the ingress node as shown in Figure 2. One control packet is sent out-of-band (OB) and the other in-band (IB). The OB control packet skips queues in the intermediate routers by passing through a separate dedicated high priority queue. Assuming the OB queues to be minimal as only other OB control packets share them, such packets experience only the forward propagation delay $d_i^f$. The IB control packet goes along with regular data packets and reaches the egress node after experiencing the current queueing delay in the network. The time interval between the OB and IB control packets measured at the egress node is a sample of the current forward trip queuing time ($ftt_q$). Considering a network with enough buffers where there is no packet loss, if flow rates at all routers do not change dramatically, then by Little's Law, the number of data packet arrivals at the egress node after the OB control packet, but before the IB control packet equals the accumulation. In Figure 2, the dashed lines cut by the forward direction OB control packet are those data packets, with each cut happening in the router $R_j$ at time $t - \sum_{k=j}^{J-1} d_k$, $\forall j \in \{1, ..., J\}$. Also observe in the figure that we can measure $rtt$ at both ingress and egress nodes and $rtt_p$ at the egress node.

In a real packet network there are some constraints which might introduce systematic accumulation estimation errors. For instance, beyond propagation and queuing delays, a packet also experiences switching and transmission delays not considered in the fluid model. But since both OB and IB control packets are of the same size (32 bytes) and sent out back-to-back from the ingress node, their switching and transmission delays cancel out. Another aspect is non-preemptive packet forwarding. When an OB control packet arrives at a router, it has to wait if there is another packet being serviced. So the delay experienced by the OB control packet is generally not constant and larger than the propagation delay $d_i^f$, so the measured time interval $ftt_q$ seems less than its true value. But as shown in Figure 2, if we just *count* the number $\hat{a}_M$ of data packet arrivals in that interval, we can at least alleviate the effect of noises included in the time interval $ftt_q$, as long as priority queue and FIFO properties are true.

Besides, we need to consider the effect of traffic burstiness. When we have a congestion window size $cwnd$, we also compute a rate based RTT estimation: $rate = cwnd/rtt$. We use this rate value to smooth incoming traffic and thus alleviate the effect of burstiness.

In practice, both data and control packets maybe lost because of inadequate router buffer size or too many competing flows. To enhance the robustness of Monaco estimator when data packets are lost, the IB control packet, identified by a control packet sequence number, carries a byte count of the number of data bytes sent during that period. If the egress node receives fewer bytes than were transmitted, then packet loss is detected. The forward OB control packet carries the same control packet sequence number as the associated IB control packet. Monaco sends congestion feedback on the reverse OB control packet, in which there is one additional piece of information: congestion feedback, i.e., a flag denoting whether the congestion window $cwnd$ should increase, decrease, or decrease-due-to-loss. The subsequent pair of forward control packets is generated after the arrival of the reverse OB control packet at the ingress node.

If either control packet is lost, then the ingress node times out and sends a new pair of control packets with a larger sequence number. The timer for control packet retransmission is similar to that of TCP [8]. These routine reliability enhancements are similar to the Congestion Manager protocol [2].

### B.2 Monaco: Congestion Response Protocol

As already noted, we use accumulation to measure network congestion and to probe available bandwidth. We keep accumulation constant for every flow by increasing/decreasing its congestion window when the accumulation is lower/higher than the target value.

For Monaco we choose a window-based instead of rate-based control policy because the former is more conservative and rate is hard to accurately measure in practice. Since pure window-based control policy might introduce burstiness we use *rate-modulated window* control to smooth incoming traffic into packet networks by employing a token bucket shaper with a rate value of $cwnd/rtt$.

We provide below the Monaco's proportional control:

$$cwnd(n+1) = cwnd(n) - \kappa \cdot (\hat{a}_M - a^*) \qquad (11)$$

where $\hat{a}_M$ is the Monaco accumulation estimation, $a^*$, set to 3 packets, is a target accumulation in the path akin to $\varepsilon_1$ and $\varepsilon_2$ used by Vegas, $\kappa$ is set to 0.5, and $cwnd(n)$ is the congestion window value at a control period $n$.

Monaco improves Vegas' control policy by utilizing the value of accumulation estimation fedback by the reverse OB control

packet, instead of regarding it as a binary information (i.e., "how congested", instead of "congested or not"). If the congestion feedback is decrease-due-to-loss, Monaco halves the congestion window size as in TCP Reno.

### C. Comparisons between Vegas and Monaco

Vegas and Monaco aim to accurately estimate accumulation, assuming different support from network routers. If $rtt_p$ can be obtained precisely and there is no reverse path congestion, then by Little's law, both of them give unbiased accumulation estimation on average. But in practice Vegas often has severe problems in achieving this objective; Monaco solves known estimation problems.

Vegas estimator operates at *sender* side. According to Equation (8) it actually calculates:

$$\hat{a}_V = \frac{cwnd}{rtt} \times ( rtt - rtt_p ) \quad (12)$$

$$= \frac{cwnd}{rtt} \times ( t_q^f + t_q^b ) \quad (13)$$

where $t_q^f$ and $t_q^b$ are forward and reverse direction queuing delays, respectively. The above equations imply that Vegas may suffer from two problems: 1) By Equation (13), if there exists reverse direction queuing delay (because of reverse direction flows), i.e., $t_q^b > 0$, then Vegas overestimates accumulation. This leads to underutilization and is hard to handle because there is no control over reverse direction flows. To show this effect we use a simple dumb-bell topology with the bottleneck of 45Mbps forward direction bandwidth shared by 7 forward direction flows and 7 reverse flows. We change the bottleneck's reverse direction bandwidth from 5Mbps to 45Mbps. As shown in [18], Vegas utilization is only 10% $\sim$ 60%. 2) By Equation (12), if $rtt_p$ is overestimated, then Vegas underestimates accumulation. This leads to extra steady queue in bottlenecks or even persistent congestion. Results for a single bottleneck of 10Mbps bandwidth and 12ms delay which is used by one flow employing Vegas and Vegas-k show that Vegas operates with very low utilization of less than 10% and Vegas-k operates with queue increase until loss occurs (see [18]).

Due to the above problems, Vegas falls short of qualifying as an effective ACC scheme, because we expect to achieve congestion control by maintaining constant accumulation for each flow at the *steady state*! In such a case, the sum of accumulations would lead to a non-zero steady state queue which is not likely to drain, and hence dynamic $rtt_p$ estimation would be impossible with in-band control packets. In summary, the sensitivity issues with Vegas point to a *fundamental* problem with the in-band techniques for accumulation estimation.

Monaco solves both problems. Monaco estimator operates at *receiver* side and thus excludes the effect of reverse path congestion. By counting the data packets arriving between in- and out-of-band control packets, Monaco does not explicitly need to estimate the forward direction propagation delay $d_i^f$. (Actually the out-of-band control packets provide implicitly this value.) More specifically, since Monaco implements a rate-paced window control algorithm to smooth out incoming traffic, the time difference between the in- and out-of-band control packets gives a sample of the current forward direction queuing delay $ftt_q$. By Little's law, the number of data packets arriving during this time period is the backlogged packets along the path. Using out-of-band control packet also makes Monaco adaptive to re-routing since it is sent every RTT. Simulation results in [18] show that, after a brief transient period, Monaco operates at around 100% utilization with no packet loss. So it's immune to $rtt_p$ estimation inaccuracy and reverse path congestion.

The above comparisons between Vegas (including Vegas-k) and Monaco suggest two important observations on how to estimate accumulation unbiasedly: 1) The key is to measure *forward direction queuing delay* (via out-of- and in-band control packets in Monaco), instead of round trip queuing delay (as in Vegas); And consequently, 2) it's better to measure accumulation at the *receiver side*, otherwise it's difficult to get rid of the effect of reverse path queuing delay which is hardly under forward direction congestion control.

## IV. Simulation and Implementation Experiments

In the last section we have shown that Monaco performs better than Vegas, so we focus on evaluating Monaco scheme by ns-2 simulation and Linux implementation in this section. Our experiments illustrate:

A) Dynamic behavior such as convergence of throughput, instantaneous link utilization and queue length in Section IV-A. We use a single bottleneck topology with heterogeneous RTTs for tens of flows periodically entering and leaving.

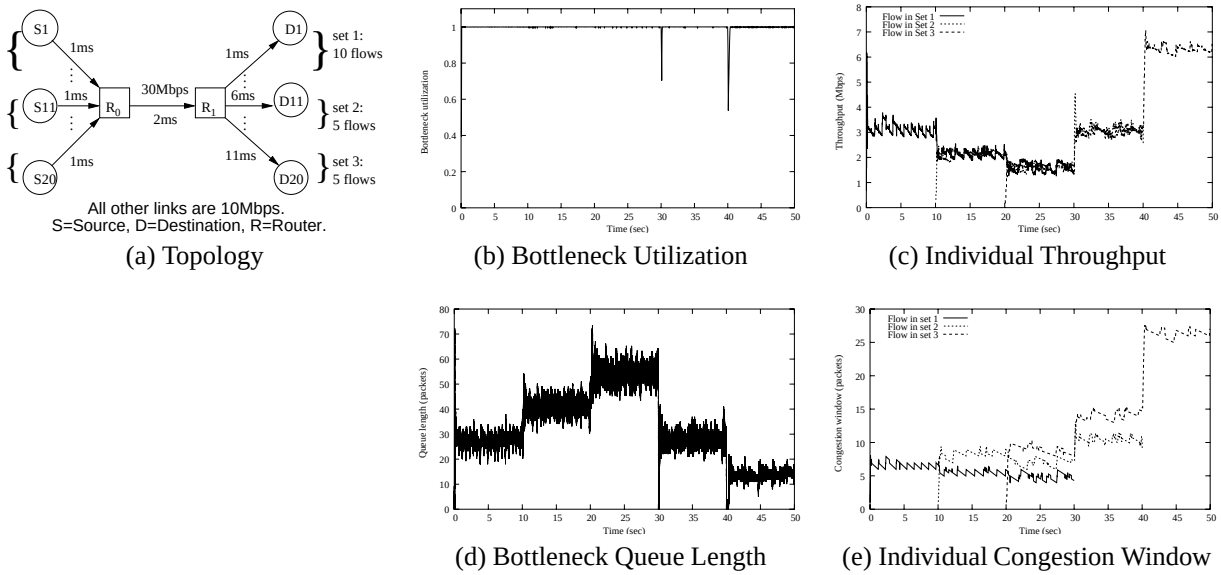
B) Steady state performance such as fairness, throughput, and throughput variance in Section IV-B. We use a linear topology of multiple congested links shared by a set of flows passing different number of bottlenecks.

In all simulation experiments we use ns-2 simulator [17] and set data packet size as 1000 bytes and target accumulation 3000 bytes. We also implement the Monaco scheme in Linux OS (kernel version 2.2.18) based on the Click configurable router [10]. Experimental results from implementation match with those of simulations. In brief, in combination with Section III-C, this section shows that the Monaco scheme satisfies all the goals outlined in Section III.

### A. Single Bottleneck with Dynamic Demands

Firstly we consider a single 30Mbps bottleneck with 2ms propagation delay shared by 3 sets of flows using the Monaco scheme. The topology is shown in Figure 3(a). Set 1 has 10 flows starting at 0s and stopping at 30s; Set 2 has 5 flows starting at 10s and stopping at 40s; Set 3 has 5 flows starting at 20s and stopping at 50s. Each source-destination pair is connected to the bottleneck via a 10Mbps 1ms link. The one-way propagation delays for the 3 sets of flows are 4ms, 9ms and 14ms, respectively. We simulate for 50s. We performed two simulations, one with enough buffer provided for the bottleneck, the other with underprovisioned buffer.

In the first simulation, the bottleneck router has enough buffer of 90 packets, as shown in Figure 3(d), there is no packet loss.




(a) Topology (b) Bottleneck Utilization (c) Individual Throughput

(d) Bottleneck Queue Length (e) Individual Congestion Window

Fig. 3. Monaco with Enough Buffer in Bottleneck

We randomly pick one flow from each set and draw its individual throughput in Figure 3(c). We observe that from 0s to 30s, the throughput is about 3Mbps, since only 10 flows are active; When the 5 flows from set 2 jump in at 10s, the throughput drops to 2Mbps, as we have 15 active flows. Similarly, when the final 5 flows from set 3 enter at 20s, the throughput changes to 1.5Mbps. Then at 30s, the 10 flows of set 1 stop, the throughput increases to 3Mbps. At 40s, the 5 flows of set 2 leave, only the 5 flows of set 3 are in the system with throughput of about 6Mbps. The congestion window dynamics is similar as shown in Figure 3(e). Bottleneck queue length is depicted in Figure 3(d) where incoming flows build up a steady queue and flows leave with queue decrease, on average 3 packets for each flow. This matches the target accumulation specified as a control parameter. During the simulation bottleneck utilization always stays around 100%, except two soon-recovered drops during abrupt demand changes at 30s and 40s as seen in Figure 3(b). From this simulation, we validate that Monaco demonstrates a stable behavior under a dynamic and heterogeneous environment and keeps a steady queue inside bottleneck.

In the second simulation, the bottleneck router buffer is underprovisioned. As shown in [18], the queue length grows to the limit of the whole buffer size, and there is a correspondent packet loss leading to halving of the congestion window. Consequently, throughput is more oscillating, but the bottleneck is still fully utilized. From this simulation, we see that without enough buffer, Monaco shows a degraded behavior under dynamically changing demands.

### *B. Multiple Bottlenecks*

Now we show the steady state performance of Monaco when flow traverses more than one bottleneck. We use a linear topology with multiple congested links depicted in Figure 4(a). We did a set of simulation experiments by changing the number of bottlenecks $N$ from 2 to 9. There are 3 "long" flows passing all bottlenecks and a set of "short" flows each using only one bottleneck. Every link has 100Mbps capacity and 4ms delay. The long flows have very different RTTs. We simulated under only one condition with enough buffer provided for the routers. As already shown in the last subsection, if buffer is not enough, the Monaco scheme degrades. For the buffer requirement scalability problem, we discuss further in the next section.

As illustrated in Figure 4(b), the throughput curve for each individual long flow is located right around the theoretic one of $100/(3+N)$Mbps. So each long flow gets roughly its fair share, for all cases of $N = 2, 3, ..., 9$ bottlenecks. The difference of throughput between the 3 long flows is measured by the Coefficient of Variance (C.O.V.) of their throughput, depicted in Figure 4(d), which is about 5% $\sim$ 10%. The bottleneck utilization for $R0 - R1$ link is shown in Figure 4(c), which is always 100% during the whole simulation of 60s.

We also did a set of experiments using our Linux implementation of Monaco. Here we show one with dynamic demands. We have 2 bottlenecks each of 1Mbps capacity as drawn in Figure 4(e). During the 80s of experiment, we have 2 short flows always active, one long flow coming in at 20s and going out at 60s, and another long flow active from 40s to 80s. After a brief transient period, each flow stabilizes at its proportionally fair share, illustrated by Figure 4(f). For instance, the first long flows' throughput starts with 0.33Mbps (its fair share) at 20s and changes to some 0.25Mbps at 40s when the second long flow shows up. After that, the second long flow gets about its fair share of 0.33Mbps.

These simulation and implementation experiments demonstrate that, with enough buffer provisioned, Monaco achieves a proportionally fair bandwidth allocation in a multiple bottleneck case, validating our theoretic results.

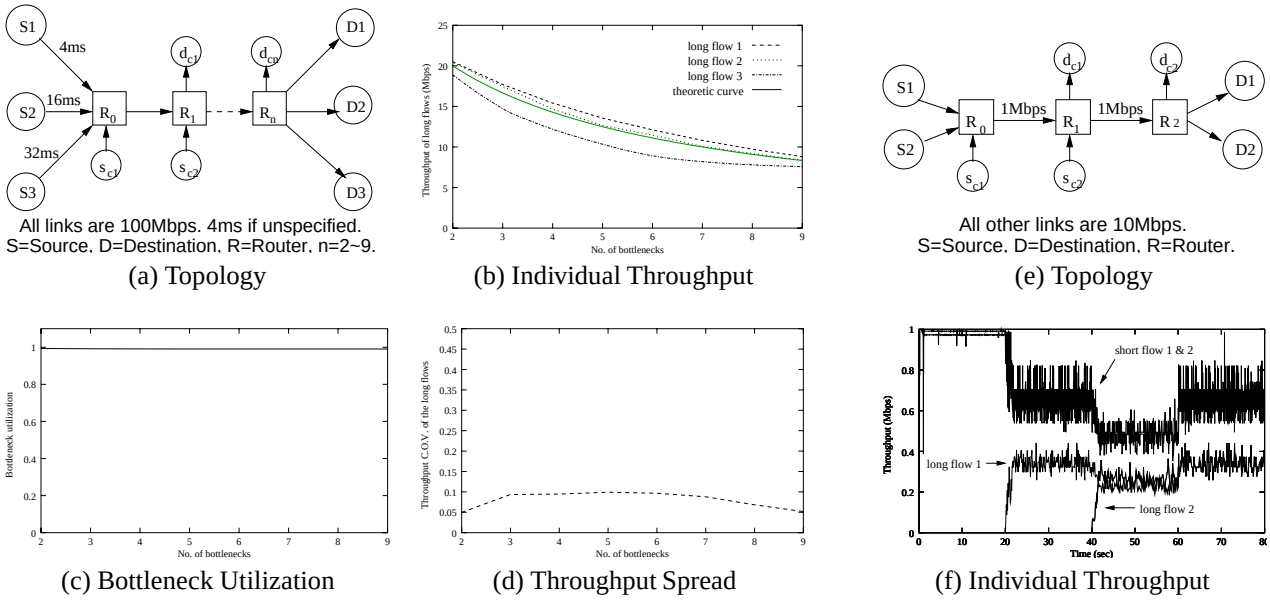



Fig. 4. Monaco Fairness via ns-2 Simulation (a-d) and Linux Implementation (e,f)

## V. Summary and Future Work

In this paper we generalize TCP Vegas and develop a general congestion control model using accumulation which is buffered packets of a flow inside network routers as a measure to detect and control network congestion. The main contributions of this paper are:

- a mathematically defined, physically meaningful concept of backlogged packets – accumulation as a measure of network congestion;

- a model of accumulation-based congestion control with provable global stability and proportional fairness based on which a family of schemes could be derived;

- a comparison between Vegas and Monaco to show that Monaco's receiver-based out-of-band accumulation measurement solves Vegas' well-known estimation problems;

- a Monaco scheme implemented as a packet network protocol which estimates accumulation unbiasedly and utilizes this value in a non-binary manner to control congestion.

One critical question that remains is the scalability of the router buffer requirement which is proportional to the number of flows passing that router. Since the ACC model can be mapped end-to-end or edge-to-edge (though we focus on the model itself and don't elaborate the architecture issues in this paper), one way to alleviate this problem is to control aggregate flow in an edge-to-edge manner, instead of end-to-end for each micro-flow of source-destination pair. A possibly better solution to keep buffer size bounded is to use an appropriate queue management mechanism such as virtual queuing [12]. Further, by keeping different accumulation for different flows, it's possible to provide service differentiation [7]. These issues are being explored in our ongoing research.

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