

SMITH'S PREDICTOR AND FEEDFORWARD DISTURBANCE COMPENSATION FOR ATM CONGESTION CONTROL

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Abstract: This paper proposes an end-to-end congestion control algorithm for ABR traffic in high-speed ATM networks. The feedback control law is designed using the Smith principle. Moreover, feed-forward disturbance compensation is added to improve performances of the control system. The control algorithm is rate based and does not require any explicit feedback from the network. We test the proposed control algorithm over the Generic Fairness Configuration 2 scenario, which is a standard benchmark with multiple bottlenecks and large round trip delays proposed by the ATM forum. Simulation results show that the proposed algorithm provides performance indexes comparable or slightly better than those obtained by the ERICA switch algorithm, which is the leading algorithm in the ATM community, even though ERICA requires explicit support from network nodes.

1. INTRODUCTION

ATM networks are a class of *virtual circuit switching* networks conceived to merge the advantages of circuit switched technology (telephone networks) with those of packet switched technology (computer networks) [6,10,14]. The goal of ATM networks is to integrate various application services, going from those that require stringent end-to-end delay and jitter guarantees, such as audio/video real time traffic, to others that are more susceptible to packet loss rather than to delay. To successfully accommodate this diversity of application requirements while guaranteeing certain quality of service (QoS), the ATM Forum has defined five ATM service categories [2]: constant bit rate (CBR); real-time variable bit rate (Rt-VBR), non-real-time variable bit rate (Nrt-VBR), unspecified bit rate (UBR), and available bit rate (ABR).

CBR provides virtual connections of a constant rate during the connection lifetime, and therefore, corresponds to circuit emulation. Typical application of this service is voice communications. Taking advantage of statistical multiplexing gain, Rt-VBR and Nrt-VBR categories can be fruitfully used for compressed video. UBR provides completely best-effort service that does not ensure any guarantees to the end users. ABR, on the other hand, guarantees a best effort service with a minimum cell rate through a closed-loop rate-based control feedback that is established between end-systems. ABR traffic is typically characterized by unspecified requirements for throughput and delay and it is conceived to rapidly "fill in" the bandwidth left unused by CBR and VBR traffic [2]. The ABR service is hence ideal for data transmission such as files, ftp, images but it could be also exploited for delivering audio and video with "elastic" quality requirements. The goal of the ABR congestion control is to avoid network congestion (buffer overflows at the ATM switches) while at the same time preserving high network utilization by a fair allocation of the available network capacity among contending ABR flows. The term "available network capacity" refers to a remainder of the link capacity that is not used by the higher priority guaranteed traffic (CBR and VBR).

To briefly summarize the algorithms proposed for ABR traffic

control, we start by recalling the binary feedback schemes. This class was the first to be introduced due to their easy implementation [4, 11, 12, 15]. In these schemes, if a switch is congested, then a binary digit is set in the control management cell. However, they suffer serious problems of stability, exhibit oscillatory dynamics, and require large amount of buffer in order to avoid cell loss. As a consequence, explicit rate algorithms have been largely considered and investigated [5,6]. Most of the existing explicit rate schemes lack of two fundamental parts in the feedback control design: 1) the analysis of the closed loop network dynamics; 2) the interaction with VBR traffic. In [3] an analytic method for the design of a congestion controller, which ensures good dynamic performance along with fairness in bandwidth allocation, has been proposed. However, this algorithm requires a complex on-line tuning of control parameters in order to ensure stability and damping of oscillations under different network conditions. In [8] an explicit rate controller for ABR service was presented. The controller uses a dual proportional-plus-derivative controller; even though the algorithm is a simplification of the one proposed in [3] it is still complex, because of the large number of control parameters that have to be tuned to ensure the stability of the control loop. The validation part considered two real ATM LANs connected via a cross continental T1 link.

The ERICA algorithm basically computes input rates dividing the measured available bandwidth by the number of active connections [7]. The algorithm was tested using the GFC-2 test bed. It is currently one of the most considered algorithms in the ATM research community. In [13] an explicit rate-flow-control algorithm was proposed, which computes explicit rate as a function of the available capacity as well as the queue's state. The number of active connections is tracked in order to compute available capacity. No simulations were reported to evaluate the effectiveness of the scheme over a wide area network with multiple bottlenecks. In [16] the control design problem is formulated as a standard disturbance rejection problem where the available bandwidth acts as a disturbance for the system. The ABR source rate is adapted to the low-frequency variation of the available bandwidth and H_2 optimal control is applied to design a controller that minimizes the difference between the source input rate and the available bandwidth. A drawback is that the design of the controller depends on the characteristic of the interacting VBR traffic and on the measurements of the available bandwidth, which is difficult to be obtained in practice. In [10] an algorithm using a Smith's predictor has been illustrated which uses *per-VC First-in-First-out (FIFO)* queueing. The switches feed the sources with queue levels as feedback information. The algorithm assures no cells loss and full utilization of network links in presence of many connections, with different round trip delays, sharing the network.

This paper enhances the algorithm proposed in [10] by introducing two basic new features: (1) switches do not supply queue length measurement to the sources, but sources estimate the bottleneck queue length via end-to-end measurements of

round trip time and available bandwidth; (2) feed-forward disturbance compensation improves the performance indices of the control system.

The paper is structured as follows. Section 2 describes the data network model. Section 3 describes the proposed control law and mathematically analyzes its properties. Section 4 provides a detailed description of the implemented algorithm. Section 5 reports simulation results, which validate the proposed approach in comparison with the leading and well-know ERICA algorithm. Finally, last section draws the conclusions.

2. THE DATA NETWORK MODEL

The network employs a store-and-forward service, i.e. cells enter the network from the source edge nodes, are then stored and forwarded along a sequence of intermediate nodes and communication links, finally reaching their destination nodes. Each output link maintains *per-VC* FIFO queuing. As a result of per-flow queuing, each flow goes through a series of buffers that are exclusively used by that flow. Per-flow buffers feed output links in accordance with a given fairness policy whereas links are shared by different flows via statistical multiplexing. An important consequence of per-flow queuing is that the network can be modeled as a set of decoupled virtual circuits [10].

Mainly following the notation reported in [3], we model the network as a graph consisting of a set $N=\{1,...,n\}$ of nodes (properly switches) connected by a set $L=\{1,...,l\}$ of communication links. Each node maintains a queue for each outgoing link where cells to be transmitted are temporarily stored. Each link i is characterized by its transmission capacity $c_i=1/t_i$ (cells/sec), where t_i is the transmission time of a packet, and by its propagation delay of t_{di} sec. Each node has a processing capacity larger than the total transmission capacity of its incoming links so that congestion is caused by transmission capacity only.

The network traffic is a set of source/destination pairs $(S,D) \in N \times N$. A Virtual Circuit (VC) is associated to each (S,D) connection, mapped on the path $p(S,D)$ [3,10,14]. The path contains one node for every network node and one directed link $e=(a,b)$ for every communication link from node a to node b . Therefore, a VC v is specified by the sequence of links

$e_1^v, e_2^v, \dots, e_k^v$ ($k \leq n$) it traverses as it proceeds through the network. A deterministic fluid model approximation of the cell flow is assumed, i.e. sources transmission rates are described by the function of time $r(t)$ measured in cells/sec. An ABR source is expected to declare only its peak cell rate PCR and minimum cell rate MCR. Moreover, it is assumed that ABR sources always have cells to send, i.e. they are persistent sources [6].

3. THE CONTROL LAW

In this section we first develop a mathematical model of the closed loop system and then we develop the controller by following the Smith principle and using feed-forward compensation.

3.1 Model of the closed loop controlled system

To implement an end-to-end closed-loop control it is necessary that a flow of control cells goes from the source to the destination and then back from the destination to the source. More specifically, we adopt the scheme that the ATM Forum recommends to implement closed-loop control [2]. It requires that each ABR source sends one control RM cell (Resource Management cell) every N_{rm} data cells. RM cells are sent back by the destination to the source to deliver feedback information, which is used to set the input rate of ABR traffic. To design the

control law we assume, for the time being, that the bottleneck queue length is fed back by the receiver to the sender through RM cells. Later, in order to implement an end-to-end control, we will remove this assumption by proposing an end-to-end estimate of the queue length.

Fig. 1 shows the block diagram that model the dynamics of a closed-loop controlled ABR flow.

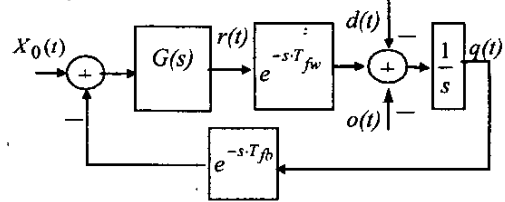


Fig. 1: Block diagram of a closed-loop controlled VC connection

It consists of:

- 1) The bottleneck queue modeled in the Laplace domain by the integrator $1/s$;
- 2) The queue length $q(t)$ that is the output of the integrator, and is bounded by the queue capacity q_c ;
- 3) The available ABR bandwidth $d(t)$ modeled as a disturbance;
- 4) The forward delay T_{fw} from the source to the bottleneck;
- 5) The backward delay T_{fb} from the bottleneck to the destination and then back to the source;
- 6) The controller transfer function $G(s)$ located at the sender;
- 7) The set point $X_0(t)$, that sets a threshold for the queue.
- 8) The overflow data rate $o(t) \geq 0$, which models the loss rate when the input rate is greater than the output rate and the buffer is full. More precisely the overflow data rate is:

$$o(t) = \begin{cases} 0 & \text{if } q(t) < q_c \\ [r(t - T_{fw}) - d(t)]^+ & \text{if } q(t) = q_c \end{cases}$$

with $[x]^+ = \max[x, 0]$. By assuming that at the initial time $t_0=0$ the queue is empty (i. e. $q(0) = 0$), the following equation model for the queue length holds:

$$q(t) = \int_0^t [r(\tau - T_{fw}) - o(\tau) - d(\tau)] d\tau$$

It is important to notice that the queue model above is an *exact* linear dynamic model of a real buffer. It has interesting features since it is not uncommon to find in literature that the finite capacity q_c of a real buffer is modeled through a nonlinear saturation function $sat_{q_c}(\cdot)$ of the queue length¹ [3,8,17].

Modeling a real buffer using saturation has the serious drawback of introducing a nonlinearity that makes exact mathematical analysis intractable [3,17]. Moreover, note that, by using the linear queue model above, the congestion control problem can be easily stated as the goal of achieving $o(t)=0$, that is, no buffer overflows.

Finally, to support our linear modeling we quote V. Jacobson [18]: "A network is, to a very good approximation, a linear system. That is, it is composed of elements that behave like linear operators -integrators, delays, gain stages, etc. Linear systems theory says that if a system is stable, the stability is exponential". Recently, a non-linear differential stochastic equation has been proposed to model the TCP [19]. Since the proposed nonlinear model is mathematically intractable, authors

¹ $sat_{q_c}(x) = \begin{cases} x & \text{if } x \leq q_c \\ q_c & \text{if } x > q_c \end{cases}$

approximate it around an equilibrium point via linearization.

3.2 Designing the rate control algorithm using the Smith principle

A well-designed congestion control algorithm for ABR traffic must achieve two goals. The first one is avoiding buffer overflow, the second is guaranteeing that the source input rate promptly uses all available bandwidth. These goals can be formally stated via the two conditions below:

I) *Stability condition:* $q(t) \leq q_c$ for $t > 0$

This condition guarantees that the bottleneck queue is always bounded, i. e. there are no cell loss;

II) *Full link utilization:* $q(t) > 0$ for $t > RTT$

that is sufficient to guarantee full utilization of the bottleneck link. The reason for having $t > RTT$ is that the feedback cannot reach the source in a time less than the backward propagation delay T_{fb} and the input rate cannot reach the bottleneck queue in a time less than the forward propagation delay T_{fw} . As a consequence, during the transient (i. e. for $t < RTT$), which is the normal operation mode of communication networks, it is never possible to reject the disturbance. For these considerations and in contrast to [7,10,16] our control goal is not disturbance rejection but is to guarantee that the bottleneck queue level is always greater than zero and less than the queue capacity.

Due to the large delays inside the feedback loop, queue level dynamics might exhibit oscillations, and even become unstable. Since the model of the communication system is known without parameter uncertainty, a controller can be designed following the Smith principle. Following this principle, the idea is to look for a controller $G(s)$ so that the input-output dynamic of the system reported in Fig. 1 becomes equivalent to the input-output dynamics of the system depicted in Fig. 2. A major feature of this system is that it is a first order system with a delay in cascade. Thus, by letting the set point $X_0(t)$ be the step function $X_0 \cdot 1(t)$, the queue level is bounded by X_0 .

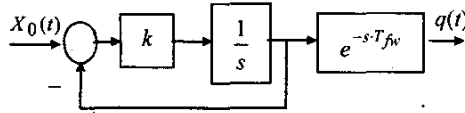


Fig. 2: Block diagram of the desired input-output dynamics.

By equating the transfer functions of the systems reported in Fig. 1 and Fig. 2, the controller $G(s)$ is easily obtained:

$$G(s) = \frac{k}{1 + k(1 - e^{-RTTs})/s}$$

The rate control equation in the time domain is

$$r(t) = k(X_0(t) - q(t - T_{fb})) - \int_{t-RTT}^t r(w)dw \quad (1)$$

This equation can be intuitively interpreted as follows: the computed input rate is proportional, through the coefficient k , to the buffer available space $X_0(t) - q(t - T_{fb})$ decreased by the number of cells released by the source during the last round trip time RTT , i. e. the "in flight cells".

Proposition 1: The system reported in Fig. 1 where the reference signal is the step function $X_0(t) = X_0 \cdot 1(t)$ and the disturbance is the step function $d(t) = D_0 \cdot 1(t - T_1)$, with $T_1 > RTT$, satisfies the stability condition $q(t) \leq X_0$.

Proof: The Laplace transform of the output $q(t)$ in response to the set point $X_0(t) = X_0 \cdot 1(t)$ is

$$QX_0(s) = \frac{X_0 \cdot k e^{-T_{fw}s}}{s(k + s)}$$

The Laplace transform of the output $q(t)$ in response to $d(t) = D_0 \cdot 1(t - T_1)$, with $T_1 > RTT$ is

$$QD_0(s) = D_0 \cdot \left(-\frac{1}{s^2} + \frac{k}{s^2 \cdot (s + k)} \right) \cdot e^{-RTT \cdot s} \cdot e^{-T_1 \cdot s}$$

where $RTT = T_{fw} + T_{fb}$.

By transforming back to time domain it follows:

$$q(t) = qX_0(t) + qD_0(t) = X_0(1 - e^{-(t-T_{fw})k}) \cdot 1(t - T_{fw}) + D_0 \cdot (t - T_1) \cdot 1(t - T_1) + D_0 \cdot (t - T_1 - RTT) \cdot 1(t - T_1 - RTT) + \frac{D_0}{k} (1 - e^{-(t-T_1-RTT)k}) \cdot 1(t - T_1 - RTT) \leq X_0$$

This completes the proof.

3.3 Feedforward disturbance compensation

In this subsection we first analyzes the impact of the threshold X_0 on the bottleneck utilization, then we show that in order to obtain full link utilization and low queue length it is necessary to set X_0 in accordance to the available bandwidth at the bottleneck. Finally, we show that this adaptive setting of X_0 can be obtained by using feedforward disturbance compensation.

Remark 1: From (2) it results that, after the transient time $RTT + 4/k + T_1$, the steady state queue length is:

$$q_{ss} = X_0 - D_0 \cdot RTT - D_0/k \quad (3)$$

By requiring $q_{ss} > 0$ in Eq. (3), in order to ensure full link utilization in steady state condition, it is easy to derive the condition $X_0 > D_0(RTT + 1/k)$.

By noting that the Round Trip Time (RTT) is sum of the propagation delay (RTT_{min}) and the queuing delay $T_q = q_{ss}/D_0$, Eq. (3) can be written as:

$$X_0 = 2 \cdot q_{ss} + D_0 \cdot (1/K + RTT_{min}) \quad (4)$$

By exploiting Eq. (4) it is possible to adaptively set X_0 by taking into account the available bandwidth D_0 in order to obtain the desired steady state queue length q_{ss} and full link utilization. In other words, the feed forward disturbance compensation depicted in Fig. 3 can be exploited.

In order to implement the feedforward disturbance compensation depicted in Fig. 3, it is necessary to estimate the disturbance $d(t)$. This is a key issue and it will be discussed in Section 4.2. In Fig. 3 the bandwidth estimate is modeled by a first order low pass filter with time constant τ_f [9].

Proposition 2: The system reported in Fig. 3, where the disturbance is the step function $d(t) = D_0 \cdot 1(t - T_1)$ with $T_1 > RTT$, satisfies the stability condition

$$q(t) \leq 2 \cdot q_{ss}$$

Proof: The Laplace transform of the output $q(t)$ in response to $2q_{ss}1(t)$ and to $d(t) = D_0 \cdot 1(t - T_1)$, with $T_1 > RTT$ is:

$$Q(s) = -\frac{D_0}{s^2} e^{-T_1} s + \frac{D_0}{s^2} e^{-(RTT+T_1)} s + \frac{D_0}{k} \frac{ke^{-(RTT+T_1)} s}{s \cdot (s+k)(1+\tau f s)} + D_0 \cdot RTT_{min} \frac{ke^{-(RTT+T_1)} s}{s \cdot (s+k)(1+\tau f s)} + \frac{D_0}{k} \frac{ke^{-(RTT+T_1)} s}{s \cdot (s+k)} + \frac{2 \cdot q_{ss} \cdot k}{s \cdot (s+k)} e^{-T_{fw}} s$$

By transforming back to time domain $Q(s)$, it follows that:

$q(t) = A_1(t) + A_2(t) + A_3(t)$, where

$$\dot{A}_1(t) = 2 \cdot q_{ss} \cdot (1 - e^{-(t-T_{fw})k}) \cdot 1(t - T_{fw}) \leq 2 \cdot q_{ss}$$

$$A_2(t) = D_0 \cdot RTT_{min} \left(1 - \frac{e^{-(t-T_1-RTT)k}}{1 - \kappa f} \right) \cdot 1(t - T_1 - RTT) + \frac{(t - T_1 - RTT)}{1 - \kappa f} \tau f$$

$$+ D_0 \cdot RTT_{min} \frac{\kappa f \cdot e^{-\tau f}}{1 - \kappa f} \cdot 1(t - T_1 - RTT) +$$

$$- D_0 \cdot (t - T_1) \cdot 1(t - T_1) + D_0 \cdot (t - T_1 - RTT) \cdot 1(t - T_1 - RTT) \leq 0$$

$$A_3(t) = \frac{D_0}{k} \left(1 - \frac{e^{-(t-T_1-RTT)k}}{1 - \kappa f} + \frac{\kappa f \cdot e^{-\tau f}}{1 - \kappa f} \right) \cdot 1(t - T_1 - RTT) +$$

$$- \frac{D_0}{k} (1 - e^{-(t-T_1-RTT)k}) \cdot 1(t - T_1 - RTT) \leq 0$$

This completes the proof.

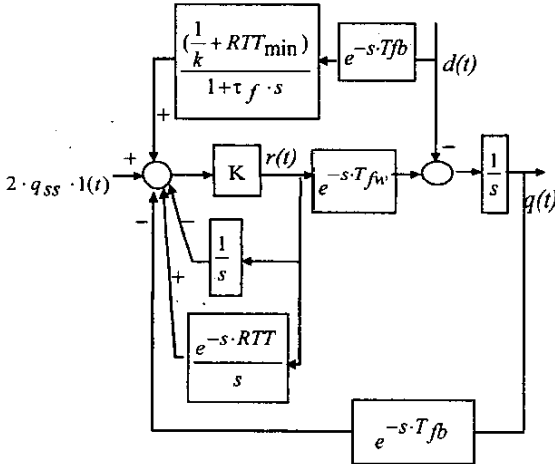


Fig. 3: Feedforward disturbance compensation.

By using the feed-forward disturbance compensation the control law becomes:

$$r(t) = k \cdot \left[2 \cdot q_{ss} + \hat{d}(t - T_{fb}) \cdot \left(\frac{1}{k} + RTT_{min} \right) - q(t - T_{fb}) - \int_{t-RTT}^t r(w) dw \right] \quad (5)$$

where $\hat{d}(t - T_{fb})$ is the delayed and filtered measurement of the available bandwidth $d(t)$.

4. ALGORITHM IMPLEMENTATION

Based on the overall control law design outlined in the previous section, we develop a detailed implementation of the algorithm

for the ABR traffic. We assume that the source time stamps RM cells and measures the RTT time when RM cells come back to the source. Network nodes do not perform neither any computation of explicit rates nor stamp any value. Notice that the feedback information is relayed in RM cells and thus not available in continuous time, but rather in sampled form. To discretize the continuous control Eq. (5), we simply invoke the Shannon theorem [1] which requires that the ratio of the system time constant $\tau = 1/k$ over the sampling time T_s must be at least greater than 2.

4.1 Initial Estimate of the available bandwidth

The source estimates the best effort available bandwidth by sending probing cells. In particular the source sets $Nrm=0$ and it transmits at its maximum rate a number N_1 of RM cells. When cells come back from destination, the source computes the minimum RTT , estimates the available bandwidth, set $k=a/RTT_{min}$, select a value for the stationary queue level q_{ss} and finally start cells transmission at rate computed according to control law (5). The parameter a allows to fix the time constant $1/k$ as a multiple of RTT_{min} . Following specifications in [2], we have selected the default value $Nrm=32$ to ensure that the control overhead does not exceed approximately 6%.

4.2 Estimate of the available bandwidth

In order to apply control equation (5), it is necessary an on-line estimation of available bandwidth. The main idea is to perform the estimate by low-pass filtering the returning flow of RM cells. By noting that, the returning rate of RM cells can never exceeds the input rate $r(t)$, the network capacity is probed by transmitting at peak cell rate (PCR) only for a "while". Moreover, in order to avoid network congestion, we must transmit by following the rate dictated by the control equation (5). Therefore, to meet both requirements stated above, we transmit at the average rate $r(t)$ over two rounds of Nrm data cells. In the first round, in order to probe network capacity, we transmit 33 cells at the peak cell rate and in the second round we transmit at a rate r_c so that the mean transmission rate of the 66 cells is equal to $r(t)$. In other words, the transmission rate $r(t)$ given by Eq. (5) involves groups of 66 cells $= 2 \cdot (Nrm + 1)$. First 33 cells are transmitted at maximum rate (PCR), other 33 cells are transmitted at a rate given by

$$r_c = \frac{r(t) \cdot PCR}{2 \cdot PCR - r(t)} \quad (6)$$

We will refer to the RM cells transmitted at the end of the burst of 33 cells at PCR as RM_1 cells, whereas we will refer to the RM cells transmitted at the end of the second group of 33 cells at a rate given by Eq. (6) as RM_2 cells.

Let Δt_k be the interarrival time at the source between an RM_2 cell and the next RM_1 cell. During this interval, cells are transmitted at peak rate

The signal Δt_k can contain high frequency components due to stochastic multiplexing with coexisting traffic such as VBR or ABR. Since congestion is due to low frequency components [9], it is necessary to filter out high frequency components. We propose the following low-pass linear time-varying filter to obtain the filtered value $\hat{\Delta t}_k$ as:

$$\hat{\Delta t}_k = \alpha_k \cdot \hat{\Delta t}_{k-1} + \frac{1 - \alpha_k}{2} \cdot (\Delta t_k + \Delta t_{k-1}) \quad (7)$$

where $\alpha_k = \frac{2\tau f - \Delta_k}{2\tau f + \Delta_k}$, Δ_k is the interarrival time between

two consecutive RM_1 cells and $1/\tau_f$ is the filter cut-off frequency. Notice that the filter is time-varying to counteract the fact that the sampling interval Δ_k is not constant. When Δ_k is constant then α_k is constant too and the filter (7) reduces to the discretization, via bilinear transformation, of a first order continuous low-pass filter with time constant τ_f [1]. It is well known from the Nyquist sampling theorem that in order to avoid aliasing effects the filter time constant τ_f must be at least two times the maximum sampling time $\Delta k_{\max}$. The maximum interarrival time between consecutive RM_1 cells is $\Delta k_{\max} = 2 \cdot (Nrm + 1) / MCR$. To be conservative we choose $\tau_f = 4 \cdot \Delta k_{\max}$. The filtered value of the interarrival time $\hat{\Delta t}_k$ is proportional to the inverse of the low-pass component of the available bandwidth. Therefore, every time an RM_1 cell is received, the available bandwidth is estimated as follows:

$$\hat{d}(t - T_{fb}) = (Nrm + 1) / \hat{\Delta t}_k$$

4.3 Estimate of the queue length

On RM cell arrival the source computes the queuing delay $T_q = (RTT - RTT_{\min})$ and then estimates the queue length using the estimate of available bandwidth $\hat{d}(t - T_{fb})$ as follows:

$$q(t - T_{fb}) = (RTT - RTT_{\min}) \cdot \hat{d}(t - T_{fb})$$

5. PERFORMANCE EVALUATION

In this section we test the proposed algorithm under strenuous conditions by simulating its performance in the Generic Fairness Configuration-2 (GFC-2) shown in Fig. 4.

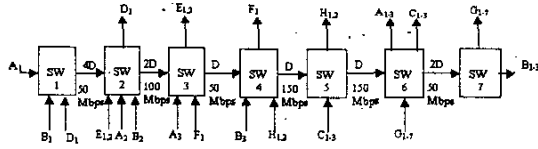


Fig. 4: Generic Fairness Configuration 2 $D = 10^3$ Km; Entry/Exit links of length D and speed 150Mbps

GFC-2 is a testbed proposed by the ATM Forum. The switches implement *per-VC FIFO* queuing. The sources are persistent and start data transmission at the same time. Parameters have been chosen as: $a=1$, $N_i=15$, $q_{ss}=100$, $MCR = PRC/100$, we have assumed that, during the connection phase, each source sets PCR according to the capacity of its own outgoing link and then switches reduce this value to the minimum bandwidth of the links encountered along the VC path. The expected results for fair ACR allocation are shown in Table I [7].

Table I: Fair allocations

VC	A	B	C	D	E	F	G	H
Fair Share (Mbps)	10	5	35	35	35	10	5	52.5

Simulation results are shown in Figs. 5 and 6. Fig. 5 shows the queue lengths at the switch output ports, whereas Fig. 6 shows the ACRs of the ABR sources.

Fair allocations are achieved in 1s and in the same time queues are drained. During the transient period queue lengths are bounded by reasonable values, that is the max queue length at SW6 is less than 15,000 cells that is equal to the bottleneck capacity (50Mbps) times the maximum Round Trip Time (130ms) of the VC_{B1}. The steady state queue levels are kept to a

constant small value (max = 1200 cells in SW6). The paper [7] reports performance of ERICA in the GFC-2 scenario: the max queue length is less than 30,000 cells, fair allocation are achieved in 400ms and queues are drained in 800ms. Comparison between the two algorithms clearly highlights that ERICA requires a buffer size much larger than the one of the proposed algorithm to avoid buffer overflow.

6. CONCLUSIONS

In this paper an end-to-end congestion control algorithm for ATM networks implementing *per-VC FIFO* buffering is designed by exploiting Smith's principle and feed-forward compensation. Its properties are first mathematically analyzed and then tested by simulating the GFC-2 scenario. Performance indices comparable or slightly better than those obtained by the leading and well-known ERICA switch algorithm are obtained, even though ERICA is more complex since it is based on explicit feedback and requires support from network nodes.

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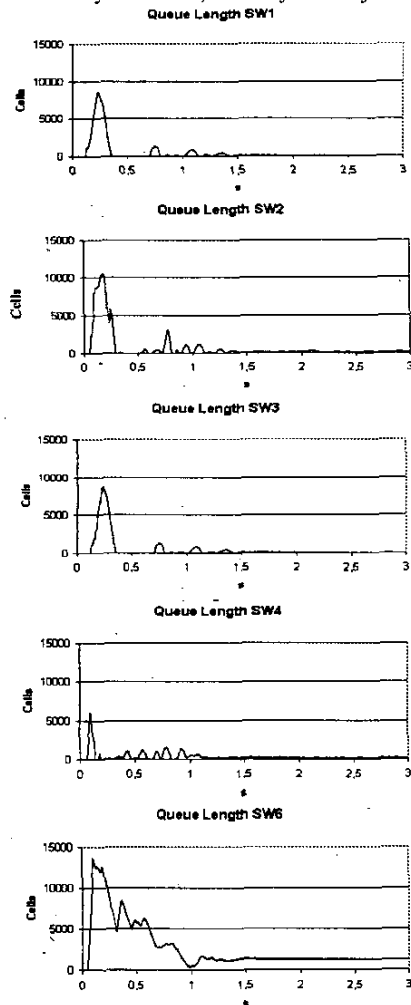


Fig. 5: Queue Lengths.

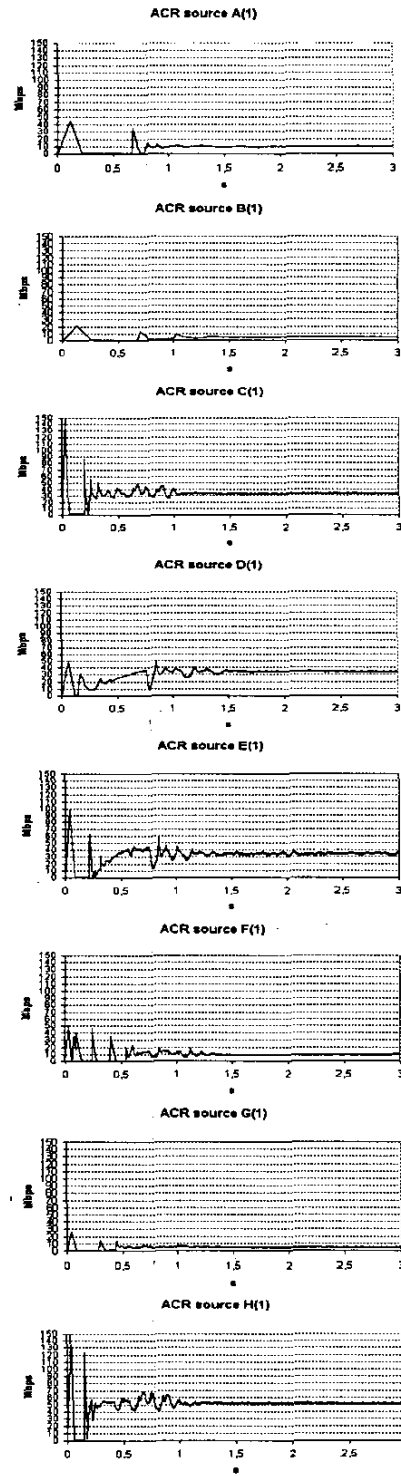


Fig. 6: ACRs of sources.