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# Design and analysis of delay control for CBR services in ATM networks

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**Abstract**

Asynchronous transfer mode (ATM) networks are designed to provide end-to-end transport of user data via virtual connections with specified quality of services (QOS), which is expected to be satisfied through effective traffic control mechanisms. This paper presents an adaptive delay-jitter control (ADJC) method that guarantees the delay bounds of constant bit rate (CBR) services in ATM networks. Our proposed method is based on a node-by-node time-frame scheme. Through both mathematical derivation and simulation, we have demonstrated that ADJC features a significant improvement over existing approaches in jitter delay, which can be reduced to one time-frame from end-to-end for CBR traffic. Simulation results have shown that CBR services receive satisfactory delay performance using our approach. © 1998 Elsevier Science B.V.

*Keywords:* ATM; CBR services; Adaptive delay-jitter control

## 1. Introduction

The asynchronous transfer mode (ATM) is selected as the basis of B-ISDN due to its flexibility and efficiency for accommodating a wide range of services. The high bandwidth offered by ATM networks makes real-time applications, such as audio/video transport, feasible. To support these applications, ATM networks are designed to provide end-to-end transport of user data with specified quality of services (QOS), which is expected to be satisfied through effective traffic control. Usually, in high-speed networks such as ATM, the propagation delay between intermediate nodes is relatively long compared with the packet (cell) transmission time, thus the reactive control method may not be effective to accommodate delay-sensitive traffic (e.g. constant bit rate (CBR) traffic, real-time traffic). On the other hand, preventive control approaches [1–7] which try to avoid network congestion in advance are more promising. The cell scheduling discipline at each intermediate node determines whether a cell is ready to be transmitted based on its QOS requirement. There are work-conserving and non-work-conserving cell scheduling disciplines. In the former the transmission is never idle as long as there is a cell waiting to be sent. Several well-known examples are weighted fair queuing (WFQ) [8], virtual clock [10], packet-by-packet generalized processor sharing (PGPS) [9,24] and self-clocked fair queuing (SCFQ) [11,12,25]. While in the latter, the transmission may be idle even when there are cells ready to send; some popular examples are stop-and-go framing (S&G) [15–17], hierarchical round robin (HRR) [18], rate control static priority (RCSP) [11,19,20] and traffic-controller rate-monotonic priority scheduling (TCRM) [21]. Work-conserving disciplines provide lower average queuing delay and larger end-to-end delay jitter than non-work-conserving disciplines. In work-conserving discipline, traffic needs to be characterized inside the network on a per connection basis to derive end-to-end delay bounds and buffer space requirements. However, the traffic pattern is usually distorted inside the network due to the cell transmission contention, and it is difficult to restore the original traffic pattern after the distortion.

The non-work-conserving disciplines have several advantages for supporting CBR services. These advantages include : (1) the end-to-end queuing delay can be easily derived through local delay analysis at each intermediate switch; (2) the required buffer space to prevent cell overflow at each switch can be reduced through traffic regulation inside the network; (3) it features shorter delay jitter performance and smaller buffer requirement for playing back at the end system. However, the non-work-conserving disciplines suffer lower bandwidth utilization because the transmission service may be idle even when there are cells

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waiting. This problem can be improved by accounting for non-delay-sensitive traffic in an integrated-services environment. A well-known non-work-conserving discipline, S&G frame strategy, has a remarkable end-to-end delay bound and delay-jitter bound compared to HRR, but its bandwidth management is inflexible and it is difficult to choose a proper time-frame size to meet various application needs. While RCSP provides a nearly optimum packet delay-jitter control for real-time traffic, however, it needs to pass the delay information (cell eligible time) from node to node and to synchronize the clock time of all intermediate nodes precisely, which is extremely difficult in an ATM wide-area network environment. TCRM provides a bounded end-to-end delay for real-time communications. It provides lower average queuing delay compared with S&G, but its delay-jitter control is worse than that in S&G.

This paper presents a feasible traffic control approach that guarantees both end-to-end delay bound and delay-jitter bound for CBR services in ATM networks. Apparently, we can reconstruct the characteristics of a CBR stream at the end system as long as its delay-jitter bound is guaranteed. We proposed a cell scheduling method based on time-frame synchronization and non-preemptive priority sorting, and demonstrated through mathematical derivation and system simulation that our cell scheduling method achieves a significant improvement over existed approaches in end-to-end delay jitter, as well as provides an adaptive delay-jitter control with end-to-end delay jitter reduced to one time-frame. The organization of this paper is as follows. Section 2 addresses the traffic control framework. In Section 3 the mathematical analysis of the proposed method is presented. The system simulation and performance evaluation are discussed in Section 4. Section 5 concludes the work.

## 2. The traffic control framework

Constant bit rate (CBR) services such as CBR audio and video are expected to comprise a significant portion of the traffic on ATM networks. Two important QOS measures for CBR traffic are the end-to-end delay and the delay-jitter, which can be guaranteed with an effective strategy for bandwidth reservation and traffic control. We use a time-frame-based technique that combines both call admission and cell scheduling policies to accomplish the task. The former decides whether to accept or to reject an incoming connection request, so that network loading can be kept below the congestion level, and the delay bounds of those accepted connections can all be satisfied, while the latter reconstructs the time property of the admitted traffic in each intermediate node, and this is essential to guarantee the queuing delay bound. The source traffic must be shaped through a regulator as smooth traffic pattern in which the cell interarrival time is restricted. Before presenting the proposed traffic control scheme, it is necessary to define the simple-smooth traffic in the ATM network as follows.

**Definition 1.** In ATM network, the cell stream of a connection is defined as *simple-smooth traffic* (SST) with time frame $T$ (in cell slot) if, during each non-trivial period $T$ (i.e. $T > 1$), there is at most one cell transmission of the same stream. A connection that carries an SST in the network is called an SST connection or an SST stream.

Assume a CBR connection has a guaranteed bandwidth $R$ with time frame $T$, which is obtained through negotiation during the connection setup based on the cell interarrival characteristics, where $T = \lceil C/R \rceil$, and $C$ is the link capacity. Fig. 1 shows an SST stream, the traffic of a CBR connection in the network will be regulated as an SST stream throughout the network using our proposed strategy.

Before we proceed to the discussion, it has to define the terminology of eligibility.

**Definition 2.** A cell of an SST connection with time frame $T$ is said to be eligible in an intermediate node if it is ready to be scheduled, and its preceding cell of the same connection was scheduled no shorter than $T$ ago.

Fig. 2 shows the CBR transmission control configuration of our traffic control architecture. The transmission controller (TC) consists of a cell dispatcher (CD), a rate controller (RC) and a priority service scheduler (PSS). The CD distributes a cell to its associated FIFO queue. The RC provides a set of regulators to accommodate each CBR connection. Each regulator shapes the input traffic of the associated connection as SST according to the eligibility rule. The PSS schedules all eligible cells for transmission. Through such traffic regulation, the TC can provide a bounded end-to-end queuing delay for CBR services. Assume that shared memory is used, then each output buffer can be logically partitioned into $n$ separate FIFO queues for holding the incoming cells of the associated CBR connections. In addition, we can assign $N$ priority levels to accommodate the eligible cells from their corresponding CBR service classes.

Our proposed strategy is similar to the S&G scheme, Zhang's RCSP [20] and TCRM [21] in the objective of transmission-control. However, our service discipline provides a significant improvement in the average end-to-end queuing delay compared with the first scheme, and features

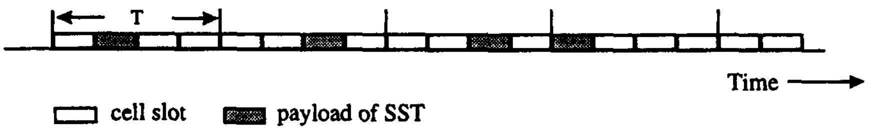



Fig. 1. An example of simple smooth traffic with time-frame $T = 4$ cell slots.

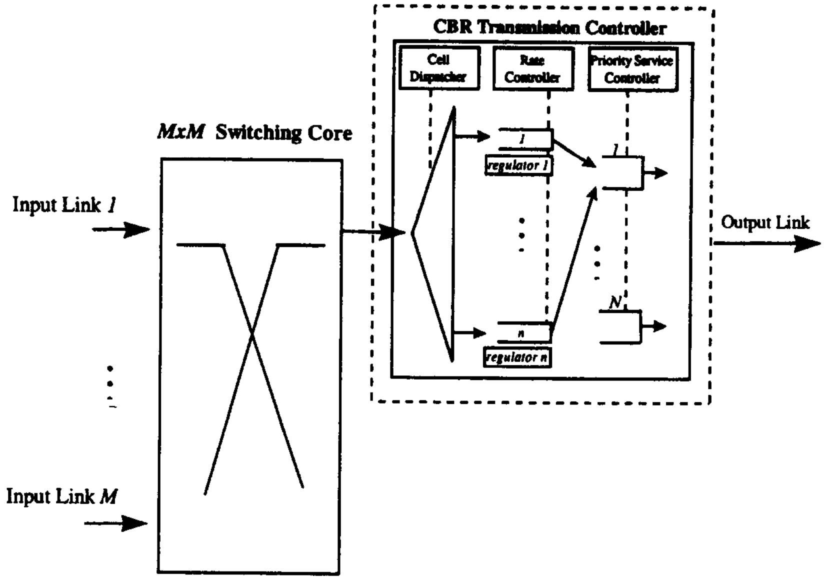



Fig. 2. Proposed CBR transmission control in an ATM switching node.

a more realistic time-frame based method for delay-jitter control compared with the latter two. The main difference between the S&G approach and ours is that ADJC avoids the extra waiting time through node-by-node synchronization, which allows arbitrary time-frame sizes that are not supported in S&G, while the difference between the delay-jitter control in RCSP and ours is the time synchronization method. ADJC does not need the cell eligibility time information from the upstream node; therefore, not only the implementation becomes feasible, but also the overhead is greatly reduced, since per cell time-stamping and time synchronization between different switching nodes are not realistic under ATM network environment. While the rate-jitter regulator in RCSP controls delays jitter by partially reconstructing the traffic pattern, it introduces large delay and buffer space requirement at a single switch node. Although the RCSP avoids the dependencies between prioritized scheduling and bandwidth allocation, its stringent admission control conditions for satisfying the delay-bound requirement restrict the resource utilization. The main difference between the TCRM approach and ours is that ADJC is time-frame based which can be realized using counters, thus it is more feasible as well as provides a better delay-jitter control.

## 3. Analysis of proposed traffic control method

First, we propose a simple delay-jitter control strategy (SDJC), then we will enhance SDJC to an adaptive delay-jitter control strategy (ADJC).

### 3.1. Simple delay-jitter control (SDJC) strategy

The SDJC deals with two issues: the resolution of transmission contention among multiple SSTS, and the maintenance of the outgoing cell stream of each connection as SST. We set three rules for intermediate nodes. (1) In order to meet the delay requirement among various service types, the priority ordering of transmission will prefer the cell with the smallest time-frame. Of course, this priority ordering can be adjusted flexibly for practical needs. (2) To keep the traffic pattern of a connection as SST throughout the network, only one cell can be granted eligibility within one time-frame interval for each connection. (3) The transmission for CBR service will be idle if there is no eligible cell.

Let $N$ be the time-frame priority level and $n_j$ be the number of connections with time-frame $T_j$, where $T_1 < T_2 < \cdots < T_N$, $0 \le n_j$ and $1 \le j \le N$. When a new CBR connection request is received by the network, a CAC procedure is executed to decide whether to accept or to reject the call. If there already existed $\sum_{k=1}^{N} n_k$ CBR connections in any node along the end-to-end path, the newly requested CBR connection with time-frame $T_i$ can be accepted by that node if the following conditions are satisfied. The first condition,

$$\frac{1}{T_i} + \sum_{k=1}^{N} \left(n_k \times \frac{1}{T_k}\right) \le 1, \tag{1}$$

ensures that the total traffic load will not exceed the link capacity. It implies that the new CBR connection can obtain the required bandwidth, while the second condition,

$$\sum_{\text{forall} j,\, T_j \le T_i} n_j \times \left(\left\lceil \frac{T_i}{T_j} \right\rceil\right) \begin{cases} \le T_i \\ > T_i \text{ and } \displaystyle\sum_{\text{forall} l,\, T_l \le T_{i-1}} n_l \times \left(\left\lceil \frac{T_{i-1}}{T_i} \right\rceil\right) < T_{i-1}, \end{cases} \tag{2}$$

is necessary to guarantee the queuing delay bound of a CBR connection in an intermediate node. Due to the transmission contention among connections with various priority levels, the maximum number of cell arrivals within time-frame $T_i$ from those connections with time-frame less or equal to $T_i$ must be not greater than $T_i(\Sigma_{T_j \le T_i} n_j \times ([T_i)/(T_j)]) \le T_i)$ to guarantee the bounded delay. However, it can also provide the bounded delay in the case that the maximum number of arriving cells within time-frame $T_i$ from those connections with time-frame less than or equal to $T_i$ is greater than $T_i$. This is because there is an idle cell slot during the $T_{i-1}$ time-interval. For example, assume there already existed three connections with $T_1 = 2$, $T_2 = 4$ and $T_3 = 8$. When a new request with $T_4 = 9$ arrives, it will be accepted. This will not deteriorate the QOS of the CBR connection because the new connection will use the idle slot within the time-frame $T_3$. In this case, the CAC must determine whether $\Sigma_{T_j \le T_{i-1}} n_j \times (\lceil (T_{i-1})/(T_j) \rceil)$ is greater than $T_{i-1}$ or not. If $\Sigma_{T_j \le T_{i-1}} n_j \times (\lceil (T_{i-1})/(T_j) \rceil) < T_{i-1}$, it means that an idle slot can be used within $T_{i-1}$ time duration. Therefore, the new request connection can be accepted as long as the number of connections with time-frame $T_i$ is no greater than $\lfloor (T_{i-1})/T_i \rfloor$ (i.e. $n_i \le \lfloor (T_{i-1})/T_i \rfloor$).

A new connection request is accepted if the network has enough resources to provide the QOS requirements of the connection without affecting the QOS already established in the network. Hence, when a new connection request satisfies the above two conditions, the CAC procedure must further decide whether admitting the new connection causes the QOS violation of the existing connection or not. The algorithm for the CAC is stated as follows:

**Algorithm Call–Admission–Control:**

Declaration:

$N$: the number of time-frame priority levels

$T_i$: the time-frame size of the new request, which belongs to class $i$ $(1 \le i \le N)$

$n_j$: the number of connections with time-frame $T_j$, where $T_1 < T_2 < \cdots < T_N$

$\sum_{k=1}^{N} n_k$: the number of existing CBR connections in an intermediate node

$R$_Flag: the recursive status flag, it is 0 if the recursive process is terminated, and 0 otherwise

$A$_Flag: the output status flag, it is 1 if the new connection is accepted. and 0 otherwise

Initialization:

$A$_Flag = 0; $R$_Flag = 1;

```
Begin
    if 1/T_i + Σ_{k=1}^{N} (n_k × 1/T_k) ≤ 1 then
        CAC(i, T_i);
    endif
end
```

```
Procedure CAC(i, T_i):
  Begin
     if (R_Flag) then
    if (i = N) then
       if Σ_{T_j ≤ T_N} n_j × (⌈T_N/T_j⌉) ≤ T_N then
       A_Flag = 1; R_Flag = 0;
       else
         if Σ_{T_j ≤ T_{N-1}} n_j × (⌈T_{N-1}/T_j⌉) > T_{N-1} and
         n_i ≤ ⌊T_N/T_{N-1}⌋ then
         A_Flag = 1; R_Flag = 0;
         endif
       endif
    else
       if Σ_{T_j ≤ T_i} n_j × (⌈T_i/T_j⌉) ≤ T_i then
       CAC(i+1, T_{i+1});
       else
         if Σ_{T_l ≤ T_{i-1}} n_l × (⌈T_{i-1}/T_l⌉) < T_{i-1} and
         n_i ≤ ⌊(T_i)/(T_{i-1})⌋) then
         CAC(i+1, T_{i+1});
         else
         R_Flag = 0;
         endif
       endif
    endif
  endif
End
```

Assume an SST cell has a higher transmission priority than a non-SST cell, we can show that an SST cell with time-frame $T_k$ can be transmitted within $T_k$ time interval once it becomes eligible.

**Lemma 1.** Given an SST connection $k$ with time-frame $T_k$, let $W_i^j$ be the delay time for its $j$th cell to be transmitted in node $i$ once it becomes eligible, then $1 \le W_i^j \le T_k$.

**Proof.** Since the TC transmits eligible cells only, thus according to the traffic regulation, each SST connection will have at most one cell ready within each associated time-frame interval. An SST connection with a smaller time-frame has higher priority than that with a larger time-frame, cell transmissions of these connections with a time-frame no greater than $T_k$ will not be affected by those with a time-frame larger than $T_k$. According to our call admission strategy and priority scheduling, the incoming cell with time-frame $T_k$ eligible at time $t$ can always be transmitted within $[t+1, t+T_k]$. This assertion is also true for an arbitrarily chosen time-frame. QED.

Given a connection $i$ with time-frame $T_i$, we use the counter $C_{tr}(i)$ to track the cell-eligibility. Once a cell of connection $i$ becomes eligible, $C_{tr}(i)$ is set to $T_i$, which then counts down to zero in $T_i$ cell-slot time. When a cell arrives, it checks whether $C_{tr}(i)$ is equal to 0 or not. If yes, the arriving cell will become eligible immediately, otherwise it has to wait. Through such a mechanism, the TC can regulate each cell stream to be an SST pattern. The algorithm for the TC is stated as follows:

**Algorithm for CBR transmission controller:**

Declaration:

$n$: the total number of connections

$C_{tr}(i)$: the traffic regulation counter of the $i$th connection $(1 \leq i \leq n)$

$N$: the number of priority levels

$T(i)$: the timeframe of $i$th connection $\left(1 \leq i \leq n, T(i) \in \{T_1, T_2, \cdots, T_N\}\right)$

$L(i)$: the $i$th FIFO queue status. It is 0 if the queue is empty, and 1 otherwise

Initialization:

```
Ctr(i) = 0; i = 1···n /* for accepted connections with
    time-frame Ti */
Begin
  for (i = 1 to n) do
    if Ctr(i) > 0
      Ctr(i) = Ctr(i) − 1;
    endif
    if (L(i) ≠ 0) and (Ctr(i) = 0) then
      insert the HOL cell of ith FIFO to the PSS for
        priority transmission service;
      Ctr(i) = T(i)
    endif
  endfor
End
```

Assuming that a connection $k$ with time-frame $T_k$ routes through $H$ hops, let $\tau_l$ represent the fixed-time overhead including the cell processing time in the $l$th switching node and the propagation delay in the outgoing link, where $l = 1 \ldots H$, and $\tau_0$ be the propagation delay of the source access link. It is obvious that the end-to-end delay of $j$th cell for connection $k$ is $D_j^k = \tau^k + QD_j^k$, where $\tau^k = \sum_{i=0}^{H} \tau_i$ is the total propagation delay between two end systems plus the total processing time in the intermediate nodes; $QD_j^k = \sum_{i=1}^{H} Q_i^j$, where $Q_i^j$ is the total queuing delay of the $j$th cell in the $i$th switching node. $Q_i^j$ includes $q_i^j$, the queuing delay (excluding the fixed processing time) of $j$th cell waiting for eligibility after it finishes the necessary processing in the $i$th node, and $W_i^j$, the delay time for the $j$th cell to be transmitted once it becomes eligible at node $i$. Then we have the following lemma.

**Lemma 2.** For any SST connection $k$ with time-frame $T_k$, the queuing delay for eligibility, $q_i^j$, of the $j$th cell in the $i$th node is bounded by $[0, T_k - 1]$.

**Proof.** Let $q_i^j$ be the queuing delay of the $j$th cell waiting for eligibility and $W_i^j$ be the delay time for the $j$th cell to be transmitted once it becomes eligible at node $i$, in other words, $q_i^j = Q_i^j - W_i^j$.

As illustrated in Fig. 3, the difference between the departure time of the $(j-1)$th cell and the $j$th cell in the $(i-1)$th node is $T_k - W_{i-1}^{j-1} + W_{i-1}^j$, thus

$$\begin{aligned} q_i^j &= T_k - \left(T_k - W_{i-1}^{j-1} + W_{i-1}^j - q_i^{j-1}\right) \\ &= W_{i-1}^{j-1} - W_{i-1}^j + q_i^{j-1}. \end{aligned} \tag{3}$$

Needless to say, $q_i^j$ must be no less than zero; thus, if Eq. (3) is no greater than zero then $q_i^j = 0$, which means that the

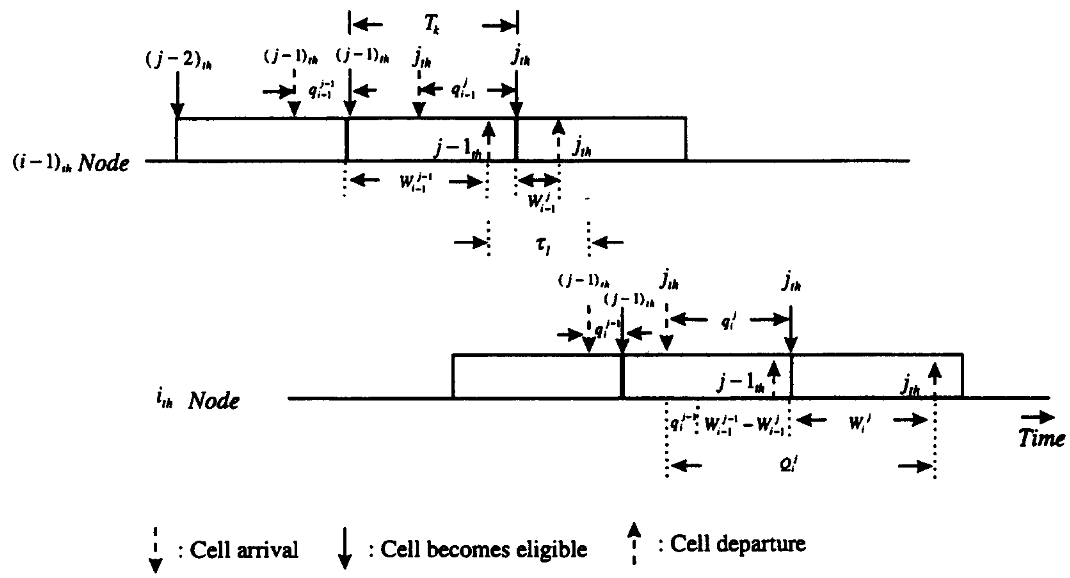



Fig. 3. Timing diagram of the cell queuing delay in a switching node.

arriving cell is eligible immediately; otherwise $q_i^j = q_i^{j-1} + W_{i-1}^{j-1} - W_{i-1}^j$, hence the following equation

$$q_i^j \begin{cases} q_i^{j-1} + W_{i-1}^{j-1} - W_{i-1}^j, & \text{if} \quad W_{i-1}^{j-1} - W_{i-1}^j + q_i^{j-1} > 0 \\ 0, & \text{otherwise} \end{cases}, \text{ where } q_i^1 = 0,\ i = 1 \cdots H,\ j = 1, 2, \cdots. \tag{4}$$

must hold for any cell $j$ of connection $k$ in the $i$th node. It is evident that $q_i^j$ should have a positive value from Eq. (3). To derive the upper bound of $q_i^j$, we expand Eq. (4) as follows

$$\begin{aligned} q_i^j &= q_i^{j-1} + W_{i-1}^{j-1} - W_{i-1}^j \\ &= q_i^{j-2} + W_{i-1}^{j-2} - W_{i-1}^{j-1} + W_{i-1}^{j-1} - W_{i-1}^j \\ &\vdots \\ &= q_i^j + W_{i-1}^1 - W_{i-1}^2 + W_{i-1}^2 - W_{i-1}^3 + \cdots + W_{i-1}^{j-2} \\ &\quad - W_{i-1}^{j-1} + W_{i-1}^{j-1} - W_{i-1}^j. \end{aligned} \tag{5}$$

Since $C_{tr}(i)$ is preset to 0, when the first cell of a connection arrives, it becomes eligible immediately, i.e. $q_i^1 = 0$, where $i = 1 \cdots H$. Therefore, the maximum value of $q_i^j$ can be derived as follows:

$$\begin{aligned} q_i^j &= W_{i-1}^1 - W_{i-1}^2 + W_{i-1}^2 - W_{i-1}^3 + \cdots + W_{i-1}^{j-2} \\ &\quad - W_{i-1}^{j-1} + W_{i-1}^{j-1} - W_{i-1}^j = W_{i-1}^1 - W_{i-1}^j \le T_k - 1, \\ &\text{where } i \ge 2. \end{aligned} \tag{6}$$

Since the traffic from the source is in SST pattern, thus $q_i^j$ will have the same property as $q_i^j$ for $i \ge 2$. From Lemma 1, it is clear that $q_i^j$ is bounded by $[0, T_k - 1]$. QED.

Theorems regarding the traffic property under the SDJC strategy are given as follows:

**Theorem 1.** For any SST connection $k$ with time-frame $T_k$, the maximum delay, $Q_i^j$ of the $j$th cell in the $i$th node is bounded by $[1, 2T_k - 1]$.

**Proof.** Since $Q_i^j$ consists of two parts, the queuing delay waiting for eligibility and the delay to complete the transmission, from Lemma 1 and Lemma 2, the following inequality

$$0 < Q_i^j = W_i^j + q_i^j \le 2T_k - 1 \tag{7}$$

must hold for any cell of connection $k$. Clearly, the queuing delay $Q_i^j$ of the $j$th cell in the $i$th node is bounded by $[1, 2T_k - 1]$. This completes the proof. QED.

If the $j$th cell becomes eligible immediately after its arrival at the $i$th node (i.e. $q_i^j = 0$), the total queuing delay of the $j$th cell will be no greater than $T_k$. Otherwise (i.e. $q_i^j \ne 0$), the total queuing delay of the $j$th cell may be larger than $T_k$ in the $i$th node. It is obvious that the end-to-end delay of the $j$th cell of a given connection $k$ is $D_j^k = \tau^k + QD_j^k$, where $QD_j^k = \sum_{i=1}^H Q_i^j$ and $Q_i^j$ is the total delay of the $j$th cell in the $i$th node.

**Theorem 2.** For any SST connection $k$ with time-frame $T_k$, which routes through $H$ hops, the end-to-end delay $D_j^k = \tau^k + QD_j^k$ is bounded by $[\tau^k + H, \tau^k + HT_k + T_k - 1]$, where $QD_j^k \Sigma_{i=1}^H Q_i^j$.

**Proof.** The delay time, $QD_j^k \sum_{i=1}^H Q_i^j$, of the $j$th cell can be substituted by

$$\begin{aligned} QD_j^k &= \sum_{i=1}^H (q_i^j + W_i^j) = q_1^j + W_1^j + q_2^j + W_2^j + \cdots + q_H^j \\ &\quad + W_H^j = q_1^j + W_1^j + (W_1^1 - W_1^j) + W_2^j + (W_2^1 - W_2^j) \\ &\quad + W_3^j + \cdots + W_{H-1}^j + (W_{H-1}^j - W_{H-1}^j) + W_H^j \\ &= q_1^j + W_1^1 + W_2^1 + W_3^1 + \cdots + W_{H-1}^1 + W_H^j. \end{aligned} \tag{8}$$

If the CBR source traffic is regulated as a constant traffic pattern (i.e. fixed interarrival), the values of $q_i^1$ and $q_1^j$ will be equal to 0. From Lemma 1 and Lemma 2, $W_i^j$ and $q_i^j$ are bounded by $[1, T_k]$ and $[0, T_k - 1]$, respectively, thus Eq. (8) becomes

$$QD_j^k = q_1^j + \left( \sum_{i=1}^{H-1} W_i^1 \right) + W_H^j \le HT_k \tag{9}$$

It should be pointed out that if a source traffic is not regulated as a constant traffic pattern, the time intervals between any two data cells in consecutive time-frames may be not identical (i.e. $0 \le q_1^j \le T_k - 1$), thus

$$H \le QD_j^k \le (T_k - 1) + HT_k. \tag{10}$$

As a sequel, the maximum end-to-end delay of the $j$th cell for connection $k$ is bounded by

$$\tau^k + H \le \tau^k + QD_j^k \le \tau^k + (T_k - 1) + HT_k. \tag{11}$$

This completes the proof. QED.

From Theorem 2, we can find out that the delay-jitter of SDJC is $(H+1)(T_k - 1)$. In order to achieve a better delay-jitter performance, we improve SDJC to the adaptive delay-jitter control described as follows.

### 3.2. Adaptive delay-jitter control (ADJC) strategy

As stated in the previous subsection, once an incoming SST cell with time-frame $T_i$ becomes eligible at time $t$, it can be transmitted within the eligible time interval $[t+1, t+T_i]$. To achieve our adaptive delay-jitter control, we introduce a specific time interval $T_\alpha (1 \le T_\alpha \le T_i)$ which is used

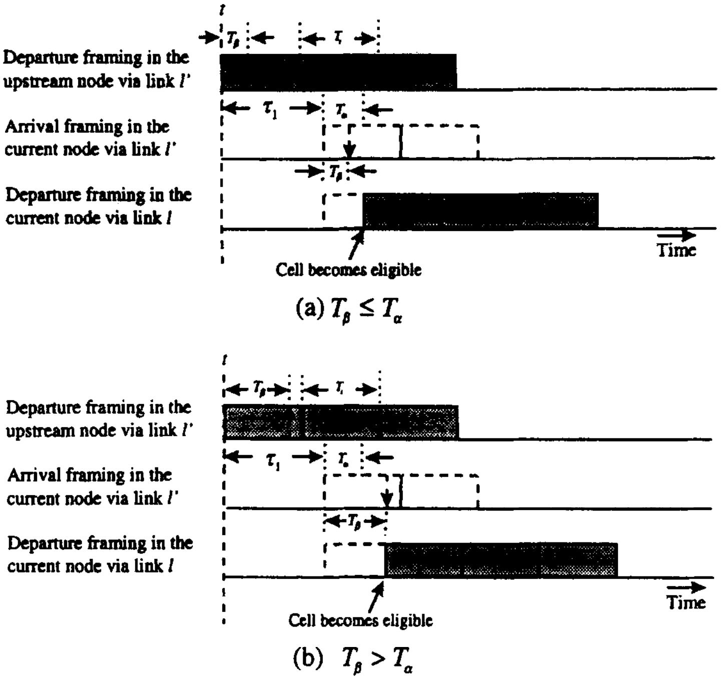



Fig. 4. Illustration of the adaptive delay-jitter control.

to determine when an incoming cell should become eligible in the current node. Assume that the current node $i$ has an outgoing link $l$, and an incoming link $l'$ connecting to the upstream node $i-1$. If an incoming cell $j$ is served at time $t' + T_\beta$ (i.e. $T_\beta = W^j_{i-1}$) in the upstream node after it became eligible at time $t'$, where $T_\beta \le T_\alpha$, then it will be delayed for a duration $T_\alpha - T_\beta$, before it becomes eligible in the current node, as shown in Fig. 4(a). Otherwise, if $T_\beta > T_\alpha$ due to the transmission contention in the upstream node, the incoming cell will become eligible immediately, as illustrated in Fig. 4(b).

The question is, how can an intermediate node know the time interval $T_\beta$ which is used to determine the delay (i.e. $T_\alpha - T_\beta$) for an incoming cell to be eligible? To resolve this problem, we investigated the behavior of any two consecutive SST cells under ADJC. Assume that two consecutive SST cells $A$ and $B$ arriving via link $l'$ and heading for link $l$ were served in the upstream node at times $t' + T_{\beta 1}$ and $t' + T_i + T_{\beta 2}$ after they became eligible at times $t'$ and $t' + T_i$ respectively. In the first case, $T_{\beta 1}$, $T_{\beta 2} \le T_\alpha$, the incoming cell $A$ with time-frame $T_i$ will be delayed a time duration $T_\alpha - T_{\beta 1}$ (i.e. $t_1 - t_0$) before it becomes eligible at time $t_1$ in the current node, then the cell $A$ can be transmitted within the time interval $[t_1 + 1, t_1 + T_i]$ and the following cell $B$ will become eligible as soon as this time interval is over. The waiting time for eligibility of cell $B$ in the current node is $T_\alpha - T_{\beta 2}$ (i.e. $t_3 - t_2$), as shown in Fig. 5(a). The second case is $T_{\beta 1} \le T_\alpha$ and $T_{\beta 2} > T_\alpha$. Owing to the transmission contention in the upstream node, cell $B$'s arrival time, $t_4$, is later than the ending time of cell $A$'s eligible region (i.e. $t_3$ or $(t_1 + T_i)$), thus it becomes eligible immediately, as shown in Fig. 5(b). Similarly, the other two cases are $T_{\beta 1} > T_\alpha$, $T_{\beta 2} \le T_\alpha$ and $T_{\beta 1}$, $T_{\beta 2} > T_\alpha$ as illustrated in Fig. 5(c) and Fig. 5(d), respectively. It can be found that no matter how early or how late cell $B$ arrives, it will not be eligible until the eligible time interval of cell $A$ is over.

Since an arriving cell immediately becomes eligible only when the eligible time interval of the preceding cell is over, the feasibility consideration of ADJC is regarding the complexity imposed on each intermediate node to set the time framing of the first data cell for each SST connection. To alleviate the extra loading of intermediate nodes, a leading RM cell can be used to indicate the start of an SST cell stream, and a reserved field can be used to carry the queuing delay information $T_\beta$ of the RM cell itself in the upstream node. Before a CBR traffic source starts the cell transmission, it needs to send an RM cell first. When the leading RM cell is received by an intermediate node, the cell transmission delay information $T_\beta$ of an SST stream in each intermediate node can be carried to the next downstream node. Since the parameter $T_\alpha$ can be set during the connection setup, the queuing delay for an RM cell waiting for eligibility after it arrives at the current node is equal to $[T_\alpha - T_\beta]^+$. Then, the following data cells only need to obey a simple rule to achieve the adaptive delay-jitter control, that is, an arriving cell immediately becomes eligible as soon as the eligible time interval of the preceding cell is over. As a consequence, our service scheduling method removes the extra waiting time through node synchronization, and it supports an arbitrary time frame. In the rest of this subsection, we examine the delay performance of the

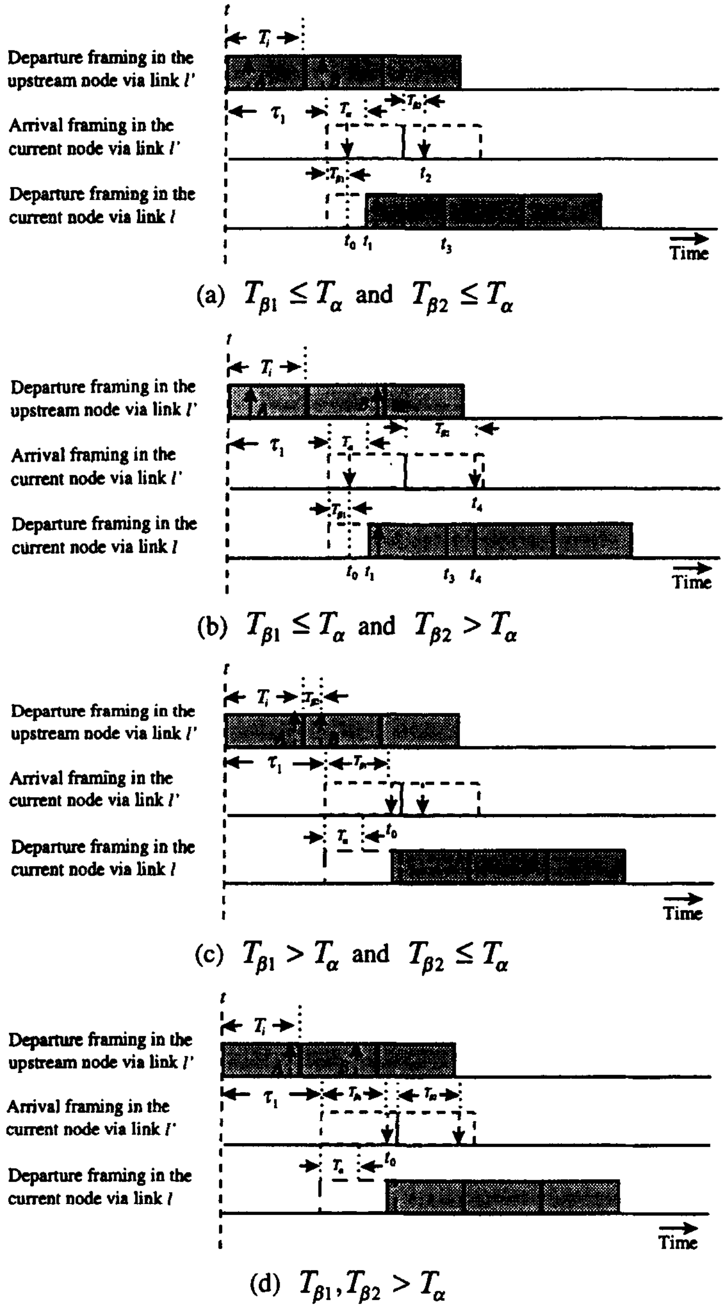



Fig. 5. Illustration of the relationship between two consecutive cells under ADJC.

adaptive delay-jitter control. Assuming an SST connection $k$ with time frame $T_k$ routes through $H$ hops, we can define the time interval $q'^j_i$ as follows:

$$q'^j_i = \begin{cases} 0 & \text{if } T_\beta \le T_\alpha \\ T_\beta - T_\alpha & \text{otherwise} \end{cases}, \text{ where } i = 1 \cdots H,\ j = 1, 2, \cdots, \tag{12}$$

where $T_\beta$ is the time interval of $j$th cell to finish the transmission once it becomes eligible at the $(i-1)$th node. Clearly, $q'^j_i$ is bounded by

$$0 \le q'^j_i \le T_k - T_\alpha \tag{13}$$

As shown in Fig. 6, the total end-to-end delay of the $j$th cell for a given connection $k$ with fixed cell interarrival time is represented as follows:

$$D^k_j = \tau^k + QD^k_j = \tau^k + \sum_{i=2}^{H} (T_\alpha + q'^j_i) + d_j = \tau^k + (H-1)T_\alpha + \sum_{i=2}^{H} (q'^j_i) + d_j \le \tau^k + HT_k, \tag{14}$$

where $d_j$ is the time interval for the $j$th cell to finish transmission after it becomes eligible at the $H$th node (i.e. $d_j = W^j_H$, $1 \le d_j \le T_k$). Since $q'^j_i$ is bounded by

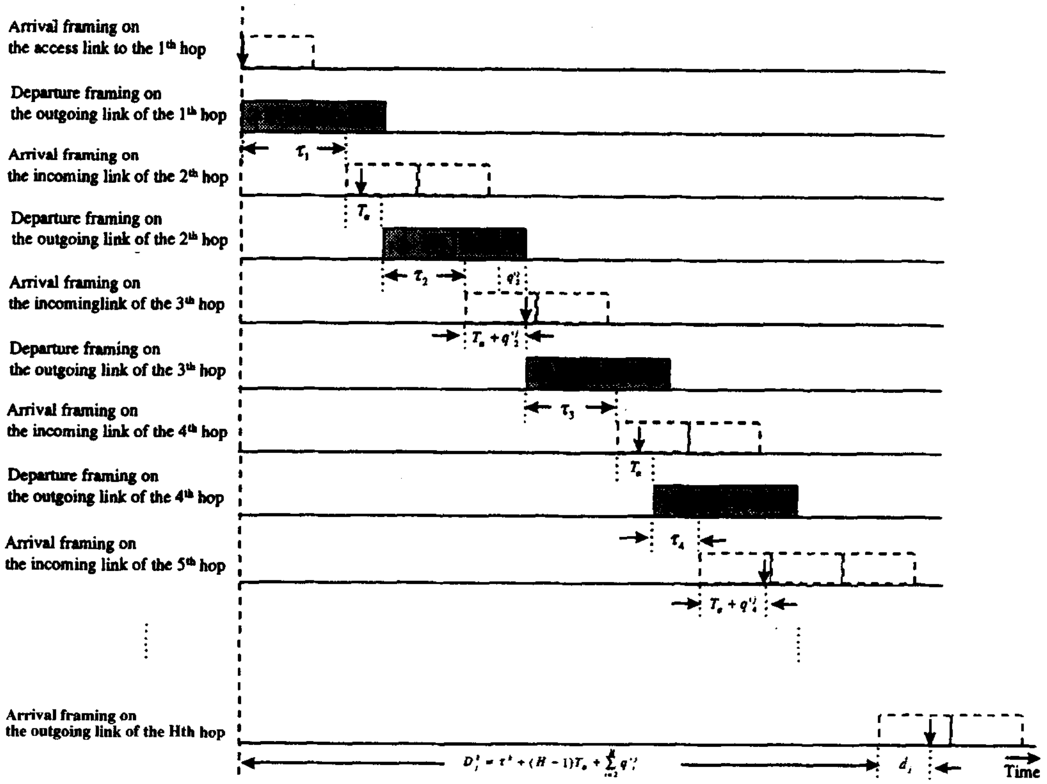



Fig. 6. Cell delays incurred during the transmission process in ATM networks.

$[0,T_k - T_\alpha]$, then the end-to-end delay is bounded by

$$\tau^k + (H-1)T_\alpha + 1 \le D_j^k \le \tau^k + HT_k. \tag{15}$$

It should also be pointed out that if a source traffic is not regulated as a fixed interarrival traffic pattern (i.e. $0 \le q_1^j \le T_k - 1$), then

$$D_j^k = \tau^k + QD_j^k = \tau^k + (T_\alpha - 1) + (H-1)T_\alpha + Hq'^j_i + d_j. \tag{16}$$

Clearly, the end-to-end delay of the $j$th cell for connection $k$ is bounded by

$$\tau^k + (T_\alpha - 1) + (H-1)T_\alpha + 1 \le D_j^k \le \tau^k + (T_\alpha - 1) + HT_k. \tag{17}$$

From the above discussion, it is apparent that we can adaptively control the delay-jitter. For example, let us set the time interval $T_\alpha$ to $T_k$ , then the end-to-end delay of a given cell of the connection $k$ is bounded by $[\tau^k + HT_k,\ \tau^k + (H+1)T_k - 1]$. If we set $T_\alpha$ to 1, the results will be the same as that in the SDJC strategy. Obviously, in ADJC the end-to-end delay-jitter can be adaptively controlled. In addition, the end-to-end delay (excluding total propagation delay) of a connection is reduced from $2HT_k$ in S&G to $(H + 1)T_k$ in our ADJC strategy. Compared to S&G, there is a significant improvement in the end-to-end queuing delay for an arbitrary time frame size. More importantly, we can use the delay parameter $T_\alpha$ to adaptively reduce the delay-jitter down to $T_k - 1$ and to precisely meet the services' requirements.

For further analysis of the smoothness property, we now investigate the relationship between the incoming cells. Given a source traffic with time frame size $T_i$, owing to the transmission contention among different connections, the cell transmission may be delayed, which may cause the time interval between two consecutive cells of the same connection smaller than $T_i$. Moreover, it is possible that two consecutive SST cells are interclustered back to back. According to the TC algorithm, it maintains the traffic smoothness for each connection based on per-VC queuing. Cells of an SST connection with time frame $T_i$ always keep the SST pattern throughout the network. Given *Num* SST connections in a switching node, for the $l$th connection ($1 \le l \le Num$), it is possible that the FIFO occupancy, $L(l)$, will be greater than 1 under the proposed service discipline. For an SST cell stream, the $k$th incoming cell should complete the transmission before the $(k + 2)$th cell arrives, thus two-cell buffer space is sufficient for each connection to prevent FIFO buffer from overflow; in other words, using our proposed method, SST streams only need a total buffer space of $2 \times Num$ cells in the output ports to avoid buffer overflow.

The proposed strategy can support a bounded queuing delay and prevent network congestion for CBR traffic mainly because the strict transmission control always keeps the traffic pattern as SST throughout the ATM network. Since non-delay-sensitive traffic can tolerate longer delay comparing with delay-sensitive traffic, the utilization problem can be compensated by accounting for non-delay-sensitive traffic, such as ABR and UBR services, in the integrated service environment. If there is no eligible CBR cell to transmit, the idle time can be used to service non-delay-sensitive traffic or other types of traffic. Such integrated service policies can offer higher bandwidth utilization. In the remainder of this section, we examine the realization of a possible approach for our proposed traffic control.

### 3.3. Realization of the proposed traffic control method

We have shown that our proposed CBR transmission control configuration has three components, a cell dispatcher, a rate controller and a priority service scheduler. At first, arriving cells are classified and written into the cell pool. Prior to this process, the associated traffic property (i.e. time frame size, TFS) is extracted by the CD and stored with its physical address (PA) into the associated queue at the RC. The RC consists of a set of regulators, each corresponding to a connection. A regulator has two functions: setting the eligibility of those HOL cells for all non-empty queues and holding a cell until it becomes eligible. The regulator can be realized by a synchronous countdown counter with parallel load, as shown in Fig. 7. This realization approach is inflexible since the number of regulators is directly proportional to the number of active CBR connections. We can conceptually decompose the RC into a set of regulators, and as each is associated with a connection, we have no need of multiple physical regulators in the implementation; a common mechanism employees a modified version of the calendar queue, which is shared by all logical regulators [19,22,21]. The PSS consists of several modules in which the TFS and PA are stored. Once the HOL cell of a queue becomes eligible, the pair (TFS, PA) of the cell will be stored into the PSS. As stated in the previous subsection, to meet the delay requirements for each service class, the transmission priority is set to prefer the smallest TFS. When cells are served by the PSS, the server compares the TFSs of those eligible cells and selects the smallest TFS for transmission since the cell with the smallest TFS value is the most urgent. The major complexity of the PSS is to select the smallest time-frame-size among all eligible cells. Using

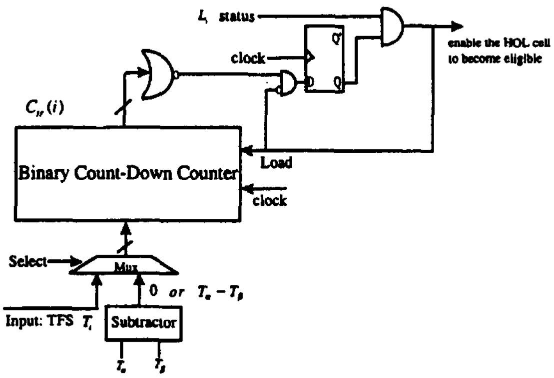



Fig. 7. A realization of the regulator.

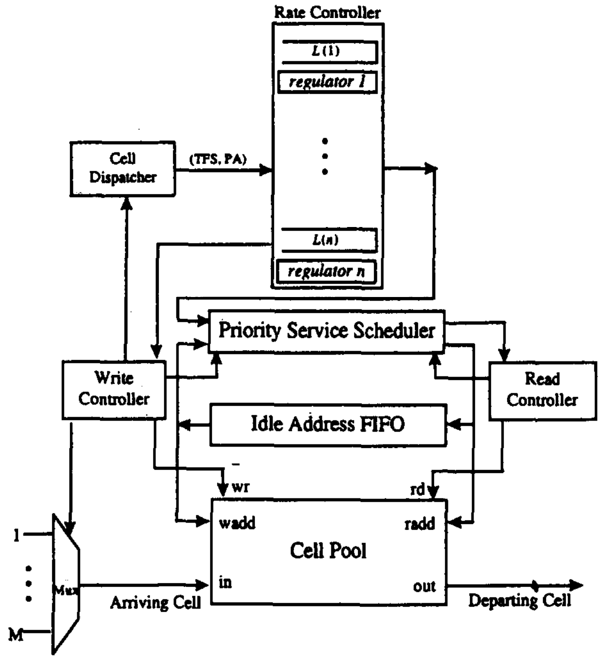



Fig. 8. Implementation architecture of the proposed traffic controller.

hardware, implementation is preferable for high speed ATM, e.g. 2.78 $\mu$s cell transmission time at 155.52 Mbps. We can use the available VLSI Sequencer chips [13,14,23] to implement the PSS module. Through this hardware realization, the proposed strategies are able to accommodate the CBR cell streams. The implementation architecture of the proposed traffic controller is illustrated in Fig. 8.

## 4. System simulation and performance evaluation

### 4.1. System simulation model

For CBR services, a traffic source can be characterized by simple parameters. In our simulation, a simple ATM subnet consists of four switching nodes and nine end-systems, which function as input sources, as shown in Fig. 9. Let these nine connections be established as follows. Sources *A*, *B*, and *C* connect to switch 1 and then through switch 2 to switch 4. Sources *D* and *E* connect to switch 2 and through the subnet to switch 4. Sources *F* and *G* connect to switch 3 and through the subnet to switch 4. Sources *H* and *I* route through node 4 to their destinations. All sources pass through the same output link of node 4 which has the heaviest load. There are frames $T$ = 4, 4, 8, 16, 32, 64, 128, 128, 128 (cell slot time) for sources *A*, *B*, *C*, *D*, *E*, *F*, *G*, *H* and *I* respectively. The switch output port buffer is organized as parallel FIFOs which share the output link's bandwidth via a priority scheduling. The major performance measure was the queuing delay. The transmission service of each output link can be modeled as a single server with multiple queues. We assume that every link has a transmission rate of 155.52 Mbps. For the sake of simplicity in

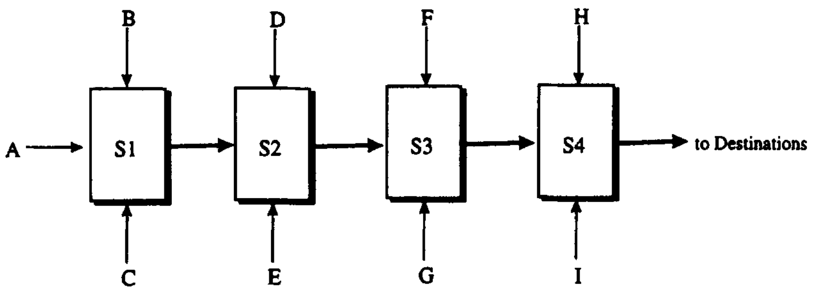



Fig. 9. System model for queuing delay analysis.

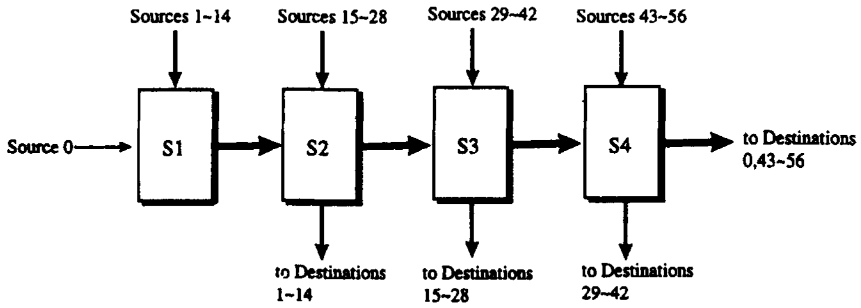
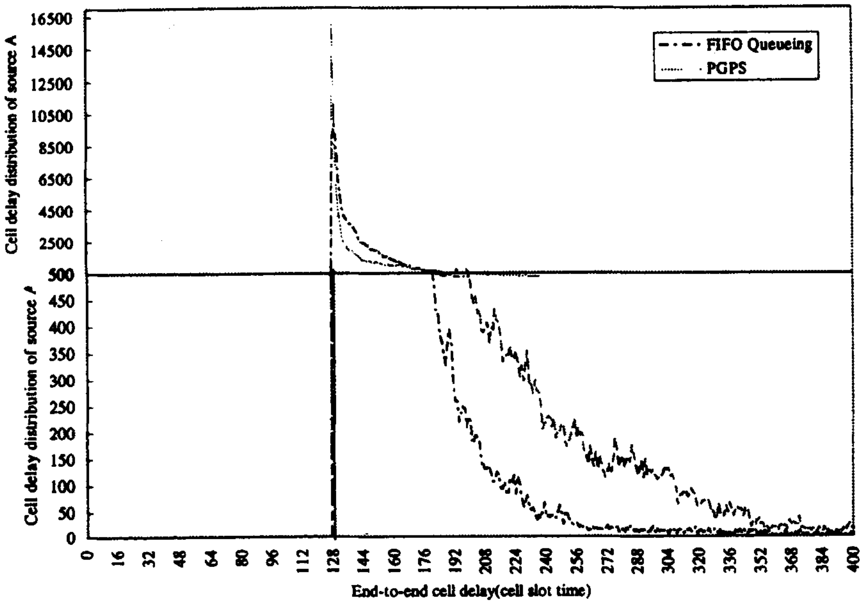



Fig. 10. System model for delay-jitter analysis.

simulating the S&G scheme, we assume that the mismatch phase, $\Theta$, of both the arrival frame and the departure frame is $T_i/2$ in a switching node.

The other model consists of four nodes and 57 sources as shown in Fig. 10. Let these 57 sources be established as follows. Source 0 connects to node 1 and then through node 2 to node 4. Sources $i$ ($i \neq 0$) connect to the $(\lceil i/14 \rceil)$th node and then through the $(\lceil i/14 \rceil + 1)$th node to its destination. These nodes were connected through links with 155.52 Mbps capacity. Each source has a time frame $T_i = 15$ cell slot time, $i = 0,1,\cdots,56$. The major performance measure considered here is the end-to-end queuing delay of source 0 in the worst case, because source 0 has the lowest priority in service contention.

As discussed earlier, the average queuing delay under work-conserving disciplines is smaller than that under non-working disciplines. However, the delay distribution of work-conserving disciplines, such as PGPS and FIFO, has much wider range, as shown in Fig. 11. In the PGPS strategy, source $A$ is allocated with a larger guaranteed bandwidth (i.e. $1/T_A = 1/2$) so that its cells have greater opportunity for transmission. As a result, the number of




Traffic intensity =0.886, $T_i$ = 2,4,8,8,8,8,8,8,8 and the on/off period is 60/40 cell slot time.

Fig. 11. End-to-end cell delay distribution of source A by using FIFO queuing and PGPS strategies.

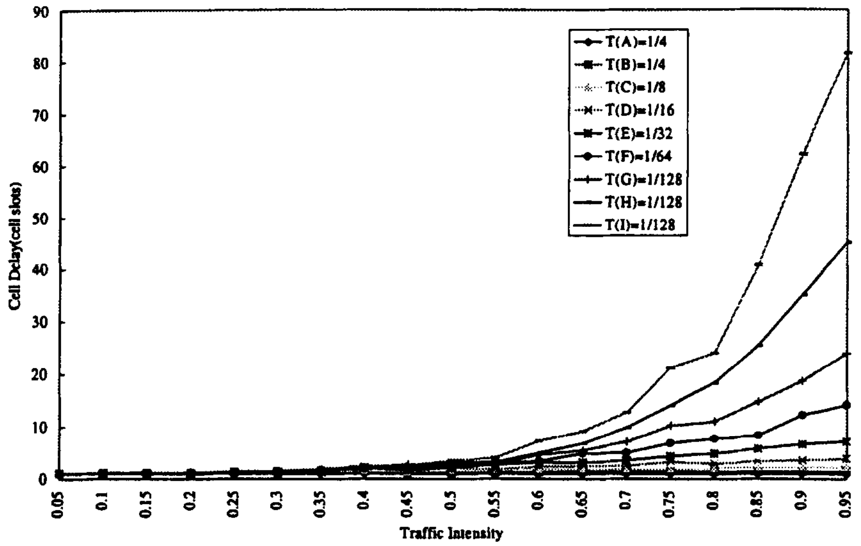



Fig. 12. Average delay of VCs with different frame sizes by SDJC at switch node 4.

cells to be sent immediately after cells arrive at a switching node is larger than that in FIFO queuing.

### 4.2. Simulation results

In the following simulation results, the end-to-end delay excludes the fixed propagation delay.

#### 4.2.1. Comparison of end-to-end delay and delay-jitter of SDJC strategy and S&G

*Average delay versus traffic intensity.* The traffic intensities vary from 0.1 to 0.95 at switch node 4. From the simulation results shown in Fig. 12, the average delay time of different time-frame sizes increases as the traffic intensity increases. We also found that the maximum cell delay, $Q_4^j$, for a given connection never exceeds its upper bound (i.e. $1 \le Q_4^j \le 2T_k - 1$) even at a high traffic intensity (0.95). Consequently, the SDJC strategy provides priority services as well as a bounded delay.

*Average end-to-end delay versus traffic intensity.* The traffic intensities vary from 0.1 to 0.9. From the simulation results shown in Fig. 13, the SDJC gives a better average end-to-end delay than S&G. The major reason is that an incoming cell of connection $i$ will become eligible immediately if $C_{tr}(i) = 0$. It removes the extra synchronization delay and leads to a smaller queuing delay.

*End-to-end cell delay distribution.* In order to evaluate the end-to-end delay under heavy load, the total traffic intensity is increased to nearly 1.0 in each node. In S&G strategy, the end-to-end cell queuing delay is given as $D_p = HT + \sum_{i=1}^{H} \Theta_{i-1,i} + d_p$, where $-T < d_p < T$ and $H = 4$. Since source 0 ($T_0 = 15$) passed through four nodes, every mismatch is equal to $T_0/2$, thus the total end-to-end delay is within the interval of [75, 104] cell slots and

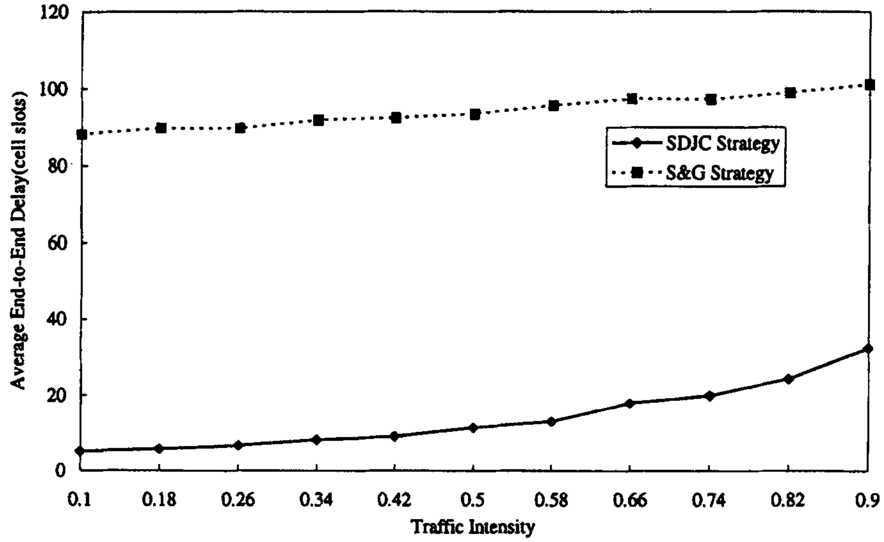



Fig. 13. Average end-to-end cell delay of source 0 under proposed traffic control and S&G.

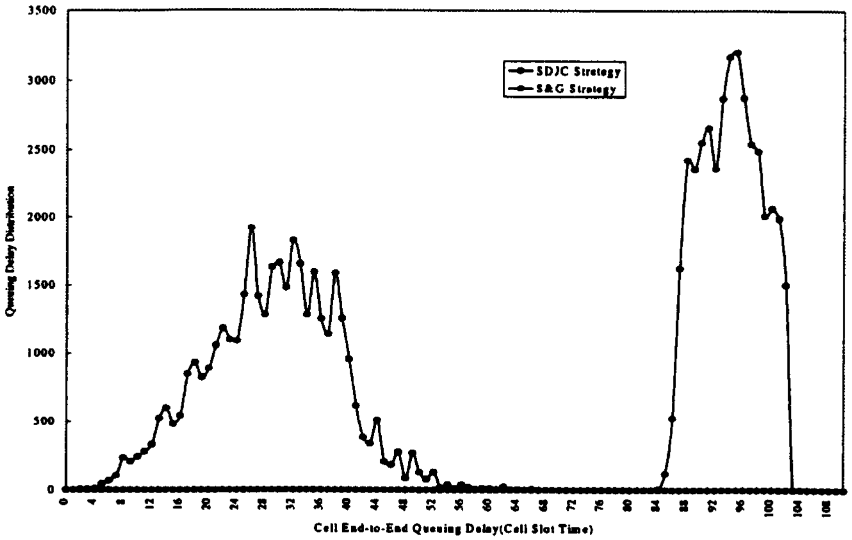



Fig. 14. End-to-end queuing delay distribution of source 0 under both SDJC and S&G strategies.

the delay-jitter is bounded by $2T_0$. In SDJC strategy, the end-to-end delay is bounded by [4, 74] cell slots and the delay-jitter is bounded by $(H + 1)(T_0 - 1)$, as illustrated in Fig. 14. The simulation results demonstrate that the SDJC strategy provides a guaranteed service for the CBR traffic. Although SDJC has higher delay-jitter than S&G, its end-to-end delay is much lower. From the viewpoint of an end-system, SDJC can provide a better end-to-end performance with lower implementation complexity.

#### 4.2.2. Comparison in end-to-end delay and delay-jitter of ADJC strategy and S&G

As illustrated in Fig. 15, the results of ADJC with different control parameter values ($T_\alpha$) show that the ADJC strategy provides a bounded end-to-end delay and supports adaptive delay-jitter control for SST connections. Since source 0 ($T_0 = 15$) passed through four nodes, if the delay control parameter $T_\alpha$ is set to 1, 5, 10, and 15, then the total end-to-end delay is within the interval of [4, 74], [20, 74], [40, 74], and [60, 74] respectively. By using ADJC strategy, we can flexibly control the delay-jitter for various CBR services in ATM networks.

### 4.3. Numerical analysis

Assume a source traffic is regulated as an SST pattern with time-frame $T$ and there are $H$ hops in the routing path, the S&G framing strategy guarantees bounded end-to-end delay, delay-jitter and cell-loss-free transmission for each connection. However, S&G may lead to extra synchronization delay. In S&G, the maximum end-to-end queuing delay is bounded by $2HT + T$ and the delay-jitter is bounded by $2T$. While the RCSP/DJ provides a nearly optimal delay-jitter control, the delay-jitter is bounded by $T$. However, it

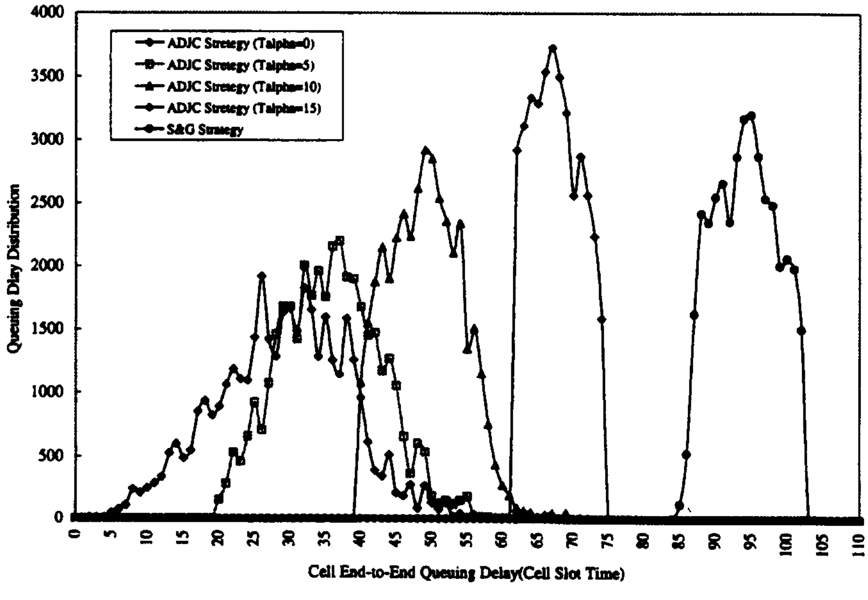



Fig. 15. End-to-end queuing delay distribution of source 0 under ADJC and S&G strategies.

Table 1
Comparison among different traffic control strategies

| | Advantages | Disadvantages |
|---|---|---|
| S&G strategy | Bounded queuing delay [$2HT - T$, $2HT + T$]. Bounded delay jitter [0, $2T$]. | Large total end-to-end delay. Extra synchronization delay is introduced. Assigned frame size is inflexible. |
| RCSP/DJ | Bounded queuing delay [0, $HT$]. Bounded delay-jitter [0, $T$]. | Very difficult to realize. |
| TCRM | Bounded queuing delay [0, $HT$]. Bounded delay-jitter [0, $HT$]. | Larger delay-jitter. |
| ADJC strategy | Adaptive bounded queuing delay. Flexible delay-jitter control. Maximum delay-jitter [0, $HT$]. Minimal delay-jitter [0, $T$]. | Introducing control parameter $T_\alpha$. Overhead of the RM cell. |

needs to pass the delay information (cell eligible time) from node to node and to synchronize the clock time of all intermediate nodes precisely, which is very difficult in a wide-area ATM network. The TCRM provides a guaranteed end-to-end queuing delay HT for real-time traffic, although it provides a lower average queuing delay HT than S&G, but the delay-jitter $HT$ is worse than that in S&G. In the ADJC strategy, the maximum end-to-end queuing delay is bounded by $HT$ and we can adaptively control the delay-jitter down to $T$. Our proposed strategy removes the extra synchronization delay and allows arbitrary time-frame sizes. The advantages and disadvantages between other schemes and our proposed method are listed in Table 1.

## 5. Conclusion

In ATM networks where flow control and error control are end-to-end, the networks may be vulnerable to overload/congestion that severely degrades their performance. It has been suggested that congestion problems can be solved by reactive control as well as preventive control strategies. The combination of access control and transmission control provides a more effective control method. Therefore, we propose a simple rate control technique and a feasible implementation method to guarantee the performance of delay-sensitive traffic in broadband networks. In our proposed strategy, the traffic pattern of each connection can be maintained as close to the smoothed source traffic (i.e. SST) as possible throughout the network. The major advantages of the ADJC strategy is the improvement of the end-to-end delay by removing the extra synchronization delay, and the flexibility to control the delay-jitter. Although the ADJC strategy needs an RM cell to carry the delay information $T_\beta$ of the upstream node, the overhead can be neglected for a CBR connection. The ADJC strategy can be implemented in most switching architectures. By using the ADJC strategy, a bounded end-to-end delay and a flexible delay-jitter can be well controlled.

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