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Giuseppe Bianchi and Vittorio Trecordi

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Abstract

A feasible and cost-effective resource management scheme is urgently needed in the Broadband Integrated Services Digital Network adopting the Asynchronous Transfer Mode (ATM) technique. In this paper we propose a simple and comprehensive strategy to manage bandwidth allocation, congestion control and quality of service (QOS) integrity in a multi-service ATM network. The proposed framework involves a core network that grants a limited number of grade of service (GOS) profiles and suggests the design of edge-adapters able to match QOS user's requirements with the GOS profiles. Three different core network behaviors are envisioned and the associated connection acceptance algorithms are presented. Also, for some particular QOS requirements, the edge-adapter dimensioning process is developed. The effectiveness of the scheme is demonstrated by numerical examples which report the network utilization performance when the proposed core network policies and edge adapter design are applied.

[0]This work was supported by CEFRIEL and ITALTEL SIT.

Proposal for a comprehensive bandwidth management scheme and connection acceptance rule for B-ISDN

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1. Introduction

Asynchronous Transfer Mode (ATM) is the standard adopted by the International Telephone and Telegraph Consultative Committee (CCITT) for implementation of wide-area broadband telecommunication networks, namely the Broadband Integrated Services Digital Network (B-ISDN) [CCI92, Leb92]. B-ISDN should be able to accommodate traffic with different Quality Of Service (QOS) requirements for any acceptable network load and achieve cost-effective, dynamic sharing of network resources (transmission, buffering and switching) among various continuous or bursty sources. To achieve such a goal a connection-oriented technique based on virtual-circuit packet switching with asynchronous multiplexing and switching of fixed-size labeled packets of information, called cells, has been assumed.

In recent years, significant research effort has been focused on effective use of network resources [CM91i, CM91ii, JSAC91], but there has been much debate and little agreement. It is widely accepted that the resource management policies differ in traffic control mechanisms (procedures aimed at preventing congestion) and congestion control mechanisms (procedures that apply when the network is congested). Many proposals have been made for traffic and congestion control mechanisms, and Connection Admission Control (CAC) policies that address the ATM scenario, but, in spite of that, some mechanisms are not well understood and, unfortunately, very few general results have been carried out.

Such study is enormously complicated by the highly heterogeneous characteristics of the traffic classes and of the QOS requirements. For example, non-real-time data traffic QOS requirements can be defined in terms of loss probability at the cell level (which can be variable from, say, 10^{-3} to 10^{-6} and more), while they tolerate quite a large delay (of the order of hundreds of ms). Furthermore the cell loss should be as concentrated as possible, in order to minimize the user frame loss probability. On the contrary, real-time uncompressed video requires very low delay and jitter, with a non-strict cell loss requirement (say 10^{-2} to 10^{-4}) and a cell loss distribution as wide as possible. Real-time compressed video requires low delay and jitter as well, but the loss requirements are usually stringent (up to 10^{-6}, 10^{-9}, according to the compression technique). Furthermore, some compression algorithms require very different loss requirements and distributions for different cells belonging to the same stream, according to the semantics of the information. In addition we have to consider the wide variety of traffic generation processes: from continuous bit rate and slight variable bit rate sources, to heavily bursty sources with burst length up to thousands of cells and burstiness up to hundreds. These few examples clearly highlight the need to define a congestion control and bandwidth management framework by taking into account the interference between a variety of traffic sources.

Furthermore, at the basis of all the resource allocation policies lies the need for a traffic descriptor, either statistical or operational, logically related to both call admission control and usage parameter control [Ber91]. The traffic descriptor should be simple enough to be extrapolated by the user's terminal equipment. Furthermore the sensitivity of network management to inaccurate descriptor specification should be limited. Moreover, when the network operation would rely on statistical resource allocation, policing and shaping techniques at the edge of the network are unavoidable to enforce declared traffic descriptors and to tailor the traffic to a less disruptive profile.

Let's look at a brief overview of some proposed bandwidth allocation techniques. Peak rate allocation offers the strongest performance guarantee and it's easy to understand and implement; the main drawback is the inefficient usage of bandwidth in the presence of bursty traffic. Minimum throughput allocation guarantees the user a certain amount of declared bandwidth, without any grant on the bandwidth exceeding the contract [Zha90]. The proposed implementation requires hardware modifications of the switching architectures, but the complexity is acceptable. The main drawback is that bursty sources with stringent requirements necessitate a high throughput, and this essentially leads to peak rate allocation. Statistical multiplexing of cells with bursty traffic specification and allocation is the most studied approach. High link efficiency is achieved at the expense of fairly high queueing (delay) inside the network nodes. One problem is that each network node requires fairly complex queueing architectures for guaranteeing different QOS performance in terms of delay and loss. However the main drawback is the absence of computational methods that can determine in real time if a new connection can be safely accepted [Ahm89, Dec90, Gal89]. Fast bandwidth/buffer reservation techniques [Tur92, Boy92] focus on the problem of burst integrity and propose a practically quite complex per-burst reservation of network resources. These approaches suffer from the need to identify bursts inside the network and offer limited maximum efficiency with respect to the theoretical limits and and high dependability of performance on the heterogeneity of traffic and QOS requirements. Rate control schemes have been proposed to keep the smoothness and the identity of the native frame through the network [Gol92], but the performance bound that can be expressed depends on the number of intermediate hops. The switching architectures must also be adapted to support such a scheme.

Many previous contributions show that control in ATM networks can only be achieved by executing several concurrent mechanisms, eventually at different levels, namely at the connection, burst and cell level [Hon91, Eck92, Wer92]. Also, some researchers expressed their concern about the trade-off of efficiency achievable by cell-granularity and the overwhelming implementation complexity of the network control system [Lea92i, Hui92].

In spite of the significant research effort, many of the open issues highlighted in [Dec91] and [Lea92ii] are not sorted out yet. In particular, the handling of heterogeneous classes and QOS requirements both at the network queueing level and at the CAC level is not well understood in any of the above proposals.

2. The proposed framework

In this section we describe a comprehensive framework for a bandwidth management scheme and connection acceptance rule for B-ISDN.

As mentioned in the introduction, all the above bandwidth allocation techniques require the network to monitor a wide spectrum of different classes and QOS, thus leading to either fairly complex CACs or to the inability to promise a QOS grant, when high network efficiency is targeted. On the other hand, it is widely accepted that convenient traffic shaping can lead to substantial gain in statistical multiplexing efficiency and most of the studies related to shaping deal with the issue of traffic tailoring so as to achieve better statistical multiplexing gain [Rig91]. However, a wholly non-segregated resource sharing technique, such as ATM, is critical in the sense that information flows belonging to different sources and with different QOS requirements interfere with each other and the goal of being able to grant the specific QOS performance is really difficult.

We suggest that the ATM network should be split in two parts: a core network able to offer a very limited set of GOS profiles and an access network, made of edge adapters able to match user's QOS requirements with network GOS profiles (Figure 1).

This concept has already been partially included in the B-ISDN draft documents. In particular the proposed ATM Adaptation Layer (AAL), performed at the User Network Interface (UNI), specifies 5 different classes of service namely CLASS 1 devoted to Circuit Emulation (CE), CLASS 2 devoted to Variable Bit Rate (VBR) service with time synchronization between sender and receiver(s), CLASS 3 Connection Oriented (CO) data service, CLASS 4 ConnectionLess (CL) data service and CLASS 5 a light version of Connection Oriented variable bit rate data service [Che92]. The AAL protocols operate above the ATM layer, performing the adaptation of user's data units to cells and handling multiplexing and cell loss detection. The specialization of AAL protocols for each class of service addresses the peculiarities of each service in terms of loss and delay sensitivity and packetization.

The main, and widely recognized, drawback of such a scenario is that all switching, multiplexing and buffering procedures must depend upon the class of each offered cell and must implement custom mechanisms. Clearly the flexibility of such an implementation for the ATM network is strongly limited. In particular, almost any future enhancement or modification of the user needs (for example the introduction of a new class of service), would cause a mayor modification of the network implementation.

Our proposal is to enhance the functionality of the AAL and to simplify the operation of the ATM core network according to the original motto of ATM. The access network is designed to:

- exploit the information on the specific traffic and the associated QOS requirements in order to tag and shape the traffic in a convenient manner before offering it to the core network;

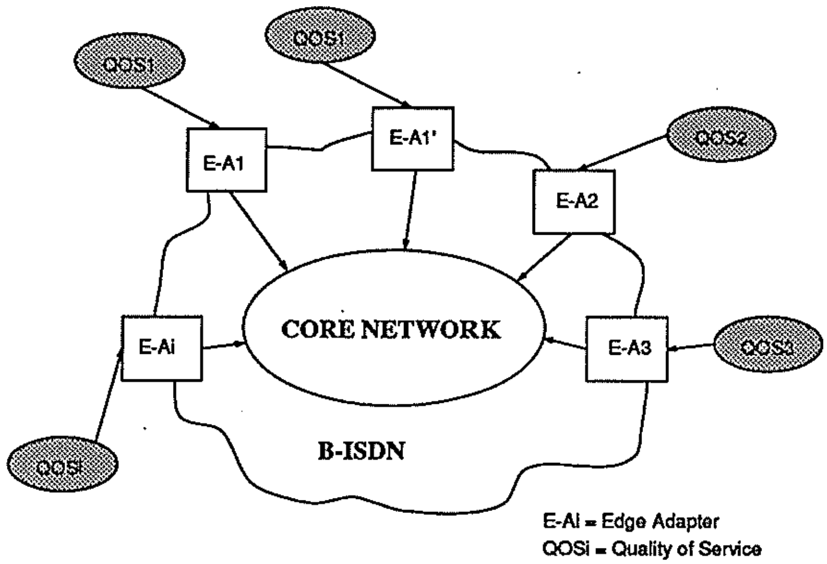

Figure 1: Broadband Integrated Service Digital Network scenario

- determine and guarantee, at the edge of the network, the effective matching to the specific QOS requirements, and take advantage of non-stringent requirements.

The enhancements of the access network interface allow the design of a simple core network, whose main properties and features should be:

- simplicity of management and simplicity of both software and hardware architectures;
- flexible upgrades, including future enhancements;
- independence of the traffic specific QOS requirements;
- management of very few GOS profiles, with simple GOS specification;
- provision for a simple connection admission control algorithm, suitable for implementing fast connection-processing.

In more detail, the access network is composed of edge-adapters that match the user's QOS requirements and the network GOS profiles. The edge-adapters should take advantage of the QOS requirements to tag and shape the traffic to a more advantageous profile from the point of view of the core network occupancy. This would lead to lower user costs, and to an increase of the number of connections that can be served at once. Basically, the more complex the QOS specifications and the wider the mismatch between the GOS profiles offered by the network and the QOS requirements of the traffic, the more sophisticated is the edge-adapter. Obviously the edge-adapters must provide a convenient set of network traffic descriptors, independent of the traffic QOS requirements. The traffic descriptor characterizes the statistical description of the shaped traffic that is used in the core network CAC process.

On the core network side, we can, in principle, consider a core network able to grant a certain number of GOS levels. However in the rest of the paper, we focus on a core network providing just two different GOS profiles, so that compatibility with the ATM cell format is guaranteed by using the Cell Loss Priority (CLP) bit in the ATM cell header to tag cells requiring different GOS profiles. The two different types of traffic (cells) are High Priority (HP) traffic and Low Priority (LP) traffic, and their expected cell loss probabilities are respectively named g_{HP} and g_{LP}, with $0 \leq g_{HP} < g_{LP}$. When a conflict arises on the occupancy of any network resource, the HP traffic can preempt the LP one. The handling of cell priorities in the network is straightforward with queueing architectures implementing suitable mechanisms, such as pushout or threshold based schemes [Gra91].

Furthermore, we impose that the amount of buffering inside the core network nodes is minimum, only the amount necessary to solve synchronous cell arrival contentions. Since the length of the queues inside the network is limited, the maximum network delay is practically given only by the network propagation delay. Thus, the GOS profiles can be simply identified by the associated expected cell loss probability. The matching of QOS delay requirements is done by the edge-adapters which can take advantage of non stringent delay requirements. Since multistage queueing can increase the peak rate of a connection, a suitable spacing policy should be taken into account [Cou92], although it is our opinion that for short queues and controlled traffic, the problem of cell clustering is marginal.

We mention here three different core network configurations obtained with different core network GOS designs:

1. If $g_{HP} = 0$ and $g_{LP} = 1$ the network acts as a Minimum Throughput Guaranteed (MTG) network on the HP traffic stream;
2. If $g_{HP} = 0$ and $g_{LP} < 1$ the network acts as an Enhanced Minimum Throughput Guaranteed (EMTG) network, with the provision of a bound on the LP traffic cell loss;
3. In the general case, with $g_{HP} > 0$ and $g_{LP} < 1$, the network acts as a Combined Statistical Multiplexing (CSM) network on the two priority levels.

Note that the definition of the core network configuration (GOS parameters and CAC policy) can be done in software for a given network hardware architecture. The three configurations above are ranked according to increasing complexity, but also increasing network performance, and could be envisioned as successive steps toward the final B-ISDN implementation.

In Case 1 the connection acceptance rule is trivial: a new connection is accepted if and only if there exists a path in the network where we can deterministically allocate the required amount of bandwidth required by the HP stream. The traffic descriptor of a connection is trivially the HP peak bit rate and the knowledge of the statistical description of the incoming traffic is of no use and thus there is no need of any policing device. The tagging as either HP or LP cell at the edge-adapter is primarily based on the traffic source generation: all the cells that can be forwarded up to the agreed MTG are tagged as HP, all the cells that exceed the same MTG are tagged as LP. Note that the tagging process could take also into account the semantics (i.e. the quantity of information carried) of data to be forwarded. Finally, notice that, in case of congestion, only the LP cells can be lost, according to the CAC rule. This approach is very interesting since it is simple and it does not rely on traffic measurements or enforcement mechanisms. The performance can be quite high for non-real-time traffic with an appropriate design of the edge-adapter. Of course, in case of real-time traffic, for which the shaping effect of the edge adaptor is marginal, the performance is similar to that achievable by peak rate allocation.

In order to overcome the performance limitation imposed by the lack of a performance grant on LP traffic, extensions 2 and 3 seem necessary for more efficient network utilization.

In Case 2 the core network is able to guarantee zero loss probability on the HP traffic, as in the case of MTG, while now a given loss probability bound for the LP traffic is granted. Note that the HP traffic is served by the network with asynchronous multiplexing and switching. No synchronous resource reservation policy applies, thus the LP traffic exploits all the unused resources left by the HP traffic. The connection acceptance rule is still quite simple. When a new connection set-up arises, the call-processors simply accept the new connection if and only if there exists a path in the network where the cumulative amount of HP peak bit rate bandwidth does not exceed the available link bit rate and the cumulative amount of LP traffic flow does not cause the violation of the GOS provided by the core network on LP traffic. Obviously, when we guarantee a bound on LP cell loss, we assume to exploit the achievable statistical multiplexing gain. As a consequence a policing function is needed to guarantee to the network the connection parameters upon which the bandwidth assignment and the connection acceptance control is performed.

In Case 3, we allow the total amount of HP traffic to overflow the link capacities, with the condition that the related HP and LP cell loss probability is limited. Statistical multiplexing gains are achieved not only by LP traffic, as in the previous case, but also by HP traffic. The connection acceptance rule requires the monitoring of the GOS compliance for both HP and LP flows. Of course, policing is still needed at the user-network interface, and, because of the different priority levels, we face the difficulties pointed out in [Gra91].

The proposed framework can be synthesized as follows (Figure 2). Given a set of QOS requirements, a tuple of user's traffic descriptors, the GOS profiles offered by the core network, and a particular edge-adapter architecture and buffer capacity, a tuple of network traffic descriptors can be determined. On the basis of these network traffic descriptors, the GOS profiles and the network occupancy status in terms of both HP and LP traffic flows, the Connection Admission Control algorithm decides whether to accept the newly offered connection.

Clearly the edge-adapter should be designed to maximize the network efficiency while respecting the traffic QOS requirements and the choice of a given core network configuration. Of course the basic step for addressing this target is the definition of a given objective function (typically the core network engagement) which can be practically used in order to optimize the edge-adapter design.

Such optimal dimensioning seems to be, in the general case, very complex, since it is necessary to understand which network traffic descriptor configuration allows us to maximize the usage of the core network resources. The analysis of the general case is thus out of the

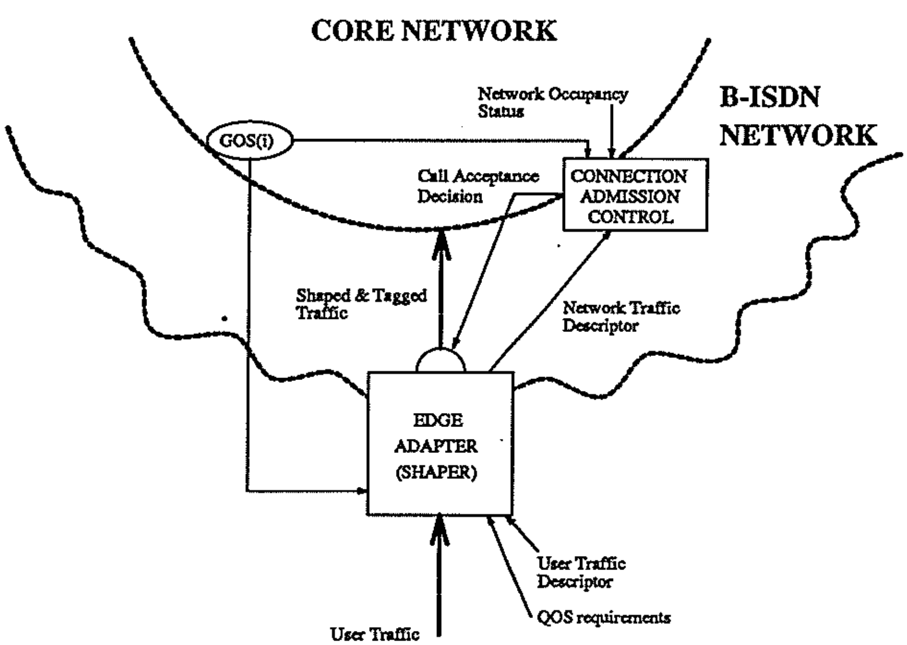

Figure 2: A sketch of the proposed framework

scopes of this paper. However, in the following section, we show how such dimensioning can be carried out for a MTG network, for some interesting edge-adapter architectures, for bursty traffic and QOS requirements given by the maximum allowed cell loss probability and the maximum delay.

3. Edge-adapter: two levels loss-free shaper

This chapter contains the definition and analysis of a specific type of edge-adapter, which we'll refer to as *two-level loss-free shaper*. Part of the analysis of the first subsection has been already developed in [Bia92] for a different problem and target, so we will show here the basic results, leaving to the enclosed appendix a brief review and extension of the cited analysis.

3.1. Shaper analysis

A generic *two-level shaper* (figure 3) is modeled as a queue with a two rate server in which (refer to figure 4 and figure 5) the maximum service capacity C_u is requested as soon as the buffer content exceeds an upper threshold level U_t (cells), while a lower capacity C_l is retained when, after transmitting a cell, the buffer content undergoes a lower threshold L_t.

Notice that, if the upper capacity is greater or equal than the peak bit rate of the source, the buffer content cannot exceed the upper threshold, which can be assumed equal to the buffer capacity Q itself. We'll refer to this case as *loss-free shaper*, since, under these assumptions, no cell can be lost due to buffer overflow.

Figure 3: Two level shaper architecture

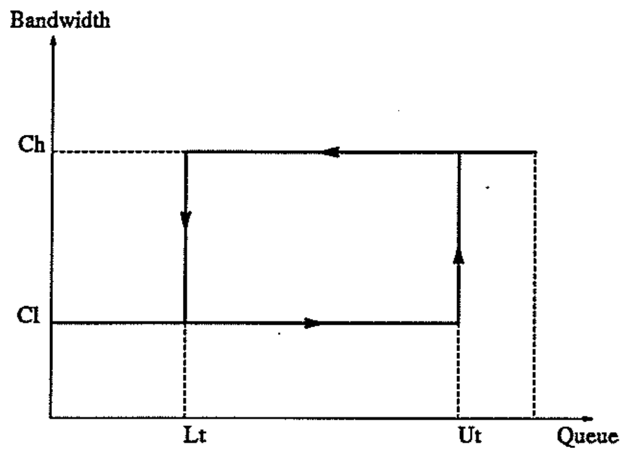

Figure 4: Server rate as a function of the queue state

Let's give some definitions and assumptions:

1. The traffic entering the buffer is modeled as a steady state arrival process composed by bursts of consecutive equally spaced cells (active periods) whose length is geometrically distributed with average B (cells). The time interval separating consecutive bursts (silent period) is also geometrically distributed with average S (time slots). The peak rate is P (cell/slot) and the average rate is $A = B/(B/P + S)$ (cell/slot).

2. The assumed shaper is a two-level loss-free shaper with upper threshold Q (cells), lower threshold $D < Q$ (cells), upper capacity P (cell/slot), equal to the peak rate of the offered traffic, and lower capacity $C < P$ (cell/slots).

3. Two extreme cases can be defined: if $D = 0$ we'll refer to the shaper as *Hysteresis shaper*, while if $D = Q - 1$ we'll refer to as *Standard shaper.*

Let $Y(t)$ be the process representing the capacity assigned to the shaper server under the previous assumptions. It is easily recognized that the times in which $Y(t)$ changes from $Y(t) = P$ to $Y(t) = C$, and conversely, are regeneration points for the process $Y(t)$. Let T_u be the average length of an *high period* (that is $Y(t) = P$), and T_l the average length of a *low period* (that is $Y(t) = C$); these periods can be evaluated as:

$$T_u = \frac{B}{P} + \frac{Q - D - 1}{P - A} \tag{1}$$

$$T_l = \phi(Q, D, C, P/A, B) \tag{2}$$

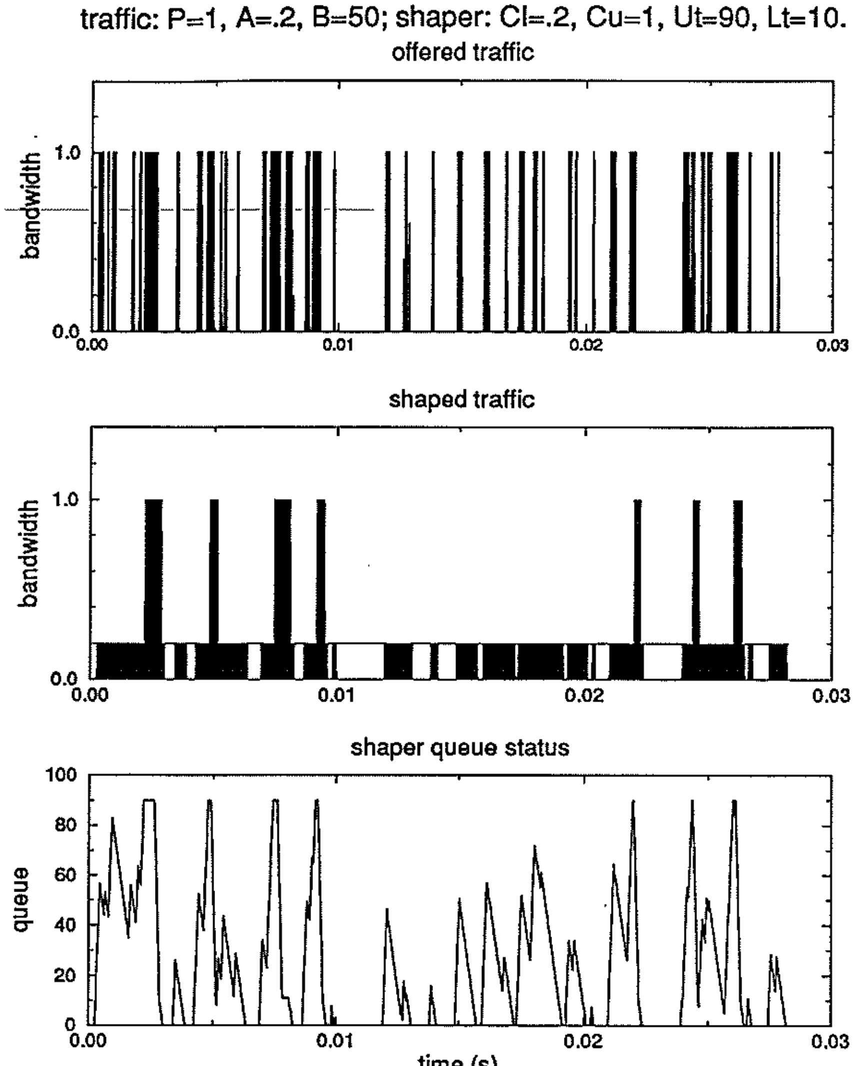

Figure 5: simulation example of two-level loss-free shaper

where ϕ is a non-closed formula and the solution of a linear system in $2Q$ variables is required.

Let's divide the cells served by the shaper server in two classes: *HP cells* are the cells served by the lower service rate C, and *LP cells* are all the remaining cells (figure 6). Clearly the LP cells are transmitted only during the high service rate periods, and during this period the fraction of LP cells is $(P-C)/P$. Define with f the total fraction of LP cells that have been served. We can express f as:

$$f = \frac{T_u(P-C)}{A(T_l + T_u)} \tag{3}$$

Let $X(t)$ be the process representing the instantaneous service rate of the shaper. $X(t)$ can assume only three values (figure 5, middle case): 0 if the shaper queue is empty, C if there are cells in the shaper queue and the server is in the low period, and P if the server is in the high period. Define with Z, L and U the probabilities of being in either state

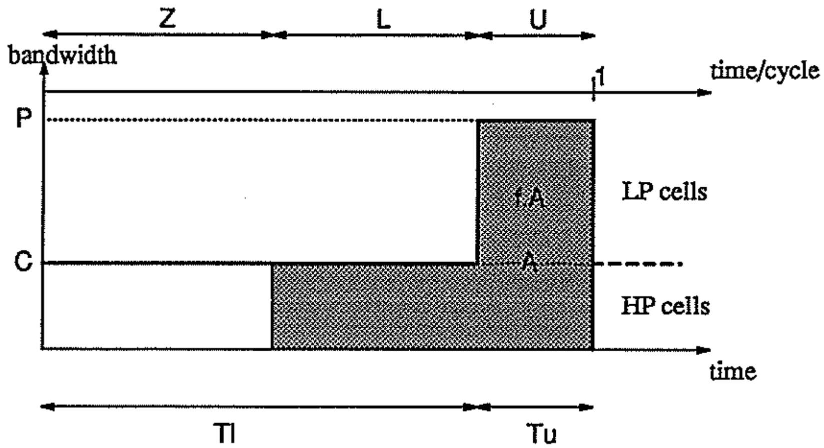

Figure 6: Graphical interpretation of the variables

respectively. It is easy to show (see figure 6) that

$$
\begin{aligned}
U &= \tfrac{Af}{P-C} \\
L &= \tfrac{A-UP}{C} \qquad (4) \\
Z &= 1 - U - L
\end{aligned}
$$

Finally, let τ_s be the maximum cell delay suffered in the shaper queue. Then

$$\tau_s = \frac{Q-1}{C} \qquad (5)$$

because the maximum delay is that suffered by a cell which, on arrival, finds itself in the $Q-1$ position in the buffer, while the buffer empties at the lower rate C.

3.2. Edge-adapter dimensioning

Suppose that the source being shaped requires the following QOS requirements: maximum delay τ_m and maximum cell loss probability λ.

Suppose now that the network is able to provide a different Grade of Service for the two different priority streams, particularly a different known cell loss rate g_{LP} and g_{HP} respectively for LP and HP cells, with $0 \leq g_{HP} < g_{LP}$. Assuming that $g_{HP} \leq \lambda \leq g_{LP}$, the source loss probability requirement can be satisfied by imposing

$$f = \frac{\lambda - g_{HP}}{g_{LP} - g_{HP}} \qquad (6)$$

Suppose furthermore that the network maximum delay is given by the fixed network propagation delay τ_p, since the queueing delay is negligible. Under the assumption that $\tau_m \geq \tau_p$, the maximum delay constraint can be satisfied by imposing $\tau_s = \tau_m - \tau_p$

The problem of guaranteeing a given QOS requirement to the source has then been transformed into the problem of dimensioning the source shaper according to the requested values f and τ_s and with the target of minimizing C when maximizing the shaper queue size. Once the type of shaper is selected ($D = 0$ for the hysteresis shaper or $D = Q - 1$ for the standard shaper), the shaper parameters Q and C can be obtained from the solution of the following non linear system:

$$\begin{cases} \phi(Q, D, C, P/A, B) = \frac{1}{Af}\left(\frac{B}{P} + \frac{Q-1-D}{P-A}\right)(P - Af - C) \\ \\ Q = 1 + C\tau_s \end{cases} \tag{7}$$

where the first equation is obtained by combining eqs. (1), (2) and (3) and the second equation is (5).

The system can be easily numerically solved by eliminating Q (or C) from the first equation and solving with any standard method the resulting non linear equation in the variable C (or Q). The termination condition is simplified by the fact that Q is an integer, so that very few solutions of the function ϕ are needed.

3.3. Application

Consider a core network configuration able to provide MTG. Thus the network doesn't guarantee any loss probability bound on the LP cells (*worst case assumption*) and, since $g_{HP} = 0$ and $g_{LP} = 1$, formula (6) becomes trivially $f = \lambda$.

The shaper dimensioning obtained by formula (7) is optimal in terms of network resource utilization, since the requested capacity C is minimized according to the constraints imposed by the traffic QOS.

Figure 7 shows, for a particular source and worst bound, how a graphical dimensioning of the shaper can be performed using such plots. Fixing the maximum tolerable shaper delay on the y-axis and the cell loss probability and the shaper type as parameter, the lower shaper service capacity can be read on the x-axis.

The use of tables, although less flexible than the direct analytical dimensioning, can be sometimes recommended for reasons of simplicity or for strict real-time dimensioning (the solution of the system (7) can require few seconds if the exact evaluation of the function ϕ is carried out).

Notice that a different table seems to be required for any different set of source parameters P, A, B and for any different value of g_{HP} and g_{LP} if the dimensioning is extended to EMTG and CSM networks. However, if $g_{HP} = 0$ or $g_{HP} << (g_{LP}, \lambda)$, it is easy to see that the worst bound table is sufficient because the dimensioning for the generic bound can be performed with a worst bound table, using, as loss parameter, the value λ/g_{LP}. Furthermore, since the function ϕ is dependent only on the average burst length and the burstiness (P/A) of the source, the number of tables can be reduced to be dependent only on these two source parameters.

Finally notice that the optimality of the dimensioning is not guaranteed for the more general cases EMTG and CSM.

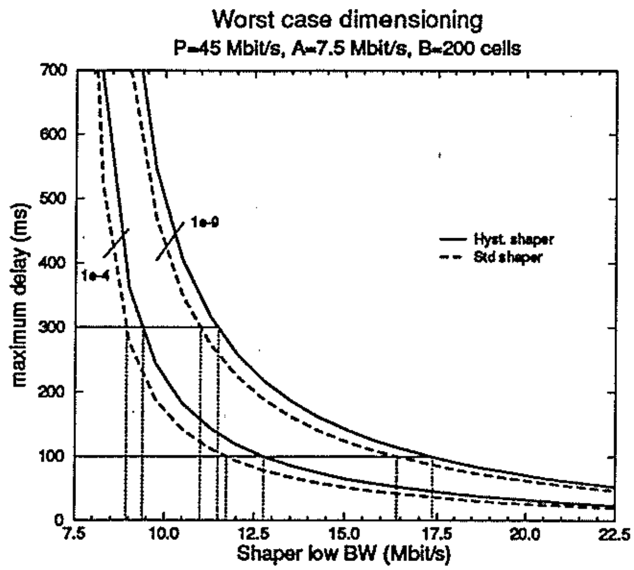

Figure 7: example of shaper dimensioning

4. Network Connection Admission Control

In the previous section we have shown that, given the cell loss GOS g_{LP} and g_{HP} for the LP and HP priority streams, a generic loss - delay QOS of a source can be guaranteed by an appropriate shaper dimensioning. Scope of this section is to present a practical Connection Admission Control (CAC) algorithm that is able to guarantee the respect of the promised GOS into the network. Particularly we will discuss the real-time computability, which is a necessary condition for any CAC algorithm.

The ideal CAC algorithm should be able to evaluate the expected cell loss probability for every connection admitted into the network, and use these values to reject or accept a new connection. This goal, as widely shown in literature, requires huge computation time, orders of magnitude over the real-time constraint imposed on the CAC.

An alternative approach, which will be used in the rest of the section, is the determination of bounds on the expected loss probability. If the algorithm to evaluate the bound is computable in real-time, it is suitable for implementation in the kernel of the CAC algorithm. Of course, the closer the bound to the exact values, the more efficient the usage of network resources and higher the network throughput.

4.1. Best bound definition

Assume that all the connections are statistically independent and the statistical description of the sources is respected during the transmission time. Then the bound is defined according to the following further assumptions:

- the evaluation of the loss bound on the HP cells is performed independently on the presence of the LP cells.

- In case of overload, HP cells have discarding priority over LP ones.
- In case of overload, the loss for a given flow of cells (HP and LP) is performed randomly and uniformly distributed among the cells of the same stream.
- A network node can absorb any contention due to synchronous arrival.
- A network node cannot absorb any contention due to instantaneous link overallocation (assumption of *null queueing*), that is, if in a certain instant the total rate of the traffic offered to the node exceeds the link capacity, then the exceeding capacity is lost.

The statistical description of the source is given by the parameters of the traffic exiting from the shaper queue. The knowledge of these statistics is used at the very best, and the bound obtained is the tightest possible once the queueing contribution at the network node is neglected, according to the null queueing assumption. Thus the name *best bound.*

Given these assumptions, the analysis is quite straightforward. Consider a mixture of n independent, generally non-homogeneous, shaped sources $s_1, s_2, ..., s_n$. The traffic offered from each shaped source is described by the shaper parameters P_i and C_i, respectively the upper and lower rate of the source shaper, and the traffic statistical parameters U_i, L_i and Z_i, respectively the probability of transmission at upper rate, lower rate and no transmission. These last three parameters can be readily evaluated, for a two-level loss-free shaper, by the knowledge of f_i, the fraction of LP cells transmitted, and A_i, the average rate of the source, by using equations (4).

Let's associate with the connections a set of couple of dependent random variables $(x_1, y_1), (x_2, y_2), ..., (x_n, y_n)$ where x_i represents the high priority rate and y_i the low priority rate. The random variables can assume only one of the following three values, with joint distribution:

$$\begin{array}{lcl} P(x_i = 0, y_i = 0) & = & Z_i \\ P(x_i = C_i, y_i = 0) & = & L_i \\ P(x_i = C_i, y_i = P_i - C_i) & = & U_i \end{array} \tag{8}$$

Consider now the superposition of these random variables: defined by $X = \sum_{i=1}^{n} x_i$ and $Y = \sum_{i=1}^{n} y_i$, X gives the total allocated bandwidth for the high priority stream, and Y the low priority bandwidth.

In order to simplify the computation, similarly to what is done in [Tur92], let's discretize the spectrum of the possible bandwidths, hence suppose that P_i, C_i are integer values in the range $0, M$, where M (say 256) is a given *granularity* parameter.

For commodity, let's consider the joint generating function:

$$f_{XY}(z_1, z_2) = \prod_{i=1}^{n} (Z_i + L_i z_1^{C_i} + U_i z_1^{C_i} z_2^{P_i - C_i}) = \prod_{i,j} K_{ij} z_1^i z_2^j \tag{9}$$

$K_{ij} = P(X = i, Y = j)$ can be numerically evaluated by knowing the traffic parameters. When a new connection acceptance request is processed, the bounds on the HP and LP cell loss probabilities for the new connection, respectively named Pl_{HP} and Pl_{LP}, can be immediately evaluated, given the new source parameters P, C, U, L, Z:

$$Pl_{HP} = \sum_{i=M-C+1}^{nM} \sum_{j=0}^{nM} K_{ij} \frac{i+C-M}{i+C} \tag{10}$$

$$Pl_{LP} = \sum_{i=0}^{M-C-1} \sum_{j=[M-P-i+1]^+}^{nM} K_{ij} \frac{i+j+P-M}{j+P-C} + \sum_{i=M-C}^{nM} \sum_{j=0}^{nM} K_{ij} \tag{11}$$

where $[x]^+ = \max(x, 0)$. When homogeneous sources are considered, the call acceptance algorithm can stop here, since all the sources are identical and with same HP and LP cell loss probability. Clearly, in the general case, it is now necessary to evaluate the effect of the new connection admission on the already accepted connections. In fact the CAC must be able to recognize if the admission of a new source leads to GOSes violation for other sources, and in this case the new source cannot be admitted although the GOS requirements for the source itself are satisfied.

The necessary steps are the following. Let's first evaluate the coefficients $K'_{i,j}$ including the contribution of the new connection. In an incremental way,

$$K'_{i,j} = K_{i,j} \cdot Z + K_{i-C,j} \cdot L + K_{i-C,j-(P-C)} \cdot U \tag{12}$$

For every background connection k, let's evaluate the coefficients $K''_{i,j}$ obtained by removing from $K'_{i,j}$ the influence of the connection itself. The evaluation is simply performed by the recurrence

$$K''_{i,j} = \frac{1}{Z_k} \left(K_{i,j}{}' - K''_{i-C,j} \cdot L_k - K''_{i-C,j-(P-C)} \cdot U_k \right) \tag{13}$$

where the coefficients with negative indexes are null. Finally the HP and LP cell loss bounds for the connection k are given by the formulas (10) and (11) using respectively $K''_{i,j}$, P_k and C_k instead of $K_{i,j}$, P and C. If, for each connection k, both the HP and LP cell loss probability bounds don't overflow the limits g_{HP} and g_{LP} respectively, the new connection can be safely admitted and the coefficient matrix updated.

When connection release occurs, no bound evaluation is necessary, and the new matrix of coefficients is simply obtained by equation (13) applied to the coefficients $K_{i,j}$.

Figure 8 shows that the best bound is very close to the simulation results (plotted with 95% confidence interval). Notice that the bound for the hysteresis shaper is much closer to the simulation than the bound for the standard shaper. This is explained considering that the small amount of buffer used in the simulation can partially absorb the effects of the overload period for the standard shaper, since the average length of a high period, B/P, is relatively short and comparable with the used multiplexer buffer size. Conversely, in the case of hysteresis shaper, the average high period length (formula (1)) is considerably longer.

The size of the $nM \times nM$ coefficient matrix $[K_{ij}]$ can be strongly reduced to a $K_1M \times K_2M$ matrix (with typical values $K_1 = 1.5, 2$ and $K_2 = 3, 4$), because the coefficients with high i and j values will be very likely zero or negligible, and can be safely omitted in the computation.

In spite of such computational complexity reduction, the best bound approach seems definitely impractical for a real-time application because of the large amount of computation required. Let's in fact approximately evaluate the requested number of arithmetic operations for each procedure:

1. about $M^2(K_1 - 1)K_2$ macro-operations, each composed by two products and four sums, for the HP loss probability evaluation;
2. about $M^2(K_2 - 1/2)$ macro-operation, each composed by two products and six sums, plus $M^2(K_1 - 1)k_2$ additional sums for the LP loss probability evaluation;
3. for each of the $M^2 \cdot K_1K_2$ matrix coefficients, three products and two sums are needed for the evaluation of the new matrix of coefficients after the acceptance of a new connection;
4. same number of operations of item 3 for the evaluation of the new matrix of coefficients after the release of a connection;

Noting that procedures 1, 2, 4 must be repeated for n active connections and procedures 1, 2, 3 must be performed for the new connection only, the total number of operations is roughly $11(n+1)K_1K_2M^2$, of which about $5(n+1)K_1K_2M^2$ are products or divisions. Clearly, with a set of parameters like $M = 256$, $K_1 = 3/2$, $K_2 = 3$ and 100 active connections, the total number of operations, restricted just to the products, is of the order of 150 millions, that require computational time of the order of several seconds, far away from the constraints imposed by a real-time computation.

In order to try to reduce or simplify the computation, some observations can be made. First, it is clear that the reduction of the granularity M allows the management of a smaller matrix and a consequent reduction of the number of operations to perform. Clearly the obtained management suffers of granularity approximations: a source with a given rate must be declared according to the superior integer of the granularity scale, thus getting a worse bound evaluation.

Another promising way to reduce the computation is to exploit the property of ordering among the offered connections, which allows us to avoid the computation of the GOS violation for part of the active connections. Such study is out of the scopes of this report: we have briefly outlined this idea in appendix B. We have also shown there how such property can be found for a simple, but interesting example.

A last observation is quite interesting. Notice that in case of large matrix and low number of connections, the coefficient matrix is considerably sparse. Hence the few non zero coefficients can be stored in a list thus reducing the memory occupancy and the number of operations to be performed. Entering in some details, let's store the coefficients K_{ij} in a list of structures, where each element is a record with three fields: the X position i, the Y position j and the coefficient value v. In order to guarantee an effective computation,

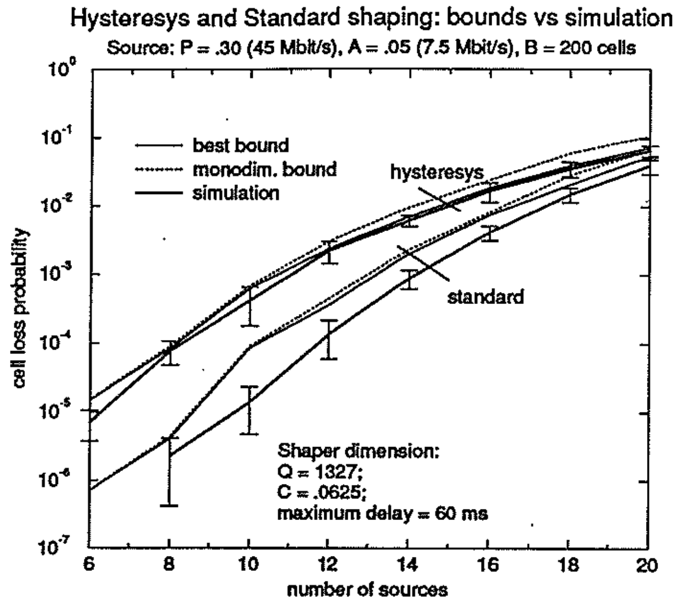

Figure 8: simulation, best bound, monodimensional bound performance

the previous formulas (10), (11), (12) and (13), referring to the coefficients K_{ij}, have to be translated in equal formulas referring to the elements in the list of the coefficients. This is not trivial, especially for (12) and (13), but it can be effectively made with some additional effort (we omit here more details).

This approach is very interesting as an alternative to the use of the matrix, particularly because, as it is easy to understand, the spectrum of offered bandwidths is no longer constrained by a predefined granularity M. The main problem is that this approach doesn't solve at all the computational problems: if the number of connections is fairly high, and the sources are heterogeneous, the sparseness of the matrix is no more an advantage. Moreover, the computational overhead of having more complex procedures leads to an evaluation time greater than that achievable in the matricial approach.

4.2. An intermediate monodimensional bound

In order to overcome the computational limitations imposed by the best bound, for practical CAC algorithm it is necessary to define *intermediate bounds* which can be evaluated in real time. Virtually a great number of intermediate bounds can be defined: here we propose a *monodimensional bound*, which allows real-time computation with a small degradation of performance with respect to the best bound.

Suppose we know the maximum HP bandwidth that can be offered to a network node: $h = \sum_{k=1}^{n} C_k$. Instead of considering all the possible different contributions of HP and LP cells (best bound), let's just consider the worst case, that is the bandwidth allocated in the node is composed by HP bandwidth up to the maximum h. Being $K_{x,j} = 0$, if $x > h$, and defined with b the overall bandwidth supposed into the node, the LP cell loss probability best bound, conditioned to h and b, can be bounded as follows:

$$Pl_{LP}/b, h = \sum_{j=[b-h]^+}^{b} K_{b-j,j} \frac{b + P - M}{P - C + j + [b - j + C - M]^+} \leq$$

$$\leq \sum_{j=[b-h]^+}^{b} K_{b-j,j} \frac{b+P-M}{P-C+[b-h]^+ + [\min(b,h)+C-M]^+} =$$

$$= \frac{b+P-M}{P-C+[b-h]^+ + [\min(b,h)+C-M]^+} \sum_{j=[b-h]^+}^{b} K_{b-j,j} =$$

$$= \frac{b+P-M}{P-C+[b-h]^+ + [\min(b,h)+C-M]^+} \cdot H_b \tag{14}$$

with the coefficients H_b depending only on the overall bandwidth and the maximum amount h of HP bandwidth.

The computation of the coefficient H_b can be directly performed by applying a procedure similar to that outlined for the best bound. Given a set of connections $s_1, s_2, \cdots, s_n$ with correspondent parameters P_i, C_i, U_i, L_i, Z_i, let's define x_i to the random variable representing the bandwidth offered by the source i and $X = \sum_{i=1}^{n} x_i$ the total offered bandwidth. It is:

$$\begin{cases} P(x_i = 0) & = & Z_i \\ P(x_i = C_i) & = & L_i \\ P(x_i = P_i) & = & U_i \end{cases} \tag{15}$$

and the generating function representing the distribution of X is given by

$$f_X(z) = \prod_{i=1}^{n} (Z_i + L_i z^{C_i} + U_i z^{P_i}) = \prod_i H_i z^i \tag{16}$$

with coefficients $H_i = P(X = i)$. Notice that now the coefficients are stored in a vector, so that we'll refer to this bound as *monodimensional bound*. Formulas equivalent to (10), (11), (12) and (13), can be now obtained for this case. Since $h = \sum_{k=1}^{n} C_k$ is the maximum offered HP bandwidth for a given set of background connections, the HP and LP loss probabilities for a new connection are given by

$$Pl_{HP} = \sum_{b=M-C+1}^{nM} H_b \frac{\min(h,b)+C-M}{\min(h,b)+C} \tag{17}$$

and

$$Pl_{LP} = \sum_{b=M-P+1}^{nM} H_b \frac{b+P-M}{P-C+[b-h]^+ + [\min(b,h)+C-M]^+} \tag{18}$$

while the formulas for including and deleting a given connection from the coefficient vector, with same notation of (12) and (13), are respectively

$$H_i' = H_i \cdot Z + H_{i-C} \cdot L + H_{i-P} \cdot U \tag{19}$$

and

$$H_i'' = \frac{1}{Z_k} \left(H_i' - H_{i-C}'' \cdot L_k - H_{i-P}'' \cdot U_k \right) \tag{20}$$

The complexity of this bound is now proportional just to M (instead of M^2), thus leading to acceptable computational complexity. Furthermore, all the simplifying considerations carried out for the best bound case still hold. Figure 8 and more detailed numerical results in the next chapter show that the degrade of performance with respect to the best bound is very small.

5. Numerical results

In this section we show the performance achievable by the proposed bandwidth management schemes and we try to compare the results with those achievable by using statistical multiplexing and peak rate allocation. The proposed framework has been applied to some simple examples, where, for simplicity, homogeneous sources have been considered. Numerical results report the maximum number of connections that can be admitted in the network when a given CAC scheme and bound evaluation method applies.

The bound for the statistical multiplexing case can be simply derived from the previous chapter analysis as a particular case, imposing that all the cells are tagged LP (or HP). Thus the new network traffic descriptor is given by $P=$ peak rate, $A=$ average rate, $C=0$ and $f=1$ ($C=P$, $f=0$). Comparison with fast buffer reservation CAC [Tur92] is implicitly assumed by the comparison with statistical multiplexing, since the cell loss performance of fast buffer reservation is lower than that achievable by statistical multiplexing.

We would like to point out that a fair CAC-based comparison among statistical multiplexing and the proposed network management schemes is quite tricky.

First of all, the performance achievable by the EMTG and CSM approaches, on one side, and statistical multiplexing, on the other, are strongly dependent on the selected network GOSes. Moreover the optimization of the GOS parameter for the EMTG and CSM methods has not been accomplished yet, while for the statistical multiplexing case the best performance is obtained with a GOS requirement exactly equal to the QOS loss requirement. The problem of the GOS selection is complicated by the assumption that the network must be able to satisfy the most stringent QOS requirement (say 10^{-9}). For the MTG and EMTG cases this is always assumed, while for the CSM case it is just necessary that $g_{HP} \leq 10^{-9}$. The performance obtained in the following examples thus exploit a choice of parameters which can support any kind of traffic. Instead, for statistical multiplexing, the network GOS must be at least equal to 10^{-9} and the network throughput is strongly reduced from the one obtained with the optimal GOS selection. Moreover, note that, even though statistical multiplexing handles two different levels of priority (with GOS, say, 10^{-3} and 10^{-9}), for a source with QOS loss requirement just slightly below 10^{-3} it is necessary to guarantee 10^{-9}. Hence, the flexibility and adaptability of our approaches to any traffic condition must be heavily taken into account for any comparative consideration.

We may point out that in the following results we have considered statistical multiplexing, without assuming any shaping. It is straightforward to obtain a monodimensional best bound for the statistical multiplexing of shaped sources, where the cells leaving the shaper queue are no more tagged. The value of such bound can be also obtained from the best bound computed for the CSM method and considering just the overall loss probability

$$Pl_{tot} = Pl_{LP} \cdot f + Pl_{HP} \cdot (1-f). \tag{21}$$

Notice that, if the GOS of the network is equal to the QOS of the sources, the performance obtained in this case are the maximum theoretically achievable by the CSM case. The explanation is straightforward if we consider that a connection request, in the CSM case, is rejected when one of the two GOS profiles is violated, while in the case of statistical multiplexing with untagged shaped traffic, rejection occurs only when the GOS is violated by the traffic as a whole. We'll refer to the performance of this case as *maximum CSM*.

A complete overview of the potentiality and performance drawbacks of the outlined approaches is out of scopes of this section. Furthermore, according to the previous considerations it is clear that a comparison among so different strategies is clearly difficult to carry out. Thus, in the rest of the section, we just consider a couple of interesting examples. Performance of statistical multiplexing, peak rate allocation and our approaches are reported. The shaper dimensioning for all our approaches is made according to the one outlined in section 3. Thus it is optimal only for the MTG case, while the performance evaluated for the EMTG and CSM cases can be enhanced by an *ad hoc* dimensioning. performance of the maximum CSM case, that is the maximum performance achievable by CSM with the given (sub-optimal) shaper dimensioning are enclosed.

5.1. Example 1

In the following we normalize all the bit rates to the link bit rate, assumed equal to 1. In this example we consider sources that have a very high peak and average bit rates compared to the link bit rate of the network links. In particular, we consider the multiplexing of homogeneous sources with peak rate $P = .25$ (i.e. 37.5 Mbit/s with a multiplexer link capacity of 150 Mbit/s), average rate $A = .05$ (7.5 Mbit/s) and average burst length $B = 200$ cells. We suppose that the sources require the same QOS, which is specified as maximum expected cell loss probability 10^{-4} and we explore three maximum QOS queueing delay requirements (τ): 10, 100 and 300 ms.

Let's first perform the dimensioning of the traffic shaper. We have stopped the computation of C at the third decimal digit, coherently with a bound evaluation granularity of $M = 1000$. The values of the shaper queue length Q and the requested HP capacity C are presented in the following table, for the case of MTG, where the fraction f of LP cells is equal to the expected maximum cell loss probability (10^{-4}), and the EMTG and CSM, supposing an LP GOS profile of 10^{-2} and neglecting (for the CSM case) the HP GOS (thus $f = .01$).

SHAPER TYPE		MTG			EMTG, CSM		
	τ (ms)	10	100	300	10	100	300
hysteresis	C	.176	.082	.061	.143	.066	.054
	Q	620	2833	6474	503	2262	5654
standard	C	.169	.075	.059	.130	.060	.052
	Q	587	2674	5820	452	2045	5121

The following table reports, for the different approaches, the value N defined as the maximum number of sources which can be allocated in a multiplexer node in the respect

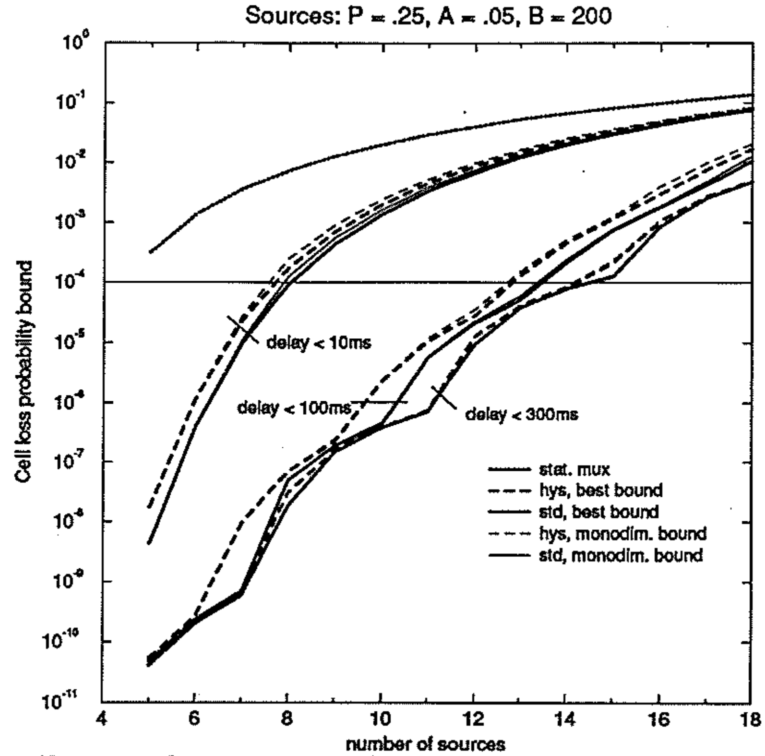

Figure 9: loss versus number of sources for the example 1

of the cell loss GOS constraints. Best and monodimensional bounds lead to practically coincident results (figure 9), and the value of N is equal for both the bounds. We have also omitted results of the CSM case, because, for $g_{HP} \leq 10^{-6}$, there is no improvement on the EMTG case. We have reported, with the name *maximum CSM*, the global performance of the CSM case disregarding the violation of LP or HP GOSes. This results are obtained using equation (21) and considering as GOS violation the overflow of just the total cell loss probability.

Method	number of multiplexed sources N					
	hys,10	std,10	hys,100	std,100	hys,300	std,300
peak rate allocation	4					
statistical mux	4					
MTG	5	5	12	13	16	16
EMTG	6	7	12	13	14	14
maximum CSM	7	8	12	13	14	14

For each policy the advantage in comparison with the statistical multiplexing or the peak allocation is clear, since the considered traffic type does not allow to draw any advantage from statistical multiplexing. Note that the shaping of traffic leads to impressive performance gains for all the methods. Noteworthy is the case of MTG with 300 ms delay, for which N is very near to the theoretical maximum of 20.

MTG achieves better performance than other policies that could exploit better statistical multiplexing since the shaper dimensioning has been performed with the target of optimal performance for the MTG method. From this example it is thus clear that the extension of the MTG optimal shaper dimensioning to the cases of EMTG and CSM is very far from the optimal.

All the policies exploiting shaping achieve very high performance compared to statistical multiplexing ones. This is primarily due to the high average rate of the considered sources. In fact it can be easily seen that sources with the same burstiness (P/A) and burst length, shaped with a same lower capacity C in proportion of P, require the same queue length Q for guaranteeing a same fraction of LP cells. On the other side, the same Q corresponds to different maximum delays, which are dependent on the absolute value of C. Thus a better shaping action, for a same set of QOS requirements, is made for sources with high average with respect to sources with small average. As extreme case consider a source with $P=37.5$ Kbit/s, $A=7.5$ Kbit/s and $B=200$ cells: the delay obtained with the same shaper parameters given in the table is exactly 1000 times greater.

Some further observations can be drawn by the graph in figure 9, which plots the cell loss probability bound for statistical multiplexing and the overall cell loss probability for the CSM approach (i.e. maximum CSM). The difference among the two different shaping techniques are relatively low. Moreover the plots are very irregular, because a very low number of high rate sources is mixed and each individual source heavily impacts on the loss probability results, and can fit better or worse with the irregular distribution of the table or vector of coefficients.

5.2. Example 2

Let's consider the multiplexing of sources with lower peak and average bit rates. A source with a lower rate will exploit shaping advantages only at the expense of maximum delay increase. So we expect MTG to be less performant than the other approaches.

Let's consider a source with $P = .1$ (15 Mbit/s), $A = .01$ (1.5 Mbit/s) and $B = 200$ cells, with same QOS requirements of the previous example (cell loss $= 10^{-4}$ and maximum shaper delay $\tau = 10$, 100 and 300 ms).

The shaper dimensioning for the cases MTG and EMTG (CSM) are reported in the following table, with the assumptions of $g_{LP} = .01$ for the EMTG and CSM dimensioning (thus $f = .01$).

SHAPER TYPE		MTG			EMTG, CSM		
	τ (ms)	10	100	300	10	100	300
hysteresis	C	.083	.042	.024	.071	.030	.018
	Q	294	1441	2522	241	1053	1791
standard	C	.082	.038	.021	.068	.025	.015
	Q	279	1290	2173	232	881	1589

The following table contains the comparative results, with two different GOS for statistical multiplexing (10^{-4}, optimum, and 10^{-9}, realistic), and g_{LP} fixed to 10^{-2} for EMTG and CSM (two cases of g_{HP}: 10^{-9} and 10^{-6}).

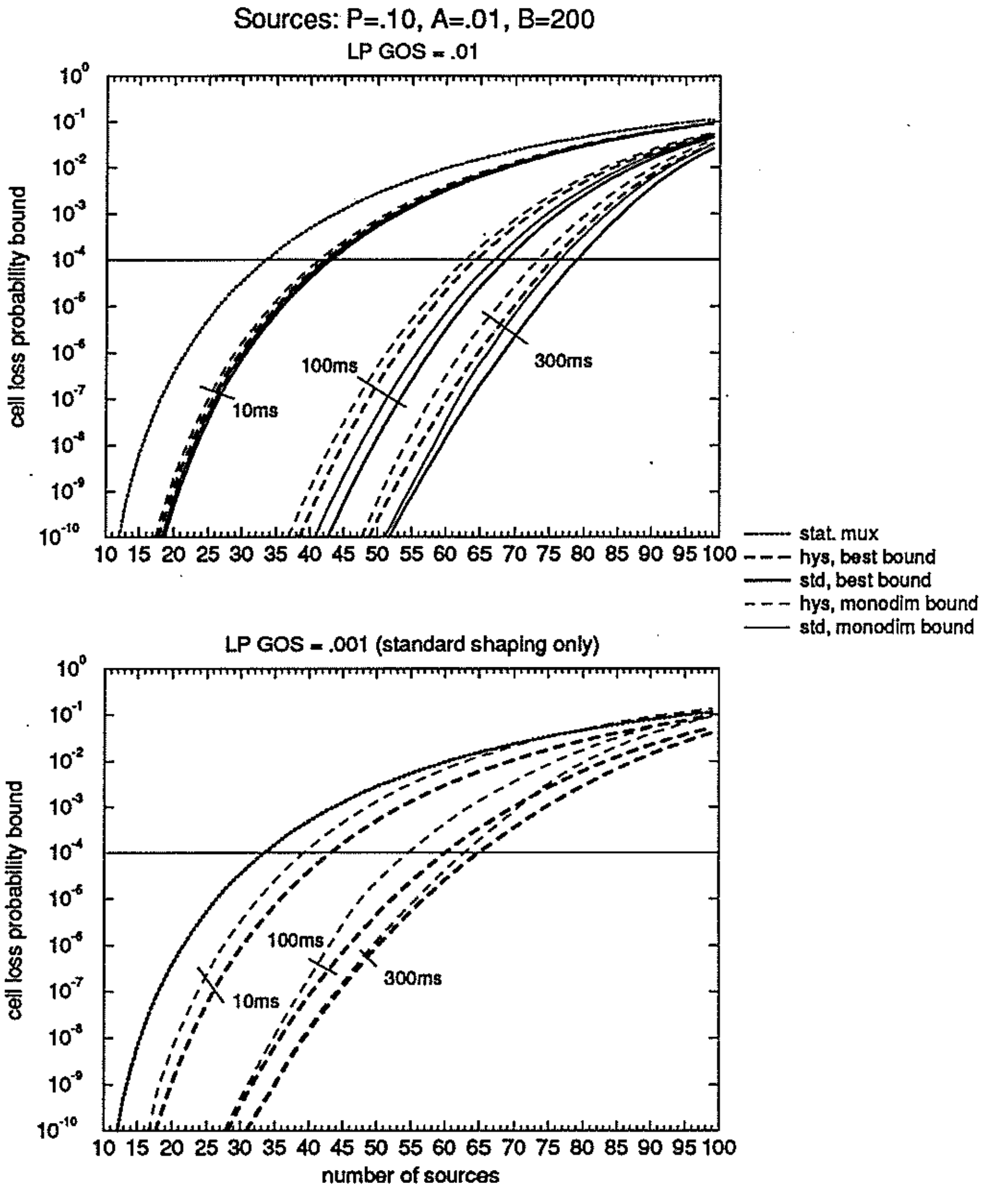

Figure 10: Loss versus number of sources for the example 2

Method	number of multiplexed sources N					
	hys,10	std,10	hys,100	std,100	hys,300	std,300
peak rate allocation	10					
stat. mux (GOS 10^{-4})	33					
stat. mux (GOS 10^{-9})	14					
MTG	12	12	23	26	41	47
EMTG	14	14	33	40	55	66
CSM ($g_{HP} = 10^{-9}$)	20 (19)	21 (20)	46 (41)	52 (46)	64 (55)	72 (66)
CSM ($g_{HP} = 10^{-6}$)	30 (29)	31 (30)	55 (53)	61 (57)	71 (65)	78 (69)
maximum CSM	42 (41)	43 (42)	64 (62)	68 (66)	75 (73)	78 (76)

It can be noticed that, unlike the previous example, MTG has always lower performance

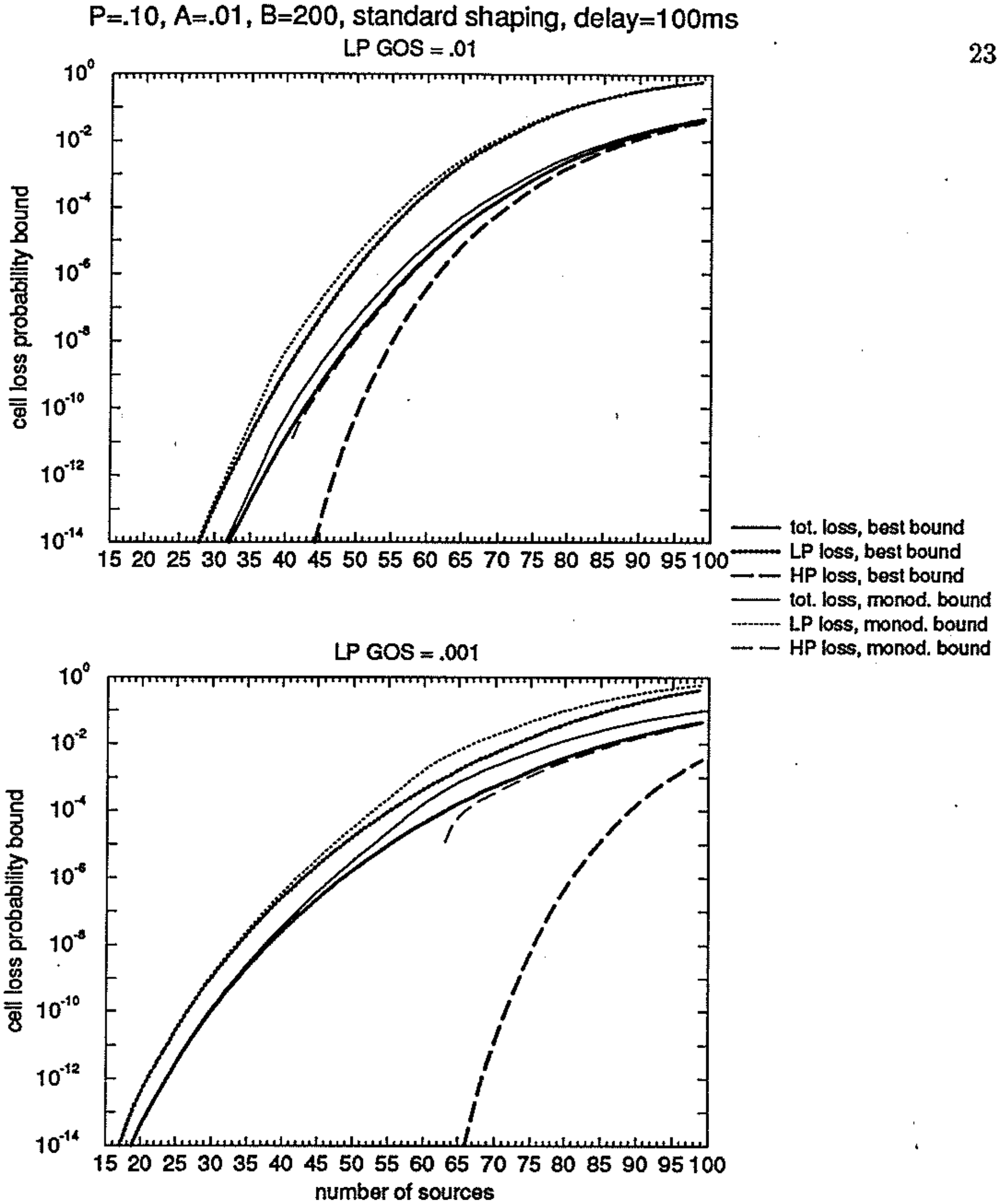

Figure 11: HP, LP and total cell loss probability, example 2

with respect to EMTG and CSM techniques. In fact a strong shaping effect for relatively low rate sources can need queue size that may lead to exceed the delay constraints. The comparison with statistical multiplexing (with optimum GOS) shows that the performance of MTG are still worse also for fairly high delay requirement (100 ms), while the performance are better only for 300 ms.

The analysis of the results of the EMTG method (with the given non-optimal shaper dimensioning) shows that, for a target of high network throughput performance, the constraint of zero loss imposed on the HP stream can be quite resource-consuming. This is evident if we compare the EMTG results with the CSM technique, whose results show that, removing the assumptions of zero loss on the HP stream, the network performance can improve and tend to the theoretical maximum CSM. Of course note in the CSM technique that, for practical considerations, a GOS profile of 10^{-6} on the HP stream can be unacceptable.

Finally, note that the best and monodimensional bounds give very close results for the EMTG and maximum CSM cases (see also figure 10 for this last case), while results of the monodimensional bound for the CSM methods are strongly impaired. For an explanation, notice in figure 11 that the LP loss probability for the two bounds is very close, and thus the overall probability, which primarily depends on the LP loss (eq. (21)). This explains the accordance of results for the EMTG method, in which the HP loss is zero, and the maximum CSM, in which the only parameter of interest is the overall loss probability. From the same figure we notice that the disagreement among the HP cell loss probability obtained with the best or monodimensional bound is of orders of magnitude. Hence the performance of the CSM method, depending directly on the HP loss values, are strongly degraded when the monodimensional bound is assumed.

The sharp approximation in the computation of the HP loss, can be easily overcome with a small augment of the computational complexity of the CAC. Instead of using the same vector coefficients for the evaluation of both LP and HP loss probabilities, the coefficient vector given in chapter 4 can be used just for the approximate computation of the LP loss, and a new coefficient vector, which allows the exact computation of the HP loss, should be introduced. The new vector is built with new traffic descriptors which neglect the contribute of the LP cells. Thus we consider all the sources as two rate processes offering only HP traffic, precisely with rate zero with probability Z and C with probability $U+L$. Notice that the CAC algorithm must manage now two vectors, with consequent increase of the computational complexity.

It is very interesting to consider what happens if the LP GOS profile changes both in the shaper dimensioning and in the EMTG and CSM network management. With $g_{LP} = 10^{-3}$, The new shaper dimensioning is:

SHAPER TYPE		EMTG, CSM		
	τ (ms)	10	100	300
hysteresis	C	.057	.021	.013
	Q	193	741	1334
standard	C	.050	.016	.011
	Q	178	567	1031

The table of performance, for the cases of EMTG, CSM and maximum CSM is modified as follows:

Method	number of multiplexed sources N					
	hys,10	std,10	hys,100	std,100	hys,300	std,300
EMTG	17	20	47	62 (58)	64 (62)	66 (65)
CSM ($g_{HP} = 10^{-9}$)	27 (20)	32 (22)	60 (47)	63 (58)	64 (62)	66 (65)
CSM ($g_{HP} = 10^{-6}$)	39 (31)	43 (34)	60 (49)	63 (58)	64 (62)	66 (65)
maximum CSM	43 (39)	46 (42)	60 (54)	63 (58)	64 (62)	66 (65)

Notice (figure 10 and table) that the maximum CSM performance is almost always reduced with respect to the previous LP GOS. On the other side, as figure 11 shows, the HP loss probability is strongly reduced. Since the performance of the CSM and EMTG approach are strongly affected by the HP loss, as previously showed, we can justify the strong performance achievable in this case, very near to the maximum CSM performance. Particularly notice (from the table and figure 11) that the condition of rejection of a new connection is very often given by the LP GOS violation. For example, in the case of 300 ms delay, the rejection of a new connection happens when the HP loss is still zero. The improve of performance of the HP traffic is of course explained by the reduction of the total amount of HP cells in the network, for a given offered load.

6. Conclusions and further investigations

In this paper we have proposed a new framework for bandwidth management, congestion control and integrated services provision in an ATM-based B-ISDN. The basic ideas of the framework can be summarized in the following items.

1. We suggest that the B-ISDN should be divided in two parts:

   (a) A simple core network able to provide and grant a small number of GOS profiles. Since we impose that the queues in each network node are very short, the queueing delay is negligible and the GOS profiles can be defined only in terms of cell loss probability.

   (b) An access network, made of network adapters, responsible for the execution of the adaptation functions and, primarily, of the matching of the QOS requirements with the provided core network GOS profiles.

2. The core network manages and controls the traffic requiring different GOS profiles, independent of the identity of the source which generated the traffic streams. Particularly, different GOS profiles can be exploited to forward the stream of cells generated by any individual source.

3. The edge adapters shape the incoming traffic, taking advantage of the QOS delay requirement, and generate, from a single traffic stream, multiple traffic flows addressed to different GOS levels in the core network.

4. The user traffic descriptor and the QOS requirements of the incoming traffic are translated by the edge-adapter in a network traffic descriptor. The core network connection acceptance algorithm relies on the network traffic descriptor to decide whether to accept the connection.

We have focused on a core network implementing two GOS profiles and we have shown how they can be managed with the standard ATM cell header format and well known queueing architectures. We have presented three different core network configuration proposals

(MTG, EMTG, CSM). They could be envisioned as subsequent steps in a soft evolution path, through more and more sophisticated bandwidth management techniques, up to the final implementation of B-ISDN. Note that policing functions are needed for the methods (EMTG, CSM) that claim to take advantage of the knowledge of the statistical characterization of the shaped traffic. For each proposal, we have outlined the connection admission control procedure. We have estimated its computational complexity for real time implementation and proposed different levels of approximation of the loss performance bound.

On the access network side, we have proposed and studied a particular class of edge-adapters. The dimensioning of this class of edge-adapters has been performed for bursty data traffic and QOS traffic requirements defined as cell loss probability and maximum end-to-end delay. The proposed dimensioning is optimum in the MTG case.

Preliminary numerical examples have been shown. The different network configuration proposals, statistical multiplexing and peak rate allocation have been compared in terms of maximum network throughput achievable with the proposed CAC algorithms. Some preliminary conclusions have been drawn:

- the network utilization performance are strongly dependent on the core network parameters and the edge adapter design and dimensioning.
- traffic that is delay tolerant can take substantial advantage of the shaping at the edge of the network;
- the higher the peak and average bit rates of the traffic with respect to the link bit rate, the higher the gain of the proposed policies with respect to statistical multiplexing.
- With equal edge-adapter dimensioning, the CSM method achieves better performance than EMTG and MTG in order. EMTG and CSM usually perform much better than statistical multiplexing, although for real-time traffic and unfavorable selection of the network parameters, they can perform worse.
- An optimal selection of the network parameters always guarantees better performance than the simple statistical multiplexing.

Final favorable conclusions can be drawn from the proposed framework:

1. The different network configuration policies are based on the same hardware platform; thus the upgrade from MTG to CSM in successive steps can be made completely in software.
2. The CAC algorithms, if the monodimensional bound is used, cope with the real-time requirement of fast connection admission.
3. the CAC algorithm is independent of the heterogeneity of the traffic and the QOS requirements.
4. All the proposed approaches guarantee a spectrum of loss requirement as wide and dense as possible.

5. The proposed approach is totally flexible and it can handle any QOS requirements, particularly those unmanageable with other techniques (like loss distribution, which can be modified with different grouping of the LP cells in the edge-adapter).

Many future developments and questions are now left open. The definition of a core network with more than two GOS profiles is recommended in order to improve the fitness to the QOS requirements and to better exploit the achievable performance of the framework. The extension of the analysis is quite straightforward, while practical problems, such as multi-priority network queues management and compatibility with the standard ATM cell header format, must be solved.

More effort is needed to understand better the performance achievable by the framework with different tuning of the parameter configurations, primarily in the critical case of just two GOS profiles in the core network.

The guidelines for the design of the edge-adapter can be improved or modified with the objective of maximizing the performance in terms of network occupancy. Moreover a study of different edge-adapters and their dimensioning, for more general types of traffic and for extended QOS requirements, is strongly recommended.

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Appendix A

Our approach to the evaluation of T_l consists in deriving an analytical model for a queue with deterministic server (rate C) and evaluate first passage times. Unfortunately this approach is not practicable if we retain the cell period as time unit. Hence, we approximate the solution by assuming a discrete-time queueing system such that:

1. One *time unit* is the time needed to transmit a cell at the capacity C.
2. Multiple cell arrivals can occur only at instants $n = 1, 2, \ldots$ (the time unit as defined above is greater than the cell time).
3. Generally the time unit is not an integer multiple of the cell period. So, if fraction of cells are present, the time unit, as far as arrivals are concerned, is considered composed of an integer variable number of cell periods with average $1/C$.

Let's solve first the problem with $P = 1$. According to the previous assumptions, the buffer content can be described by the following chain:

$$q(n+1) = \min(Q, \max(q(n) - 1, 0) + a(n+1)) \tag{22}$$

where $a(n)$ is the r.v. representing the number of cells arrived during the n-th slot. This chain is not Markovian; however the bidimensional chain $(q(n); s(n))$, where $s(n) = 0, 1$ represents the state of the bursty source (inactive, active) is indeed Markovian.

We are interested in describing the arrival process $(a(n); s(n))$. To the purpose, let m be the discrete time representing cell arrival instants and $N(m, m+r)$ be the number of cell arrivals between m and $m + r$. We are interested in deriving the steady-state probability distribution:

$$\alpha_{i,r}(j,k) = P(N(m; m+r) = k;\ s(m+r) = j/s(m) = i) \tag{23}$$

Denoting $p_{01} = 1/S$, $p_{11} = (S-1)/S$, $p_{10} = 1/B$, $p_{00} = (B-1)/B$, it is easily proved that the above distribution obeys to the following recursion:

$$\alpha_{i,r}(j,k) = \sum_{h=0}^{1} p_{ih}\alpha_{h,r-1}(j, k-i) \tag{24}$$

with starting condition

$$\alpha_{i,1}(j,k) = \begin{cases} p_{ij} & k = i \\ 0 & \text{otherwise} \end{cases}$$

The distribution α exactly describes the number of cells arrived in a transmission slot if the slot length $1/C$ is an integer multiple of the cell time. If this is not the case, let c and d be respectively the integer and the fractional part of $1/C$. By assumption 3, we assume the slot time as composed of either c or $c+1$ cell times with probabilities $d-1$ and d respectively. Arrivals in the $c+1$ cell of a slot are described by the following distribution σ, whose indexes are defined as those of α:

$$\begin{array}{ll}
\sigma_{0,d}(0,0) = 1 - dp_{01} & \sigma_{0,d}(1,0) = dp_{01} \\[1em]
\sigma_{1,d}(0,0) = 0 & \sigma_{1,d}(1,0) = 1 - d \\[1em]
\sigma_{0,d}(0,1) = 0 & \sigma_{0,d}(1,1) = 0 \\[1em]
\sigma_{1,d}(0,1) = dp_{10} & \sigma_{1,d}(1,1) = dp_{11}
\end{array}$$

The distribution $\gamma_i(j,k)$ representing the probability that k cell arrive in a service slot and the source is in state j given that the state in the previous slot was i, is finally obtained as:

$$\gamma_i(j,k) = \sum_{h=0}^{1}\sum_{r=0}^{1} \alpha_{i,c}(h, k-r)\sigma_{h,d}(j,r) \tag{25}$$

Let now $t_{q,s}$ be the average transition times from states (q,s), $q < Q$ to state Q, whichever s is. These times are obtained by the following system of equations:

$$\begin{aligned} t_{0,s} &= 1 + \sum_{k=0}^{Q-1} [\gamma_s(0,k)t_{k,0} + \gamma_s(1,k)t_{k,1}] \\ t_{q,s} &= 1 + \sum_{k=0}^{Q-q} [\gamma_s(0,k)t_{q-1+k,0} + \gamma_s(1,k)t_{q-1+k,1}] \qquad q \geq 1 \end{aligned} \tag{26}$$

Finally, the period length T_l that we need is then obtained, for a generic lower threshold D, as:

$$T_l = (p_{00}t_{D,0} + p_{01}t_{D,1})\frac{1}{C} \tag{27}$$

The coefficient matrix of system (26) is a quasi-triangular band matrix that can be easily reduced to triangular band by a particular operation of the Gauss-Jordan elimination method. This technique, which complexity is of the order of Q (instead of Q^3 for standard Gauss-Jordan elimination or LU decomposition), allowed to solve in very short time huge systems (for example, a 100000×100000 matrix, corresponding to $Q = 50000$, is solved in less than 10 seconds on a sparc station 2), although the resolution algorithm is limited by round-off errors which become a constraint when $C < A$.

Again about computational complexity, the time required for the evaluation of (26), although short, can be a constraint for real-time evaluations. Fortunately it can be noticed that two systems whose parameters are related by $B' = B/x,\ S' = S/x,\ Q' = Q/x,\ x > 1$, being the others unchanged, may correspond to the same physical system but different cell units. Thus, the changes in the investigated parameter T_l (in seconds) are due to quantization effects and we have numerically verified that these changes are negligible in first approximation, especially if the quantization step is small with respect of the average burst size B and the buffer size Q. Thus systems with $Q = 10000,\ B = 200$ can be approximated by the solution of the equivalent case $Q = 1000,\ B = 20$ (or less).

The solution for the general case $P \leq 1$ can be directly obtained by the previous analysis with rescaling, that is using the value $S = S/P$ for the determination of the values p_{ij} and $C = C/P$ for the determination of c and d, and the consequent distribution γ. Eqs. (26) and (27) are unchanged.

The evaluation of T_u, that theoretically can be performed writing a system similar to (26), is enormously simplified by the following considerations. Being P the rate of service, let's define as *service slot* the time interval ($1/P$ time slots) among two consecutive services. In service slots, the average length of an active period for the source is B and the average length of an inactive period is SP. Notice that at the beginning of the high period T_u we observe an activity period whose average length is $B - 1$ service slots, because we know that, a service slot before the start of the high period, the source is active. In order to reach the lower threshold D starting from the upper threshold Q we necessitate exactly $Q - D$ empty service slots, which are separated by the insertion, after each empty service slot, of an active period with probability $1/SP$. The average number of activity periods inserted is thus $(Q - D - 1)/SP$ and the average high period length, converted in time slots, sums to:

$$T_u = \left(B - 1 + (Q - D) + (Q - D - 1)\frac{B}{SP}\right)\frac{1}{P} = \frac{B}{P} + \frac{Q - D - 1}{P - A} \qquad (28)$$

which is the equation used in the paper.

Appendix B

Let's assume, for simplicity of presentation (generalization is straightforward), that the traffic offered by a source is composed by only one class of cells (for example LP cells) or, alternatively, that, among the priority classes composing a stream, only one is subjected to cell loss probability.

Consider a generic not null set of sources $S = \{s_1, s_2, \cdots, s_n\}$. Selected $s_1, s_2 \in S$, define with $\mathcal{T}(s_1, s_2) = \{S | s_1, s_2 \in S\}$ the space of all the possible sets of sources including s_1, s_2. Suppose furthermore that a generic algorithm for the evaluation of the cell loss probability bound of a generic source s, given a background traffic S, with $s \in S$, is defined by the function $\psi : (s, S) \rightarrow \Re$, and let's indicate with

$$\begin{aligned} l_1(S) &= \psi(s_1, S) \\ l_2(S) &= \psi(s_2, S) \end{aligned} \qquad (29)$$

the cell loss probability bound for the sources s_1 and s_2 according to a given background traffic $S \in \mathcal{T}(s_1, s_2)$.

Generally will be $l_1(S) \leq l_2(S)$ for some $S \in \mathcal{T}(s_1, s_2)$ and $l_1(S) > l_2$ for the other S. If

$$\forall S \in \mathcal{T}(s_1, s_2), \quad l_1(S) \leq l_2(S) \qquad (30)$$

we say that $s_1 \leq_\psi s_2$

Notice that the relation $\leq_\psi$, according to the bound algorithm ψ, defines a partial order on any given set of sources S. Once this relation is found, it can be actively used in the CAC algorithm as follows.

Define with $S(t)$ the set of sources admitted to a given network node in the generic time t, and define with $S_{sup}(t) = \sup(S(t), \leq_\psi)$ the extreme superior of the set $S(t)$. Define with g the maximum bound for the cell loss probability for every source in $S(t)$. Suppose now that a new source s_* requests the admission to the network node: generally the source s_* can be accepted only if

$$\forall s \in S(t) \cup s_*, \quad \psi(s, S(t) \cup s_*) \leq g \qquad (31)$$

but, according to the defined relation $\leq_\psi$, it is immediate to understand that the acceptance condition (31) can be reduced to the restricted condition

$$\forall s \in S_{sup}(t) \cup s_*, \quad \psi(s, S(t) \cup s_*) \leq g \qquad (32)$$

Owing to the last relation, we can practically reduce the number of sources to monitor during the acceptance of a new source, and such reduction will be greater as more the relation $\leq_\psi$ is near to a total order (in this case it is necessary just to monitor one background source, thus leading to a CAC algorithm independent on the number of sources admitted).

Of course a long term control is necessary for updating the set $S_{sup}(t)$ at the varying of the set $S(t)$ with every set-up and tear-down of the connections, but in this case the time constraints are not so stringent.

A non-trivial problem is the determination of such relation. In some cases, as the following example shows, the relation is quite straightforward, but in other cases (like the best bound algorithm), it seems difficult to obtain. Furthermore if the relation is weak, the reduction of sources to monitor can be relatively small without an effective practical advantage.

We think that a few time should be spent on such problem, particularly addressing the following two topics: first of all find the relation (30) for a practical bound algorithm; then try to augment its strength in order to get a substantial gain from the strong reduction of sources to monitor. The rationale about this second consideration is based on the fact that (30) is defined for all the space $\mathcal{T}(s1, s2)$, thus, also in the case the relation is found, very likely it won't be a strong relation. An alternative practical approach could consists in the definition of a set of relations (30), but defined on different subsets of $\mathcal{T}(s1, s2)$, so that, hopefully, the strength and efficiency of the relations will be increased. A long term control in the CAC algorithm will take care of the selection and switching of the relations according to the particular set $S(t)$ of sources admitted to a network node in the time t.

We conclude this appendix with a very simple example which shows how this relation can be found for a particular bound algorithm, taken by the paper [Tur92]. In this work Turner describes a method for the evaluation of the burst loss probability for bursty source in the hypothesis that a whole burst is not accepted in a network node in case the bandwidth required by the burst, added to the background allocated bursts, overflows the channel capacity.

Let's consider, reviewing the contents of [Tur92] with adapted symbology, a generic source i described by the parameters p_i, the probability that the source is active, and the integer $P_i \leq M$, the rate of transmission during the activity period referred to the maximum link capacity M. Being x_i the r.v. representing the state of the source, with distribution $P\{x_i = 0\} = 1 - p_i$, $P\{x_i = P_i\} = p_i$, let's define, for a set $(s_1, s_2, \cdots, s_n)$ of sources, $X = \sum_{i=1}^{n} x_i$, $\tilde{X} = \sum_{i=3}^{n} x_i$, $P\{\tilde{X} = i\} = \tilde{H}_i$ and $\tilde{f}(x) = P\{\tilde{X} > M - x\}$.

The burst loss probability of a generic source is given, according to the buffer management method described in [Tur92], by $l_i = P\{X - x_i > M - P_i\}$. Consider now two particular sources s_1, s_2. Indicating with $\bar{p}_i = 1 - p_i$, it is immediately

$$\begin{aligned} l_1 &= P\{X - x_1 > M - P_1\} = P\{\tilde{X} + x_2 > M - P_1\} = \\ &= p_2 P\{\tilde{X} > M - P_1 - P_2\} + \bar{p}_2 P\{\tilde{X} > M - P_1\} = \\ &= p_2 \tilde{f}(P_1 + P_2) + \bar{p}_2 \tilde{f}(P_1) \end{aligned} \tag{33}$$

and analogue for l_2. Impose now that $l_1 \geq l_2$. Then

$$p_2 \tilde{f}(P_1 + P_2) + \bar{p}_2 \tilde{f}(P_1) \geq p_1 \tilde{f}(P_1 + P_2) + \bar{p}_1 \tilde{f}(P_2) \rightarrow$$

$$\rightarrow \bar{p}_2(\tilde{f}(P_1+P_2)-\tilde{f}(P_1)) \leq \bar{p}_1(\tilde{f}(P_1+P_2)-\tilde{f}(P_2)) \rightarrow$$

$$\rightarrow \bar{p}_2 \sum_{M-P_1-P_2+1}^{M-P_1} \tilde{H}_i \leq \bar{p}_1 \sum_{M-P_1-P_2+1}^{M-P_2} \tilde{H}_i \qquad (34)$$

Relation (34), that is $l_1 \geq l_2$, is always satisfied for any background traffic $\tilde{X}$ and thus for any $X \in \mathcal{T}(s_1, s_2)$, if

$$\begin{cases} p_1 \leq p_2 \\ P_1 \geq P_2 \end{cases} \qquad (35)$$

which defines the partial ordering relation among the sources we were looking for.

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Glossary

QOS : Quality Of Service

GOS : Grade Of Service

CAC : Connection Admission Control

HP : High Priority (stream, traffic, cells)

LP : Low Priority (stream, traffic, cells)

MTG : Minimum Throughput Guaranteed (network management)

EMTG : Enhanced Minimum Throughput Guaranteed (network management)

CSM : Combined Statistical Multiplexing (network management)

Most common notations

P : Peak rate, upper shaper capacity (normalized; integer in section 4)

A : Average rate (normalized)

B : average burst length (ATM cells)

S : average silence length (slots)

C : lower shaper capacity (normalized; integer in section 4)

Q : shaper queue size (ATM cells)

D : lower shaper threshold (ATM cells)

T_u : average shaper high period length (slots)

T_l : average shaper lower period length (slots)

f : fraction of LP cells leaving the shaper

U : high period probability

L : active lower period probability

Z : inactive lower period probability

g_{HP} : GOS for HP traffic (loss probability)

g_{LP} : GOS for LP traffic (loss probability)

M : granularity of discretization for the bound evaluation

K_{ij} : coefficient of the matrix for the best bound evaluation

H_b : coefficient of the vector for the monodimensional bound evaluation