Hierarchical CAC and routing in IP/ATM networking

Raffaele Bolla°, Franco Davoli°, Mario Marchese*, Marco Perrando°

° Department of Communications, Computer and Systems Science (DIST) – University of Genoa
* Italian Consortium for Telecommunications (CNIT) – University of Genoa Research Unit
Via Opera Pia 13, I-16145 Genova, Italy

E-mail: (lelus, franco, perr)@dist.unige.it, mario.marchese@cnit.it

Abstract- In the context of an ATM access and transport network, carrying guaranteed quality (CBR, rt-VBR) services as well as IP datagrams (as ABR or UBR traffic classes), we consider the joint problems of Call Admission Control (CAC), bandwidth allocation and routing. The presence of distributed access multiplexers is assumed, which are both geographically dispersed (e.g., at the user premises) and hierarchically structured. Such multiplexers are intelligent devices with decision making capabilities that operate jointly, in order to make the best possible use of the transport capacity of the access network and to maintain the Quality of Service (QoS) requirements of different users and service classes. Following the physical system organization, a hierarchical control structure is defined, where the admission of calls for real-time traffic classes (or different users) is performed by independent controllers; the latter are parametrized by the bandwidths allocated by a common agent, playing the role of a coordinator in the hierarchical control scheme. This decision maker aims at minimizing a global cost that captures QoS requirements both at the call-level (call blocking probability) for QoS-aware, connection-oriented services and at the cell-level (cell loss probability) for connectionless, best-effort, ones. The control architecture also reflects the multilayer hierarchy introduced by the presence of multiple teletraffic time scales, by essentially decoupling the above problem from that of ensuring QoS at the cell-level for services of the first type. We derive the optimal parameters setting by means of a mathematical programming procedure, and show an example for an ATM-PON (Passive Optical Network). Then, the same structure is applied to link multiplexers of the transport network nodes, which are supposed to posses both ATM and IP switching/routing capabilities, and a possible organization of the routing strategies at both ATM and IP levels is outlined.

I. INTRODUCTION

Key factors for multimedia telecommunication services distribution to business and residential customers on a large scale are the presence of a broadband access network [1-5] and of an integrated high-speed transport. In general, a similar framework can be recognized in several access networks, where distributed multiplexing units are present, which are both geographically dispersed (e.g., at the user premises) and hierarchically structured. With current computing technology, such multiplexers can become intelligent devices with decision making capabilities, which operate jointly in order to make the best possible use of the transport capacity of the access network, while at the same time maintaining the Quality of Service (QoS) requirements of different users and service classes. In this respect, a first distinction can be made between QoS-aware, connection-oriented services, and connectionless, best-effort, ones. At the same time, whenever cell-level statistical multiplexing is present for services of the first type, the control architecture should also reflect the multilayer hierarchy introduced by the presence of multiple teletraffic time scales, by coping with the problem of ensuring QoS at the cell-level also in this case (cell loss probability and, possibly, cell transfer delay). A similar structure for service and time-scale decoupling can be recognized also at link multiplexers within a switching node in the transport network, be it based on IP only or on IP-over-ATM.

These features suggest the application of dynamic hierarchical control structures, similarly as in the context of hybrid TDM [6], where the admission of calls for real-time traffic classes (or different users) is performed by independent controllers; the latter are parametrized by the bandwidths allocated by a common agent, playing the role of a coordinator in the hierarchical control scheme. This agent’s decisions are based on the (constrained) minimization of a global cost, whose form tries to capture QoS requirements, both at the call-level (call blocking probability) for QoS-aware, connection-oriented services, and at the cell-level (cell loss probability) for connectionless, best-effort, ones. At the same time, whenever statistical multiplexing at the cell level is present for services of the first type, the control architecture should also reflect the multilayer hierarchy introduced by the presence of multiple teletraffic time scales, by coping with the problem of ensuring QoS at the cell-level also in this case (cell loss probability and, possibly, cell transfer delay).

In this paper, we consider the general structure of such multiplexers, loaded with different service classes, with the goal of defining management and control laws and algorithms of the type mentioned above in this specific context. To this aim, we suppose to operate in an ATM context, and to have a mix of Continuous Bit Rate (CBR) and Variable Bit Rate (VBR) service classes, which share a common bandwidth with Available Bit Rate (ABR) or even Unspecified Bit Rate (UBR) ones; the latter are generated by a flow of data packets at the network layer (e.g., IP datagrams). Given the
bandwidth allocated to the CBR and VBR sources, we derive the region in the space of connections of the various classes of this type, within which cell-level QoS is satisfied (Service Separation with Dynamic Partitions [7]): this concept essentially decouples the problem of cell- and call-level QoS. Above this region, the dynamics of connection-oriented, QoS-aware traffic will be described by Markov chains at the call level, and Call Admission Control (CAC) strategies will be defined. On the other hand, a self-similar traffic model can be used to characterize connectionless, best-effort, traffic (e.g., "short-lived" IP flows), which may be given the "residual" bandwidth over the link, with a constraint on the minimal allocation (in order to avoid TCP congestion control to drastically reduce throughput). We use the above models to also construct a coordination structure, which is based on a hierarchical decomposition between "local" admission controllers and a link bandwidth allocation controller that plays the role of the coordinator. In the access area, this hierarchical structure may be applied to realize a decomposition among both service classes and users. At the transport nodes, only decomposition and coordination among the services for each outgoing link is present.

After the introduction and analysis of this CAC and bandwidth allocation paradigm, we also attempt to define routing strategies, both for connection-oriented, QoS-aware traffic at the ATM call-level (including "long-lived" IP flows) and for connectionless "short-lived" IP flows. To do so, we suppose switching nodes to possess both IP and ATM switching capabilities; ATM VP/VC routing is used for traffic of the first type, whereas datagrams of the second type are segmented into cells and transferred over ATM VPs between IP routers, where the datagram is reconstructed and routed accordingly.

The report is organized as follows. We outline our model and the cell-level requirements in the next section. The third section is dedicated to the admission control level, and to the definition of the cost function of the bandwidth allocation level. Section 4 reports some numerical results; Section 5 contains the initial proposals to deal with the routing problem and Section 6 contains the conclusion.

II. TRAFFIC MODELS AND SERVICE SEPARATION

We suppose the traffic in the network to be categorized into H+1 service classes. The first H contain either CBR or bursty VBR (on-off) sources, characterized by statistical parameters like peak rate, average transmission rate and average burst length, as well as by QoS requirements, like cell loss probability and cell delay. We indicate with \( B^{(h)} \) the average burst length, \( P^{(h)} \) the peak bit rate, the average bit rate and the burstiness, respectively, of a source of the \( h \)-th class, \( h=1,...,H \) (obviously, CBR sources are included in this description, with \( B^{(h)}=1 \)). We let \( \lambda^{(h)} \) and \( \mu^{(h)} \) represent the average arrival rate and the average duration of connections of class \( h \), respectively, and \( P^{(h)} = \frac{\lambda^{(h)}}{\mu^{(h)}} \). The channel time is slotted, and a slot carries an ATM cell (424 bits).

Moreover, we suppose to have an asynchronous packet flow, which represents the traffic generated by connectionless, best-effort, services; this flow is supposed to originate from the superposition of a number of on-off sources, whose sojourn time \( Y \) (expressed in "source time units", to be defined below) in the active state follows a Pareto distribution, i.e.,

\[
\Pr\{Y = y\} = cy^{-(\alpha+1)}, \quad 1 < \alpha < 2, \quad y \geq 1
\]  

where \( c \) is the normalization constant and \( \alpha \) is a parameter. In our setting, the "source time unit" \( T \) is defined as the packetization delay of the above-defined sources, i.e., the time to generate a cell at the source speed.

The Pareto distribution is well known for its "heavy-tail" property, and has actually been used to model self-similar traffic: more specifically, the aggregation of a large number of sources of the above mentioned type has been shown to give rise to self-similar traffic [8]. The packets (after segmentation into ATM cells) receive a variable rate service (ABR or even UBR). We model the queueing of cells generated in this way as a synchronous Z/D/C_{asy}/Q^{(H+1)} system, where Z is the aggregate self-similar process (discretized over a "source time unit"), \( C_{asy} \) the capacity available for the connectionless traffic, and \( Q^{(H+1)} \) is the dimension of the buffer dedicated to it. In the following, the upper bound on the overflow probability derived in [8] will be used.

At each ATM multiplexer, traffic class \( h, h=1,...,H \), is assigned a separate buffer of length \( Q^{(h)} \) cells, whose output is statistically multiplexed on the outgoing link by a scheduler, which substantially divides the part of channel capacity assigned to connection-oriented traffic \( C_{co} \) bits/s into "virtual" partitions \( C^{(h)} \) among the classes, whose sum amounts to \( C_{co} \). The partitions may be maintained by serving the buffer in a Weighted Round Robin fashion, or by using a technique like Generalized Processor Sharing [9]. The overall queueing system at the cell level is depicted in Fig. 1.

Connection requests are also processed on a per-class basis.

Fig. 1. Cell-level multiplexing under Service Separation.
Given a model for the traffic sources of a class, the cell-level performance requirements (e.g., in terms of average cell loss and delayed cell rate) allow to define a region in call-space (which will be referred to as "Feasibility Region" or FR), where they are certainly satisfied. This region corresponds to the CAC method named “service separation with dynamic partitions” in [7, p. 147]. In a network, one such region can be associated with each link. 

Clearly, the points on the boundary of the FR correspond to the maximum numbers of Virtual Circuit (VC) connections \([N_{\text{max}}^{(1)}, \ldots, N_{\text{max}}^{(H)}]\) that are compatible with the given cell-level QoS constraints. We can associate each \(N_{\text{max}}^{(h)}\) with the minimum amount of bandwidth \(C_{\text{min}}^{(h)}\) that is necessary to support that number of connections with the given QoS guarantees. 

The computation of the FR has been the object of several studies and can be effected in different ways, either by analysis, given a model of the traffic sources, or by simulation. Using any approach based on equivalent bandwidth (involving, in our Service Separation context, only homogeneous sources) yields a straightforward boundary of the FR. In any case, it is worth noting that, in the context of the methods to be considered in the next section, the FR itself will be just a tool to describe the CAC schemes. The specific technique to ensure QoS satisfaction at the cell-level might be changed (always within the framework of Service Separation), without affecting the access control general procedure. 

However, to fix ideas, we refer here for the computation of the FR to the model we have used in [10] and in previous work, where a maximum threshold value is set for the average cell loss rate \((P_{\text{loss}}^{(h)}(n,C^{(h)}))\) and for the average delayed cell rate \((P_{\text{delay}}^{(h)}(n,C^{(h)}))\), with \(n\) calls in the active state out of \(N_{\text{max}}^{(h)}\) accepted calls and a bandwidth \(C^{(h)}\) assigned to traffic class \(h\); more specifically 

\[
\sum_{n=1}^{N_{\text{max}}^{(h)}} P_{\text{loss}}^{(h)}(n,C^{(h)}) v_{n,N_{\text{max}}^{(h)}} \leq \varepsilon^{(h)} 
\]

\[
\sum_{n=1}^{N_{\text{max}}^{(h)}} P_{\text{delay}}^{(h)}(n,C^{(h)}) v_{n,N_{\text{max}}^{(h)}} \leq \delta^{(h)} 
\]

where \(\varepsilon^{(h)}\) is an upper limit on the average value of the cell loss rate and \(\delta^{(h)}\) has the same meaning for the average value of cells that suffer a delay longer than a fixed upper bound \((\Delta^{(h)}\) in Section 4). The quantity \(v_{n,N_{\text{max}}^{(h)}}\) is the probability of having \(n\) active connections of class \(h\), with \(N_{\text{max}}^{(h)}\) accepted connections of the same class, which is given by a binomial distribution; an Interrupted Bernoulli Process (IBP) [11] is used to model the state of a call, from which \(P_{\text{loss}}^{(h)}(n,C^{(h)})\) and \(P_{\text{delay}}^{(h)}(n,C^{(h)})\) are derived. \(N_{\text{max}}^{(h)}\) and \(C_{\text{min}}^{(h)}\) are derived. \(N_{\text{max}}^{(h)}\) and \(C_{\text{min}}^{(h)}\) are derived. \(N_{\text{max}}^{(h)}\) and \(C_{\text{min}}^{(h)}\) are derived.

can be computed from (2) and (3). As an alternative approach, any one based on equivalent bandwidth (e.g., [12]) could be used, yielding similar results (see [10]). Obviously, in the present case, the FR is parametrized by the capacity \(C_{\text{co}}\) actually assigned to the connection-oriented traffic. 

We let 

\[
N_{\text{A}}(k) = \text{col}[N_{\text{max}}^{(h)}(k), h=1,\ldots,H] 
\]

where \(N_{\text{max}}^{(h)}(k)\) is the number of connections in progress at the generic instant (slot) \(k\) for class \(h\). The vector in (4) represents the state of the system at instant \(k\) (the VC-profile in [7]). 

### III. CALL ADMISSION CONTROL AND CAPACITY ALLOCATION 

At this point, we are ready to consider QoS performance measures and constraints at the call level, having essentially decoupled this problem from the lower level one. As in our previous work in the ATM context [10], we have chosen to adopt admission control policies for our connection-oriented traffic that belong to the class of Complete Partitioning (CP) ones; such policies define a “rectangular” sub-region within the FR, by dividing the available capacity among traffic classes in a “static” way. In other words, given a point \(N_{\text{max}}^{*}\) of the rectangular acceptance region [10]; in the present case, an additional constraint may be added by the requirements of the connectionless traffic. Actually, even though this traffic will be mostly best-effort, an upper bound on the cell loss probability may be considered in the bandwidth allocation, in order to avoid some undesired effects (e.g., too many TCP retransmissions). 

Before considering the choice of \(N_{\text{max}}^{*}\), we may note that the connectionless traffic can always be allocated all the bandwidth unused by the connection-oriented classes, in a way analogous to movable boundary schemes in TDM networks [6]. We can do this through the knowledge of the VC-profile \(N_{\text{A}}(k)\), by calculating (with any valid method, as mentioned in the previous section) the minimum bandwidth \(C_{\text{min}}^{(h)}(k)\) that is necessary to ensure cell-level QoS to the \(N_{\text{A}}^{(h)}(k)\) connections of class \(h\) in progress. 

Letting 

\[
C_{\text{min}}^{(h)}(k) = \sum_{h=1}^{H} C_{\text{min}}^{(h)}(k) 
\]

we can assign (through the scheduler) the connectionless
traffic the residual bandwidth

\[ R(k) = C - C_{\min}(k) \]

(7)

where \( C \) is the total transfer capacity of the link; this assignment can last until either a new call is accepted or a connection terminates. It can be noted that, given the connection-oriented traffic characteristics and the bandwidth \( C_{\text{co}} \) globally assigned to it, the corresponding FR can be constructed off-line, and the residual bandwidth \( R(k) \) can be determined for each of its points.

Even with the above described assignment, it is however clear that different “static” partitions (determined by the value of \( C_{\text{co}} \)) may be necessary to combine the requirements of the various classes. These will be determined by the optimization procedure that leads to the choice of \( N_{\text{opt}}^{*} \). By first setting \( C_{\text{co}} = C \) and then gradually decreasing (in discrete steps) the bandwidth globally allocated to connection-oriented services, we obtain a family of FRs, with decreasing areas. For each corresponding boundary \( S(C_{\text{co}}) \), we can compute the point \( N_{\text{max}}^{*}(C_{\text{co}}) \) that minimizes a cost function involving the stationary call blocking probabilities \( p_{\text{block}}^{(h)}(N_{\text{max}}^{*}) \); owing to the service separation assumption, these probabilities are given simply by the Erlang B formula [7], where \( N_{\text{max}}^{*} \) is the number of servers. Among the various possibilities, we have chosen the following cost function (named Balanced Erlang Scheme (BES) in [10], which tends asymptotically with \( Q(H+1) \) [8]. We define

\[ P_{\text{loss}}(R_{s}) = \begin{cases} \frac{c\lambda}{(\alpha-1)(R_{s} - a\lambda)}(Q^{(H+1)})^{-\alpha+1} & \text{if } R_{s} > a\lambda \\ 1 & \text{otherwise} \end{cases} \]

(12)

where \( R_{s} \) represents the residual capacity (expressed in slots per “source time unit”) corresponding to the value \( s \) of the VC-profile \( N_{A}(k) \) (the system’s state) and \( S_{R}(N_{\text{max}}^{*}) \) is the “rectangular” region within the FR, whose vertex on the FR’s boundary is \( N_{\text{max}}^{*} \). With the given CAC rule, the state of the system \( N_{A}(k) \) is a multidimensional Markov chain over \( S_{R}(N_{\text{max}}^{*}) \), whose stationary distribution has been indicated by \( \{\pi(s)\} \).

We can now turn to the determination of the “globally” optimal point \( N_{\text{max}}^{*} \) that minimizes a global cost function, to be defined. Among a few possible choices (see [14] for a different one), we have found that the following allows to obtain the most straightforward control over the desired tradeoff between the two traffic types; specifically, we have chosen

\[ J_{1}(N_{\text{max}}) = \max_{h} \left\{ J_{1}(h) \right\} \]

(8)

If, for each service class, we also want to take into account a constraint on the blocking probability, say

\[ p_{\text{block}}^{(h)}(N_{\text{max}}^{*}) \leq \Gamma^{(h)}, \quad h = 1, \ldots, H \]

(9)

we can modify the cost function (8) as follows. Let \( \bar{N} \) be the point whose coordinates satisfy relations (9) with equality; then we can consider

\[ J_{1}(N_{\text{max}}) = \begin{cases} \tau J_{1}(N_{\text{max}}^{*}), & N_{\text{max}} < \bar{N} \\ J_{1}(N_{\text{max}}), & N_{\text{max}} \geq \bar{N} \end{cases} \]

(10)

where \( \tau \) is a constant (the larger \( \tau \), the higher the penalty for not matching the constraint).

Given \( N_{\text{max}}^{*}(C_{\text{co}}) = \arg \min J_{1}(N_{\text{max}}^{*}(C_{\text{co}})) \), we consider, for each value of \( C_{\text{co}} \), the following cost function for the asynchronous, connectionless traffic:

\[ J_{2}(N_{\text{max}}^{*}) = \sigma \pi_{\text{loss}}^{(H+1)}(N_{\text{max}}^{*}) = \sigma \sum_{s \in \mathbb{S}} P_{\text{loss}}(R_{s}) \pi(s) \]

(11)

Essentially, in determining both \( P_{\text{loss}} \) and the stationary distribution \( \{\pi(s)\} \), we use the same decoupling procedure as in [6] and [13]: \( \{\pi(s)\} \) is readily computed from the transition probability of the multidimensional birth-death process describing the call dynamics [6].

Here, we may want to set a desired level of (rather than a strict constraint on) the maximum cell loss that is tolerable for the connectionless service, i.e.,

\[ \frac{P_{\text{loss}}^{(H+1)}(N_{\text{max}}^{*})}{\sigma} \leq e^{(H+1)} \]

(14)
The above form of the “global” cost tends to equalize the two values at the minimum point; in any case, their relative importance is weighted by the parameter $\sigma$, whose choice can take into account the “desired” upper bound in (14).

All calculations related to the FRs only depend on the cell-level parameters, i.e., on the statistical characteristics of the sources, which determine their categorization into specific classes, but not on the call-level ones (e.g., offered load); for a given classification, all the FRs can be computed off-line.

The same would be possible for the optimization we have described, if the offered load of each traffic class remained constant: actually, this is the situation we have considered in the numerical examples to be discussed in the next section. However, slow variations in the offered load of some class could be followed by an on-line application of the algorithm, possibly by adopting some simplified numerical procedure, along the same lines as in [6].

It is also worth noting that the case considered in this paper is limited to the inter-class bandwidth allocation (e.g., within the premises of a single user); our procedure can be extended to the multi-user case, and can distinguish explicitly the downlink and uplink directions. This requires a further (third) level of coordination, which allocates the corresponding bandwidth shares, according to a new global cost function.

This problem has been touched in [14], with respect to an ATM-Passive Optical Network (ATM-PON) model [5] and is also the subject of further investigation.

IV. NUMERICAL RESULTS

We present now some numerical results regarding the above described optimization and some evaluations of the performance indices under different traffic loads. The data values adopted are taken from the ATM-PON application that is examined in [14].

$H=2; \quad C=150$ Mbits/s

**Connection-oriented traffic (VBR) - Cell level parameters**

(for the definition of the FR)

<table>
<thead>
<tr>
<th>Class</th>
<th>Class 1</th>
<th>Class 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Peak bandwidth</td>
<td>5 Mbits/s</td>
<td>2 Mbits/s</td>
</tr>
<tr>
<td>Average bandwidth</td>
<td>0.3 Mbits/s</td>
<td>1 Mbit/s</td>
</tr>
<tr>
<td>Delay constraint</td>
<td>2200 slots</td>
<td>2200 slots</td>
</tr>
<tr>
<td>$P_{\text{loss}}$ upper bound</td>
<td>$10^{-8}$</td>
<td>$10^{-5}$</td>
</tr>
<tr>
<td>$P_{\text{delay}}$ upper bound</td>
<td>$10^{-5}$</td>
<td>$10^{-5}$</td>
</tr>
<tr>
<td>Buffer length</td>
<td>10 cells</td>
<td>10 cells</td>
</tr>
</tbody>
</table>

Fig. 2 shows the sequence of FRs resulting from different values of the capacity $C_{\text{co}}$ dedicated to the connection-oriented traffic, along with the “local” optimal points $N_{\text{max}}^*$, the global optimum $N_{\text{max}}^\text{opt}$, the corresponding Complete Partitioning region and the region where constraints (9) are satisfied. The call level parameter values used in this case are $\rho^{(1)}=10$ Erlangs and $\rho^{(2)}=18$ Erlangs; moreover, $\lambda=0.2$ bursts/[source time unit], $T=(424 \text{ bits})/(2.048 \text{ Mbit/s})=0.207$ ms, $\alpha=1.4$, $\tau=10$, $\theta^{(1)}=\theta^{(2)}=1$, $\Gamma^{(1)}=\Gamma^{(2)}=0.1$, $Q^{(1)}=1000$, and $\sigma=10^4$. The latter parameter value has been chosen in order to make a blocking probability of 0.1 comparable to a desired buffer overflow level $e^{-1}=10^{-3}$. With these figures for the traffic intensities, the bandwidth is partitioned, on the average, so that about 85 Mbits/s are assigned to the connection-oriented traffic and 65 Mbits/s to the connectionless one.

We now present some results obtained by varying the traffic intensities, and then computing the ensuing optimal assignments and the corresponding overall blocking and buffer overflow probabilities. The load values in the following two figures correspond to a reference one, given by $\rho^{(1)}=7$ Erlangs, $\rho^{(2)}=10$ Erlangs, $\lambda=0.2$ bursts/[source time unit], multiplied by the number on the x-axis. The following configurations of weight ($\sigma$) and constraints apply:

A) $\sigma=0$, $\Gamma^{(1)}=\Gamma^{(2)}=1$;
B) $\sigma=5 \cdot 10^{-3}$, $\Gamma^{(1)}=\Gamma^{(2)}=1$;
C) $\sigma=5 \cdot 10^{-5}$, $\Gamma^{(1)}=\Gamma^{(2)}=0.1$.

Case (A) has no constraints and no weight on the buffer overflow probability. Its blocking (Fig. 3) is the lowest and its overflow (Fig. 4) the highest one, for all load values (no control is exerted, except for the attempt to equalize blocking among the two VBR classes that is inherent in the BES criterion). At the other extreme, case (B) shows the effect of weighting the loss of cells from the best-effort flow, but still without explicitly constraining the blocking probability. The more “balanced” situation is shown in case (C), where both constraints and weight play a role. In this situation, the blocking satisfies the constraints up to the load value 3, after which the link is heavily saturated.
V. COMBINING CAC AND ROUTING

The same technique of bandwidth allocation among the different classes of traffic and the asynchronous data flow may be applied, at the higher hierarchical control level, even at routing nodes of a transport network; at the lower control level, the nodes use some kind of algorithm for the admission control and routing of the calls and the routing of the data flows. We have taken into account the Distributed Least Congested Path (DLCP) algorithm [15, 16] for routing of incoming calls at set-up time, and some different algorithms for the routing of asynchronous traffic.

The DLCP routing works as follows. At connection set-up, a call request packet is forwarded, hop by hop, from node to node. At each node traversed, the call request of a certain class is first assigned a subset of outgoing links towards the destination, upon which the resources necessary to maintain the required level of QoS are available; then, a routing decision is taken among these links, by choosing the least “cost” one, according to the DLCP criterion.

The cost of the link is formed by two different terms: the first states the actual (local) link congestion, the second takes into account, in same way, an aggregate (global) congestion measure of the successors of the link. A node \( t \) chooses the link to which to forward a class \( h \) connection request, by minimizing (over the subset of outgoing links \( j \) that lead toward the destination required) the quantity

\[
c_{ij}^{(h)}(k, s) = c_{ij,L}^{(h)}(k) + \xi_j c_{ij}^{(h)}(s)
\]

where \( c_{ij,L}^{(h)}(k) \) is the actual link congestion and \( c_{ij}^{(h)}(s) \) is the aggregate measure of node \( j \) and its successors; \( \xi_j \in [0, 1] \) is a weighting coefficient, used to balance the influence of the local and global cost. For the local congestion of link \( ij \), we choose the following form:

\[
c_{ij,L}^{(h)}(k) = \frac{1}{N_{ij,A}^{(h)}(k) - N_{ij,j}^{(h)}(k)}
\]

where \( N_{ij,j}^{(h)}(k) \) and \( N_{ij,A}^{(h)}(k) \) correspond to the same quantities defined in sections II and III above, indexed by the link indices, having dropped the * symbol where present, and \( Z \) is a very large number (enough to ensure that no saturated links will be chosen if non-congested ones are available).

The aggregate cost of node \( j \) is formed by two terms

\[
c_{ij}^{(h)}(s) = c_{ij,L}^{(h)}(s) + \xi_j c_{ij,A}^{(h)}(s)
\]

where \( \xi_j \) is a weighting coefficient, \( c_{ij,L}^{(h)}(s) \) represents the average situation of the node with respect to the congestion state of its links, and \( c_{ij,A}^{(h)}(s) \) is an aggregate information on the average congestion of its adjacent nodes. More specifically, we define

\[
c_{ij,L}^{(h)}(s) = \frac{1}{L_j} \sum_{k \in \text{Succ}(j)} c_{jk,L}^{(h)}(s)
\]

\[
c_{ij,A}^{(h)}(s) = \frac{1}{L_j} \sum_{k \in \text{Succ}(j)} c_{kj}^{(h)}(s)
\]

\( \text{Succ}(j) \) being the set of nodes that are successors of node \( j \), and \( L_j \) its cardinality.

In order to avoid highly oscillatory behaviors due to the routing of data flows, we choose to adopt for them an
algorithm that works on the mean residual capacity of each link left by the calls, instead of the instantaneous residual capacity. By the way, the routing strategy must be reapplied whenever the mean residual bandwidth changes. This occurs either when the traffic intensity of the calls varies, or the bandwidth allocation changes the value $N_{ij,max}^{(h)}(k)$. Within this context there are still many ways to route data flows. One possibility is to apply a distributed version of Dijkstra or Belmann-Ford algorithms, while another one is to divide the flows and route the single fragments on different links, in order to maximize the utilization of the residual capacity, but choosing only among the links the lead the flow toward its destination in a number of hops that stands between the minimum and the minimum plus an offset, in order to avoid paths crossing too many hops, and circular routing.

This routing technique works together with the CAC and bandwidth allocation, by means of the term $N_{ij,max}^{(h)}(k)$ that is chosen by the latter according to the previously stated scheme: $N_{ij,max}^{(h)}(k) = N_{ij,max}^{(h),opt}$, adopting, for example, the constrained BES algorithm. It must be noted that now the traffic intensities (both of calls and data), viewed at the input of the links used by the algorithm, are strictly dependent from the routing decisions. In practice this means that they must be estimated over a time window, and the corresponding partitions must be recomputed when the estimates fall out of a range around the value that was previously used for the bandwidth allocation.

This effect can give rise to an oscillatory behaviour of the system. An investigation by simulation will clarify this point.

VI. CONCLUSIONS

We have introduced and analyzed a control scheme suited for multimedia access multiplexers, operating with ATM and IP over ATM service classes. The presence of multiple classes with possibly conflicting requirements has been taken into account and a joint strategy for resource (bandwidth) allocation and call admission control has been defined. The hierarchical control approach we have followed decouples cell- and call-level requirements and makes use of service separation with optimized and adaptive bandwidth partitioning. In particular, the presence of connectionless, best-effort, traffic with self-similar properties has been explicitly incorporated in the model. The situation considered here can model a single multiplexing unit, which is assigned a portion of bandwidth in the access network, but it can be extended to cover multiple units sharing a single link, and to include uplink and downlink asymmetric bandwidth assignments. A further extension has been considered, as an initial stage of future work, to include link multiplexers at switching nodes within the network, along with routing strategies at both IP and ATM levels.

ACKNOWLEDGMENT

This work was supported by the Italian Ministry of the University and Scientific and Technological Research (MURST), under the national Research Program on Techniques for QoS Guarantees in Multiservice Telecommunication Networks.

REFERENCES


