On-line Bandwidth Control for Quality of Service Mapping over Satellite Independent Service Access Points

Mario Marchese, *Senior Member, IEEE*, Maurizio Mongelli, *Student Member, IEEE*
CNIT - Italian National Consortium for Telecommunications
University of Genoa
Via Opera Pia 13, 16145 Genoa, Italy
{ Mario.Marchese, Maurizio.Mongelli }@cnit.it

***Abstract*—In the *Broadband Satellite Multimedia* (BSM) architecture, developed by the *European Telecommunications Standardization Institute* (ETSI), the physical layers are isolated rom the rest by a *Satellite Independent-Service Access Point* (SI-SAP). For technological reasons, *Satellite Independent* (SI) traffic classes, requiring different QoS guarantees, must be aggregated together within the *Satellite Dependent* (SD) core and may change the encapsulation format. Moreover, SD layers must assure proper fading countermeasures to face satellite channel degradation. In this work, we investigate a novel control algorithm to tackle the envisaged problems. Simulation results validate the proposed approach.**

***Index Terms*—QoS Interworking, Equivalent Bandwidth, Measurement-based Control.**

I. INTRODUCTION

ETSI-BSM (*European Telecommunications Standardization Institute-Broadband Satellite Multimedia*) protocol stack [1] separates the layers identified as *Satellite Dependent* (SD) and *Satellite Independent* (SI). SD layers strictly depend on the physical implementations and are often covered by industrial copyright. Usually, ATM or DVB queues are deployed at the SD core. SI ayers, on the other hand, are composed of IP and upper layers (such as TCP/UDP) and, for SI QoS management, the DiffServ paradigm is used. The interface between SI and SD is defined by ETSI through SI-SAPs (*Satellite Independent – Service Access Points*).

The paper envisages the QoS management issues arising at the SI-SAP interface and proposes a novel control scheme for the optimization of the bandwidth provision at the SD layer.

The interworking between SI and SD layers reveals to be a hot topic of research. There are two main problems: **1)** the change of information unit (*encapsulation*, e.g., from IP to ATM or DVB) and **2)** the need to *aggregate traffic* (e.g., different SI DiffServ flows are conveyed together over a single SD tunnel).

The problem is how much bandwidth must to be assigned to each SD queue so that the SI (IP-based) *Service Level Agreement* (SLA) is guaranteed.

II. THE SI-SAP QOS MAPPING PROBLEM

We formalize here the mentioned issues in a proper optimization framework. The chosen mathematical framework is based on a *Stochastic Fluid Model* [2] of the traffic buffers. We consider the SI-SAP interface, composed of N queues at the SI layer and of a single queue at the SD layer.

Before detailing the optimization problem related to the SI-SAP QoS mapping, it is necessary to consider an additional problem, peculiar of satellite networks, represented by the fading degradation affecting the satellite channel and to specify the mathematical model used to describe it in this work.

Let $\theta^{SD}(t)$ be the service rate assigned to a traffic buffer at the SD layer at time t. The effect of fading can be modelled as a reduction of the bandwidth actually "seen" by the buffer. The reduction is represented by a stochastic process $\phi(t)$. At time t, the "real" service rate $\hat{\theta}^{SD}(t)$ (available for data transfer) is:

$$\hat{\theta}^{SD}(t)=\theta^{SD}(t)\cdot\phi(t);\quad \phi(t)\in[0,1] \tag{1}$$

The model may be associated with fading countermeasure mechanisms located at physical layer due to *Forward Error Correction* (FEC) mechanisms, implemented at the physical layer.

Let $\alpha_i^{SI}(t)$ be the inflow rate process entering the i-th traffic buffer at the SI layer at time t, $i=1,\dots,N$. After entering one single buffer (with service rate $\theta_i^{SI}(t)$) at the SI layer, each $\alpha_i^{SI}(t)$ process is conveyed to a single SD buffer (whose service rate is $\theta^{SD}(t)$) at the SD layer. We denote by ${}^{i}L_V^{SI}(\alpha_i^{SI}(t),\theta_i^{SI}(t))$ the *loss volume* of the i-th IP buffer

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according to the bandwidth allocation $\theta_i^{SI}(t)$. Let $\alpha^{SD}(t)$ be the inflow rate process of the buffer at the SD layer at time t. The $\alpha^{SD}(t)$ process derives from the outflow rate processes of the SI buffers (or directly from the $\alpha_i^{SI}(t)$ processes, if no buffering is applied at the SI layer). In any case, a change in the encapsulation format is applied when the $\alpha^{SD}(t)$ is produced. We denote by ${}^{i}L_V^{SD}(\alpha^{SD}(t),\theta^{SD}(t)\cdot\phi(t))$, the *loss volume* of the i-th traffic class within the SD buffer. It is a function of the following elements: the SD inflow process $\alpha^{SD}(t)$ (deriving from the aggregation of the SI inflow processes $\alpha_i^{SI}(t)$, $i=1,...,N$, and the transport technology change), the fading process $\phi(t)$ and the SD bandwidth allocation $\theta^{SD}(t)$.

IP *packet loss probability* (PLP) is the chosen performance metric.

We suppose that SI resource allocation can satisfy the required QoS at the SI level. Here, the key problem is to "equalize" the QoS measured at the SD layer in dependence of the QoS imposed at the SI layer.

The optimization problem can now be stated. **QoS Mapping Optimization** (QoSMO) **Problem**: find the optimal bandwidth allocation ${}^{Opt}\theta^{SD}(t)$, so that the following cost function $J(\cdot,\theta^{SD}(t))$ is minimized:

$$
\begin{gathered}
{}^{Opt}\theta^{SD}(t)=\arg\min_{\theta^{SD}(t)} J(\cdot,\theta^{SD}(t));\; J(\cdot,\theta^{SD}(t))=\underset{\omega\in\Theta}{E}\; L_{\Delta V}(\cdot,\theta^{SD}(t)) \\
L_{\Delta V}(\cdot,\theta^{SD}(t))=\sum_{i=1}^{N}\left[{}^{i}L_V^{SI}(\alpha_i^{SI}(t),\theta_i^{SI}(t))-{}^{i}L_V^{SD}(\alpha^{SD}(t),\theta^{SD}(t)\cdot\phi(t))\right]^2
\end{gathered}
\tag{2}
$$

We denote by ω a sample path of the system, i.e., a realization of the stochastic processes involved in the problem ($\phi(t)$, $\alpha_i^{SI}(t)$, $i=1,...,N$, $\alpha^{SD}(t)$).

Traditionally, QoS mapping operations are heuristically performed by the network manager. We now summarize two possible operative proposals, alternative to the optimization framework proposed here.

A. The CellTax approach

The first one is dedicated to IP over ATM, it is much used in industry and acts as follows. The increase in bandwidth necessary for SD layer can be foreseen by means of the so-called "cell-tax effect" due to the change of encapsulation format (e.g., when applying AAL5 encapsulation to produce the ATM frame). Since, during the generation of the SD frame, two octets (for the AAL5 overhead) need to be added to each IP packet, the number of SD cells for each IP packet is:

$$
\#ATMCells=\left\lceil\frac{DimPacket+2}{48}\right\rceil \tag{3}
$$

where *DimPacket* denotes the IP packet's size (or average packet size in case of a variable packet size at the SI layer) in bytes and 48 is the payload of an ATM cell in bytes. Hence, it is possible to compute the overall overhead due to the encapsulation format of the SD frame and the percentage bandwidth increase in the SD core, denoted in the following with *CellTax*:

$$
CellTax=\frac{\#ATMCells\cdot 53-DimPacket}{DimPacket} \tag{4}
$$

where 53 is the overall size of an ATM cell in bytes. Then, a possible forecast for the SD bandwidth allocation (called in the following **CellTax**) is ruled by the heuristic allocation law (5), each time the SI rate provision θ^{SI} changes.

$$
\theta^{SD}={}^{CellTax}\theta^{SD}=(1+CellTax)\cdot\theta^{SI} \tag{5}
$$

A similar heuristic can be employed when the transport technologies of interest are different, e.g., in case of a QoS mapping involving IPv6 over IPv4 or IPv4 over MPLS or DVB. The only concern is related to the knowledge of the change in the encapsulation format and of the SI rate provision for each traffic class.

As we will show in the experimental part of the paper, CellTax allocation underestimates the necessary SD rate provision. A deeper insight into the statistical behaviour of the flows is necessary.

B. The equivalent bandwidth approach

The second approach is based on the concept of *equivalent bandwidth.* A popular equivalent bandwidth technique (called **Equivalent Bandwidth** approach (EqB) in the following), actually applicable in this context, is ruled by (6) below.

Being: $k=1,2,...$ the time instants of the SD rate reallocations, $m_{\alpha^{SD}}(k)$ and $\sigma_{\alpha^{SD}}(k)$ the mean and the standard deviation respectively of the SD inflow process measured over the time interval $[k,k+1]$; the bandwidth provision ($\theta^{SD}(k+1)$) at the SD layer, assigned for time interval $[k+1,k+2]$, may be computed as a function of the measured statistics $m_{\alpha^{SD}}$ and $\sigma_{\alpha^{SD}}$:

$$
\theta^{SD}(k+1)=m_{\alpha^{SD}}(k)+a\cdot\sigma_{\alpha^{SD}}(k) \tag{6}
$$

where $a(\varepsilon)=\sqrt{-2\ln(\varepsilon)-\ln(2\pi)}$ and ε represents the upper bound on the allowed PLP.

C. The Sensitivity Estimation algorithm

In our proposal, we capture the temporal behaviour of each single performance level ${}^{i}L_V^{SD}(\cdot)$ through on-line measurements and perform the SD rate reallocations ccordingly to them. To reach the aim, we exploit the cost function $L_{\Delta V}(\cdot)$ derivative that can be obtained from (7):

$$\frac{\partial L_{\Delta V}(\cdot,\theta^{SD}(t))}{\partial\theta^{SD}(t)}=2\cdot\phi(t)\cdot\sum_{i=1}^{N}\frac{\partial^{i}L_{V}^{SD}(\hat{\theta}^{SD}(t))}{\partial\hat{\theta}^{SD}(t)}[{}^{i}L_{V}^{SD}(\hat{\theta}^{SD}(t))-{}^{i}L_{V}^{SI}(\theta_{i}^{SI}(t))] \quad (7)$$

Due to the application of *Infinitesimal Perturbation Analysis* (IPA), recently developed in the field of *Sensitivity Estimation* techniques for *Discrete Event Systems* (see, e.g., [2] and references therein), each $\frac{\partial^{i}L_{V}^{SD}(\hat{\theta}^{SD}(t))}{\partial\hat{\theta}^{SD}(t)}$ component can be obtained in real time only on the basis of some traffic samples acquired during the system evolution. The IPA-based derivative estimator (7) is then used to optimally tune the SD rate provision. The proposed optimization algorithm (called hereinafter **Reference Chaser Bandwidth Controller** (RCBC)) is based on the gradient method, whose descent step is ruled by (8).

$$\theta^{SD}(k+1)=\theta^{SD}(k)-\eta_k\cdot\frac{\partial L_{\Delta V}(\cdot,\theta^{SD}(k))}{\partial\theta^{SD}(k)};\ k=1,2,\ldots \quad (8)$$

III. PERFORMANCE ANALYSIS

In this section, we investigate our rate control mechanism through an ad-hoc C++ simulator. We firstly compare RCBC and CellTax with respect to the encapsulation change issue. Then, RCBC and EqB are compared with respect to the aggregation problem.

A. Encapsulation change

One IP queue and one ATM queue are taken into account for now and only the encapsulation problem is analyzed. We consider the case of a *Voice over IP* (VoIP) traffic, guaranteed at the SI layer, and carried along the ATM SD layer.

Each VoIP source is modeled as an exponentially modulated on-off process, with mean on and off times (as for the ITU P.59 recommendation) equal to 1.008 s and 1.587 s, respectively. All VoIP connections are modeled as 16.0 kbps flows voice over RTP/UDP/IP. The IP packet size is 80 bytes. The required performance objective of a VoIP flow is less than 2% of PLP. We suppose that SI bandwidth θ^{SD} has been already dimensioned in order to guarantee the required PLP constraint.

We compare CellTax with RCBC with the same size of IP and ATM buffers (fixed at 20 VoIP packets corresponding to 31 ATM cells) and by progressively increasing, from 70 to 110, the number of VoIP sources in the flow. The step is of 10 sources to stress the working conditions. The time interval between each change in the traffic flow is fixed to 3000 s to highlight the changes in the stochastic environment. Similar results can be obtained by imposing a mean interarrival time of connection requests (this was validated by simulation results not reported here). To help convergence, the CellTax heuristic (4) is used to initialize the gradient descent of RCBC. Each time the number of VoIP sources changes (every 3000 seconds), the SD bandwidth allocation of RCBC is initialized through (7), and, every 30 s, a new SD bandwidth allocation θ_k^{SD} is performed through (8). Particular attention is necessary for the gradient stepsize in (8) to avoid strong bandwidth oscillations or a low convergence. We verified through simulation inspection that the convergence speed is optimized if the gradient stepsize is tuned as $\eta_k=\frac{\partial L_{\Delta V}(\theta_k^{SD})}{\partial\theta_k^{SD}}\cdot 6\cdot 10^{-7}$.

In this way, we obtain a decreasing gradient stepsize to assure convergence, as usually required by stochastic approximating algorithms [3]. Note that $\eta_k\xrightarrow{k\to+\infty}0$ since the sequence θ_k^{SD}, $k=0,1,\ldots$, driven by (8), minimizes the cost function $L_{\Delta V}(\cdot)$ in (2).

Fig. 1 shows the performance of CellTax, which is not always able to guarantee the required PLP. On the other hand, the performance of RCBC (reported in Fig. 2) is very satisfying. The two allocation techniques are compared with respect to the bandwidth provision in Fig. 3.

CellTax clearly underestimates the required SD rate provision to correctly carry the VoIP flows. The rate provisions at both the SI and the SD layers are compared in Fig. 4 to highlight the bandwidth shift arising at the SI-SAP interface.

B. Traffic Aggregation

We consider now the case of two SI traffic buffers. The first one conveys the traffic of 30 on-off VoIP sources requiring PLP$<10^{-2}$. The second buffer is dedicated to a video service. "Jurassic Park I" video trace (taken from [4]) is used. It requires PLP$<10^{-3}$. Both the outputs of the SI buffers are conveyed towards a single queue at the SD layer. DVB encapsulation (header 4 bytes, payload 184 bytes) of the IP packets through the LLC/SNAP (overhead 8 bytes)) is implemented in this case.

In Fig. 5, we compare the SD bandwidth provision produced by RCBC and EqB. The loss probability bound ε for EqB is set to 10^{-3}, being the most stringent PLP constraint imposed at the SI level. The time interval between two consecutive SD bandwidth reallocations (denoted by "T" in Fig. 5) is 7.0 minutes for RCBC. Different values of T are used for EqB.

We must note from fig. 5 that RCBC captures the bandwidth need of the SD layer in a single reallocation step. On the other hand, the EqB produces strong oscillations in the SD rate assignment. SD buffer video PLP (whose performance threshold is 10^{-3}), averaged over the entire simulation horizon, is shown in Fig. 6. The corresponding bandwidth allocations (averaged over the simulation duration), are shown in Fig. 7. RCBC effectively finds the optimal operation point of the system, namely, the minimum SD bandwidth provision needed to track the SI PLP thresholds.

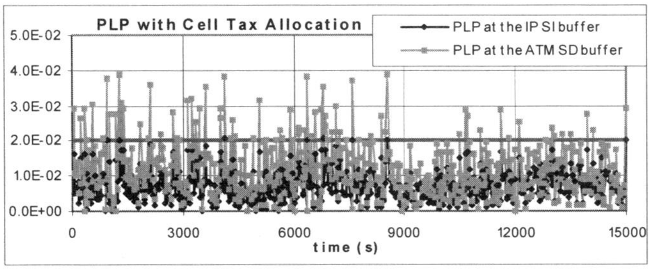

Fig. 1. Encapsulation change. PLP with CellTax.

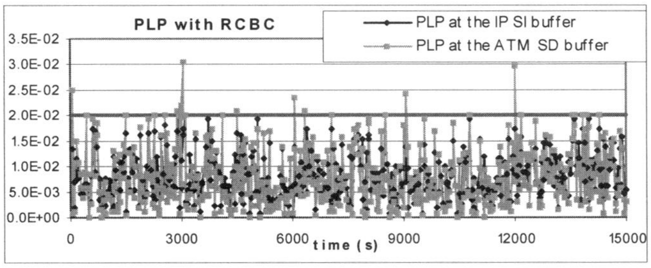

Fig. 2. Encapsulation change. PLP with RCBC.

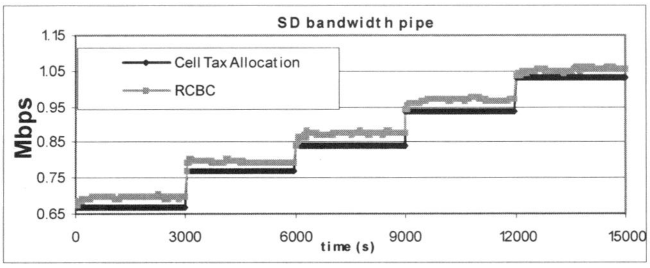

Fig. 3. Encapsulation change. SD allocation.

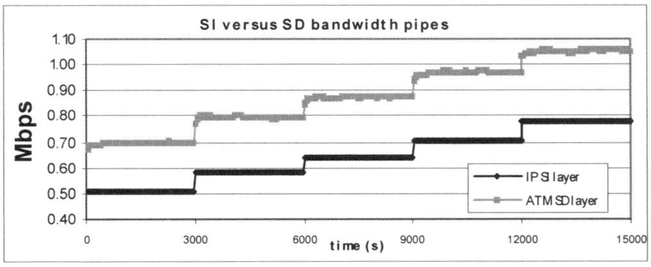

Fig. 4. Encapsulation change. SI versus SD allocations.

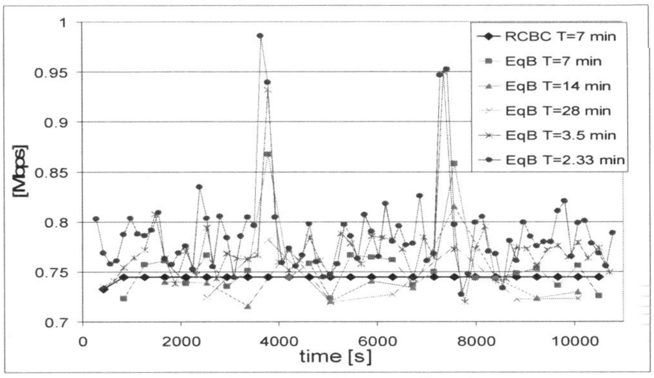

Fig. 5. Aggregation of VoIP and Video. SD allocations. RCBC vs EqB.

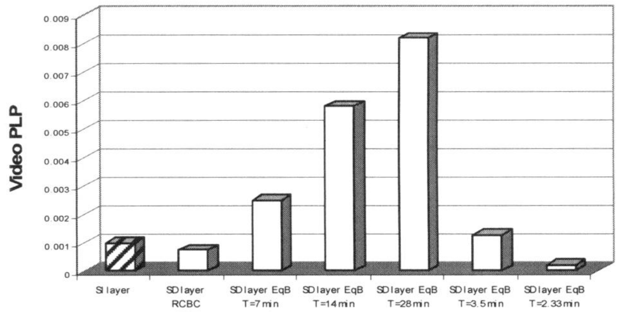

Fig. 6. Aggregation of VoIP and Video. Video PLP.

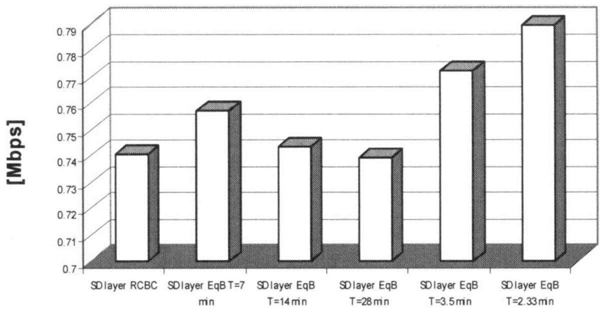

Fig. 7. Aggregation of VoIP and Video. Average SD layer provision.

IV. Conclusion and Future Work

The challenging problem of *Broadband Satellite Multimedia* architecture is the communication between *Satellite Independent* (SI) and *Satellite Dependent* (SD) layers. SD stack should offer QoS guarantees to upper layers in dependence of the *Service Level Agreements* fixed at the SI layer (in terms of *packet loss probability*). A control scheme, based on *Infinitesimal Perturbation Analysis*, has been studied to allow bandwidth adaptation at SD layer and consequent tracking of the loss performance metric. The results have shown a good efficiency. For a deeper insight into the proposed control structure, the reader is referred to [5], where RCBC is validated with respect to the encapsulation change and the fading countermeasure issues.

Directions for future research may be: a deep investigation of the traffic aggregation problem in dependence of different traffic categories, the study of the RCBC applied to elastic traffic (e.g., TCP/IP) when it is encapsulated in a DVB-based SD core and the application of the SI-SAP principles in other wireless environments. Further simulation scenarios, including other transport technologies (e.g., IPv6), are currently the subject of ongoing research, as well as the application of the control mechanism for the delay and delay jitter constraints.

References

[1] ETSI. Satellite Earth Stations and Systems (SES). Broadband Satellite Multimedia. Services and Architectures. *ETSI Technical Report*, TR 101 984 V1.1.1, Nov. 2002.

[2] C. G. Cassandras, G. Sun, C. G. Panayiotou, Y. Wardi, "Perturbation Analysis and Control of Two-Class Stochastic Fluid Models for Communication Networks," *IEEE Trans. Automat. Contr.*, vol. 48, no. 5, May 2003, pp. 23-32.

[3] H. J. Kushner, G. G. Yin, *Stochastic Approximation Algorithms and Applications*, Springer-Verlag, New York, NY, 1997.

[4] http://www-tkn.ee.tu-berlin.de/research/trace/trace.html.

[5] M. Marchese, M. Mongelli, "On-line Bandwidth Control for Quality of Service Mapping over Satellite Independent Service Access Points," Computer Networks, to appear.