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Adaptive rate allocation and resource planning for service level agreement maintenance in satellite communications

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Abstract

The paper deals with protocol architectures to support mapping of Quality of Service (QoS) between protocol layers of telecommunications networks. The Satellite Independent – Service Access Point, currently under study within the European Telecommunications Standardization Institute (ETSI), is taken as technological reference. The key elements of the ETSI solution are outlined in detail. A novel control algorithm is also proposed to tune bandwidth dimensioning within the envisaged architecture. The joint optimization of loss and delay performance metrics, together with the presence of mapping operations, introduce the generalization of the regular concept of "equivalent bandwidth". The generalization leads to the study of a proper measurement-based tool, able to capture the most stringent QoS requirement over time (between loss and delay) and the related instantaneous bandwidth need at the lower layer of the protocol stack. The control methodology is tested through real fading and traffic traces to highlight its effectiveness to support bandwidth dimensioning during mapping operations.

Keywords: Satellite Service Access Point; QoS architectures; QoS mapping; Measurement-based control; Infinitesimal Perturbation Analysis

1. Introduction

Protocol stacks in telecommunications networks are composed of functional layers. *Quality of Service* (QoS) provision depends on the performance achieved at each layer and is based on functions performed at layer interfaces. Having in mind the OSI paradigm, QoS derives from reliable physical and link layers that can offer specific transport services to the upper network layers. The flows generated by the network layers (or bundles of them) are forwarded down to a physical interface that transports the information along a channel. Fig. 1 summarizes some possible examples.

Even if network layer implements efficient QoS mechanisms (e.g., IP IntServ, IP DiffServ, MPLS), it is topical that layers below can assure connection to the channel with specific degrees of performance. Otherwise the implementation of complex QoS mechanism is useless. As a consequence, the QoS requirements must flow vertically and need to be received and satisfied by all the layers of the protocol stack. More specifically, the link layer must implement appropriate mechanisms to support the *Service Level Agreement* (SLA) defined at the network layer. In some cases, in particular in wireless environments, the link layer acts in cooperation with the physical layer through the application of specific cross-layer design solutions. The interaction between the layers in this context is called here "QoS mapping". It leads to some technological problems that nowadays constitute open areas of standardization and research.

The paper proposes an insight into QoS mapping issues from different viewpoints. Firstly, a detailed analysis is reported for the protocol architectures necessary to support QoS mapping. Recent results of the *European Telecommunications Standardization Institute* (ETSI) are considered in detail. Then, a solution is proposed in this context to support mapping operations in cooperation with resource control.

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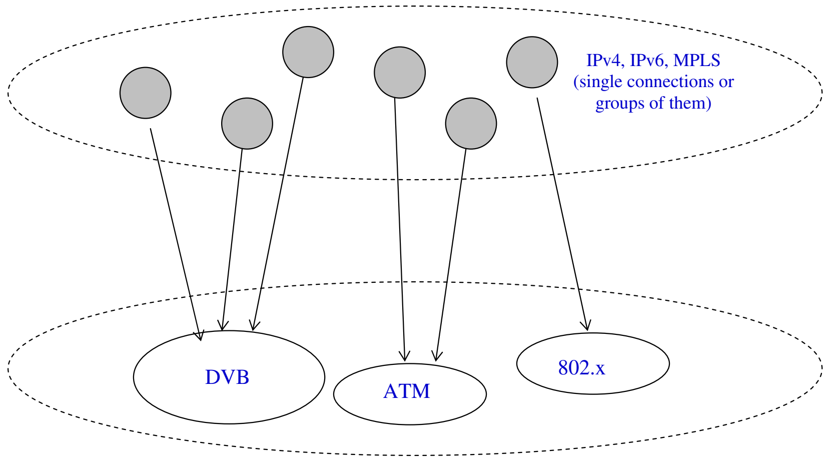

Fig. 1. Network layers over transport technology (link and physical layers).

The remainder of the paper is organized as follows. The next section introduces the technological elements of the ETSI architecture at both user and control plane levels. Section 3 specifies the QoS mapping operations inherent to this solution. Section 4 formalizes an optimization problem modeling the mapping operations. Section 5 contains the algorithm used to solve the problem with respect to the joint control of loss and delay constraints. Section 6 outlines a possible alternative approach, based on traditional equivalent bandwidth techniques. Performance analysis is proposed in Section 7. Conclusions are summarized in Section 8, with emphasis on possible directions of future research.

2. ETSI BSM protocol architecture

The protocol stack considered here is the ETSI BSM (*Broadband Satellite Multimedia*) architecture [1] shown in Fig. 2. It is related to satellite communication, but it may have a wider application. This is the motivation to choose it as reference for the investigation of QoS mapping.

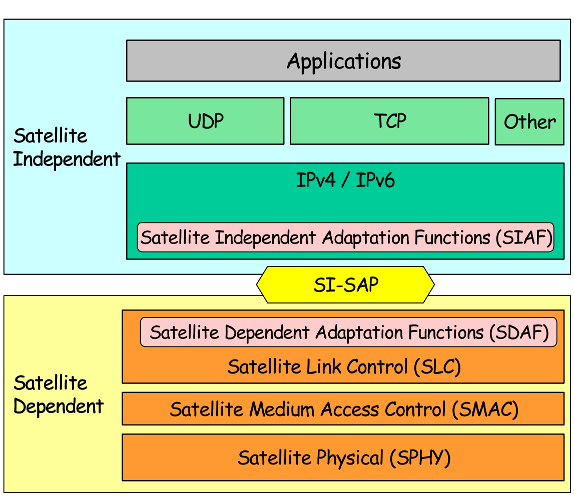

Fig. 2. ETSI Satellite Independent – Service Access Point (SI-SAP) [1].

The considered protocol stack separates the layers between *Satellite Dependent* (SD) and *Satellite Independent* (SI). The interface between SI and SD is defined as *Satellite Independent – Service Access Points* (SI-SAPs). QoS requirements must flow through SI-SAP and be implemented at SD layers. The terrestrial portion of the network interconnects with the satellite portion through an earth station. The SI-SAP is located in the earth station.

The QoS paradigm on the terrestrial portion is supposed to be DiffServ [2]. Thus, at SI level, DiffServ classes are classified using *Differentiated Services Code Point* (DSCP) field. Then, DSCP are mapped to SD queues. Since SD classes are system dependent, there is the need to abstract from the lower layers and to provide the SI layers with a common and agnostic interface. This concept leads to the definition of *Queue Identifiers* QIDs.

QID represents an abstract queue at the SAP level. Each QID is formally a relationship between IP queues and SD queues. Each QID defines a class of service for IP packets into the SD core. The application of QID principle allows hiding to SI layers the local implementation of the QoS within the SD systems, where, for instance, DVB may be applied. A QID is "virtual" in that it represents how a specific subset of IP queues are mapped over a specific subset of SD queues. For example, all *Assured Forwarding* (AF) queues may be aggregated together in a single SD queue or traffic shaping may be applied to IP flows before being forwarded to the SD queues. The application of shaping or other policing depends on the SLA between the satellite and terrestrial networks providers. On the other hand, the QID is "real" because it characterizes the real properties of the SD queues, for example, in terms of available SD resource reservations.

Fig. 3 depicts the relationship between traditional DiffServ IP queue management, which is performed above the SI-SAP, and the lower layers queues (QIDs and SD queues, from top to bottom). The relationship is shown for user plane on the left side of Fig. 3 and for control plane on the other side. IP-to-QID mapping means the definition of which outgoing flow from IP queues is forwarded

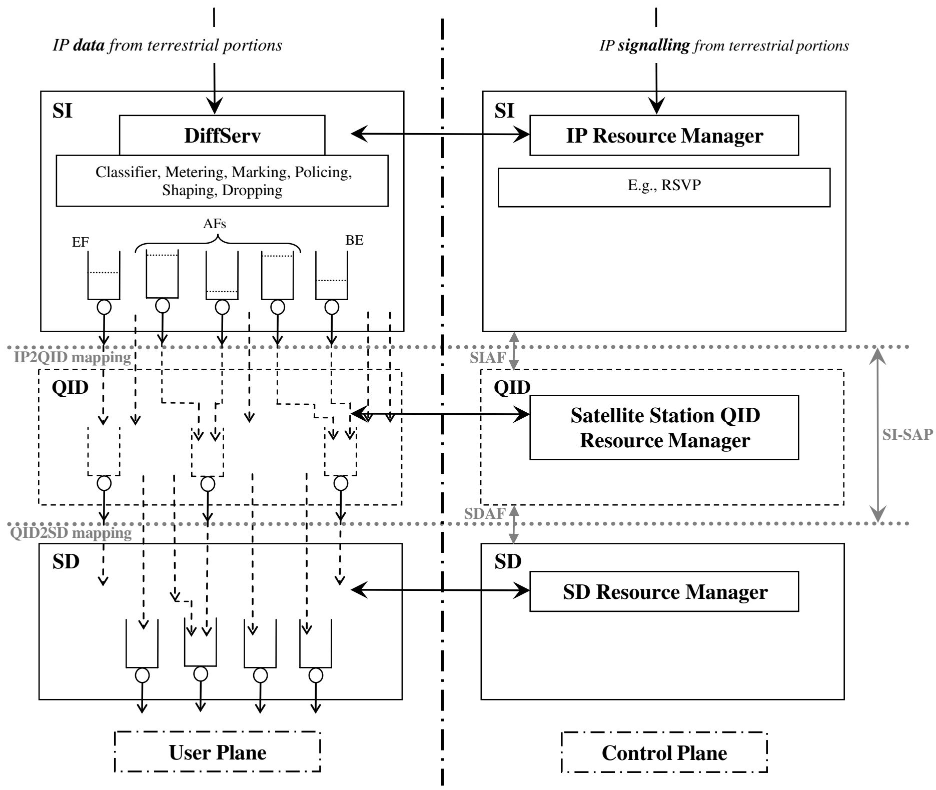

Fig. 3. ETSI SI-SAP protocol architecture.

down to a specific subset of QID queues. The same definition holds true for QID-to-SD mapping when forwarding traffic towards SD queues.

The control plane consists of one module above the SI-SAP (the IP resource manager), one module below (the SD resource manager), and one interfacing module (the *Satellite Terminal QID Resource Manager*, STQRM). An appropriate set of primitives [3] (called *Satellite Independent Adaptation Functions*, SIAF, and *Satellite Dependent Adaptation Functions*, SDAF, see Figs. 2 and 3) is defined to support STQRM resource invocation. More specifically the STQRM:

(1) receives requests from the IP module to allocate, to release or to modify resources through SIAF;
(2) translates these requests into QID allocation/release/modification actions;
(3) checks with the SD module how the upper layer requests can be mapped to real resources through SDAF.

The control plane may be statically or dynamically configured.

2.1. Static configuration

No specification to dynamically reserve resources or to receive indications of network resource availability may be implemented on the terrestrial portion of the network. In this case, user traffic entering the SI-SAP may be served through static trunks. The term static denotes the manual management of network resources over large time scales. For instance, the pre-allocation of a bandwidth trunk for a specific traffic class. Neither traffic prediction nor real time reaction to congestion, including *Call Admission Control* (CAC), is provided. The QIDs present to higher layers static resource reservations.

2.2. Dynamic configuration

Another approach consists of designing a dynamic control plane. It means triggering resource reservation on the basis of time varying SI traffic and satellite channel conditions. Thus, the control plane is based on some signalling scheme. RSVP may be used on the SI side. Usually, RSVP is applied in conjunction with IntServ. In case aggregation operations like IntServ over DiffServ are applied directly at the SI-SAP, RSVP is explicitly interfaced with DiffServ BSM [4]. So, SI-SAP can react to RSVP requests by allocating resources based on aggregated SI flows using regular RSVP agents as in common IntServ routers. The IP resource manager is therefore responsible for:

- translating RSVP requests into QID allocation requests;
- forwarding the QID allocation requests to the STQRM;
- receiving the reply from the STQRM and configure the IP queuing module in the user plane;
- sending appropriate RSVP signalling.

In turn, the STQRM, which received a QID allocation request, performs the following operations:

- check for available SD resources to accommodate the request;
- if resources are available, make the required changes in the IP-to-QID or QID-to-SD mappings, and notify the IP resource manager;
- if SD resources are not available, request allocation of new resources to the SD resource manager, wait for response, make the required changes in the IP-to-QID or QID-to-SD mappings, and notify the IP manager.

3. QoS mapping and resource allocation

In both static and dynamic configurations, the topical point is how SI resource invocations are mapped over the SI-SAP control plane.

In the presence of static configuration, a consistent IP-to-QID mapping is created and statically maintained. No signalling is required across the SI-SAP. A set of DiffServ queues is pre-configured according to the SLA. The QID-to-SD has to be static, too. The way in which QIDs are associated to SD queues is left to the implementers and to the specifics of each SD technology.

On the other hand, dynamic configuration means changing both IP-to-QID and QID-to-SD mappings and, as a consequence, the bandwidth allocations at SD level. Two entities may introduce IP-to-QID mapping modifications: the IP resource manager and the STQRM. The interaction between the two entities is supported by SIAF invocations.

(1) *IP Resource Manager action*. Requests about allocating or releasing resources may come from IP resource manager on the basis of explicit signalling coming from terrestrial portion (e.g., RSVP as mentioned above) or by monitoring the IP queue occupancy. The request is forwarded down to the STQRM, which, if accepted, adapts the QIDs to the new needs of SI layers.
(2) *STQRM action*. Even without explicit messages coming from IP Resource Manager, STQRM can adapt QIDs to time varying incoming IP traffic by means of feed-forward information received on SI queues state.

In both cases, the STQRM sends feedback to the SD manager through SDAF primitives to trigger resource reallocation if needed. The reallocation may introduce also a QID-to-SD mapping change, for example in the case the number of QIDs and/or of SD queues changes.

In the dynamic configuration case, a complex resource control problem arises for the SD manager. Fig. 4, similarly used in [7], summarizes a general case of QID abstraction. The QID-to-SD mapping "hides" the following mapping operations.

(1) *Change of information unit*. It is the consequence of IP traffic transport over a SD portion that implements a specific technology, e.g., ATM as often done in industrial systems [5], or DVB.
(2) *Heterogeneous traffic aggregation*. The due association of IP QoS classes to SD transfer capabilities is limited by hardware implementation constraints. As outlined in [6], "*it is accepted in the BSM industry that at the IP level (above the SI-SAP interface) between 4 and 16 queues are manageable for different IP classes. Below the SI-SAP these classes can further be mapped into the satellite dependent priorities within the BSM which can be from 2 to 4 generally*". A queue model describing the aggregation operation is reported in Fig. 4. The problem is how much bandwidth must be assigned to each SD queue so that the SI IP-based SLA is guaranteed.
(3) *Fading counteraction*. Finally yet importantly, many transmission environments, as well as satellite an wireless link, needs to tackle time varying channel conditions due to fade.

The mapping operations make the SD bandwidth dimensioning a hard task. *Equivalent bandwidth* (EqB) modeling may be used. According to the scientific literature, EqB is the traditional mathematical instrument to "capture" the statistical behaviour of traffic sources while

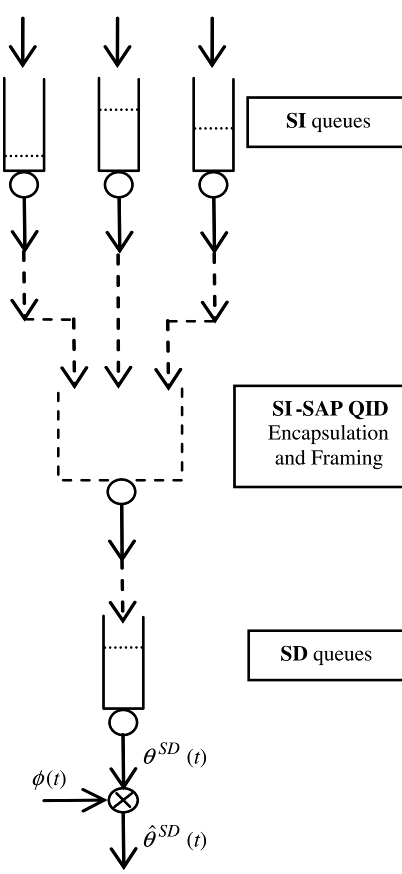

Fig. 4. Reference example of IP-to-QID and QID-to-SD mappings.

driving bandwidth dimensioning. A "generalized" EqB concept underlies the SD bandwidth provisioning problem. EqB is usually defined as *the minimum bandwidth allocation necessary to guarantee a specific QoS to a traffic flow*. What is needed here is the minimum bandwidth provision that satisfies all the QoS levels required by the different classes aggregated in the same SD trunk. Moreover, traffic entering SD queues is statistically heterogeneous due to the aggregation operation described at point (2) above. Many studies confirm the efficiency of aggregating homogeneous traffic, but EqB for non-homogeneous trunks (from the statistical behaviour and QoS requirement viewpoints) is still an open issue. More details concerning the application of regular EqB techniques to the present architecture are reported in Section 6.

4. The SI-SAP QoS mapping problem

To summarize, optimal SD bandwidth dimensioning faces the definition of a new EqB paradigm. To this aim, a novel control scheme for the optimization of the bandwidth provision is formalized. The algorithm was firstly introduced in [9]. Differing from [9], multiple SI traffic classes are considered along with the joint performance metric composed of packet loss probability and average delay. The joint control of loss and delay was firstly presented in [10,11].

First of all, it is necessary to mathematically express fading counteraction (operation (3) above). Let $\theta^{SD}(t)$ be the service rate assigned to a traffic buffer at the SD layer at time t. The effect of fading is modeled as a reduction of the bandwidth actually "seen" by the buffer [8], due, e.g., to the application of *Forward Error Correction* (FEC) redundancy applied at the physical layer. 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 thus obtained as:

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

Then, the following formulation of SI-SAP queues allows capturing the effect of QoS mapping on SD bandwidth dimensioning (see also Fig. 4).

4.1. Stochastic fluid model and optimization problem

IP Packet Loss Probability (PLP) and *IP Packet Average Delay* (AD) are the chosen SLA performance metrics. The mathematical framework is based on *Stochastic Fluid Models* (SFM) [12] of the SI-SAP traffic buffers. SFM constitutes the mathematical background for the analytical derivation of buffer metrics derivative estimators used in the following (for example with respect to loss and workload quantities). N SI queues and, without loss of generality, one single SD queue are considered for the analytical formulation. Each independent DiffServ flow entering a SI queue is called *traffic class*. Fig. 5 updates Fig. 4 with respect to the notation used.

Let $\alpha_i^{SI}(t)$ and $\beta_i^{SI}(t)$ be, respectively, the *inflow* and *outflow* rate processes of the ith traffic class at SI layer at time t, $i = 1, \ldots, N$. The service rate of buffer i is θ_i^{SI}. It is straightforward that:

$$\beta_i^{SI}(t) = \begin{cases} \alpha_i^{SI}(t), & \text{if } x_i(\cdot, t) = 0 \\ \theta_i^{SI}, & \text{if } x_i(\cdot, t) \neq 0 \end{cases} \tag{2}$$

Let $\gamma_i^{SI}(\cdot, t)$ and $x_i^{SI}(\cdot, t)$ be the *overflow* rate and the *workload* (i.e., the fluid volume in the buffer) processes of the ith traffic class at SI layer, respectively, $i = 1, \ldots, N$. The same quantities are defined for the SD layer, namely, the overflow rate and the workload processes measured at SD layer are $\gamma_i^{SD}(\cdot, t)$ and $x_i^{SD}(\cdot, t)$, for each traffic class $i = 1, \ldots, N$. Let $\alpha^{SD}(t)$ be the inflow rate process of the buffer at the SD layer at time t. It 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) and from the change of the encapsulation format at the SI-SAP.

No a-priori assumptions are made on the statistical behaviour of the inflow processes $\alpha_i^{SI}(t)$, $i = 1, \ldots, N$. As outlined above concerning EqB modeling, even if some certainty equivalent assumptions were feasible for them, the QoS mapping operations would let the assumptions not

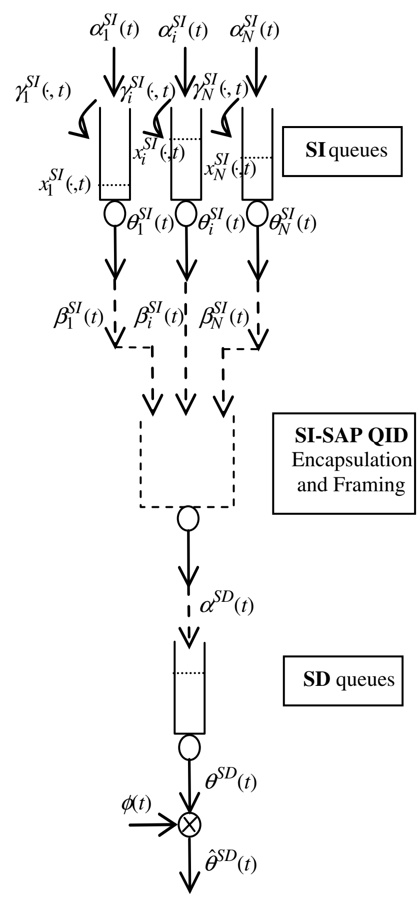

Fig. 5. Stochastic fluid model of the reference example.

applicable for $\alpha^{\mathrm{SD}}(t)$. No analytical expression for QoS achieved by traffic classes at SD layer is available, since there are no instruments for the mathematical description of the statistical behavior of the packets belonging to a specific flow (i.e., $\alpha_i^{\mathrm{SI}}(t)$ for a given i) within an aggregated trunk (i.e., $\alpha^{\mathrm{SD}}(t)$). The necessary information may be got measures. To this aim, two SFM cumulative metrics are defined to match PLP and AD modeling and to drive reallocations on the basis of on-line measures. They are the results of integrating $\gamma_i^{\mathrm{SD}}(\cdot,t)$ and $x_i^{\mathrm{SD}}(\cdot,t)$ quantities, $i=1,\ldots,N$, over a given *observation horizon* (OH). The cumulative quantities of interest are: ${}^{i,k}L_V^{\mathrm{SD}}\left(\alpha^{\mathrm{SD}}(t),\theta^{\mathrm{SD}}(k)\cdot\phi(t)\right)$ and ${}^{i,k}L_W^{\mathrm{SD}}\left(\alpha^{\mathrm{SD}}(t),\theta^{\mathrm{SD}}(k)\cdot\phi(t)\right)$, defined as the *loss volume* and the buffer *cumulative workload*, respectively, of the ith traffic, class within the SD buffer over the kth OH (i.e., for $t\in[k,k+1]$).

$$ {}^{i,k}L_V^{\mathrm{SD}}(\alpha^{\mathrm{SD}}(t),\theta^{\mathrm{SD}}(k)\cdot\phi(t)) = \int_k^{k+1}\gamma_i^{\mathrm{SD}}(\cdot,t)\mathrm{d}t \tag{3} $$

$$ {}^{i,k}L_W^{\mathrm{SD}}(\alpha^{\mathrm{SD}}(t),\theta^{\mathrm{SD}}(k)\cdot\phi(t)) = \int_k^{k+1}x_i^{\mathrm{SD}}(\cdot,t)\mathrm{d}t \tag{4} $$

They are functions of the following elements: the SD inflow process $\alpha^{\mathrm{SD}}(t)$ (deriving from the aggregation of the SI inflow processes $\alpha_i^{\mathrm{SI}}(t)$, $i=1,\ldots,N$, encapsulated in a specific transport technology), the fading process $\phi(t)$. The SD bandwidth allocation decided at time k is $\theta^{\mathrm{SD}}(k)$ and remains fixed over the kth OH. For example, the SD queue may receive a service rate of 1.0 Mbps over the kth OH and 2.0 Mbps over the $k+1$th OH to counteract the addition of a double amount of traffic during the $k+1$th OH.

The quantities ${}^{i,k}L_V^{\mathrm{SD}}(\cdot)$ and ${}^{i,k}L_W^{\mathrm{SD}}(\cdot)$, $k=0,1,\ldots,$ allow capturing the performance level of each traffic class i for PLP and AD, respectively:

$$ \begin{aligned} {}^{i,k}\mathrm{PLP}^{\mathrm{SD}}\left(\alpha^{\mathrm{SD}}(t),\theta^{\mathrm{SD}}(k)\cdot\phi(t)\right) &= \frac{\int_k^{k+1}\gamma_i^{\mathrm{SD}}(\cdot,t)\mathrm{d}t}{\int_k^{k+1}\beta_i^{\mathrm{SI}}(\cdot,t)\mathrm{d}t} \\ &= \frac{{}^{i,k}L_V^{\mathrm{SD}}\left(\alpha^{\mathrm{SD}}(t),\theta^{\mathrm{SD}}(k)\cdot\phi(t)\right)}{\int_k^{k+1}\beta_i^{\mathrm{SI}}(\cdot,t)\mathrm{d}t} \end{aligned} \tag{5} $$

$$ \begin{aligned} &{}^{i,k}\mathrm{AD}^{\mathrm{SD}}\left(\alpha^{\mathrm{SD}}(t),\theta^{\mathrm{SD}}(k)\cdot\phi(t)\right) \\ &= \frac{\int_k^{k+1}x_i^{\mathrm{SD}}(\cdot,t)\mathrm{d}t}{\theta^{\mathrm{SD}}(k)} + \frac{\overline{DimPacket_{i,k}}}{\theta^{\mathrm{SD}}(k)} \\ &= \frac{{}^{i,k}L_W^{\mathrm{SD}}\left(\alpha^{\mathrm{SD}}(t),\theta^{\mathrm{SD}}(k)\cdot\phi(t)\right)}{\theta^{\mathrm{SD}}(k)} + \frac{\overline{DimPacket_{i,k}}}{\theta^{\mathrm{SD}}(k)} \end{aligned} \tag{6} $$

being $\int_k^{k+1}\beta_i^{\mathrm{SI}}(\cdot,t)\mathrm{d}t$ the amount of traffic (in bit) served for the ith traffic class at SI level, ${}^{i,k}\mathrm{PLP}^{\mathrm{SD}}(\cdot)$ and ${}^{i,k}\mathrm{AD}^{\mathrm{SD}}(\cdot)$ the PLP and AD measured over the kth OH at SD level for traffic class i and $\overline{DimPacket_{i,k}}$ the average size of the packet belonging to the ith traffic class and entering the SD buffer during the kth OH. The choice to make use of ${}^{i,k}L_V^{\mathrm{SD}}(\cdot)$ and ${}^{i,k}L_W^{\mathrm{SD}}(\cdot)$ to develop the optimization problem of next section (in place of ${}^{i,k}\mathrm{PLP}^{\mathrm{SD}}(\cdot)$ and ${}^{i,k}\mathrm{AD}^{\mathrm{SD}}(\cdot)$) is related to the application of their SFM derivative estimators that are based on on-line measures. This topic is addressed in the next section.

In theory, the OH should be sufficiently large to capture the average value of the performance metrics defined over an infinite horizon. A too short OH introduces excessive noise to measurements performed over the real system. On the other hand, OH should be sufficiently short to track on line the behavior of the system and to quickly drive reallocations in time varying conditions. Thus, the OH size is the compromise between these competitive needs. In general, when applying measurement-based techniques, OH is heuristically dimensioned on the basis of experience available over the system to be controlled. In general, when working on QoS parameters, values of the OH size are typically between fractions of minute up to 10 min [12,13].

The key idea of the proposed formulation is to "equalize" the QoS measured at the SD layer in dependence of the QoS imposed by the SI layer. To capture this concept, it is useful to think at a "penalty cost function", whose values can be interpreted as an indication about the current inability of the SD layer to guarantee the required QoS. In practice, ${}^{i,k}L_V^{\mathrm{SD}}(\cdot)$ and ${}^{i,k}L_W^{\mathrm{SD}}(\cdot)$, $k=0,1,\ldots,$ chase the dynamic variations of the quantities (representative of the SI layer requests) identified in the following as ${}^{i,k}L_{V-\mathrm{Thr}}$ and ${}^{i,k}L_{W-\mathrm{Thr}}$, for PLP and AD, respectively. Concerning this issue, the paper proposes two alternatives, reported below:

(a) *SLA reference thresholds.* ${}^{i,k}L_V^{\mathrm{SD}}(\cdot)$ and ${}^{i,k}L_W^{\mathrm{SD}}(\cdot)$ chase, for each $k=0,1,\ldots,$ threshold values that flow from SI to SD layer and are representative of the SLA PLP_i^* and AD_i^* of the ith traffic class:

$$ {}^{i,k}L_{V-\mathrm{Thr}} = \mathrm{PLP}_i^*\cdot\int_k^{k+1}\beta_i^{\mathrm{SI}}(t)\mathrm{d}t \tag{7} $$

$$ {}^{i,k}L_{W-\mathrm{Thr}} = \theta^{\mathrm{SD}}(k)\cdot\mathrm{AD}_i^* - \overline{DimPacket_{i,k}} \tag{8} $$

(b) *SI performance reference.* ${}^{i,k}L_V^{\mathrm{SD}}(\cdot)$ and ${}^{i,k}L_W^{\mathrm{SD}}(\cdot)$ chase the performance measured at SI layer for each $k=0,1,\ldots$ Let ${}^{i,k}L_V^{\mathrm{SI}}\left(\alpha_i^{\mathrm{SI}}(t),\theta_i^{\mathrm{SI}}\right)$ and ${}^{i,k}L_W^{\mathrm{SI}}\left(\alpha_i^{\mathrm{SI}}(t),\theta_i^{\mathrm{SI}}\right)$ be the measured *loss volume* and the *cumulative workload* of the ith IP buffer having service rate θ_i^{SI}, respectively:

$$ {}^{i,k}L_V^{\mathrm{SI}}(\alpha_i^{\mathrm{SI}}(t),\theta_i^{\mathrm{SI}}) = \int_k^{k+1}\gamma_i^{\mathrm{SI}}(\cdot,t)\mathrm{d}t \tag{9} $$

$$ {}^{i,k}L_W^{\mathrm{SI}}(\alpha_i^{\mathrm{SI}}(t),\theta_i^{\mathrm{SI}}) = \int_k^{k+1}x_i^{\mathrm{SI}}(\cdot,t)\mathrm{d}t \tag{10} $$

over the kth OH. In this case, the aim is the equalization of the PLP and AD of each traffic class between the SI and SD layers:

$$ {}^{i,k}L_{V-\mathrm{Thr}} = {}^{i,k}L_V^{\mathrm{SI}}(\cdot) \tag{11} $$

$$ {}^{i,k}L_{W-\mathrm{Thr}} = {}^{i,k}L_W^{\mathrm{SI}}(\cdot) \tag{12} $$

It is an interesting alternative to point (a): chasing the performance of the layer above, even if it makes worst the overall performance because the two layers are in cascade, may help save bandwidth when the performance of the SI layer is not satisfying. Actually, if the SI layer cannot guarantee a specific level of QoS, probably it is useless to provide effort at SD to assure the PLP and AD thresholds. Moreover, tracking the behavior of the layer above (or a fraction of it) allows operating without the knowledge of the SLAs at SD layer.

The optimization problem, called QoS mapping Optimization (QoSMO) Problem, can now be stated. Stochastic processes driving SD queue evolution over time (α^{SD} and ϕ) are assumed to be ergodic. The QoSMO problem consists of finding the optimal bandwidth allocations $L_{\Delta V}{}^{\theta}_{\mathrm{Opt}}{}^{\mathrm{SD}}$ and $L_{\Delta V}{}^{\theta}_{\mathrm{Opt}}{}^{\mathrm{SD}}$, so that the cost functions $J_{L_{\Delta V}}(\cdot,\theta^{\mathrm{SD}})$ and $J_{L_{\Delta W}}(\cdot,\theta^{\mathrm{SD}})$ (defined below) are minimized:

$$
\begin{aligned}
&L_{\Delta V}{}^{\theta}_{\mathrm{Opt}}{}^{\mathrm{SD}} = \arg\min_{\theta^{\mathrm{SD}}} J_{L_{\Delta V}}(\cdot,\theta^{\mathrm{SD}}); \quad J_{L_{\Delta V}}(\cdot,\theta^{\mathrm{SD}}) = \underset{\alpha^{\mathrm{SD}},\phi}{E}\, {}^{k}L_{\Delta V}(\cdot,\theta^{\mathrm{SD}});\\
&{}^{k}L_{\Delta V}(\cdot,\theta^{\mathrm{SD}}) = \sum_{i=1}^{N} {}^{i,k}L_{\Delta V}(\cdot,\theta^{\mathrm{SD}}); \qquad (13)\\
&{}^{i,k}L_{\Delta V}(\cdot,\theta^{\mathrm{SD}}) = \left[{}^{i,k}L^{\mathrm{SI}}_{V-\mathrm{Thr}} - {}^{i,k}L^{\mathrm{SD}}_{V}(\cdot,\theta^{SD})\right]^2; \quad k=0,1,\ldots
\end{aligned}
$$

$$
\begin{aligned}
&L_{\Delta V}{}^{\theta}_{\mathrm{Opt}}{}^{\mathrm{SD}} = \arg\min_{\theta^{\mathrm{SD}}} J_{L_{\Delta W}}(\cdot,\theta^{\mathrm{SD}}); \quad J_{L_{\Delta W}}(\cdot,\theta^{\mathrm{SD}}) = \underset{\alpha^{\mathrm{SD}},\phi}{E}\, {}^{k}L_{\Delta W}(\cdot,\theta^{\mathrm{SD}});\\
&{}^{k}L_{\Delta W}(\cdot,\theta^{\mathrm{SD}}) = \sum_{i=1}^{N} {}^{i,k}L_{\Delta W}(\cdot,\theta^{\mathrm{SD}}); \qquad (14)\\
&{}^{i,k}L_{\Delta W}(\cdot,\theta^{\mathrm{SD}}) = \left[{}^{i,k}L^{\mathrm{SI}}_{W-\mathrm{Thr}} - {}^{i,k}L^{\mathrm{SD}}_{W}(\cdot,\theta^{\mathrm{SD}})\right]^2; \quad k=0,1,\ldots
\end{aligned}
$$

where ${}^{i,k}L_{V-\mathrm{Thr}}$ and ${}^{i,k}L_{W-\mathrm{Thr}}$ are the reference thresholds defined either in (7) and (8) or in (11) and (12). Note that apex k is used for ${}^{i,k}L_{\Delta V}(\cdot)$ and ${}^{i,k}L_{\Delta W}(\cdot)$ functions since they depend on the specific measures performed over the kth OH. On the other hand, assuming α^{SD} and ϕ processes being ergodic make the apex k unnecessary for $J_{L_{\Delta V}}(\cdot)$, $J_{L_{\Delta W}}(\cdot)$, $L_{\Delta V}{}^{\theta}_{\mathrm{Opt}}{}^{\mathrm{SD}}$ and $L_{\Delta V}{}^{\theta}_{\mathrm{Opt}}{}^{\mathrm{SD}} \cdot J_{L_{\Delta V}}(\cdot) = \underset{\alpha^{\mathrm{SD}},\phi}{E}\, {}^{k}L_{\Delta V}(\cdot)$, $J_{L_{\Delta W}}(\cdot) = \underset{\alpha^{\mathrm{SD}},\phi}{E}\, {}^{k}L_{\Delta W}(\cdot)$ and their minima are independent of k.

The main idea of the QoSMO formulation is trying to equalize ${}^{i,k}L_{V-\mathrm{Thr}}$ and ${}^{i,k}L_{W-\mathrm{Thr}}$ with ${}^{i,k}L_{V}(\cdot)$ and ${}^{i,k}L_{W}(\cdot)$ for each traffic class.

It is important to remind that one QoS constraint (e.g., PLP) may reveal to be more stringent than the other one (e.g., AD) in dependence of the specific traffic condition. Being the QoS metric currently greedier in terms of bandwidth, its satisfaction automatically assures the other constraint. For this reason, it is expected that the distinct solutions of (13) and (14) are in general different (i.e., $L_{\Delta V}{}^{\theta}_{\mathrm{Opt}}{}^{\mathrm{SD}} \neq L_{\Delta W}{}^{\theta}_{\mathrm{Opt}}{}^{\mathrm{SD}}$). Without *a-priori* knowing the most stringent constraint, the control scheme needs to "learn" it in real time and automatically adapt the bandwidth.

To envisage this concept, the quantity ${}^{\mathrm{Opt}}\theta^{\mathrm{SD}} = \mathrm{Max}\left\{L_{\Delta V}{}^{\theta}_{\mathrm{Opt}}{}^{\mathrm{SD}}, L_{\Delta W}{}^{\theta}_{\mathrm{Opt}}{}^{\mathrm{SD}}\right\}$ is defined to state the *optimal solution* of the QoSMO problem, i.e., the minimum rate provision to assure the SLA within the SD core. To summarize, the final aim is to get a control algorithm able to counteract time varying system conditions, tracking $L_{\Delta V}{}^{\theta}_{\mathrm{Opt}}{}^{\mathrm{SD}}, L_{\Delta W}{}^{\theta}_{\mathrm{Opt}}{}^{\mathrm{SD}}$ and performing the SD rate provision according to ${}^{\mathrm{Opt}}\theta^{\mathrm{SD}}$.

Finally yet importantly, it is worth noting that a QoSMO problem arises for each SD queue and may be exploited by the SD manager to support resource invocations as outlined in Section 4. On the basis of the QoSMO problem solution, the SD manager may update the SD queues reservation state. It may also trigger feedback up to the STQRM. For example, in case the SD manager is not able to support the desired levels of performance (i.e., PLP^*_i and AD^*_i, $i=1,\ldots,N$). In turn, the STQRM may perform its specific reallocation decisions in terms of QID-to-SD and IP-to-QID mappings and/or forward feedback up to the IP manager.

5. The joint control of loss and delay

5.1. Derivative estimators

Without any analytical expression of $J_{L_{\Delta V}}(\cdot,\theta^{\mathrm{SD}})$ and $J_{L_{\Delta W}}(\cdot,\theta^{\mathrm{SD}})$, the designed control scheme follows the principle of *Perturbation Analysis* (see, e.g., [12] and references therein), by sampling on line the cost functions and estimating their derivatives used in the gradient descent explained later on. Considering the PLP constraint first, the cost function ${}^{i,k}L_{\Delta V}(\cdot)$ derivative can be obtained for the kth OH and each i as:

$$
\left.\frac{\partial^{k}L_{\Delta V}(\cdot,\theta^{\mathrm{SD}})}{\partial\theta^{\mathrm{SD}}}\right|_{\theta^{\mathrm{SD}}=\theta^{\mathrm{SD}}(k)} = 2\cdot\phi(t)\cdot\sum_{i=1}^{N}\left.\frac{\partial^{i,k}L^{\mathrm{SD}}_{V}(\theta^{\mathrm{SD}})}{\partial\theta^{SD}}\right|_{\theta^{\mathrm{SD}}=\theta^{\mathrm{SD}}(k)} \times\left[{}^{i,k}L^{\mathrm{SD}}_{V}(\theta^{\mathrm{SD}}(k)) - {}^{i,k}L^{\mathrm{SI}}_{V}(\theta^{\mathrm{SI}}_{i})\right] \tag{15}
$$

As far as the AD is concerned, the derivative can be similarly obtained:

$$
\left.\frac{\partial^{k}L_{\Delta W}(\cdot,\theta^{\mathrm{SD}})}{\partial\theta^{\mathrm{SD}}}\right|_{\theta^{\mathrm{SD}}=\theta^{\mathrm{SD}}(k)} = 2\cdot\phi(t)\cdot\sum_{i=1}^{N}\left.\frac{\partial^{i,k}L^{\mathrm{SD}}_{W}(\theta^{\mathrm{SD}})}{\partial\theta^{SD}}\right|_{\theta^{\mathrm{SD}}=\theta^{\mathrm{SD}}(k)} \times\left[{}^{i,k}L^{SD}_{W}(\theta^{SD}(k)) - {}^{i,k}L^{\mathrm{SI}}_{W}(\theta^{\mathrm{SI}}_{i})\right] \tag{16}
$$

Due to the application of *Infinitesimal Perturbation Analysis* (IPA), recently developed in the field of *Sensitivity Estimation* techniques for *Discrete Event Systems* ([12]), each $\left.\frac{\partial^{i,k}L^{\mathrm{SD}}_{V}(\theta^{\mathrm{SD}})}{\partial\theta^{\mathrm{SD}}}\right|_{\theta^{\mathrm{SD}}=\theta^{\mathrm{SD}}(k)}$ and $\left.\frac{\partial^{i,k}L^{\mathrm{SD}}_{W}(\theta^{\mathrm{SD}})}{\partial\theta^{\mathrm{SD}}}\right|_{\theta^{\mathrm{SD}}=\theta^{\mathrm{SD}}(k)}$ component can be obtained in real time on the basis of traffic samples acquired during the system evolution during each OH $k=0,1,\ldots$

Such derivative estimators are derived in [12] for a single traffic class and are heuristically adapted to the multi-classes case in this paper. Both for loss and delay, the key idea is to measure the contribution of the IP packets belonging to the ith traffic class to the SD components of (15) and (16). These quantities allow "measuring" the QoS received by the ith traffic class within the aggregated trunk.

5.2. Reference chaser bandwidth controller

From now on, the notation $\Psi = V, W$ is used to avoid notational burden. The proposed optimization algorithm, called *Reference Chaser Bandwidth Controller* (RCBC), performs a sequence of bandwidth reallocations $\theta^{\mathrm{SD}}(k)$, $k = 0, 1, \ldots$, based on the gradient method, whose descent steps are ruled by:

$$\theta^{\mathrm{SD}}_{L_{\Delta\Psi}}(k+1) = \theta^{\mathrm{SD}}_{L_{\Delta\Psi}}(k) - \eta^{k}_{L_{\Delta\Psi}} \cdot \left.\frac{\partial^{k} L_{\Delta\Psi}(\cdot, \theta^{\mathrm{SD}})}{\partial\theta^{\mathrm{SD}}}\right|_{\theta^{\mathrm{SD}}=\theta^{\mathrm{SD}}(k)} \tag{17}$$

$$\theta^{\mathrm{SD}}(k+1) = \theta^{\mathrm{SD}}_{L_{\Delta\Psi^{\mathrm{Max}}_{k}}}(k+1) \tag{18}$$

$$\Psi^{\mathrm{Max}}_{k} = \arg \underset{\Psi}{\mathrm{Max}} \left\{ \left.\frac{\partial^{k} \bar{L}_{\Delta\Psi}(\cdot, \theta^{\mathrm{SD}})}{\partial\theta^{\mathrm{SD}}}\right|_{\theta^{\mathrm{SD}}=\theta^{\mathrm{SD}}(k)}, \Psi = V, W \right\} \tag{19}$$

where $\frac{\partial \bar{L}_{\Delta\Psi}(\cdot,\theta^{\mathrm{SD}})}{\partial\theta^{\mathrm{SD}}}$ in (19) denotes the normalized cost derivative, namely:

$$\frac{\partial \bar{L}_{\Delta\Psi}(\cdot, \theta^{\mathrm{SD}})}{\partial\theta^{SD}} = \frac{\frac{\partial L_{\Delta\Psi}(\cdot,\theta^{SD})}{\partial\theta^{\mathrm{SD}}}}{\frac{\partial L^{\mathrm{Max}}_{\Delta\Psi}(\cdot,\theta^{\mathrm{SD}})}{\partial\theta^{\mathrm{SD}}}}, \quad \Psi = V, W \tag{20}$$

being $\frac{\partial L^{\mathrm{Max}}_{\Delta\Psi}(\cdot,\theta^{\mathrm{SD}})}{\partial\theta^{\mathrm{SD}}}$ the maximum value achievable for $\frac{\partial L_{\Delta\Psi}(\cdot,\theta^{\mathrm{SD}})}{\partial\theta^{\mathrm{SD}}}$ over a given decision period, $\eta^{k}_{L_{\Delta V}}, \eta^{k}_{L_{\Delta W}}$ the two gradient stepsizes and k the reallocation instant.

The rationale behind (19) relies on the need to find out the most stringent QoS requirement at the moment. Intuitively, this decision can be taken by exploiting the current values of the performed sensitivity estimators (15) and (16), and by choosing the largest one as an indication of the most suffering QoS constraint. It is worth noting that $\theta^{\mathrm{SD}}(k+1)$ is not defined in (18) as $\theta^{\mathrm{SD}}(k+1) = \underset{\Psi}{\mathrm{Max}}\left\{\theta^{\mathrm{SD}}_{L_{\Delta\Psi}}(k+1), \Psi = V, W\right\}$ because of the presence of the gradient stepsizes in (17). Being $\eta^{k}_{L_{\Delta\Psi}}$ tuned as tradeoff between convergence speed and rate oscillations, could affect the identification of the most stringent QoS constraint, which needs to be identified independently of the $\eta^{k}_{L_{\Delta\Psi}}$ tuning process. The direct comparison of the normalized sensitivity estimators $\frac{\partial \bar{L}_{\Delta\Psi}(\cdot,\theta^{\mathrm{SD}})}{\partial\theta^{\mathrm{SD}}}, \Psi = V, W$ reveals to be a precise tool to detect the most sensitive QoS threshold. An example is reported in the performance evaluation section.

Technical conditions for RCBC convergence to global optimum are: (1) ergodic stochastic processes, (2) decreasing behavior of gradient stepsizes, (3) boundness of gradients within the control domain $\theta^{\mathrm{SD}} \in \Re^{+}$, and (4) nonexistence of local optima. (1) and (2) are assumptions. As outlined when formulating the QoSMO problem, point (1) is the technical condition necessary to capture the (unknown) average performance (i.e., $\underset{\alpha_{\mathrm{SD}},\phi}{E}\left[{}^{k}L_{\Delta\Psi}(\cdot)\right], \Psi = V, W$) only on the basis of the measures (i.e.,${}^{k}L_{\Delta\Psi}(\cdot), \Psi = V, W$) over the OH. The same concept holds true with respect to the (unknown) derivative of the functional costs $\left(\frac{\partial E_{\alpha_{\mathrm{SD}},\phi}\left[{}^{k}L_{\Delta\Psi}(\cdot,\theta^{\mathrm{SD}})\right]}{\partial\theta^{\mathrm{SD}}}, \quad \Psi = V, W\right)$ estimated by sampling IPA-based estimators $\frac{\partial^{k} L_{\Delta\Psi}(\cdot,\theta^{\mathrm{SD}})}{\partial\theta^{\mathrm{SD}}}$, $\Psi = V, W$ (15) and (16). As to further details regarding the analytical derivation of IPA estimators, the reader is referred to [12], where IPA principles and their technical conditions are detailed. Concerning point (3), the lengths of buffer busy periods are bounded by OH size. Thus, measured loss rate at the end of each OH cannot be infinite. Concerning point (4), the loss rate of a traffic queue can be reasonably assumed to be continuous, differentiable, with a negative derivative in the service rate, so the cost function is also continuous, differentiable with a unique minimum. The convergence speed depends on the length of OH and on the gradient stepsize. In this perspective, the assumption of process ergodicity may be relaxed and limited to the time the sequence $\theta^{\mathrm{SD}}(k)$, $k = 0, 1, \ldots$ needs to converge to ${}^{\mathrm{Opt}}\theta^{\mathrm{SD}}$. When α^{SD} or ϕ change their statistical behavior, a new QoSMO problem is solved by starting RCBC algorithm again.

6. Alternative approach: equivalent bandwidth heuristic

As mentioned at the end of Section 3, regular equivalent bandwidth techniques may be also used to support bandwidth dimensioning during mapping operations. Equivalent bandwidth is based on the statistical characterization of the traffic by means of descriptors as *peak rate*, *mean rate*, *maximum burst size* and so on. The inflow process $a^{\mathrm{SD}}(t)$ is the result of the aggregation of the outflow processes of the SI buffers and the mentioned traffic descriptors may be hardly exploited, but the technique may be still applied. The mean value $m_{\alpha^{\mathrm{SD}}}$ and the standard deviation $\sigma_{\alpha^{\mathrm{SD}}}$ of the $a^{\mathrm{SD}}(t)$ process may be used. The approach is called here *Equivalent Bandwidth* (EqB). As well as RCBC, EqB exploits on-line measures without assuming any a-priori knowledge of traffic statistics and buffer size. Being $k = 0, 1, \ldots$ the time instants of the SD rate reallocations, $m_{a^{\mathrm{SD}}}(k)$ and $\sigma_{a^{\mathrm{SD}}}(k)$ the mean and the standard deviation, respectively, of the SD inflow process measured over the time interval $[k, k+1]$, the bandwidth provision $\theta^{\mathrm{SD}}(k+1)$ at the SD layer, assigned over the time interval $[k+1, k+2]$, is computed as in (21).

$$\theta^{\mathrm{SD}}(k+1) = m_{\alpha^{\mathrm{SD}}}(k) + z(\mathrm{PLP}^{*}_{\mathrm{EqB}}) \cdot \sigma_{\alpha^{\mathrm{SD}}}(k) \tag{21}$$

where $z(\rho) = \sqrt{-2\ln(\rho) - \ln(2\pi)}$ and $\mathrm{PLP}^{*}_{\mathrm{EqB}}$ is the PLP upper bound.

Some important observations should be outlined with respect to the concept of equivalent bandwidth in this context.

It is important to remind that there is no EqB tool for the bandwidth computation of the joint combination of PLP and AD in the literature. EqB techniques are in general suited for PLP only. The delay control may be also matched in parallel to the applied EqB method by properly dimensioning the buffer size (e.g., [14]). The EqB technique here is also the one of [15], where EqB is applied in similar heterogeneous conditions to drive CAC decisions. It is chosen as performance comparison because it is the only applicable closed-form expression in heterogeneous traffic conditions.

Another important note concerns the definition of EqB in the presence of QoS heterogeneity within SD layer aggregate. EqB is the minimum rate allocation necessary to guarantee a specific QoS to a single statistically homogeneous flow. The definition is generalized to be applied in the SI-SAP, since the minimum θ^{SD} satisfying all the QoS levels of the SI classes is the target. EqB approach may be adapted to this need, as done here, by choosing the parameter PLP^{*}_{EqB} as the most stringent PLP^{*} required at the SI layer and by properly dimensioning the buffer size to control the most demanding AD^{*} constraint. An example is reported in the next section.

Having in mind the static and dynamic configuration scenarios outlined in Section 2, it is remarkable that both EqB and RCBC can support both real time control in dynamic conditions and resource planning in static scenarios. Some examples are reported in the following.

7. Performance analysis

To test the proposed control methodology, a C++ simulator has been develop for the SI-SAP queues, having in mind the aggregation architecture shown in Fig. 5. The simulator has been developed from scratch to this aim. However, it exploits some validated software packages available in our department to support scientific works in the field of simulation analysis of discrete event systems.

The aims of the performance evaluation are to check: (1) *RCBC real-time response*, to highlight both QoS preservation and adaptive reaction of RCBC to real fading variations, (2) *loss versus delay bandwidth need*, to outline RCBC capability for the exact computation of the SD rate when PLP and AD contend bandwidth resource, (3) *RCBC versus EqB*, to test RCBC bandwidth optimization with respect to other possible EqB approaches and (4) *RCBC planning support*, as an example of bandwidth provision sensitivity to traffic statistics.

7.1. Fading counteraction

An aggregate trunk of 50 VoIP on-off sources composes the traffic at the SI-SAP interface. For now, the cascade of a single IP buffer and a single ATM buffer is considered. 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 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 end-to-end performance objective of a VoIP flow for ITU P.59 is composed of: end-to-end loss below 2% and maximum delay below 150 ms. ATM is used as SD transport technology. The time horizon of the simulation scenario is 133.0 min. The SLA is $PLP^{*}_{VoIP} = 0.02$ and $AD^{*}_{VoIP} = 20$ ms. The buffer size is set to 1600 bytes (20 VoIP packets) at the SI layer and to 70 ATM cells for the SD layer. SI resource allocation has been dimensioned by simulation analysis to satisfy the given degree of QoS accuracy at the SI buffer. The OH lasts 1 min. The gradient stepsizes are kept constant, $\eta^{k}_{L_{\Delta\Psi}} = 1, \Psi = V, W$, without affecting convergence as in [16].

A real trace of time varying channel degradation (taken from [8]) affects the SD buffer service rate as depicted in Fig. 6. The process generates peaks of channel degradation, especially in the time interval [4800, 6000]. PLP and AD measured versus time at the SD layer are depicted in Figs. 7 and 8, respectively. Only 4 peaks of performance degradation (PLP and AD above the corresponding QoS requirements) appear and only in correspondence of reduction factor step change. It is important to remind that, in this case, when the fading shape varies, RCBC scheme is started from the beginning (i.e., giving the same bandwidth previously allocated in clear sky conditions), to stress the algorithm effort. In these hard conditions, the quick reaction and adaptation to fading variations is evident.

For almost all the time of the simulation, RCBC reallocations (17) are driven by the loss constraint, which reveals

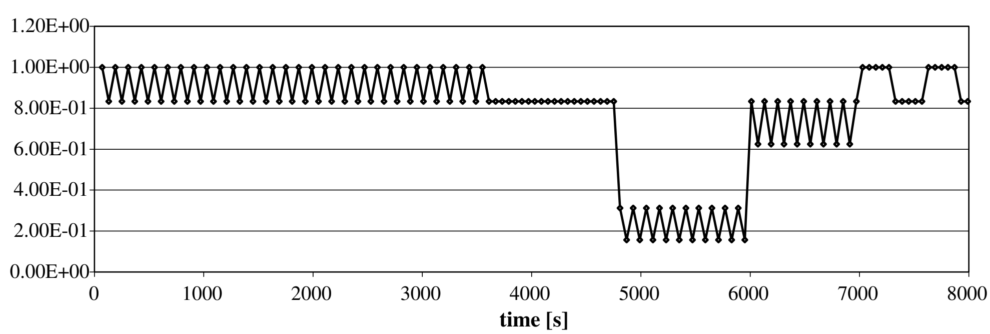

Fig. 6. RCBC real-time response scenario. Bandwidth reduction $\phi(t)$ taken from [8].

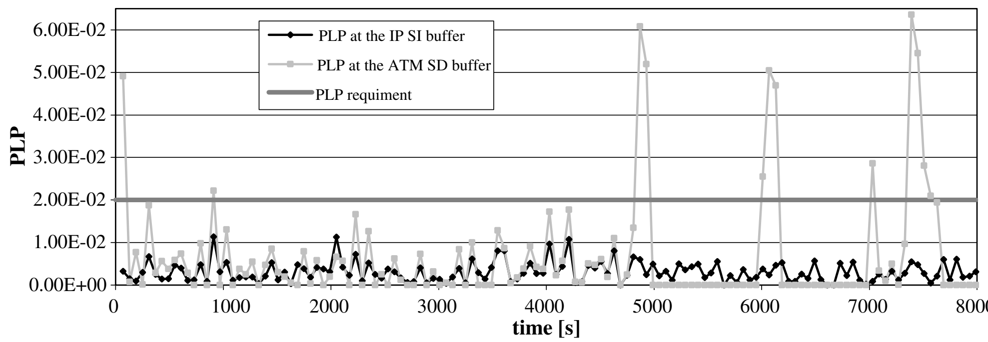

Fig. 7. RCBC real-time response scenario. PLP at the SI and SD layers.

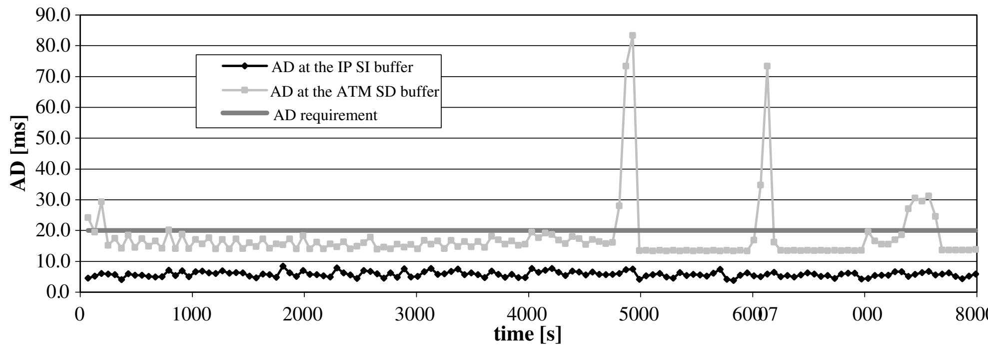

Fig. 8. RCBC real-time response scenario. AD at the SI and SD layers.

to be the most stringent QoS requirement through (19). The real operative thresholds, followed by RCBC, in this case, are not the fixed values 2×10^{-2} (for PLP) and 20 ms (for AD), but the PLP and AD really measured at the SI buffer, according to (11) and (12). This choice does not violate the performance constraints because PLP and AD at SI buffer are always below the required requirements, but highlights the dynamic behaviour of RCBC and its ability to track highly time varying targets. The bandwidth allocations of the SD and SI layers are compared versus time in Fig. 9. RCBC allocations include the additional bandwidth assigned to match fading counteraction.

From the results presented, it is clear that RCBC effectively produces a quick adaptive response to the channel variations and is able to keep the desired QoS. SD, PLP and AD, averaged over the entire simulation horizon, are 1.62×10^{-2} and 17 ms, respectively, which are below the performance requirements.

It is important to note that this scenario allows highlighting the RCBC's real time "active learning" capability. Actually, optimal rate allocations may be "memorized", in real time, in correspondence of experienced fading values and with given number of connections in the flows. Hence, SD allocation can be optimally tuned for the rest of time without further exploration steps (and consequent performance degradation) until a "new" traffic or fading configuration takes place.

7.2. PLP versus AD

Fig. 10 compares the normalized cost derivatives expressed in (20), as function of SD buffer size and by increasing from 10 to 90 the number of VoIP calls. The

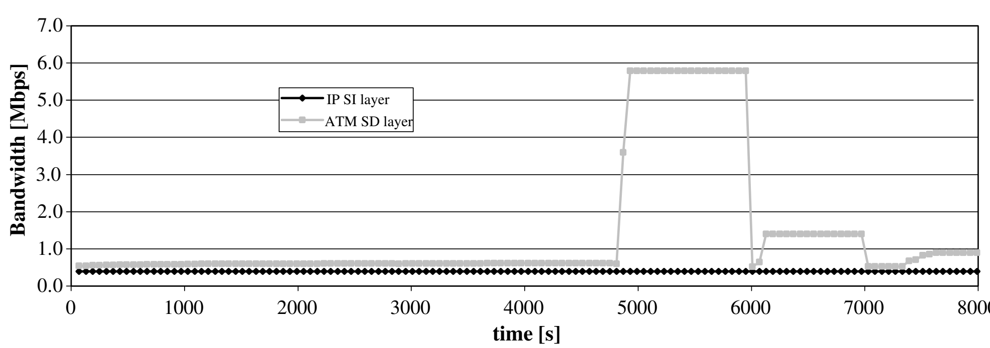

Fig. 9. RCBC real-time response scenario. SD allocations.

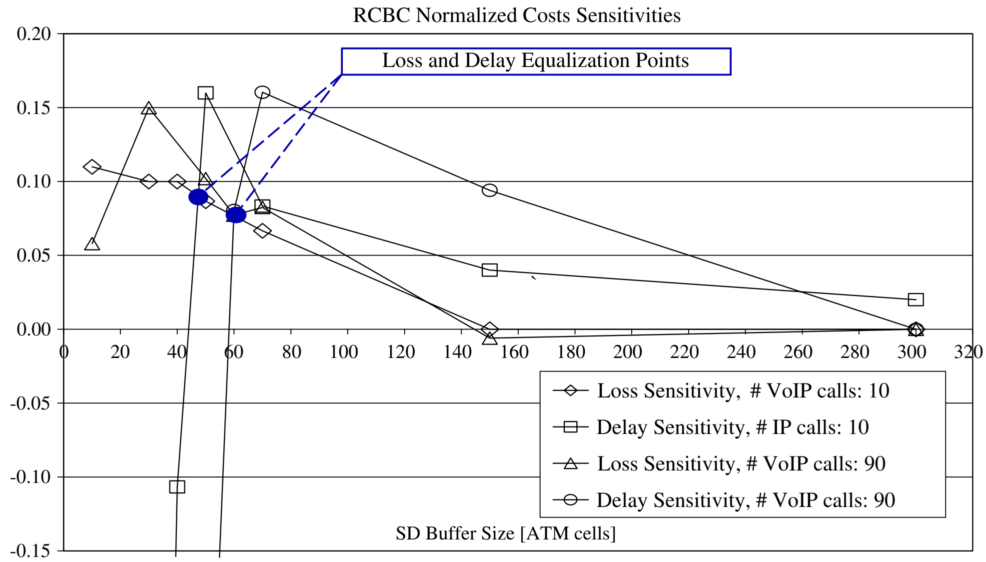

Fig. 10. PLP versus AD scenario. Loss and delay cost sensitivities as function of SD buffer size.

quantities reported are the result of averaging the instantaneous values of the cost derivatives achieved during the gradient descent over different repetitions of 1500 s of simulation to achieve a confidence interval less than 1% for 95% of the cases. No fading affects the satellite channel.

The analysis outlines the impact of the chosen QoS metrics on bandwidth dimensioning, for the specific VoIP over ATM case. As expected, the loss volume derivative function increases when the SD buffer size decreases. On the other hand, the delay derivative function shows the opposite behaviour, it increases in the SD buffer size. Thus, it is easily observable that for each combination of VoIP calls in the trunk, there is an intersection point (depicted in Fig. 10 as a black bullet) between the loss and delay cost derivatives. It is called *Loss and Delay Equalization* (LDEq) point. It represents the value of the buffer size where PLP and AD meet the same cost sensitivity. The location of the LDEq points is an increasing function of the traffic intensity. The curves corresponding to the range from 30 to 80 VoIP calls (not reported in Fig. 10 for the sake of picture clarity) confirms the analysis. The behaviour explicates the underlying principle of (19): for larger values of the buffer size above a given LDEq point, the gradient descent is driven by the delay derivative and, for smaller values, by the loss one. At the best of the authors' knowledge, there is no analytical tool in the literature capable to obtain the performance curves of Fig. 10 for PLP and AD metrics.

7.3. RCBC versus EqB

In this scenario, the presence of two traffic buffers at SI layer is considered without fading degradation at SD physical level. The first buffer offers the same VoIP service mentioned above. The second one is dedicated to video service. "The Firm" video trace, taken from the web site referenced in [17], is used. Video data are H.263 encoded and have an average bit rate of 64 kbps as well as a peak bit rate of 360 kbps (burstiness 5.7, average packet size 1132 bytes having standard deviation 411). SI rate allocation (240 kbps) for video (fixed off line through simulations analysis) assures $PLP^{*}_{video} = 10^{-3}$ and $AD^{*}_{video} = 50$ ms, which composes the video SLA (SI video buffer size is 75,000 bytes). 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. SD buffer is of 50 DVB cells (about 8 average video packets). Differing from the previous simulation scenario, RCBC tracks the SLA performance thresholds, independently to the measured SI performance by following (7) and (8).

The OH amounts of 3.0 min. Gradient stepsizes are fixed to 0.095. These values, found by simulation inspection, allows obtaining quick RCBC convergence while limiting bandwidth oscillations.

Fig. 11 depicts the RCBC bandwidth saving (in percentage) with respect to an EqB technique matching loss and delay together heuristically. The heuristic fixes the *Maximum Transfer Delay* (MTD) to achieve a given degree of accuracy for the delay performance. In this case, the required MTD amounts of 20 ms, having in mind the most stringent AD requirement between VoIP and video. Actually, there is no way for the heuristic to differentiate the QoS performance achieved by the SI flows (VoIP and video) in the SD trunk. The SD allocation is updated by the known EqB formula (21) presented in Section 6 and by properly dimensioning the buffer size to guarantee MTD around 20 ms. Being PLP^{*}_{EqB} the upper bound on the allowed PLP, again, it must be chosen as the lowest PLP of the SLAs (10^{-3} in this case). As said, no other EqB closed forms are applicable in place of (21), since no other traffic descriptors (such as *peak bandwidth*, *burstiness*)

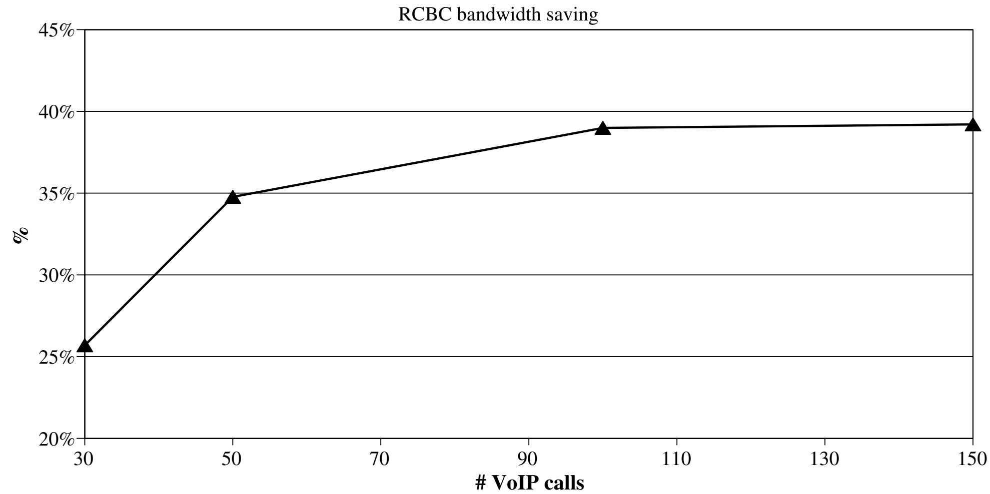

Fig. 11. RCBC versus EqB scenario. RCBC bandwidth saving over EqB heuristic.

can match the statistical heterogeneity of the α^{SD} process (see also [15] for a similar aggregation approach in a DiffServ environment).

To counteract the estimation error introduced by (21) and to avoid excessive rate oscillations (both dependent on the dimension of the OH), the EqB computation should be accurately tuned. Simulation analysis demonstrated that VoIP and video PLPs are matched, in the presence of small buffer sizes, if (20) is computed with respect to the maximum achievable values of $m_{\alpha^{SD}}$ and $\sigma_{\alpha^{SD}}$ (obtained in correspondence of the maximum number of active VoIP connections), thus ignoring the multiplexing gain obtained by the aggregation of the VoIP sources. The approach, in any case, is more bandwidth saving than a simple per peak bandwidth allocation.

To summarize, the heuristic performs a sequence of overprovisioning steps: it keeps the most stringent QoS requirements from SI SLAs and takes as references both the MTD (instead of the AD) and the worst traffic conditions. As a consequence, the bandwidth saving obtained by RCBC is outstanding (see Fig. 11). It is an increasing function of the traffic intensity because keeping small the buffer size to achieve the MTD target has a larger impact of the bandwidth need as the number of VoIP calls increases. Fig. 11 highlights that bandwidth dimensioning for MTD metric is much more expansive than optimizing the AD bandwidth need chased by RCBC. Moreover, the optimal solution of the QoSMO problem is automatically obtained by RCBC, while dimensioning both the buffer size and the bandwidth need with the heuristic requires a sequence of trial-and-error steps, thus implying a difficult planning of SD resources that hardly can be applied in real time.

Finally yet importantly, it is worth noting that the performed performance evaluation highlights a possible limitation of RCBC scheme in the presence of traffic classes having performance requirements differing of one order of magnitude (e.g., $PLP^*_{video} = 10^{-3}$ versus $PLP^*_{VoIP} = 2 \cdot 10^{-2}$). In this case, the convergence towards the satisfaction of the most stringent requirement may be affected by the (over) satisfaction of the other requirements. In the case presented, the over satisfaction of PLP for VoIP compensates the dissatisfaction of video PLP during the computation of the gradient estimation component of (17). The RCBC convergence towards the satisfaction of PLP^*_{video} is therefore slowed down. Such a problem may be solved by the application of a slightly different RCBC descent, for example by exploiting only the most stringent requirement of each QoS metric in (17). This topic is left open for future research.

7.4. RCBC planning support

Fig. 12 depicts the increase in percentage (with respect to the SI layer) of the bandwidth provision at the SD layer necessary to guarantee the SLAs outlined in the previous simulation scenario. The simulation setting is identical to the one above, too. This time, however, different video traces are applied (again taken from [17]). Video data are H.263 encoded and have an average bit rate of 260 kbps as well as a peak bit rate ranging from 1.3 to 1.5 Mbps, depending on the specific trace. The SI rate allocation for video is around 350 kbps (depending on the video trace in play) to assure the mentioned QoS thresholds.

The increase is computed as:

$$\text{SD additional bandwidth} = \frac{{}^{\text{Opt}}\theta^{SD} - (\theta^{SI}_{VoIP} + \theta^{SI}_{Video})}{\theta^{SI}_{VoIP} + \theta^{SI}_{Video}} \quad (22)$$

The ${}^{\text{Opt}}\theta^{SD}$ quantity is obtained through RCBC and guarantees the maintenance of the required SLA for VoIP and video services within the DVB portion.

As mentioned in Sections 2 and 3, the QoS guaranteed over terrestrial portions (external to the satellite core) should be guaranteed at SD level. For example, the SLA between providers may consist of assuring the same levels

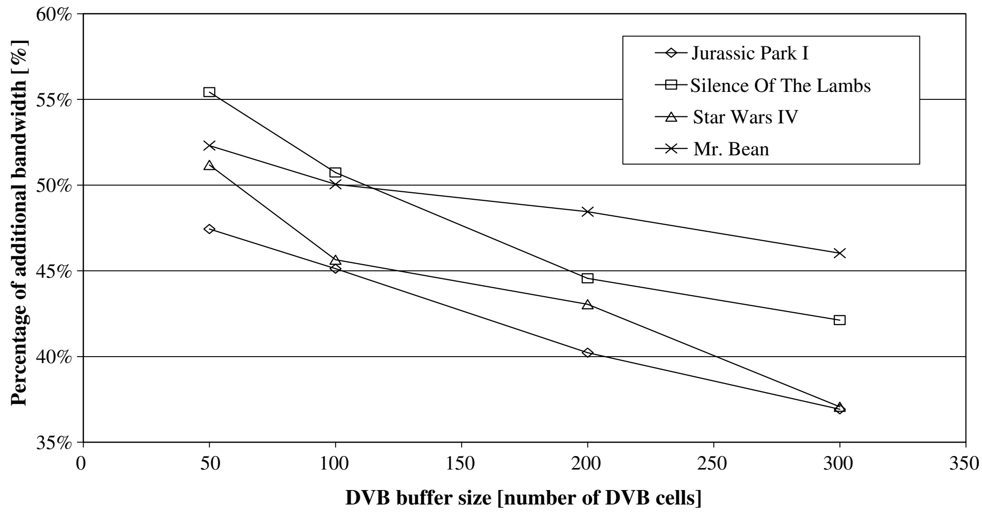

Fig. 12. RCBC planning support scenario. Bandwidth increase at the SD layer, measured by RCBC-based simulation inspection.

of QoS, despite technologies changes at SI-SAP (in this case from IP to IPoDVB). Technology changes should be transparent to end users.

In this perspective, RCBC reveals to be a suitable instrument for network planning, too. From the operative viewpoint, if a SD static trunk must be provisioned for the aggregation case presented, the bandwidth provision of the traffic entering the SD queue must be increased up to 50% in order to assure the SI QoS.

It is worth noting that the 50% of bandwidth increase holds true in the worst case, namely, with 50 DVB cells in the SD queue. For other buffer allocations, the increase is lower and depends on the specific trace in play. The application of different traces is studied by simulation inspection. The simulation analysis is driven by RCBC to speed up the QoS equalization between SI and SD queues. The analysis may be extended by considering a variable number of video and VoIP sources. The procedure is performed off line and may help the network operator to measure resource reservation sensitivity as a function of video statistical behaviour. The results may be used to deploy a static trunk serving the aggregation of VoIP and video flows. Overprovisioning may be applied by considering the worst statistical behaviour, thus simplifying resource reservation over a large time scale.

8. Conclusions and future work

The paper has presented the key elements of a protocol architecture used to map the QoS between the layers of a communication protocol stack. The solution of the *European Telecommunications Standardization Institute* (ETSI), related to satellite environments, has been taken as technological reference. A control scheme has been studied to allow bandwidth adaptation and consequent tracking of loss and delay performance metrics. The results have shown a good efficiency.

Directions for future research may rely on: (1) study of the control scheme applied to elastic traffic (e.g., TCP/IP) encapsulated in DVB frame, (2) application of the QID queues abstraction principle for cross-layer design in other wireless environments and (3) management of the delay jitter metric using the proposed control scheme.

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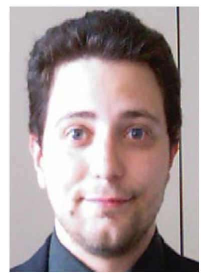

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