

# A Priority-based Processor Sharing Model for TDM Passive Optical Networks

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**Abstract**—The use of passive optical networks (PONs) enables access rates of multi-Gbit/sec bandwidth and provision of quality of service for high definition multimedia services. In this paper, we consider and analyse a generic multi-priority dynamic bandwidth allocation (DBA) algorithm for TDM PONs serving multimedia traffic in an upstream link. PON traffic is served strictly according to its priority. We consider this DBA algorithm using two approaches: (i) the algorithm assigns a fixed service quantum to each priority service and (ii) different service quanta are assigned to different priority services. The mean message delay is evaluated using a multiqueue processor sharing (MPS) model and an MPS with Heterogeneous Traffic (MPS-HT) model for the two approaches respectively. The MPS model is a classical processor sharing model limited by the critical assumption that there is egalitarian service sharing among all users, which is inefficient for multimedia applications in PONs. We extend the MPS model to a general MPS-HT model that enables the analysis of message delay performance in the case where the service quanta may be different for different services.

**Index Terms**—Passive optical network (PON), dynamic bandwidth allocation (DBA), multimedia traffic, performance evaluation, service quanta, delay.

## I. INTRODUCTION

A passive optical network (PON) is a point-to-multipoint access network laid between a central office (CO) of the network provider and customer premises. It consists of an optical line terminal (OLT) located at the CO, multiple optical network units (ONUs) at customer premises, and the fibers and passive optical splitters between them, forming the optical distribution network in a tree topology [1].

With the proliferation of multimedia applications in the Internet and advanced broadband technologies, subscribers are willing to pay more for more bandwidth to obtain better quality of service (QoS) from network operators. Within the access network, this growth in traffic demand is well addressed by

PONs. The use of optical fibre enables PONs to provide higher bandwidth than that of wireless technologies such as WiFi or WiMAX. Also, because of the shared passive optical network architecture and no active elements at public premises, its relatively low establishment and operating cost makes PONs economically viable.

The International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) and the IEEE Ethernet in the First Mile Task Force (IEEE EFM TF) have worked in parallel to develop their standards for PONs. The ITU-T first released the G.983.x series of recommendations [2] to specify the asynchronous transfer mode (ATM)-based PON (APON) and its enhanced version, Broadband PON (BPON). Both APON and BPON employ ATM cells to encapsulate the data transmitted between the OLT and ONUs. Later on, BPON evolved to Gigabit PON (GPON) under recommendation series G.984.x [3]. Two major improvements in progressing from BPON to GPON are higher data rates and better support of layer 2 protocols. The maximum transmission speed over GPON is 2.448 Gb/s for both upstream and downstream flows. Other than ATM cells, GPON supports Ethernet frames by using the GPON-encapsulation-method (GEM) frames. From the IEEE 802.3 working group, they have completed the IEEE 802.3ah standard [4] for Ethernet PONs (EPONs), which carry Ethernet frames with a 1 Gb/s symmetric transmission speed. Currently, under different standards, EPON and GPON are the commonly adopted technologies. A summary and comparison of these two PON technologies can be found in [5], [6].

All these PON technologies fall under the category of time division multiplexed (TDM) PONs where access to the medium is shared based on allocation of different time-slots to different users [7]. Due to the point-to-multipoint architecture, the transmission modes of downstream and upstream traffic are different. In the downstream mode, a wavelength is used to transmit data from the OLT to different ONUs during different time-slots. Data is broadcast to each ONU and each ONU reads only the data destined for it according to the packet's logical link ID (LLID). In the upstream direction, another wavelength is shared among all ONUs to send data to the OLT. Since only one ONU is permitted to transmit data during a time-slot, a medium access control (MAC) protocol is required to arbitrate the access to the shared upstream link by multiple ONUs. The OLT is responsible for this access control by assigning time-slots to individual ONUs for upstream transmission.

Multimedia traffic exhibits different traffic characteristics and QoS requirements. In order to satisfy diverse QoS re-

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quirements and utilize the bandwidth efficiently, a dynamic bandwidth allocation (DBA) algorithm is needed in the MAC protocol [8], [9]. However, DBA algorithms are not specified in the standards to give vendors/network operators flexibility to implement their preferred algorithms.

In general, there are two forms of DBA: status-reporting (SR) and non-status reporting (NSR). As there is no information about the ONU queues in NSR DBA, the system performance can be very poor. In SR DBA, the OLT polls ONUs for the queue status of each ONU traffic class and allocates bandwidth based on the reporting messages received. In EPON, the multipoint control protocol (MPCP) was developed for the MAC layer to enable the implementation of DBA algorithms [10]. MPCP is only a handshaking procedure with two messages: *REPORT* and *GATE*. The *REPORT* message is sent by the ONU with a request for the number of time-slots that the ONU needs. The *GATE* message is sent by the OLT with a grant of the transmission start time and duration to the corresponding ONU. For the status-reporting mechanism of GPONs, readers are referred to [11] for a detailed description. Due to control message formats and guard times are different for the EPON and GPON standards, research on DBA algorithms were mostly focus on either EPON or GPON.

For EPON, Kramer *et al.* [12], [13], [14] proposed a DBA algorithm, called Interleaved Polling with Adaptive Cycle Time (IPACT). The bandwidth is dynamically distributed based upon the results of interleaved polling. There is a difficulty in providing heterogeneous traffic QoS guarantees. To handle demands in a multi-service environment, a differentiation between services for QoS provision has been introduced in some DBA algorithms. Ma *et al.* [15] proposed a bandwidth guaranteed polling (BGP) scheme by dividing ONUs into two groups, those with guaranteed service and those with best-effort service respectively. Zhang *et al.* [16] developed a deterministic effective bandwidth - generalized processor sharing (DEB-GPS) scheduler to manage ONU flows. Each flow owns either a QoS queue or a best effort queue. In order to guarantee an upper bound on the delay for high-priority traffic, Kamal *et al.* [17] proposed a scheme which provides separate *GATE* messages for different priority classes. A DBA algorithm was proposed for multimedia traffic by Choi and Huh [18]. Strict priority queueing and control message formats were introduced to handle classified bandwidth using MPCP. Bandwidth is allocated to each flow of the ONUs which is assigned one of three priorities (high, medium and low) and prioritized scheduling is performed at the OLT level. Pereira *et al.* [19] proposed a novel scheduling policy, called Proportional Sharing with Load Reservation, which provides bandwidth guarantees on a per-flow basis and redistributes the unused bandwidth among active flows according to their priority levels. Moreover, based on [20], the bounds for the backlog and delay on a per-flow basis have been derived, thus enabling a network to support multimedia traffic with absolute performance guarantees. When strict priority scheduling is deployed, low priority queues may be starved by high priority queues. To alleviate this problem, Kramer *et al.* [21] proposed a two-stage queueing scheme in which packets are put in the second stage after a *REPORT* message is sent. Upon the

receipt of a *GATE* message, the second stage is served first. In [22], this technique is combined with a bandwidth guaranteed polling strategy to form a hierarchical scheduling scheme to support differentiated services. Dhaini *et al.* [23] proposed a DBA scheme based on the Deficit Weighted Round Robin service discipline to ensure fairness among different traffic classes. In the intra-ONU scheduling scheme proposed by Zhu *et al.* [24], packets are scheduled according to their delay bound requirements. Whenever it is not urgent to transmit the delay-sensitive packets, packets of best-effort traffic are scheduled first. Assi *et al.* [25] suggested a fixed cycle time-based DBA for QoS and a prioritized scheduling that is performed at the ONU level. Xie *et al.* [26] proposed a two-layer DBA scheme with a weight-based priority implementation and service differentiation to handle bandwidth demands of all ONUs. A comprehensive survey of EPON DBA algorithms has been provided in [27].

Compared with the research work conducted on DBA over EPON, publications that report on GPON DBAs are few. In [28], a DBA scheme using a new procedure for bandwidth reporting and a novel balance transferring mechanism was proposed for the GPON upstream link resource sharing by differentiated services. In [29], candidate technologies for next-generation PONs were evaluated from the perspective of bandwidth allocation.

In this paper, we consider and analyse a generic multi-priority DBA algorithm for TDM PONs serving multimedia traffic in the upstream link. Our approach can be applied in EPONs or GPONs. The algorithm is only based on status reporting. An ONU assigns different traffic types to different priorities and queues packets on a per-connection basis. Based on the reported queue status rather than queue sizes of each queue, the DBA algorithm executed at the OLT schedules the transmission times of packets from different connections. Traffic is served strictly according to its priority. We consider this DBA algorithm using two approaches. In the first one, the algorithm assigns a fixed service quantum (i.e. a fixed number of time-slots) to each priority service. In the second one, a different service quantum is allocated to different priority services. We then analyse the delay performance using a multiqueue processor sharing (MPS) model [30] and an MPS with Heterogeneous Traffic (MPS-HT) model for the two approaches respectively. Our new MPS-HT model is an extension of the MPS model, where users of different priorities receive a different amount of service at a given time instead of a fixed service portion as in the MPS model. An accurate closed-form approximation for the mean message delay is provided. Given that our closed-form approximation is simple, easily computable and captures traffic heterogeneity, it can be incorporated as part of a connection admission function and can be useful for PON dimensioning. Note that the term *message* refers to an application layer data-unit (e.g., http transaction or a video frame) which is broken down into a number of packets for transmission. In this way, we focus on user-perceived delay performance of admitted connections.

To support differentiated services in PONs, the approach addressed in previous work [15] - [28] is to grant bandwidth to different services according to their class requirements and

provide certain guarantees. The difference of these schemes is the way to determine the grant bandwidth for service classes. The performance analyses of these algorithms were based only on simulation studies. Such an approach normally requires a dynamic bandwidth allocation scheme that resembles a connection admission control (CAC) at the access. The benefit of such a scheme is that the access rate can be guaranteed for admitted calls. However, one drawback is that such a guarantee does not extend beyond the access point and calls may still suffer loss and delay elsewhere in the network. Furthermore, so far the Internet community has applied an approach that resembles processor sharing rather than that used in traditional telecommunication networks' CAC whereas our approach is essentially a compromise between these two approaches. For example, in [22], the guaranteeing bandwidth at the access does not always guarantee end-to-end QoS. However, since the bottleneck often occurs at the access, guaranteeing bandwidth at this point has its advantages. The scheme of [22] involves a cyclic scheduler whilst in our proposal we consider a processor sharing. Fundamentally, a processor sharing will provide better fairness over small time scales. However, if QoS is not compromised under the cyclic scheduler scheme, this may not be a significant advantage. The performance in [22] was studied via simulation. We aim here for an analytical solution and therefore our model is made simpler than the system of [22] for tractability.

The remainder of this paper is organized as follows. We describe a connection-oriented QoS architecture in the PON MAC layer involving service differentiation in Section II. Section III describes a generic priority-based DBA algorithm using a fixed service quantum. The MPS model is introduced to analyse the delay performance. In Section IV, we provide a range of numerical results of the DBA algorithm. In Section V, we improve on the DBA algorithm by using different service quanta and provide an analysis that leads to an accurate evaluation of the mean message delay, which is then validated using simulation. Conclusions are presented in Section VI.

## II. QoS ARCHITECTURE IN PON MAC

PONs provide QoS at the MAC layer. Fig. 1 depicts our proposed MAC QoS architecture in which a new connection needs to be policed based on the initialised request. Before a logical connection can be established between an ONU and an OLT, the ONU sends a connection request message to the OLT. The request includes the traffic characteristics and QoS-related information such as the requested bandwidth and the maximum tolerable delay. Each established connection will be assigned a unique LLID by the OLT, which will be used in status reports at the MAC layer and in the MAC protocol data-unit (PDU) headers. Then, source policing enforces the incoming traffic to be compliant with the characteristics specified in the traffic contract.

For service differentiation, we define four PON traffic classes: constant-bit-rate (CBR) service, real-time variable-bit-rate (rt-VBR) service, non-real-time variable-bit-rate (nrt-VBR) service, and best effort (BE) service. The main characteristics of these service classes are briefly reviewed below:

- **CBR** is designed to support real-time applications with a constant bandwidth requirement, such as T1/E1 and VoIP without silence suppression.
- **rt-VBR** is designed to support real-time applications with variable bit rates, such as Moving Pictures Expert Group (MPEG) video and VoIP with silence suppression.
- **nrt-VBR** is designed for applications without any specific delay requirement but with the need for a minimum amount of bandwidth.
- **BE** is designed for applications that are delay-tolerant and do not require a minimum bandwidth.

The four classes of traffic can be managed by means of the differentiated services (DiffServ) architecture [31] which allows traffic streams to be given different treatment from each other by a great variety of mechanisms. It allows our own defining mechanism. The term used to describe the DiffServ treatment at each ONU/OLT is "Per Hop Behaviour" (PHB). This could be achieved by assigning DiffServ tags to IP packets in accordance with the four categories of traffic we refer to – CBR, rt-VBR, nrt-VBR and BE.

Since multiple ONUs share a single uplink, only one ONU is permitted to transmit at any given time. To achieve such an exclusive access at any time, each ONU needs to be granted time-slots before it can transmit. A polling mechanism is deployed to allow each ONU to report the traffic backlog of each connection. Based on the status report of each connection, the DBA algorithm in the OLT allocates the upstream grants and schedules the transmission time of each ONU. The ensuing grant allocation for the current polling cycle is conveyed to the ONUs. Although an upstream grant is allocated according to individual requests from each connection, the grants are aggregated and given to an ONU to be distributed among its connections at its discretion. Therefore, upon receiving a grant, the ONU upstream scheduler revises scheduled access for one or more of its connections.

## III. PRIORITY-BASED BANDWIDTH ALLOCATION USING FIXED SERVICE QUANTUM

### A. Algorithm

Based upon the proposed QoS architecture, once the connection is established, according to the received connection request message combining information about the traffic characteristics, the OLT polices the upstream traffic arriving at an ONU and classifies it into the one of four service classes. As the CBR traffic is allocated dedicated bandwidth, it does not share the granted bandwidth with other service classes. Accordingly, the DBA algorithm only needs to handle the other three services classes, i.e., rt-VBR, nrt-VBR and BE. Moreover, these three service classes are naturally assigned to high, medium and low priorities, respectively. Each connection has its own queue at its respective ONU. When a message arrives, it will be broken down into a number of packets and will join a queue. Each packet fits into one time-slot of an upstream frame. In our priority-based DBA algorithm, using the assigned LLID at each connection, an OLT always grants rt-VBR connections to be served first. Then, only if all rt-VBR connections have no packet waiting, will the nrt-VBR

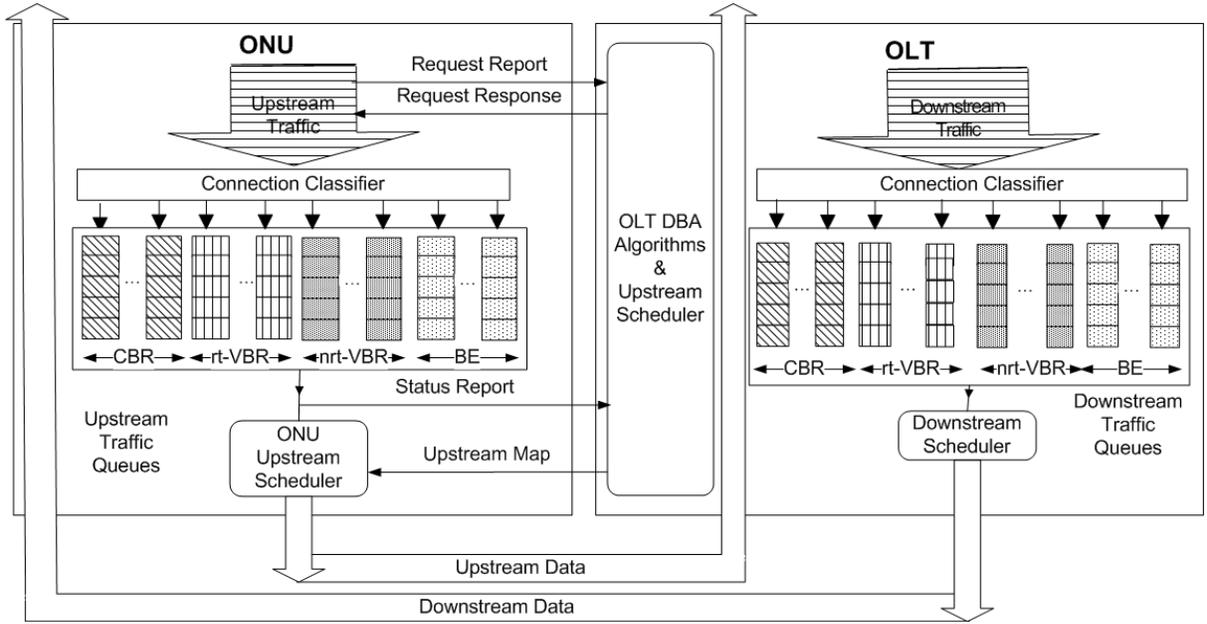


Fig. 1. PON upstream QoS architecture.

connections be served. Similarly, only if all rt-VBR and nrt-VBR connections have no packet waiting, will BE connections be served. Whenever there are multiple active connections of the same priority, the algorithm serves one packet from each connection in a round robin fashion. Since the lowest priority traffic is very tolerant and constitutes a large proportion of the traffic, a PON can be dimensioned to provide the required level of service to all traffic classes.

### B. Analytical Model

The delay performances of different service classes under our proposed DBA algorithm can be analysed by the MPS model [30]. Here, we analyse the message delay of each service class, which is the time between the arrival and departure of a message at an ONU, excluding the propagation delay for the transmission to the OLT. Introducing priorities to Kleinrock's round robin processor sharing model [32], the MPS model consists of a number of groups of distributed local queues (LQs) and a central server with a processor sharing (PS) queue. A LQ corresponds to a buffer queue of an ONU connection and the central server corresponds to an OLT upstream scheduler. Group- $p$  LQs contain  $M_p$  LQs with priority  $p$ ,  $p = 1, 2, \dots, P$ , where 1 is the highest priority. The central server performs prioritized round robin processor sharing among LQs by allowing no more than one message from each LQ to be present in the PS queue. Only when the service of an entire message is completed, is its LQ allowed to transfer another message into the PS queue. Fig. 2 shows an example of the MPS model for a two-priority case. LQs 1 and 2 belong to the low priority and LQs 3 and 4 belong to the high priority. When the MPS model is applied to a PON, the PON traffic is modelled by different priorities.

MPS is a discrete-time model whereby time is divided into equal-length time-slots. It assumes that messages arriving

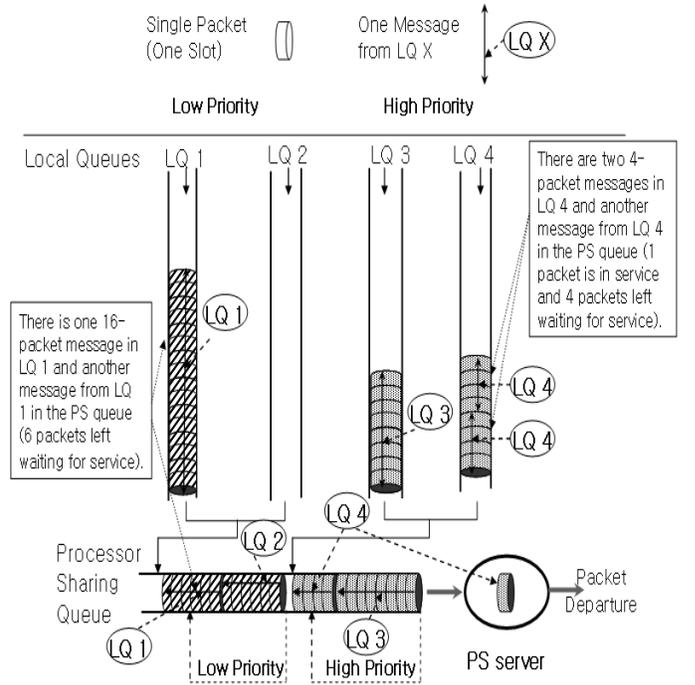


Fig. 2. The MPS model for the case of two priorities.

at the LQs consist of an integral number of packets, each requiring a service time of a single time-slot. A packet here corresponds to a MAC layer data unit representing an uninterrupted quantum of service time received by the message from the MAC PS server. A MAC layer normally operates using units called slots each of which consists of a fixed number of bytes. In the MPS model, one packet is equal to one slot. In the snapshot, the shared processor serves one packet from an LQ 4 message. Note that only one packet, i.e. only

one slot, can be served at a time when a message is holding the service token. It further assumes the following:

- 1) All messages arrive at a time-slot boundary.
- 2) For each  $p$ , the number of priority  $p$  messages arriving at an LQ within each time-slot is independent and identically distributed (i.i.d.) and is also independent of arrivals to other LQs.
- 3) The number of packets contained in a message (the message length) are discrete i.i.d. for each priority. The distribution of message lengths may be different for different priorities.
- 4) The transmission of a message can only be interrupted by messages from higher priorities or from other connections of the same priority after the current packet is completely transmitted, i.e. until at the end of this time-slot.

The mean delay  $D_p(n)$  (in units of time-slots) of a priority  $p$  message of length  $n$  packets is simply given by

$$D_p(n) = L_p + S_p(n), \quad (1)$$

where  $L_p$  is the mean time spent by a priority  $p$  message in its LQ, and  $S_p(n)$  is the mean time of a priority  $p$  message, consisting of at least  $n$  packets, spends in the PS queue to complete the services of  $n$  packets. Note that  $L_p$  is not related to the message length  $n$ .

Let the random variable  $a_p$  represent the number of priority  $p$  message arrivals within a time-slot to any priority- $p$  LQ. We denote the mean of  $a_p$  as  $\bar{a}_p$ . Let the random variable  $b_p$  be the priority  $p$  message length with mean  $\bar{b}_p$ . Since a packet transmission requires a time-slot,  $b_p$  also represents the message transmission time in units of one time-slot. Let  $C_{a,p}^2$  and  $C_{b,p}^2$  represent the squared coefficients of variation for  $a_p$  and  $b_p$  respectively. The total arrival rate for priority- $p$  traffic  $\lambda_p = M_p \bar{a}_p$ ; the total load of priority- $p$  traffic  $\rho_p = \lambda_p \bar{b}_p$  and let  $\varepsilon_p = \sum_{i=1}^p \rho_i$ . According to [30], we have:

$$D_p(n) = \frac{\nu_p / \rho_p + \sum_{i=1}^p \nu_i / (1 - \varepsilon_p)}{2(1 - \varepsilon_{p-1})} + \frac{n - [\bar{b}_p(1 + C_{b,p}^2) + 1]/2}{1 - \varepsilon_{p-1} - \frac{M_p - 1}{M_p} \rho_p} + \frac{1}{2}, \quad (2)$$

where  $\nu_p = \rho_p \bar{b}_p (C_{b,p}^2 + \lambda_p C_{a,p}^2 / M_p)$ . The overall mean priority  $p$  message delay is simply given by  $D_p(\bar{b}_p)$ .

We assume a stable PON in the following discussion. Then, this MPS model can be used directly to calculate the mean message delay under our priority-based bandwidth allocation scheme. As CBR traffic receives a constant bandwidth, we do not need to consider it for dynamic bandwidth allocation. Connections belonging to rt-VBR, nrt-VBR and BE are assigned with priorities 1, 2 and 3, respectively. So  $P = 3$ .

Using Eq. (2), we are able to examine the impact on the mean message delay due to multiple PON service classes. Clearly, even when the nrt-VBR or BE loads are changed, rt-VBR messages still receive the same service rate because of priority protection. In other words, the mean message delay of rt-VBR service is only affected by its traffic characteristics

such as arrival rate, message length and their variation. However, for nrt-VBR and BE, their delay performance will also be affected by higher priority traffic.

## IV. NUMERICAL RESULTS

In this section, the MPS model will be validated by simulation. We first present the traffic model of a typical application for each PON service class defined in Section II. We then describe the simulation environment used to validate the MPS model. Based on these traffic models, simulation and analytical results for different traffic loads are compared.

### A. Traffic models

A considerable amount of research on traffic modelling has been carried out to investigate the characteristics of different traffic sources for various communication networks [33], [34], [35], [36]. Here, we refer to their results.

A typical application of an rt-VBR service is variable-bit rate video communication. With the large amount of bandwidth available in PONs, it is expected that high quality video communications will be commonly deployed. Therefore, we use MPEG-2 video trace files as video traffic sources in our simulation. For nrt-VBR, an example application is web access. A possible model for web access is to have the message arrivals modelled as a Poisson process and the message size following a Pareto cut-off distribution. Finally, the commonly used Poisson arrival process with exponentially distributed message sizes is chosen as a model for a BE traffic source. The parameters of each traffic source used in this paper are listed in Table I.

TABLE I  
ARRIVAL PROCESSES AND MESSAGE SIZE DISTRIBUTIONS OF THE TRAFFIC SOURCES WITH A PRIORITY ARRANGEMENT

Traffic sources	Arrival process	Message size distribution	Priority level
MPEG-2 trace files	Deterministic: 30 frames/s.	As per trace files [37].	1
Web source	Poisson, mean inter-arrival time: 0.2s.	Pareto-cut-off: $\alpha=1.1$ , minimum message = 4.5k bytes, maximum message = 2M bytes.	2
BE traffic	Poisson.	Exponential, mean = 5000 bytes.	3

### B. Simulation model

We have developed a special-purpose discrete event simulation program using C++ to validate the MPS model. The traffic models described in Section IV-A are used to generate the input traffic. Packet traffic from different service classes is processed according to the QoS architecture as described in Section II. The class-based buffer sizes in the simulation are unlimited in accordance with the MPS model. The simulation has the following settings:

- 1) An aggregate bandwidth of 1.24 Gbits/s is assumed at the upstream link and shared by all connections of an OLT [5].

- 2) The bandwidth allocation in the simulation follows the algorithm described in Section III-A.
- 3) The overall utilisation is limited to be less than unity during the simulation in order to ensure that the system remains stable.
- 4) The packet/slot size is set at 100 bytes.

### C. Simulation and numerical results

We consider three scenarios according to different traffic loads. Analytical results and simulation results are compared in each scenario and presented in Fig. 3 to Fig. 8, respectively. Confidence intervals of 95% based on a t-test are obtained for all of the simulation results. The range of the confidence interval of each point of the rt-VBR simulation curves is within 4.5% which is hardly noticeable on the figures.

Firstly, we fix the load of nrt-VBR and BE traffic, and investigate the effect of the load of rt-VBR on the mean message delay by increasing the number of video sources. Traffic of lower priorities contribute about 65% of the load by 250 nrt-VBR sources and 150 BE sources, respectively. We set the mean inter-arrival time of the BE traffic to be 0.01s. The mean and variance of message sizes for different service classes are measured in the simulation and fed into the MPS model. The number of video sources increases from 50 to 200 and each source uses the MPEG2 trace file [37] with a random start frame. Results obtained from the simulation and analytical models are shown in Fig. 3 and Fig. 4. It can be seen that the simulation and analytical results are in good agreement, particularly for the rt-VBR and nrt-VBR traffic.

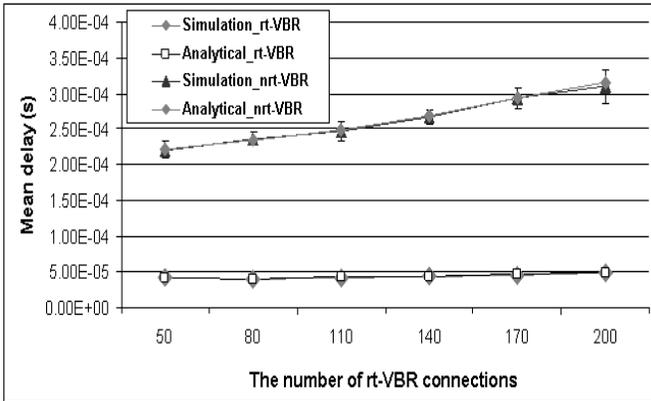


Fig. 3. rt-VBR and nrt-VBR message delays under different rt-VBR traffic loads.

In the second scenario, rt-VBR and BE traffic are fixed but the nrt-VBR traffic is changed by increasing the number of connections. We have 150 video sources and 150 BE sources in this scenario. Parameters of the BE traffic are set to be the same as in the first scenario: the mean inter-arrival time is 0.01s and the mean message size is 5000 bytes. The number of web sources increases from 50 to 320. Fig. 5 and 6 show that the delay for rt-VBR traffic is not affected much by the change of nrt-VBR loads while the increase of nrt-VBR connections leads to a dramatic growth in the BE traffic delay.

In the third scenario, we fix the load of rt-VBR and nrt-VBR traffics, with 150 video sources and 160 web sources

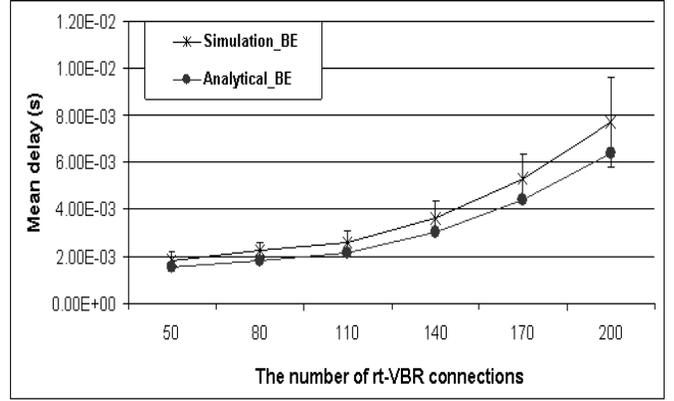


Fig. 4. BE message delays under different rt-VBR traffic loads.

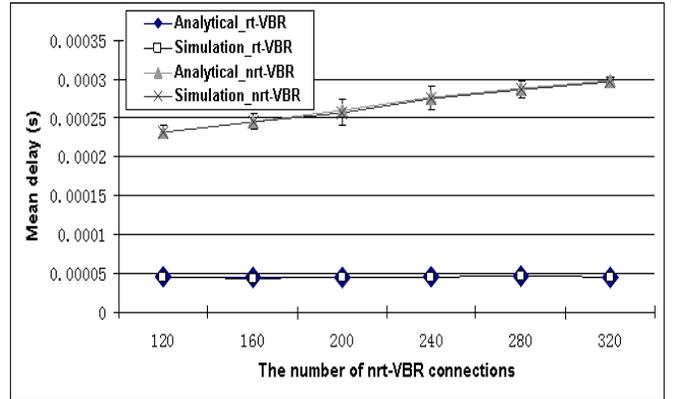


Fig. 5. rt-VBR and nrt-VBR message delays under different nrt-VBR traffic loads.

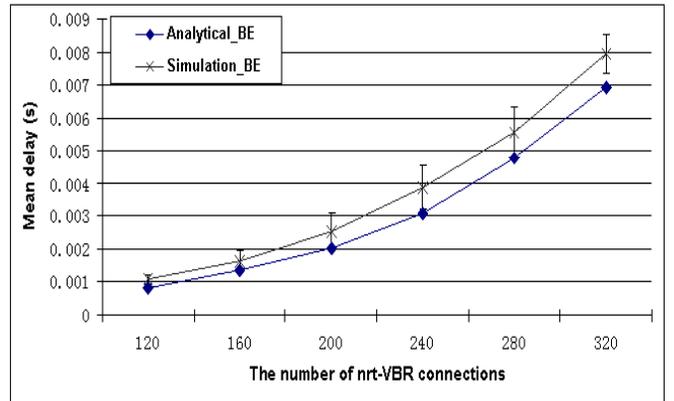


Fig. 6. BE message delays under different nrt-VBR traffic loads.

respectively. We vary the load of the 150 BE connections by varying their message inter-arrival times. Fig. 7 and 8 are plotted for this case. As expected, the loading of BE traffic does not affect the message delays of other higher priority traffic.

### V. IMPROVEMENT BY USING DIFFERENT SERVICE QUANTA

In traditional TDM systems, the allocation of bandwidth is quantum-based, i.e., a connection of any service priority can only receive a fixed amount of service quantum (a fixed number of time-slots). It is inefficient for multimedia applications with heterogeneous traffic characteristics as the quantum

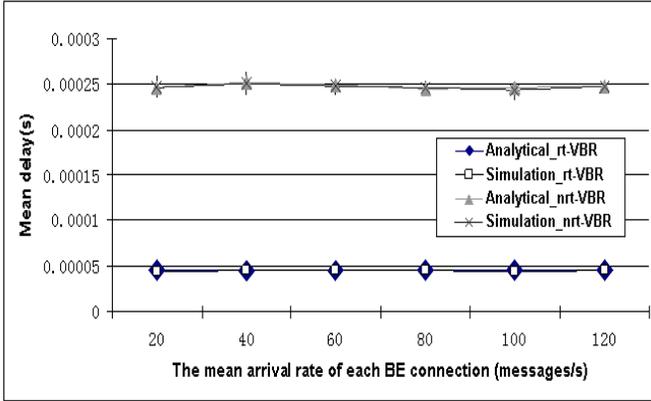


Fig. 7. rt-VBR and nrt-VBR message delays under different BE traffic loads.

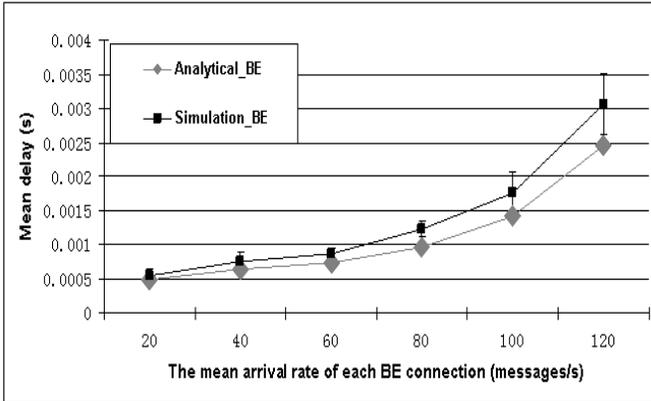


Fig. 8. BE message delays under different BE traffic loads.

is normally set according to a relatively small size request. We can improve such an allocation scheme by using different service quanta for different services. In this scheme, when a message arrives, it will be broken down into a number of packets the size of which are priority related. Each packet fits into a variable number of time-slots. As in the above described priority-based DBA algorithm, rt-VBR connections are always served first and the scheduler serves one packet from each connection in a round robin fashion among multiple active connections of the same priority. Moreover, when the traffic arrivals follow a Poisson process, we can extend the MPS model to the MPS-HT model and derive an accurate approximation for the mean message delay.

#### A. MPS-HT model

We illustrate our MPS-HT model by a two-priority (high and low) example depicted in Fig. 9. Consider a centralized processor shared by four LQs in a prioritized PS manner. LQs 1 and 2 are exclusively loaded with low priority messages and LQs 3 and 4 with high priority messages. Each LQ is assumed to have an associated infinite buffer.

Message lengths for each priority are discrete i.i.d.. Each message is assumed to consist of an integral number of packets; for example, the message in LQ 1 has two packets. As for the MPS model, a packet here still corresponds to a MAC layer data unit representing an uninterrupted quantum of service time received by the message from the MAC PS server

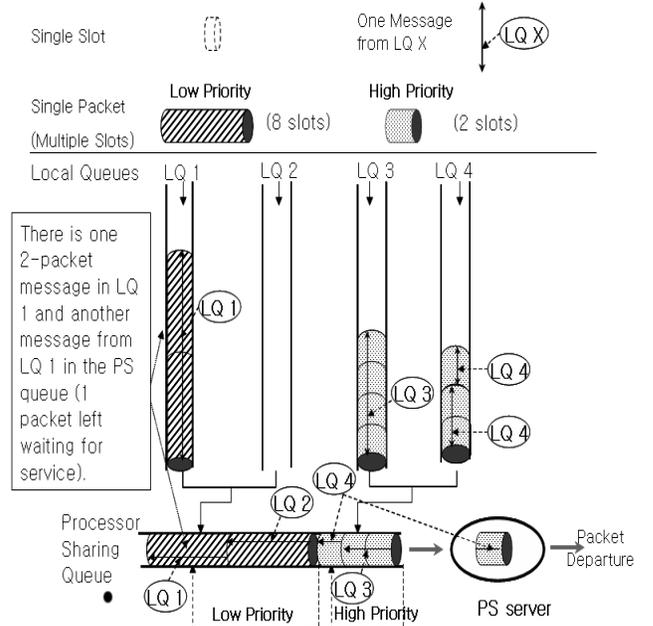


Fig. 9. The MPS-HT model for the case of two priorities.

and “slot” is still a MAC layer operating unit each of which consists of a fixed number of bytes. Slot sizes for packets may vary between priorities but, within a given priority, all packets consist of the same integral number of slots. For each priority  $p = 1, 2, \dots, P$  (where a smaller number indicates a higher priority), the length of a priority  $p$  packet is denoted by  $N_p$  [slots],  $1 \leq N_p \leq N$ , where  $N$  represents the maximum possible packet size. In the example presented in Fig. 9, the size of the low priority packet is eight slots and the size of the high priority packet is two slots. At any time, at most one message from each LQ can be present in the PS queue, i.e. a message from an LQ cannot move into the PS queue till the previous message from the same LQ has been completely served.

As for the MPS, time is divided into consecutive equal-length time units called *time-slots* which are related to the service time of *slots*. The service time of a packet is known as a *packet-time*. Let the time points at the beginning of each time-slot be designated by  $1, 2, 3, \dots$ , so that the  $k$ th time-slot is the time interval  $[k, k + 1)$ . Messages arriving at a priority  $p$  LQ within any time-slot follow a Poisson process with a mean arrival rate of  $\bar{a}_p$  [messages/time-slot]. Thus, the squared coefficient of variation for the number of arriving messages  $C_{a,p}^2$  is equal to  $1/\bar{a}_p$ . Let  $M_p$  still be the number of priority  $p$  LQs, so the total arrival rate of priority  $p$  messages is given by  $\lambda_p = M_p \bar{a}_p$ .

The PS server provides one-packet service to the message at the head of the PS queue and recycles the incomplete message to the tail of its own priority group, but ahead of all lower priority messages. In the snapshot presented in Fig. 9, a packet from LQ 4 is in service and the remaining packet of the same message is sent to the end of the high-priority group in the PS queue; and one packet of the incomplete LQ 3 message will be served next. We consider non-preemptive priority scheduling

at the packet level, i.e., any new message arrival, even from a higher priority, cannot interrupt the current packet's service.

Let  $b_p$  be a discrete random variable representing the priority  $p$  message size in packets. The message size is i.i.d.. Denote  $\bar{b}_p = E[b_p]$  and  $C_{b,p}^2$  is the squared coefficient of variation of  $b_p$  given by  $C_{b,p}^2 = \text{Var}[b_p]/\bar{b}_p^2$ . Thus, the mean size in slots of priority  $p$  messages is given by  $\bar{b}_p N_p$  and the traffic load of priority  $p$  messages is given by  $\rho_p = \lambda_p \bar{b}_p N_p$ ,  $0 < \rho_p < 1$ .

### B. Mean Message Delay

In the MPS model [30], the delay of an arriving message is obtained by summing up its waiting time in the LQ and the sojourn time in the PS queue. Notice that the method of [30] relies on the following two assumptions that we do not adopt here: (1) all messages arrive at a time-slot boundary; and (2) packets of different priorities are of the same size. Nevertheless, because each priority is considered separately, we can still develop an accurate approximation for the mean message delay by applying the results of [30] and then correcting the result using a compensation term. Accordingly, the mean delay of a priority  $p$  message consisting of  $n$  packets, denoted by  $D_p(n)$  (in units of priority  $p$  packet-time), is given by

$$D_p(n) = L_p + S_p(n) + \Delta_p, \quad (3)$$

where  $L_p$  and  $S_p(n)$  are as defined in III-B, and  $\Delta_p$  is the compensation term defined as the difference between the delay for a priority  $p$  message and the delay predicted by the MPS model. In the following, we derive these three components separately.

Since the transmission time of each packet in the MPS model is equal to one time-slot, to use that model in our approximation, it is convenient to consider the basic time unit to be a packet-time. In particular, when we evaluate the mean delay of a priority  $p$  message, we consider time to be measured in priority  $p$  packet-time units and the amount of traffic that arrives in units of priority  $p$  packet-time. It should be noted that packets of other priority messages may be neither the same size as priority  $p$  packets nor even an integer multiple of the priority  $p$  packet size. Furthermore, these other priority messages arrive during the priority  $p$  packet-time ( $N_p$  time-slots) and not all at once as in the MPS model. For our approximation, we do not consider the actual arrival process but instead we simply fit the first two moments of our arrival process and this has been found to give a reasonable approximation in practice, since MPS is also only based on two moments. We re-describe the arrival process of priority  $i$  messages in Table II.

The analysis in [32] (p.168) and Appendix I of [30] shows that  $S_p(n)$  is linearly increasing with  $n$ , which is based on the previous studies of the round robin processor sharing (RRPS) model [38], [39]. Moreover,  $S_p(0)$  in [30] was considered to be zero following the notation in the original round robin (RR) queue, representing the starting time that a customer enters the RR system. However, notice that the MPS model and our MPS-HT model, unlike round robin (RR) with a single

TABLE II  
PRIORITY  $i$  TRAFFIC LOAD CONVERTED INTO UNITS OF PRIORITY  $p$   
PACKET AND PACKET-TIME

The mean length of messages in units of priority $p$ packets	$E[b_i N_i / N_p] = \bar{b}_i N_i / N_p$ .
The mean number of message arrivals during a priority $p$ packet-time	$\lambda_i N_p$ .
The variance of the number of message arrivals during a priority $p$ packet-time	$N_p M_i \text{Var}(a_i) = N_p M_i \bar{a}_i^2 C_{a,i}^2$ , as $C_{a,i}^2 = 1/\bar{a}_i$ for Poisson arrivals, $N_p M_i \bar{a}_i^2 C_{a,i}^2 = N_p M_i \bar{a}_i = N_p \lambda_i$ .
The second moment of the number of priority $i$ messages arriving during a priority $p$ packet-time	$N_p \lambda_i + N_p^2 \lambda_i^2$ .

queue, is assumed to consist of LQs and a PS queue, where the propagation delay from an LQ to PS queue is assumed to be zero. A message arrives at LQ first, then enters the PS directly if there is no message belonging to the same LQ in the PS queue; otherwise, if such a message exists in the PS queue, the new arrival waits at the LQ, and then moves forward to the head of line (HOL) position of the LQ. For the second case, the HOL message is still waiting at its LQ when the last packet of the previous message belonging to the same LQ in the PS queue is moving out. However, following the consideration of the traditional RR, the PS queue was looked as the single queue [30]; the start time of the LQ HOL message moving into the PS queue was calculated from the start time that the last packet of the previous message belonging to the same LQ in the PS queue is served. For such a case the LQ HOL message was supposed to be in the PS queue already, but this is incompatible with the original assumption of the MPS model which requires that the HOL message should still be in the LQ. Therefore,  $S_p(n)$  is overestimated by (2) of [30]. This period should be taken into account in  $L_p$  rather than in  $S_p(n)$ . Let  $\delta_p$  denote the difference between the mean time that messages spend in PS queue for the MPS or the MPS-HT model and for the RRPS model. In the MPS or the MPS-HT model,

$$S_p(n) = n \left( \frac{Q_p(0)}{\lambda_p} \right) - \delta_p, \quad (4)$$

where  $Q_p(0)$  is the mean number of priority  $p$  messages in the PS queue which has not received any service. Note that the total delay  $L_p + S_p(n)$  in the MPS model has been proved correct [30]. In this paper, we demonstrate just the apportionments to LQ ( $L_p$ ) and to PS queue ( $S_p(n)$ ) are not quite accurate.

We derive  $S_p(x)$  by considering a priority  $p$  "test" message of length  $x$  packets as in [30], whose probability of occurrence does not affect the overall statistics. Then the time  $S_p(x)$  spent in the PS queue by this message must approach its own service requirement  $x$ , plus the time required by the total work for all messages, which arrive to the PS queue during its service and waiting time but before its last packet (i.e. the  $x$ th packet) starts service (the duration is  $S_p(x) - 1$ ).

These arriving messages include the arrivals to priority  $p$  LQs other than the one owning this test message (given by  $(S_p(x)-1)\rho_p(M_p-1)/M_p$ ), and the arrivals to all local queues of priorities higher than  $p$  (given by  $(S_p(x)-1)\sum_{i=1}^{p-1}\rho_i$ ). We have

$$S_p(x) \rightarrow x + (S_p(x)-1)\left[\frac{M_p-1}{M_p}\rho_p + \sum_{i=1}^{p-1}\rho_i\right], \quad \text{as } x \rightarrow \infty. \quad (5)$$

Defining  $\varepsilon_p = \sum_{i=1}^p \rho_i$ , we obtain:

$$\lim_{x \rightarrow \infty} S_p(x) = \frac{x}{1 - \varepsilon_{p-1} - \frac{M_p-1}{M_p}\rho_p}. \quad (6)$$

When  $x \rightarrow \infty$ , the slope of the mean PS delay is:

$$\frac{Q_p(0)}{\lambda_p} = \frac{1}{1 - \varepsilon_{p-1} - \frac{M_p-1}{M_p}\rho_p}, \quad (7)$$

which is identical to that shown for the MPS model [30].

Substitution of the slope in (4), we obtain

$$S_p(n) = \frac{n}{1 - \varepsilon_{p-1} - \frac{M_p-1}{M_p}\rho_p} - \delta_p. \quad (8)$$

Thus,

$$D_p(n) = L_p + \frac{n}{1 - \varepsilon_{p-1} - \frac{M_p-1}{M_p}\rho_p} - \delta_p + \Delta_p. \quad (9)$$

In a similar fashion to [30], we can also derive  $L_p$ . We use a discrete-time non-preemptive priority queueing model [40] as the equivalence to the MPS-HT model in terms of the average packet delay. Both queueing systems follow a strict priority discipline and are work-conserving based on the service disciplines at the packet level. If the total packet arrival process into both systems are identical and knowing that the two systems have equal average queue sizes and equal average arrival rates, by Little's formula, the mean delay of packet at each priority for both systems is equal. Notice that the packet delay distributions are different even though they have equal means, as the distribution of delay usually depends on the service order. So  $L_p$  can be obtained by equivalence in terms of the mean of priority  $p$  packet delay for the MPS-HT model and for the discrete-time non-preemptive priority model.

According to Appendix II of [30], we have

$$R_p(k) = \frac{1 - F_{b,p}(k-1)}{\bar{b}_p} \quad \text{for } k = 1, 2, 3, \dots, b_p,$$

where  $R_p(k)$  for  $k = 1, 2, \dots, b_p$  is the probability that a randomly selected priority  $p$  packet is the  $k$ th in its own message. Then, the mean delay of priority  $p$  packets is obtained as

$$E[D_{seg(p)}] = \sum_{k=1}^{b_p} [L_p + S_p(k)] R_p(k).$$

Substituting of (8) in the above equation yields

$$E[D_{seg(p)}] = L_p + \frac{[\bar{b}_p(1 + C_{b,p}^2) + 1]/2}{1 - \varepsilon_{p-1} - \frac{M_p-1}{M_p}\rho_p} - \delta_p. \quad (10)$$

For the discrete-time non-preemptive priority queueing model, the packets arriving in message batches with higher

priority than those currently being served, are assumed to enter service in the next packet interval and preempting service of lower priority packets which might be partially completed. Therefore, it is equivalent to the MPS-HT model which is preemptive resume at the packet level. For these packet-sized batch arrivals to the packets queue, using (6.5) of [40], we obtain the following cumulants of the i.i.d. packet arrival processes with the first two moments from the results in Table III. For the arrival processes of priority  $i$  messages on priority  $p$  packet level, we have

- mean =  $\lambda_i \bar{b}_i N_i = \rho_i$  [priority  $p$  packets / priority  $p$  packet-time],
- variance =  $\rho_i \bar{b}_i N_i (1 + C_{b,i}^2) / N_p$ .

Let  $\nu_i = \rho_i \bar{b}_i N_i (1 + C_{b,i}^2) / N_p$ . The total packet arrival process for priority  $p$  messages or higher has

- mean =  $\sum_{i=1}^p \rho_i = \varepsilon_p$ ,
- variance =  $\sum_{i=1}^p \nu_i$ .

Using these mean and variance expressions with (6.17) of [40] and adding one packet transmission time, we obtain the mean delay of priority  $p$  packets in units of  $p$  packet-time, given by

$$E[D_{seg(p)}] = \frac{\nu_p / \rho_p + \sum_{i=1}^p \nu_i / (1 - \varepsilon_p)}{2(1 - \varepsilon_{p-1})}. \quad (11)$$

The corresponding parameters for our case using (6.17) are shown in Table III. Note that  $\bar{B}_y$  in (6.17) of [40] is equal to 1 when we calculate the mean *packet* waiting time.

TABLE III  
CORRESPONDING PARAMETER NOTATIONS USED IN (6.17) OF [40]

Parameters in (6.17) of [40]	Parameters in MPS-HT model
$\rho_{I,y}$ : the traffic load from those priorities higher than $y$ .	$\varepsilon_{p-1}$ : the traffic load from those priorities higher than $p$ .
$E(\bar{N}^{(F_y)})$ and $E([\bar{N}^{(F_y)}]^2)$ : the first two moments of the number of packets from priorities $i \leq y$ .	$\varepsilon_p$ and $\sum_{i=1}^p \nu_i + \varepsilon_p^2$ : the first two moments of the number of packets from priorities $p$ or higher arriving during a priority $p$ packet-time, in the units of priority $p$ packet size.
$N_i \bar{B}_i$ : the mean arrival rate of packets from the priorities $i$ .	$\rho_i$ .
$\bar{B}_y$ : the mean message/packet size depends on the equation used for calculating the mean message/packet waiting time.	1: as we calculate the mean <i>packet</i> waiting time.
$\bar{N}_y$ and $N_y^2$ : the first two moments of the arriving process for priority $y$ messages.	$\rho_p$ and $\nu_p + \rho_p^2$ : the first two moments of priority $p$ packets arriving during a priority $p$ packet-time.

Equating (11) with (10), we obtain the following expression for  $L_p$ :

$$L_p = \frac{\nu_p / \rho_p + \sum_{i=1}^p \nu_i / (1 - \varepsilon_p)}{2(1 - \varepsilon_{p-1})} - \frac{[\bar{b}_p(1 + C_{b,p}^2) + 1]/2}{1 - \varepsilon_{p-1} - \frac{M_p-1}{M_p}\rho_p} + \delta_p + \frac{1}{2}. \quad (12)$$

Then, summing (8) and (12), we obtain

$$L_p + S_p(n) = \frac{\nu_p / \rho_p + \sum_{i=1}^p \nu_i / (1 - \varepsilon_p)}{2(1 - \varepsilon_{p-1})} +$$

$$\frac{n - [\bar{b}_p(1 + C_{b,p}^2) + 1]/2}{1 - \varepsilon_{p-1} - \frac{M_p - 1}{M_p} \rho_p} + \frac{1}{2}. \quad (13)$$

For the first part of the compensation, henceforth denoted  $\Delta_p(1)$ , when a priority  $p$  message arrives, it has to wait until the packet in progress completes its transmission – regardless of its priority – because we do not allow preemptions. At the same time, due to this delay, there are higher priority messages which arrive during the time, so the total work required by all these packets should be taken into account using  $\Delta_p(1)$ . Thus, denoting the waiting time as  $W_p$ , we have

$$\Delta_p(1) \rightarrow W_p + \Delta_p(1) \sum_{i=1}^{p-1} \rho_i. \quad (14)$$

We can evaluate  $W_p$  for a priority  $p$  message due to the transmission time of a priority  $i$  packet currently found in service. Since the message arrival process follows a Poisson process, this occurs with probability  $\rho_i$  (by the PASTA principle). The time that the priority  $p$  message has to wait until the priority  $i$  packet completes its service is estimated as half of  $N_i$  time-slots. We estimate

$$E[W_p] = \frac{\sum_{i=1}^P \rho_i N_i}{2N_p}. \quad (15)$$

Thus, we have

$$\Delta_p(1) = \frac{\sum_{i=1}^P \rho_i N_i}{2N_p(1 - \varepsilon_{p-1})}. \quad (16)$$

The part  $\Delta_p(1)$  only considers compensation associated with the initial delay. It is important, however, to notice that for the same reason that the high priority message experiences additional delay relative to the MPS model (because it has to wait until a low priority packet completes its transmission), the low priority message experiences a reduction of delay (relative to the MPS model) if a high priority message arrives during the transmission of the last packet of a low priority message. Notice that this reduction of delay, which we henceforth denote by  $\Delta_p(2)$ , is only relevant to the last packet. If the high priority message arrives during transmission of an earlier packet, the reduction of delay gained by the low priority message will be offset by a later delay of transmission of subsequent packets as they can only be transmitted after the transmission of the high priority message is completed and since the latter incurred initial delay, it will also be delayed in completing its transmission. Let  $J_p(x)$  for any  $p > 1$  be a function defined by

$$J_p(x) = \begin{cases} 1 & \text{if } x < cN_p, \\ 0 & \text{otherwise,} \end{cases}$$

where the factor  $c$ ,  $0 < c \leq 1$ , will allow us to disregard messages of priorities higher than  $p$  that their packet size is close to that of  $p$ . Let  $\lambda(p) = \sum_{i=1}^{p-1} \lambda_i J_p(N_i)$ . The rate  $\lambda(p)$  is the arrival rate of messages that we would like to consider in evaluating  $\Delta_p(2)$ . In particular,  $\Delta_p(2)$  is evaluated by the time elapsed from the moment of the first occurrence of a Poisson process with rate  $\lambda(p)$  within the last packet-time of our priority  $p$  message until the end of that packet-time. Since  $\Delta_1(2)$  is relevant to the arrival rate  $\lambda(p)$ , if  $\lambda(p) = 0$ ,

$\Delta_1(2) = 0$ . It is known that conditioning on the number of Poisson arrivals within a packet-time, the arrival times have the same distribution as the order statistics of the same number of uniformly distributed random variables within that packet-time. Conditioning and unconditioning on the number of arrivals in an interval of the packet-time of a priority  $p$  message  $N_p$ , we obtain,

$$\Delta_p(2) \cong 1 - \frac{1}{\lambda(p)N_p} \left(1 - e^{-\lambda(p)N_p}\right), \quad (17)$$

where  $p > 1$  and  $\lambda(p) \neq 0$ . Since  $\Delta_p(2)$  is relevant only for  $p > 1$ , we set  $\Delta_1(2) = 0$ . Overall,  $\Delta_p$  is estimated by

$$\Delta_p \cong \Delta_p(1) - \Delta_p(2), \quad p \geq 1. \quad (18)$$

Due to possible large variations in packet size for the different priorities, the compensation term  $\Delta_p$  is key to an accurate evaluation of the overall mean message delay. Especially,  $\Delta_p(1)$  can have a major effect on the result, as  $\Delta_p(2) \leq 1$ . We can multiply the result of  $D_p(n)$  by  $N_p$  to convert it back to the common “currency” of time-slots.

### C. Model Evaluation

In this section, we validate our approximation for the mean message delay using the simulation program described in Section IV-B which also incorporates the variable service quanta. PON traffic is served by the DBA algorithm using different service quanta as described before. The bandwidth is allocated in the units of packet size for each priority. The packet sizes are 300, 3000 and 500 bytes for priority 1, 2 and 3, respectively. The other settings are the same as described in Section IV-B.

The same scenarios for the MPS model in Section IV-C are considered here. Using a 100 byte slot, we calculate the mean message delay for priority  $p$  traffic,  $D_p[\bar{b}_p]$ , using (3), (13) and (18) with  $c = 0.8$ . Analytical and simulation results with 95% confidence intervals (which are too small to be noticed in many cases) based on a Student's t-test are presented in Figs. 10 to 15. The analytical results are in good agreement with the simulation results, especially for rt-VBR and nrt-VBR traffic.

Comparing with the results of the MPS model, the mean message delay of highest priority traffic is slightly increased and that of the lower priorities is decreased as we introduce a bigger service quantum for the lower priority traffic. The change in values depends on the selection of service quanta.

## VI. CONCLUSIONS

We have developed a multiservice multiqueue PS model for a generic DBA algorithm for TDM PONs serving multimedia traffics in the upstream link. Our model allows service quanta to be different for different services and provides an easily computable approximation for the mean message delay in a closed-form. We have generally observed agreement between analytical and simulation results for QoS guaranteed services. However, the approximation is somewhat inaccurate for best-effort service, especially under heavy loading conditions. Our simple closed-form approximation that captures traffic heterogeneity can be used in admission control functions and dimensioning of PONs.

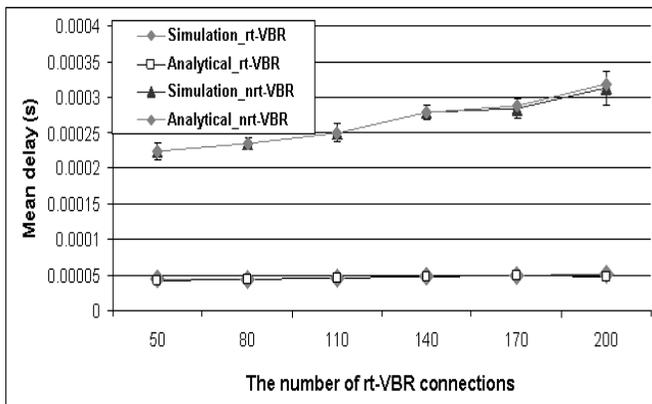


Fig. 10. rt-VBR and nrt-VBR message delays under different rt-VBR traffic loads.

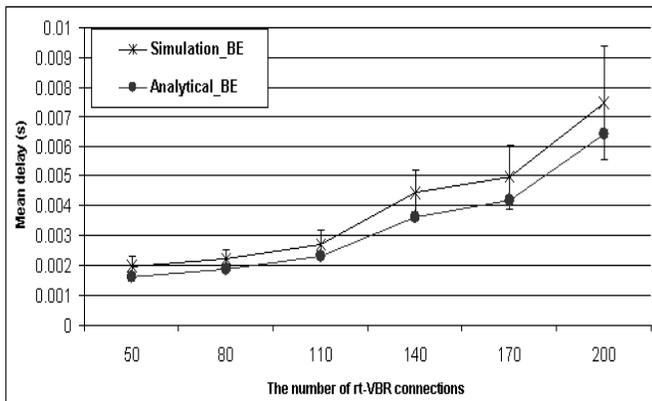


Fig. 11. BE message delays under different rt-VBR traffic loads.

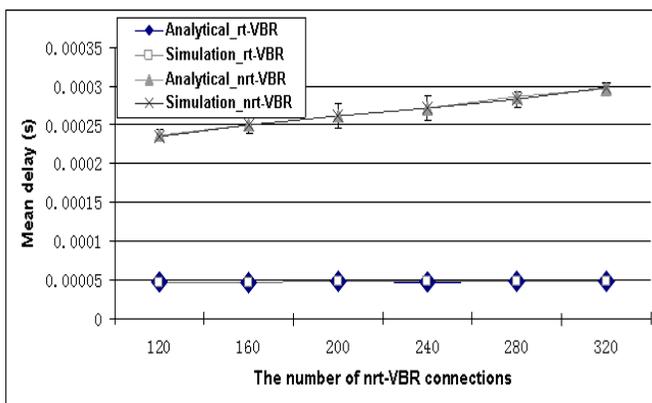


Fig. 12. rt-VBR and nrt-VBR message delays under different nrt-VBR traffic loads.

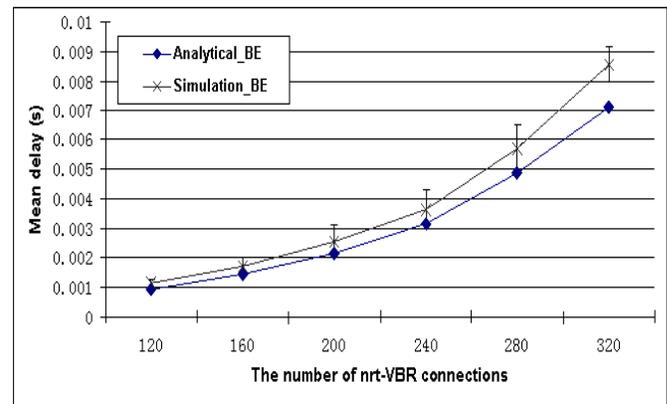


Fig. 13. BE message delays under different nrt-VBR traffic loads.

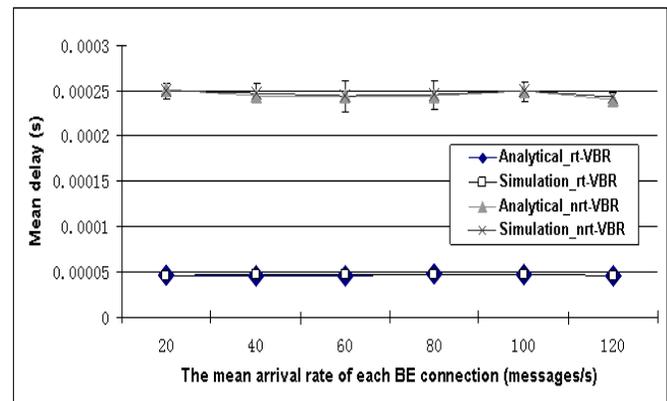


Fig. 14. rt-VBR and nrt-VBR message delays under different BE traffic loads.

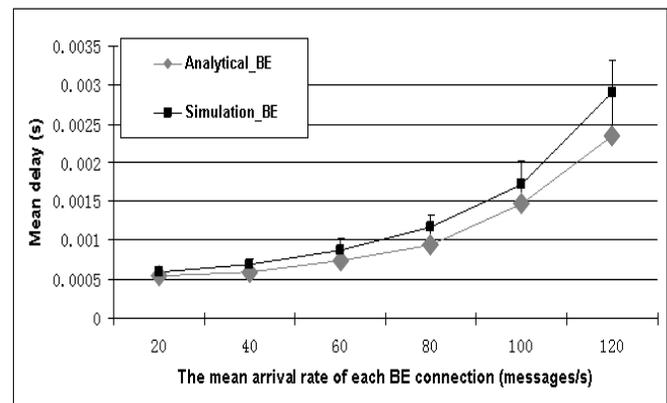


Fig. 15. BE message delays under different BE traffic loads.

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