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End-to-End Statistical Delay Guarantees in MPEG Video Transmission Over the Next Generation Internet

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ABSTRACT

In this paper, we analyze the end-to-end statistical delay performance of Motion Pictures Experts Group (MPEG) compressed video flows in the next generation Internet when the flows are served by earliest deadline first (EDF) scheduling discipline at each switch. We develop a model for an isolated scheduler fed by shaped MPEG sources, and extend the model for end-to-end statistical delay analysis for multiple hops. The statistical framework allows realizing much larger schedulable regions than the deterministic case. Significant gain in network utilization is observed for as low a delay violation probability as 10^{-5} compared to the deterministic case. Our framework could be highly useful in the design of connection admission control (CAC) schemes for MPEG video transmission over the next generation Internet.

I. INTRODUCTION

In order to provide quality of service (QoS) guarantee to real-time applications on Internet Protocol (IP) networks, the Internet Engineering Task Force (IETF) is actively developing new service models for the future generation networks. The first result of these efforts is the *Integrated Services* (IntServ, or IS) model, which comprise of two categories of service: the Guaranteed Service [19] and the Controlled Load Service [22]. However, deployment of IntServ requires state information about each flow to be stored in every intermediate switch, which does not scale well for large networks. Due to this scalability problem, the IETF is currently working on the *Differentiated Services* (DS, or DiffServ) model [4, 14]. In DiffServ, flows are not individually provided any QoS guarantees. Rather, different classes of service are offered with different QoS. Each of the classes can support the aggregate traffic generated by multiplexing a number of flows requiring similar quality of service. As a result of these developments, it is likely that the future QoS-enabled Internet will be composed of many IntServ domains connected by a backbone DiffServ domain as shown in Fig. 1.

In case of MPEG flows, the aggregate multiplexed traffic must be carried by a service class which can ensure that the QoS guarantees provided by the DiffServ backbone network is adequate for providing QoS guarantees to individual MPEG flows. In this paper, we develop a model to study the end-to-end delay for MPEG video transmission over the next-generation Internet where the intermediate network nodes provide a mechanism for differentiation of service. To support different types of QoS guarantees to real-time traffic as well as best-effort traffic, scheduling disciplines that are more sophisticated than the simple first in first out (FIFO) discipline are necessary at each switch in a QoS-enabled network. Generalized processor sharing (GPS) [15, 16] and earliest deadline first (EDF) [7, 21] are two such scheduling schemes.

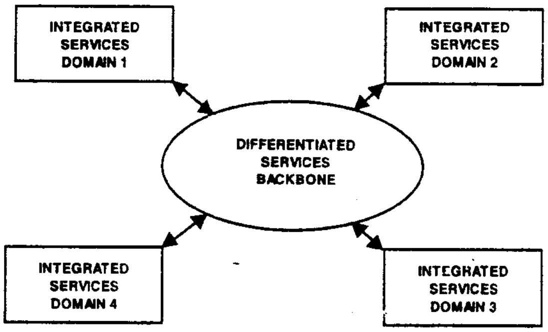

Fig. 1. An architecture for QoS-enabled next generation Internet.

In a QoS-enabled network, a connection admission control (CAC) mechanism must exist so that a new flow is admitted only if all admitted flows can be served without violating QoS requirements. The number of flows that can be supported with deterministic guarantees on delay bound have already been studied in the literature for GPS, EDF, and other service disciplines [8, 16]. However, these deterministic frameworks are excessively conservative in admitting flows into the network, since they account for the worst-case scenarios that can be encountered in the switches, even though such worst cases may occur with exceedingly low probabilities. Real-time applications are typically resilient to infrequent violations in their delay bounds. As a result, the need for a *statistical framework* arises where end-to-end delay bound is statistically guaranteed with a reasonable high probability. Such statistical frameworks allow the links to operate with much higher utilization, admit more flows, and still meet the QoS requirements of all the admitted flows.

Recently, statistical frameworks have been developed for priority scheduling [12] and GPS [6]. Also, statistical frameworks for EDF scheduling have been developed in a number of recent works [3, 17, 20]. We choose the EDF scheduling scheme because EDF has been shown to provide the optimal delay performance under deterministic setting [9, 13], and the advantages of EDF are also expected to be observed in the statistical setting [3, 20]. However, none of these papers address the issue of MPEG video transmission over the next generation Internet incorporating IntServ and DiffServ. In this paper, we develop an end-to-end model for MPEG transmission by combining a *traffic shaper* or *rate controller* model developed by the authors [1, 2] and an *EDF scheduler* model based on [3] applied to MPEG traffic flows in IntServ and DiffServ. We choose the model in [3] as our basis for EDF analysis due to the fact that this model is rather general and can be applied to any specific traffic arrival pattern.

The rest of the paper is organized as follows. In Section II, we discuss some fundamental concepts on MPEG video streams, traffic shaping, and EDF scheduling. In Section III, we develop an analytical model for studying the end-to-end delay characteristics. Section IV discusses analytical model and simulation results. We conclude this paper in Section V.

II. BACKGROUND

In this section, we discuss the characteristics of MPEG streams, traffic shaping of MPEG flows, and EDF scheduling.

A. MPEG Streams

MPEG video sequences are arranged into Group of Pictures (GoP). Each GoP contains three different types of frames: Intra (I), Bi-directional (B) and Predictive (P). At the beginning of a GoP, an I-frame is transmitted. After the I-frame, a number of B-frames are transmitted with P-frames inserted between the B-frames. A typical sequence of frames, for example, is IBB-PBB-PBB-PBB. The GoP structure is usually described as *MmNn* where *n* is the total number of frames in a GoP, and *m* is the I-P or P-P frame interval. During the transmission of the I-frame, a complete image is transmitted which makes the I-frame much larger in data content than other frames.

A real-time stream over the IntServ is established when an end application reserves bandwidth from the network using the Resource ReSerVation Protocol (RSVP) [5]. IETF has defined the Real-time Transport Protocol (RTP) for delivering real-time traffic on IP networks. Once a flow is set up by RSVP, an MPEG stream is carried as RTP payload encapsulated in IP packets [11].

B. Traffic Shaping of MPEG Flows

In an IntServ network, flows are policed by token bucket rate controllers [19, 22]. The token bucket rate controllers are characterized by three parameters: the peak rate p, the average rate r, and the bucket size b. In our model, we assume that the MPEG flows are shaped at the ingress of each node by a token bucket rate controller or traffic shaper. We also assume that all of the traffic shapers are identical since no advantage is gained by using shapers that are not identical [8]. In a multi-hop transmission, a properly shaped MPEG flow suffers from delay only once at the first traffic shaper, and no additional delays are introduced due to per node traffic shaping [10].

The delay suffered by a single *MmNn* GoP sequence in an MPEG flow while passing through a token bucket traffic shaper can be expressed as [1, 2]

$$d_{shaper} = \begin{cases} \dfrac{1}{f}\left[\tau n - 1 + R_I \dfrac{(1-\beta)}{r}\right] & \beta < 1 \\[2ex] \dfrac{1}{f}\left[\dfrac{\tau n}{\beta} - 1\right] & \beta \ge 1 \end{cases} \qquad (1)$$

where r is the average data rate of the GoP, which depends on the I-frame data rate R_I, the B-frame data rate R_B, and the P-frame data rate R_P as follows:

$$r = \frac{R_I + (n - n/m)R_B + (n/m - 1)R_P}{n}. \qquad (2)$$

The parameters β and τ in Eq. (1) are specified in terms of some other traffic parameters as follows. Let us assume that the number of frames per second is f, the peak rate of transmission is p, and the burst volume, which is the area under the peak-rate transmission (at rate p) in the shaper, is B. Then, the *normalized burst length* τ is defined as:

$$\tau = \frac{\text{Burst length}}{\text{GOP period}} = \frac{B/p}{n/f} \qquad (3)$$

which is the fraction of time the traffic shaper transmits at the peak rate compared to the time taken to transmit one GoP. Also, the *normalized burst volume* β is defined as:

$$\beta = \frac{\text{Burst volume}}{\text{I frame data size}} = \frac{B}{R_I/f} \qquad (4)$$

which is the ratio of the burst volume to the I-frame data size in the MPEG video sequence.

C. EDF Scheduling

Each switch in the path of the MPEG flows is assumed to serve the flows by EDF scheduling discipline. The EDF scheduling principle [7, 21] is as follows. Each flow i at switch m is associated with a local delay bound T_i^m. An incoming packet of flow i arriving at the scheduler at time a_p, is stamped with a deadline $d_p = a_p + T_i^m$, and the packets in the scheduler are served by increasing order of their deadline.

In the deterministic setting, EDF is known to be the optimal scheduling policy at a single switch [9]. Let the function $A_i(s, t)$ represent the total traffic inflow in the interval $[s, t]$. Given N flows with traffic arrival envelopes $A_i(0, t)$ $(i = 1, 2, \dots N)$ sharing an output link with delay bounds $T_1, T_2, T_3, \ldots T_N$ respectively, the authors in [9, 13] show that the schedulable region for EDF scheduling satisfies

$$\sum_{i=1}^{N} A_i(0, t-T_i) \le Ct \qquad t > 0 \qquad (5)$$

where C denotes the link rate, and it is assumed that packet transmission time is negligible compared to the queuing delay. In case of a token bucket (r, b, p) controlled source as discussed in Section II(B), the traffic envelope is bounded by $A_i(s, t) = \max\{p(t-s), b + r(t-s)\}$.

III. ANALYTICAL MODEL

In this Section, we develop our analytical model to study the end-to-end behavior of MPEG flows. We assume that each scheduler supports K number of service classes. Figure 2 shows a typical EDF scheduler multiplexing a number of flows of different service classes to a single output link of capacity C.

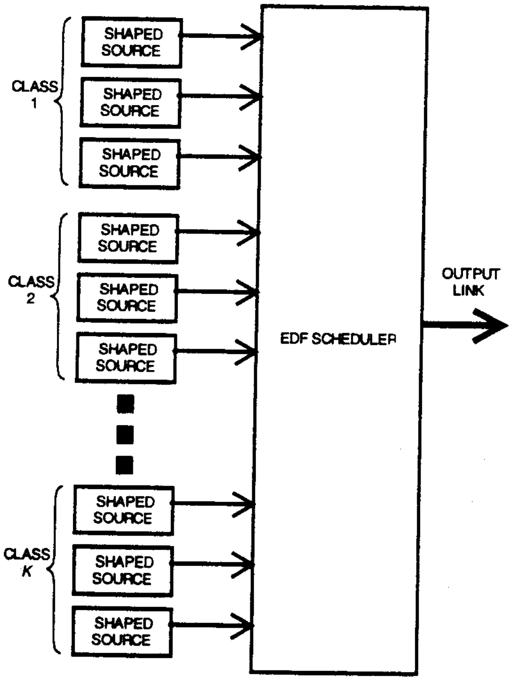

Fig. 2. An EDF scheduler.

We further assume that in each service class k, all sessions arrive based on a single deterministic on-off pattern A^k that is periodic with period V_k and on-period v_k. During the on-period, packets of length L_k arrive at intervals of time h_k. Figure 3 shows the traffic arrival pattern of each session. Each session i in class k is defined by a phase shift θ_i chosen uniformly at random in $[0, V_k]$. Thus, for all values of t, the traffic envelope of flow i is, $A_i(s, t) = A^k(s-\theta_i, t-\theta_i)$. We assume that the delay bound for all flows in each class k is identical.

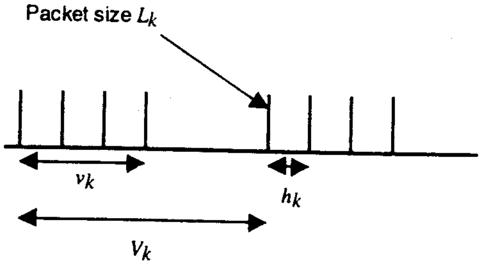

Fig. 3. The periodic arrival process.

Let us assume that a packet p arrives at time a_p. With a delay bound T_k, the deadline for this packet is $d_p = a_p + T_i$. For any time t, let let $P(t, d_p)$ be the set of packets that arrive at the scheduler no earlier than t and that have a deadline no later than d_p. A sufficient condition for the packet p to be able to meet its deadline is that the size of the packets in the set $P(t, d_p)$ is at most $C(d_p - t)$ for all $t \le a_p$. Thus, the packet deadline violation probability is upper bounded by the probability that the size of the packets in $P(t, d_p)$ is at most $C(d_p - t)$.

The main difficulty in calculating such a bound is that we must check the size of the packets in $P(t, d_p)$ for all values of $t \le a_p$. In order to address this problem, the methodology developed in [3] is to first choose a decreasing sequence of time values $\{y_i\}$ such that $d_p \le y_0 \le a_p$ and $y_i \to -\infty$. Now, for all $n \ge 1$, if the size of the packets in $P(y_n, d_p)$ is at most $C(d_p - y_{n-1})$, the packets do not violate their deadline. Using this discrete condition, we can use a union bound to obtain an upper bound on the probability that packet p violates its deadline. A session-i packet is in the set $P(y_n, d_p)$ if and only if it arrives in the interval $[y_n, d_p - T_i]$. Hence the total size of packets in $P(y_n, d_p)$ is equal to $\sum_i A_i(y_n, d_p - T_i)$ where the sum is taken over all sessions that pass through the scheduler. By a union bound, the packet deadline violation probability P_{VIO} is bounded by,

$$P_{VIO} \le \sum_{n \ge 1} \Pr\left[\sum_i A_i(y_n, d_p - T_i) \ge C(d_p - y_{n-1})\right]. \quad (6)$$

We can bound each term in Eq. (6) by a Chernoff bound. The Chernoff bound states that if a random variable X has a well-defined moment generating function $M_X(s) = E[e^{sX}]$, then

$$-\log \Pr[X \ge x] \ge sx - \log M_X(s) \qquad (7)$$

for all $s \ge 0$. By a Chernoff bound applied to Eq. (6) and the fact that the sessions are independent, we get

$$-\log P_{VIO} \geq sC(d_p - y_{n-1}) - \sum_i \log E[e^{sA_i(y_n,d_p-T_i)}]. \quad (8)$$

By calculating $\log E[e^{sA_i(y_n,d_p-T_i)}]$, we can perform a one-dimensional optimization on s and find the probability bound. Since the phase shift θ_i for each flow i is uniformly distributed over $[0, V_k]$, we can easily calculate the expectation value by counting the number of packets that fall in the interval $[y_n, d_p-T_i]$ for different values of θ_i.

In case of multi-hop transmission, the end-to-end delay bound for a particular flow is given by,

$$d_{total} = d_{shaper} + \sum_{m=1}^{M} T_i^m \qquad (9)$$

where the total delay is the sum of the traffic shaping delay bound d_{shaper} given by Eq. (1) and the delay bounds at each scheduler T_i^m.

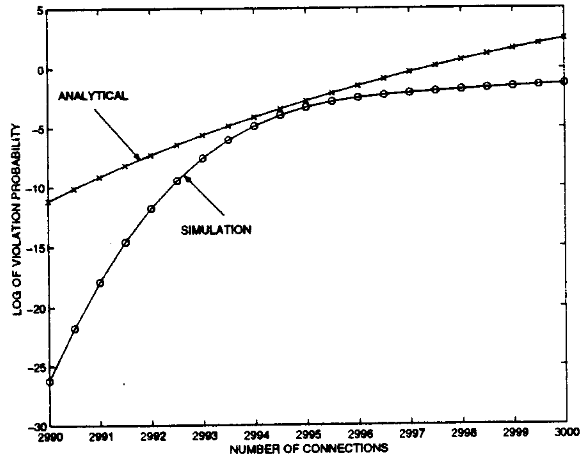

Fig. 4. Analytical and simulation results for packet deadline violation probability as function of the number of connections.

IV. RESULTS

In order to study the end-to-end delay performance in our model, we first investigate a single class traffic source and perform a simulation experiment by multiplexing a number of connections of an MPEG movie clip from the movie Jurassic Park at a single EDF scheduler. The clip contains 40,000 frames and the frame sizes were obtained from [18]. The frames have an average data rate of 0.33 Mbit/s with a peak rate of 1.3 Mbit/s controlled by a traffic shaper. The shaped MPEG sources are equivalently modeled in the analytical model of Section III by an on-off process with a periodicity equal to the GoP period, V=0.48 sec. The on period is v=0.12 sec, during which packets of length L=12075 bits arrives at intervals of h = 10 ms. The packet length is almost the size of an Ethernet frame. The output link operates at 1 Gbit/s and the stable maximum number of connections is 3000.

Figure 4 shows the packet deadline violation probability as a function of the number of connections. The analytical model results show upper bounds on the deadline violation probability, while the simulation results show actual fraction of packets that violated the deadline. The actual fraction of packets that violate the deadline is always less than the predicted upper bound from the analytical model.

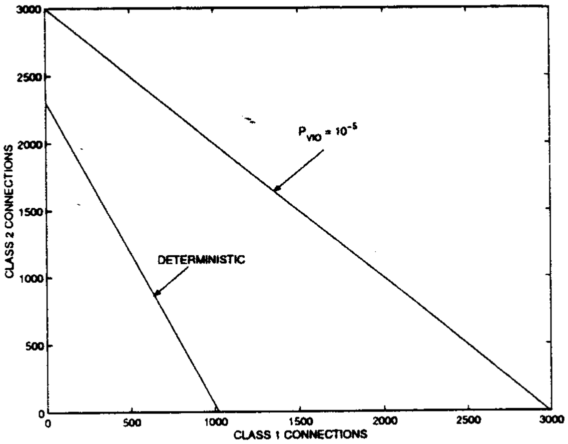

Fig. 5. Two-class schedulable region.

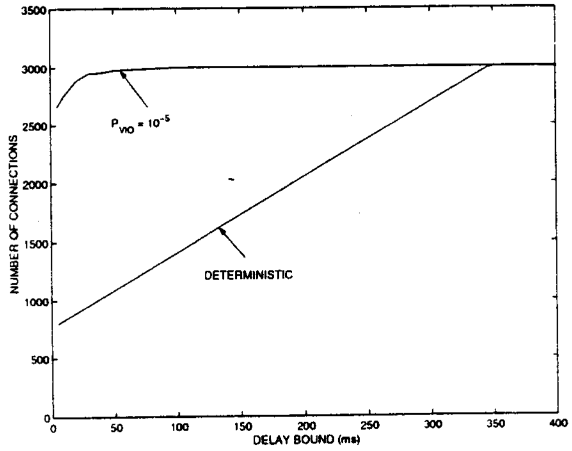

Fig. 6. Number of connections vs. delay bound - single hop.

Next, we investigate a two-class scenario where class-1 consists of MPEG sources with a delay bound of T=40 ms whereas class-2 consists of MPEG sources with a delay bound of T=480 ms. Both classes are characterized by the same on-off parameters as that was used for Fig. 4. For the two-class scenario, we show the schedulable region in Fig. 5 for both deterministic and statistical cases. For the statistical case, the packet deadline violation probability is fixed at 10^{-5}

and the numbers of class-1 and class-2 sources are varied. As can be seen from Fig. 5, the schedulable region for statistical case is much larger than the deterministic case.

In Fig. 6, we show the number of connections that can be admitted as a function of delay bound for a single hop when a single class of traffic (as that of Fig. 4) is present. Both the deterministic delay bound and the statistical delay bound with a delay violation probability of 10^{-5} are shown in Fig. 6. For low delay bounds, the number of flows that can be admitted under deterministic setting is far less than that under the statistical setting.

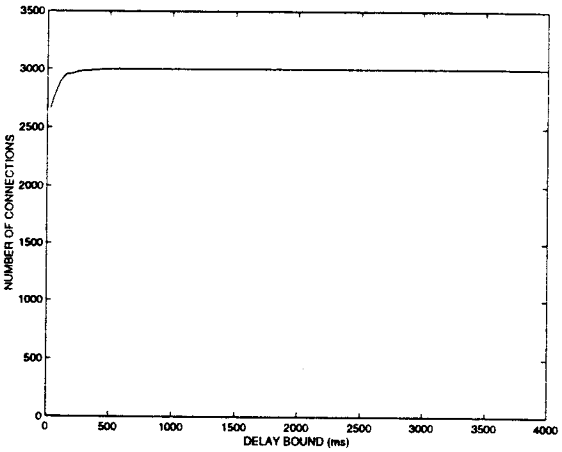

Fig. 7. Number of connections vs. delay bound - 5 hops.

In Fig. 7, we show the number of connections (of the same traffic as in Fig. 6) that can be supported as a function of delay bound for 5-hops. The packet delay violation probability is fixed at 10^{-5}. In the multi-hop setting, we assume that the total delay bound is equally distributed over all hops. The multi-hop case shows that with a low reasonable packet violation probability, the network utilization is high under a large range of end-to-end delay bounds.

V. CONCLUSION

In this paper, we develop a statistical framework for analyzing the end-to-end delay bounds for MPEG video flows over the next generation Internet. We develop a model for an isolated scheduler that multiplexes several different traffic classes using EDF scheduling discipline, and extend the model to multi-hop scenario. The statistical framework allows realizing much larger schedulable regions than the deterministic case. Significant gain in network utilization is observed for as low a delay violation probability as 10^{-5} compared to the deterministic case.

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