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Improving Utilization for Deterministic Service In Multimedia Communication

Hui Zhang
Lawrence Berkeley Laboratory
MailStop: 50B-229
Berkeley, CA 94720
hzhang@george.lbl.gov

Domenico Ferrari
University of California at Berkeley
and
International Computer Science Institute
ferrari@icsi.berkeley.edu

Abstract

Network-based real-time multimedia applications require guaranteed performance communication services. To provide guaranteed service, resources have to be reserved within the network. If reservation is based on the peak rate of each connection, the network will be under-utilized by guaranteed service traffic when the traffic is bursty. In this paper, we first show that local deterministic delay bounds can be guaranteed over a link for bursty traffic even when the sum of the peak rates of all the connections is greater than the link speed. Compared to previous admission control conditions, the new result allows a multi-fold increase in the number of admitted connections when the traffic is bursty. We then show that this new result can be efficiently extended from a single switch to a network of arbitrary topology by using rate-controlled service disciplines at the switches.

1 Introduction

High speed networking has introduced opportunities for new network-based multimedia applications such as video conferencing, scientific visualization and medical imaging. These applications have stringent performance requirements in terms of throughput, delay, delay jitter and loss rate. The best-effort service provided by the current packet-switching networks is not adequate. New services are needed.

Two types of new services have been proposed in the literature to support real-time multimedia applications: *guaranteed* service [9] and *predicted service* [3]. In a guaranteed service model, client-specified *a priori* performance bounds are guaranteed to each connection *regardless of* the behaviors of other connections. In a predicted service model, a network dictated *post facto* delay bound and playback point are provided to the communication client. The playback point may vary and the service may be disrupted due to the network load fluctuation. It is assumed that applications using the predicted service can adapt to the changing of the playback point and tolerate infrequent service disruptions.

There are two important factors to consider in choosing between these two service models: the quality of the service and the cost of providing such a service. From a communication client's point of view, the quality of the guaranteed service is better since the performance bounds are guaranteed and there are no service disruptions. From the network's point of view, the cost of providing a service depends on the maximum network utilization that can be achieved. Although it is argued that the predicted service would allow a higher network utilization by real-time traffic than guaranteed service [3], since there is no complete description of the algorithms for the predicted service, especially the admission control algorithms, there are no quantitative comparisons available.

In [6], two types of guaranteed services are proposed: *deterministic service* and *statistical service*. In deterministic service, performance bounds are guaranteed for *all* packets on a connection even in the worst case. In statistical service, *probabilistic* performance bounds are guaranteed. Although the quality of deterministic service is better, statistical service allows the network to achieve a higher utilization by exploiting statistical multiplexing.

It is clear from the above discussion that achieving high network utilization is one of the most important considerations in providing guaranteed service, especially deterministic service. To provide guaranteed service, resources have to be reserved within the network. In general, resource reservation schemes can achieve high network utilizations for smooth traffic. However, many clients requiring performance guarantees have bursty sources, for example, compressed video. If reservation is based on the peak rate of each connection, new requests will be rejected when the sum of the peak rates of all the connections reaches link speed. In this case, the network will be under-utilized by guaranteed service traffic when the peak-to-average-rate ratio is high.

Therefore, it is important to derive solutions that can provide performance guarantees even when the sum of the peak rates of all the connections is greater than the link speed, and to understand the relationship between the server utilization and traffic burstiness. In this paper, we discuss these issues within the framework of the Tenet real-time channel scheme. We show that deterministic guarantees can be provided even when the sum of peak rates of all the connections is greater than the link speed, and reasonable average network utilization can be achieved for deterministic service even when traffic is bursty. Compared to previous admission control algorithms for deterministic service, the new result allows a multi-fold increase in the number of admitted connections for a single server when the traffic is bursty.

Providing local performance bounds at a single switch only solves part of the problem. In a networking environment, packets from different connections are multiplexed at each of the switches. Even if the traffic can be characterized at the entrance to the network, complex interactions among connections will destroy many properties of the traffic inside the network, and the traffic model at the source may not be applicable inside the network. Since local performance bounds can be guaranteed for a connection only if the connection's input traffic to the switch satisfies certain traffic characterization, traffic pattern distortion may make it difficult to guarantee local performance bounds at switches inside the network. We address this problem by using rate-controlled service disciplines inside the network to reconstruct traffic patterns. Rate-controlled service disciplines allows the result for a a single switch be efficiently extended to a network of arbitrary topology.

The reminder of the paper is organized as follows. Section 2 briefly reviews the real-time channel scheme and shows the limitations of previous admission control tests for the deterministic service. Section 3 presents the analysis which provides delay bounds for a First-Come-First-Served (FCFS) scheduler. Section 4 gives numerical examples to illustrate the results derived in Section 3. Section 5 discusses issues on providing end-to-end delay bounds in a networking environment. Section 6, discusses the implications of the results. Finally, Section 7 concludes the paper with a summary.

2 Background

In this section, we give a brief overview of the current version of the Tenet resource management algorithms [9, 8].

The Tenet algorithms are based on a communication abstraction called a *real-time channel* [9]. A real-time channel is a network connection associated with traffic and performance parameters. The parameters are provided by the clients to specify their traffic characteristics and performance requirements. The traffic specification consists of parameters $(Xmin, Xave, I, Smax)$, where $Xmin$ is the minimum packet inter-arrival time, $Xave$ is the worst-case average packet inter-arrival time over an averaging interval, I is the averaging interval, and $Smax$ is the maximum packet size. In such a specification, it is easy to see that $Xave/Xmin$ is the peak-to-average-rate ratio, which is an indicator of the traffic burstiness. It should be also noticed that I also affects the traffic burstiness. For the given values of $Xmin$ and $Xave$, I determines how long the source can continuously send packets at the peak rate in the worst case. The larger $Xave/Xmin$ and I, the burstier the traffic. Table 1 shows the model and the interpretations of some of the formulas. In the table, l denotes the link speed.

$\frac{Smax}{Xmin \times l}$	peak rate
$\frac{Smax}{Xave \times l}$	upper bound on average rate over I
$\frac{Xave}{Xmin}$	peak-to-average-rate ratio or burst ratio
$\frac{I}{Xave}Xmin$	maximum burst length

Table 1: $(Xmin, Xave, I, Smax)$ Traffic Model

Tenet algorithms provide both deterministic and statistical guarantees. For the purpose of this paper, we only consider deterministic guarantees. For a channel with deterministic delay bound $\overline{D}$, the network guarantees that delays of all packets on that channel will be less than $\overline{D}$ as long as the channel does not violate its traffic specification.

A channel needs to be *established* before data can be transmitted. This channel establishment is achieved in the following manner: a real-time client specifies its traffic characteristics and end-to-end performance requirements to the network; the network determines the most suitable route for the channel, translates the end-to-end parameters into local parameters at each node, and attempts to reserve resources at these nodes accordingly. This is done in a distributed manner during a round-trip communication.

In order to provide performance guarantees, two levels of controls are needed: at the channel level, channel admission control algorithms reserve resources for each of the channels and limit the maximum utilization of the network by real-time traffic; at the packet level, the service discipline at each of the switches determines the multiplexing policy and allocates resources to different channels according to the reservations.

As shown in [7, 21], many service disciplines can be used to provide real-time service. However, different service disciplines require different admission control algorithms. In [9], conditions are given for a variation of the Earliest-Due-Date discipline. Three tests need be satisfied before a new channel request can be accepted, which are: (a) *deterministic test*, which limits the peak utilization of the server to be less than 1; (b) *delay bound or schedulability test*, which avoids the case of scheduling saturation; and (c) *buffer test*, which ensures enough buffer space is available. For the purpose of the discussion in this paper, we will focus on the deterministic test[1], which requires

$$\sum_{j=1}^{n} \frac{1}{Xmin_j} \times \frac{Smax_j}{l} < 1 \qquad (1)$$

where l is the link speed and n is the number of channels traversing the link.

Although, the tests guarantee that deterministic delay bounds can be provided, they are rather restrictive in the sense that the sum of the peak rates of all deterministically guaranteed connections on any link has to be less than the link speed. If the peak-to-average-rate ratio is high, the average link utilization by real-time traffic will be low.

3 Delay Analysis

In this section, we show that deterministic delay bounds can be obtained even when the sum of peak rates of all real-time channels is greater than the link speed, though the sum of the average rates of all real-time channels has to be less than the link speed. The result holds even for a simple discipline like FCFS. We use the bounding techniques developed by Cruz [4]. In [4], a fluid traffic model (σ, ρ) is used. A channel satisfies traffic specification (σ, ρ) if, for any time interval of length u, the number of bits arriving during the interval is no more than $\sigma + \rho u$. The model we use is the discrete $(Xmin, Xave, I, Smax)$ model.

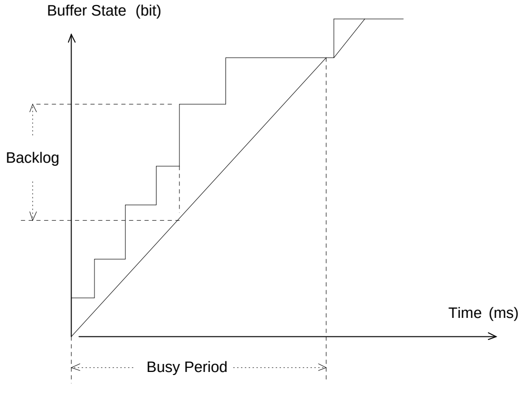

Figure 1: Concepts: Delay, Backlog and Busy Period

Figure 1 illustrates some of the concepts used in the analysis. The horizontal axis is the time. The upper curve in the figure is the sum of the bits that have arrived since the origin of time. Each arrival of a packet causes an upward jump in this curve. The lower curve is the number of bits that have been transmitted. The difference between the two curves is the *backlog* function. A work-conserving server always transmits packets at a constant rate when there is a backlog. The points where the arrival curve and the service curve meet, or where the backlog is zero, divide the time axis into *busy periods* and *idle periods*. Within such a framework, the following two propositions immediately follow, where $\overline{Smax}$ is the maximum packet size that can be transmitted over the link, and l is the link speed.

Proposition 1 *For a work-conserving real-time scheduler, if for* any *realization of the input traffic that satisfies a given traffic constraint, the maximum length of a busy period is no greater than* $\overline{d}$*, then* $\overline{d} + \overline{Smax}/l$ *is an upper bound for the delays of all packets.*

Proposition 2 *For a FCFS real-time scheduler, if for* any *realization of the input traffic that satisfies a given traffic specification, the maximum real-time traffic backlog divided by the link speed is no greater than* $\overline{d}$*, then* $\overline{d} + \overline{Smax}/l$ *is an upper bound for the delays of all packets.*

To take into account of the effect of non-real-time packets, which have a lower priority than real-time packets, but cannot be preempted after the beginning of their transmission, $\overline{Smax}/l$ is included in the delay bounds.

Notice that in both propositions, in order to derive the delay bound, certain upper bound values (the length of busy period or the backlog) need to be calculated for *any* realization of the input traffic. It is impossible to calculate these upper bounds for every realization of the input traffic because there are infinite such realizations. If we can find one "worst-case" realization of the input traffic, and prove that this realization gives the maximum upper bound values, we can just analyze the scheduler for this realization and compute the delay bound.

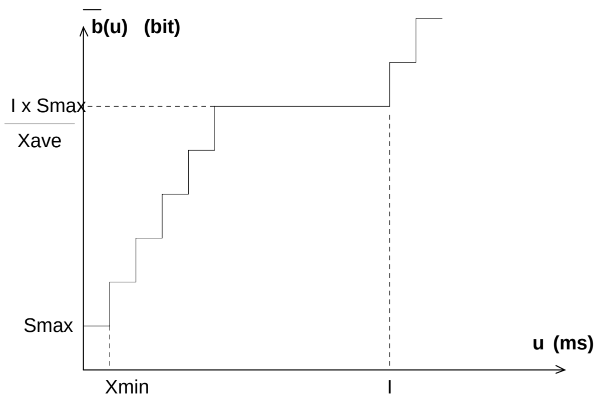

Figure 2: Traffic Constraint Function for (Xmin, Xave, I, Smax) Specification

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[1] We consider a network with non-blocking switches, where queueing happens at the output link of each switch. The processing time of each packet is thus equal to the transmission time, and is bounded by $\frac{Smax_j}{l}$

Central to the analysis is the concept of *traffic constraint function* $\overline{b}(.)$ [2]. $\overline{b}_j(u)$ is defined to be the maximum number of bits that can arrive on channel j during *any* interval of length u. For a traffic source that obeys the model $(Xmin_j, Xave_j, I_j, Smax_j)$, it is easy to show that $\overline{b}_j(u)$ is

$$(min(\lceil\frac{u\ mod\ I_j}{Xmin_j}\rceil, \lceil\frac{I_j}{Xave_j}\rceil) + \lceil\frac{u}{I_j}\rceil\lceil\frac{I_j}{Xave_j}\rceil)Smax_j$$

Figure 2 illustrates the function $\overline{b}(.)$ for a channel with traffic specification $(Xmin, Xave, I, Smax)$.

The following theorem bounds the delay for a FCFS scheduler.

Theorem 1 *Let there be n channels multiplexed on a link with a FCFS scheduler and link speed l. If for $j = 1, \cdots, n$, the traffic on channel j is bounded by $\overline{b}_j(.)$, then the delays of packets on all the channels are bounded by $\overline{d}$, where $\overline{d}$ is defined by*

$$\overline{d} = \frac{1}{l}\max_{\forall u \geq 0}\{\sum_{j=1}^{n}\overline{b}_j(u) - l \times u + \frac{\overline{Smax}}{l}\} \qquad (2)$$

Proof.

Let ξ be a realization of the input traffic. If s is the starting time instant of a busy period, and t is a time instant within the busy period, define $b_j^\xi(s,t)$ to be the number of bits that arrive during the interval (s,t) on channel j. The maximum backlog in this realization ξ is $\max_{\forall s,t}\{\sum_{j=1}^{n} b^\xi(s,t) - l \times (t-s)\}$, We have

$$\begin{aligned}
& \tfrac{1}{l}\max_{\forall s,t}\{\textstyle\sum_{j=1}^{n} b^\xi(s,t) - l \times (t-s))\} \\
\leq\ & \tfrac{1}{l}\max_{\forall s,t}\{\textstyle\sum_{j=1}^{n} \overline{b}_j(t-s) - l \times (t-s))\} \\
\leq\ & \tfrac{1}{l}\max_{\forall u \geq 0}\{\textstyle\sum_{j=1}^{n} \overline{b}_j(u) - l \times u\}
\end{aligned}$$

Since ξ is an arbitrary realization of the input traffic, from Proposition 2. $\overline{d}$ as defined by Equation (2) is a delay bound for packets from all channels. **Q.E.D.**

In the following two corollaries, we give closed-form solutions for two special cases. Corollary 1 considers the case when $\sum_{j=1}^{n}\frac{Smax_j}{Xmin_j} \leq l$, Corollary 2 considers the case of homogeneous sources when $\sum_{j=1}^{n}\frac{Smax_j}{Xmin_j} > l$.

Corollary 1 *Let there be n channels multiplexed on a link with a FCFS scheduler and link speed l. Assume channel j obeys the traffic specification $(Xmin_j, Xave_j, I_j, Smax_j)$, $(j = 1, \cdots, n)$. If $\sum_{j=1}^{n}\frac{Smax_j}{Xmin_j} \leq l$, then the delays of packets on all the channels are bounded by $\overline{d}$, where $\overline{d}$ is given by*

$$\overline{d} = \frac{1}{l}\sum_{j=1}^{n} Smax_j + \frac{\overline{Smax}}{l} \qquad (3)$$

Proof.

$$\begin{aligned}
\overline{d} &= \frac{1}{l}\max_{u \geq 0}\{\sum_{j=1}^{n}\overline{b}_j(u) - l \times u\} + \frac{\overline{Smax}}{l} \\
&\leq \frac{1}{l}\max_{u \geq 0}\{\sum_{j=1}^{n}\lceil\frac{u}{Xmin_j}\rceil Smax_j - l \times u\} + \frac{\overline{Smax}}{l} \\
&\leq \frac{1}{l}\max_{u \geq 0}(\sum_{j=1}^{n}(\frac{u}{Xmin_j} + 1)Smax_j - l \times u) + \frac{\overline{Smax}}{l} \\
&= \frac{1}{l}\sum_{j=1}^{n} Smax_j + \frac{\overline{Smax}}{l} + (\sum_{j=1}^{n}\frac{Smax_j}{Xmin_j} - l)\frac{u}{l} \\
&\leq \frac{1}{l}\sum_{j=1}^{n} Smax_j + \frac{\overline{Smax}}{l}
\end{aligned}$$

Q.E.D.

In the proof, the first inequality holds due to the assumption that the minimum inter-packet spacing is $Xmin_j$. All three inequalities become equalities when $u = 0$, i.e., the worst case backlog is at the beginning of a busy period, when every channel sends out a maximum length packet.

Corollary 2 *Let there be n homogeneous channels with traffic specification (Xmin, Xave, I, Smax) multiplexed on a link with the FCFS service discipline and link speed l. If $n \times \frac{Smax}{Xave} \leq l$ and $n \times \frac{Smax}{Xmin} > l$, the delays of packets from all the channels are bounded by $\overline{d}$, where $\overline{d}$ is given by*

$$d = Xmin + (\mu_{ave} - \frac{1}{burstRatio})I + \frac{\overline{Smax}}{l} \qquad (4)$$

where $\mu_{ave} = n \times Smax/Xave \times l$, and $burstRatio = Xave/Xmin$.

Proof.

Let $\hat{I}$ be the length of the longest interval that packets from one channel can arrive at the scheduler with $Xmin$ spacing, we have,

$$\begin{aligned}
\hat{I} &= \min\{u \mid \overline{b}(u) = \overline{b}(I)\} && (5) \\
&= (\frac{I}{Xave} - 1)Xmin && (6)
\end{aligned}$$

It is easy to see that $\sum_{j=1}^{n}(\overline{b}_j(u) - l \times u)$ is maximized at $u = \hat{I}$.

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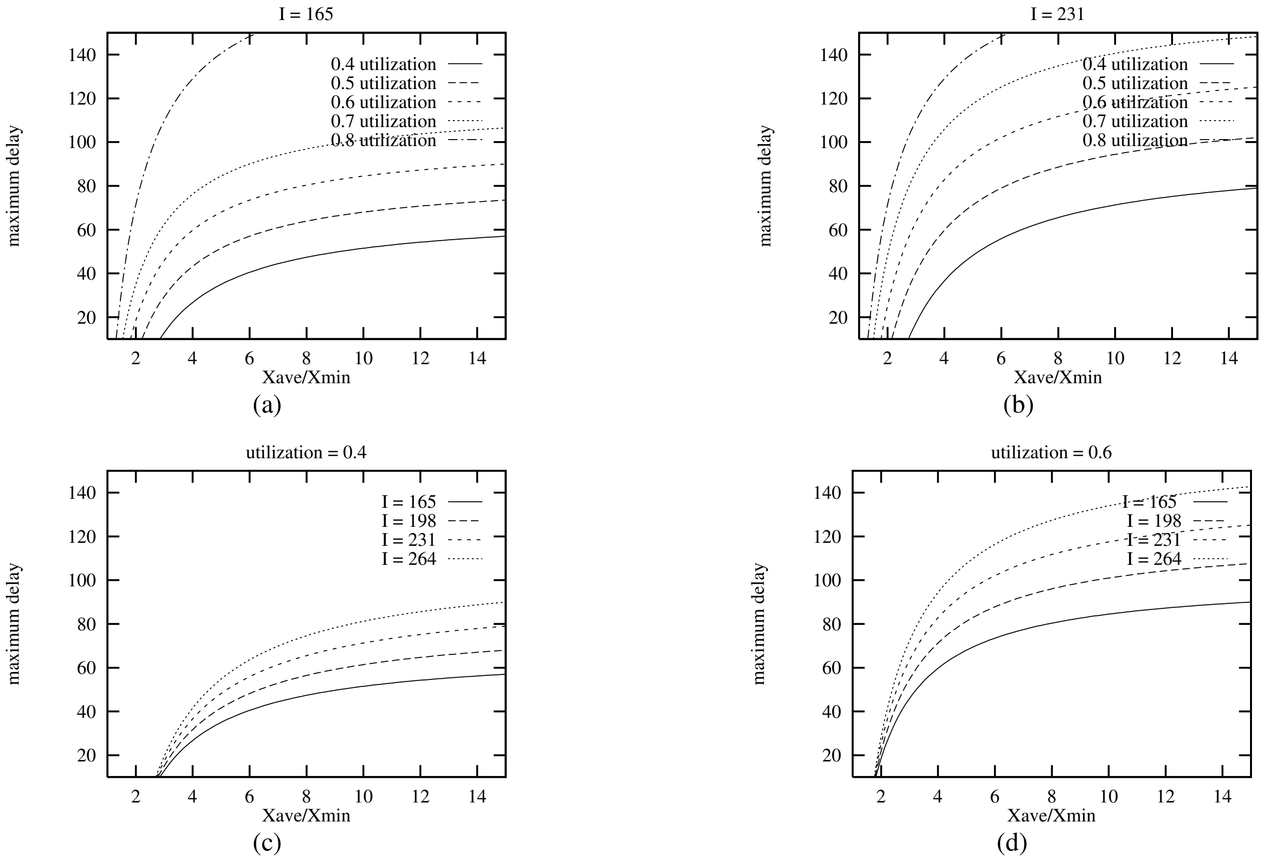
[2] We use a notation that differs slightly from that in [4], where $b(.)$ is used to denote the traffic constraint function. In this paper, $\overline{b}(.)$ is used to denote the traffic constraint function, and $b^\xi(s,t)$ denotes the number of bits that arrive between time instants s and t for an input traffic realization ξ.



(a) (b)

(c) (d)

Figure 3: Effects of Burstiness and Utilization on Delay Bound

From Theorem 1, we have,

$$\begin{aligned}
\overline{d} &= \frac{1}{l}\sum_{j=1}^{n}(b_j(\hat{I}) - l \times \hat{I}) + \frac{\overline{Smax}}{l} \\
&= \frac{1}{l}(\sum_{j=1}^{n} Smax\frac{I}{Xave}) - \hat{I} + \frac{\overline{Smax}}{l} \\
&= \mu_{ave} \times I - \hat{I} + \frac{\overline{Smax}}{l} \\
&= \mu_{ave} \times I - (\frac{I}{Xave} - 1)Xmin + \frac{\overline{Smax}}{l} \\
&= Xmin + (\mu_{ave} - \frac{1}{burstRatio})I + \frac{\overline{Smax}}{l}
\end{aligned}$$

Q.E.D.

In (4), μ_{ave} is the average utilization of the link, and $Xave/Xmin$ and I represent the burstiness of the channels. Intuitively, the more loaded the link, and the burstier the channel traffic, the larger the delay bound. This is exactly what is shown by (4). It is important to see that the averaging interval I affects the traffic burstiness. For the given values of $Xmin$ and $Xave$, I determines how long the source can continuously send packets at the peak rate in the worst case. This is illustrated in Equation (6).

The previous two corollaries consider only two special cases. More general result is given by the following corollary:

Corollary 3 *Let there be n channels multiplexed on a link with a FCFS scheduler and link speed l. Assume channel*