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Understanding TCP Dynamics in an Integrated Services Internet

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Abstract

A large number of Internet applications are sensitive to overload conditions in the network. While these applications have been designed to adapt somewhat to the varying conditions in the Internet, they can benefit greatly from an increased level of predictability in network services. We propose minor extensions to the packet queueing and discard mechanisms used in routers, coupled with simple control mechanisms at the source that enable the network to guarantee minimal levels of throughput to different network sessions while sharing the residual network capacity in a cooperative manner. The service realized by the proposed mechanisms is an interpretation of the controlled-load service being standardized by the IETF. Although controlled-load service can be used in conjunction with any transport protocol, our focus in this paper is on understanding its interaction with TCP. Specifically, we study the dynamics of TCP traffic in an integrated services network that simultaneously supports both best-effort and controlled-load sessions. In light of this study, we propose and experiment with several variations to TCP's control mechanisms with the objective of fine-tuning them for an integrated services environment. We then investigate the overheads associated with these enhancements and the benefits they provide. Finally, we show how the service mechanisms proposed here can be embedded within more elaborate packet and link scheduling frameworks in a fully-evolved integrated services Internet.

1 Introduction

A large class of Internet applications, referred to as tolerant playback applications in [1,15], can greatly benefit from an increased level of predictability in network services. These applications typically buffer a portion of the data on the client before starting the playback, and then operate in a streaming mode to keep the buffer from underflowing. Examples include increasingly popular applications such as PointCast, RealAudio, and VDOnet which stream text/image, audio, and video data over the Internet. The quality of playback for each of these applications can vary from excellent to intolerable, depending on the network load. These applications can benefit tremendously from a network service that guarantees a minimum level of throughput at all times, but allows for the possibility of higher throughput during periods of light loads. Such a service is also useful to more traditional, elastic applications [1,15], such as `ftp` and `telnet`. Tasks such as booting a disk-less workstation over the network, backing up remote files, and synchronizing web proxies can be performed with more predictability and within a bounded time by guaranteeing a minimal bandwidth to the underlying sessions. This service can also be used to set up a virtual overlay network in the Internet, connecting business-critical servers and clients with virtual links of a minimum guaranteed bandwidth.

The controlled-load service [17] currently being standardized by the Internet Engineering Task Force (IETF) fits very well into the scenarios sketched above. It is part of an ambitious goal of defining a service architecture that is suitable for a diversity of applications. Our objective in this paper has been to follow an evolutionary path towards this goal. That is, to enhance network services within the framework of the IETF-defined service architecture, but with minimal enhancements to the network infrastructure, especially the routers.

We propose a simple extension to the queueing mechanisms used in today's routers coupled with a control mechanism at the source to guarantee a minimal level of end-to-end throughput to different network sessions. The residual network capacity is shared in a socially cooperative fashion, in a manner similar to the one in use in the Internet today by applications using TCP. In this scheme, each reserved session is associated with a traffic envelope. Traffic is policed at the source and packets conforming to the envelope are marked. Non-conformant traffic and best-effort traffic is injected into the network unmarked. At the routers we use an enhanced random early detection (RED) [4] and discard mechanism. Both marked and unmarked packets share the same FIFO queue. When the queue length at the router exceeds a certain threshold, packets are dropped randomly as done in RED gateways. However, unlike standard RED gateways where all packets have the same drop probability, in the enhanced RED (ERED) gateway, marked packets have a lower drop probability than the unmarked packets.

The service realized by the mechanism described above is an interpretation of the controlled-load service. By definition, traffic belonging to a controlled-load session and conforming to the associated traffic envelope sees very little loss and very little queueing delay through the network. Non-conformant controlled-load traffic is treated as best-effort traffic. By using a common queue for best-effort and conformant controlled-load traffic, we essentially relax the recommended delay targets for conformant controlled-load traffic. This laxity not only simplifies the implementation and reduces packet handling overheads at the routers, but also helps maintain packet ordering. Note that ERED ensures a low loss rate to conformant controlled-load traffic. We argue that elastic and tolerant playback applications can withstand a reasonable amount of queueing delay and can fully exploit the guarantees on throughput to improve performance.

Although controlled-load service can be used in conjunction with any transport protocol, our focus in this paper is on TCP. We examine TCP because: (1) an overwhelming number of applications use TCP as the transport protocol of choice, and (2) TCP has a well-developed congestion and flow control mechanism that makes it an interesting case study. While some of the tolerant playback applications may not use TCP, the mechanisms described here can easily be applied to other transport protocols, such as RTP. Our objective in this paper is to understand and tune the end-to-end control mechanisms used in TCP in an integrated services Internet that supports both best-effort and controlled-load services.

Integrated services in the Internet is a relatively new area of research. To the best of our knowledge, no published work addresses the specific issues discussed in this paper. In a more general sense, studies on supporting TCP over ABR/UBR services in ATM networks [8,12] address similar issues. However, due to significant differences between the service architectures of ATM and the Internet, the nature and the focus of these studies are quite different. In [8], the authors propose using a modified switch buffer allocation policy in order to obtain peak throughput and fairness among TCP connections over ATM. In particular, each connection is effectively provided with a weighted fair share of the buffers in the switch. The switch uses per-connection accounting to determine which connections are overloading their allocation and drops cells accordingly. In contrast, most of the modifications we propose are in the TCP sender. This is in line with the Internet design philosophy of providing sophisticated end-to-end control in end hosts, coupled with a relatively simple control inside the network. Moreover, since these modifications are required only for senders, they may be deployed incrementally.

The rest of the paper is organized as follows. In Section 2 we briefly describe the proposed service architecture and the ERED mechanism. Performance of TCP in an integrated services environment and the effects of different service parameters are investigated in Section 3. In Section 4, we propose simple modifications to TCP's transmission and windowing mechanisms to exploit reservations in the network. Section 5 examines the overhead such modifications incur and the benefits which they provide. Section 6 is devoted to a discussion and preliminary results from experiments on the possible integration of the ERED mechanism into a more elaborate packet and link scheduling architecture, such as class-based queueing. We conclude in Section 7.

2 Network and Service Models

The RSVP and the INTSERV working groups in the IETF are responsible for defining protocols and standards to support integrated services in the Internet. In this section we briefly review the relevant aspects of these standards and show how the proposed enhancements fit into the IETF-defined framework.

To avail itself a reservation, a connection has to specify a traffic envelope, called *Tspec*. Tspec includes a long-term average rate r_m, a short-term peak rate r_p, and the maximum size of a burst b of data generated by the application. For example, for an application generating MPEG-encoded video, the average rate could be the long-term data rate, the peak rate could be the link bandwidth at the source, and the burst size could be the maximum size of a frame. Tspec also specifies the maximum and minimum packet sizes to be used by the application. Connections are monitored and policed at the network entry points. This could be either at the source, or at the boundary between the corporate or campus intranet and the Internet. Packet classification and service differentiation also takes place at the routers. The service priority given to a packet is a function of the Tspec, and in the case of some service classes, a separate service specification known as *Rspec*. For controlled-load service, no Rspec is specified.

In order to police traffic at the source, we use token buckets [13]. The token generation process follows the Tspec advertised by the source. That is, the long-term average rate of token generation is t_m, the short-term peak rate of token generation is t_p, and the depth of the token bucket is b. Each time a packet is injected into the network, if sufficient tokens are available, an equivalent number of tokens are considered consumed. If there aren't enough tokens present at the time of transmission, the packet is treated as non-conformant.

In addition to policing, we also propose to mark packets at the network entry point. Conformant controlled-load traffic is marked before being injected into the network. Non-conformant controlled-load traffic and best-effort traffic is injected into the network unmarked. Although marking is not currently supported in IP networks,

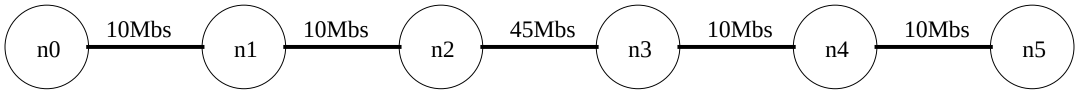


Figure 1: Network Topology

there are sufficient hooks (specifically the type of service bits) in the IP header to add this feature easily. Note that marking is not a mandatory requirement, it just facilitates traffic classification at the routers. In the absence of a marking facility, IP datagrams have to be passed through a classifier at the source, as well as at the routers, to determine which flows they belong to and to determine whether they are in violation of, or in conformance with, the advertised Tspecs of the flows. In the presence of a marking facility, classification is only required at the network entry point and not at interior routers. In the rest of the paper, we assume that a marking facility is available.

The routers perform admission control for controlled-load connections. Admission control algorithms are not discussed in this paper, but, for the purpose of the experiments, we assume that the aggregate reservation levels at the routers are within their capacities. In addition to performing admission control, the routers also need to support service differentiation between marked (conformant controlled-load) and unmarked (non-conformant controlled-load and best-effort) packets. One obvious approach to providing different services to marked and unmarked packets is to maintain separate queues for each class and serving them according to their scheduling priority. However, we propose to use a common queue for both compliant and non-compliant traffic and serve them in FIFO order. A common FIFO queue not only simplifies the scheduling functionality at the router, it also helps maintain packet ordering in controlled-load connections. Although maintaining packet ordering is not a requirement, failure to do so may have serious performance impacts on transport protocols such as TCP. Our approach to service differentiation between marked and unmarked packets relies on a selective packet discard mechanism. We use an enhanced version of the RED (Random Early Detection) algorithm for this purpose. In classical RED routers, a single FIFO queue is maintained for all packets. Packets are dropped randomly with a given probability when the queue length exceeds a certain threshold. The drop probability itself depends on the queue length and the time elapsed since the last packet was dropped. Enhanced Random Early Detection (ERED) is a minor modification to the original RED algorithm. In ERED, the drop probabilities of marked packets are lower than that of unmarked packets.

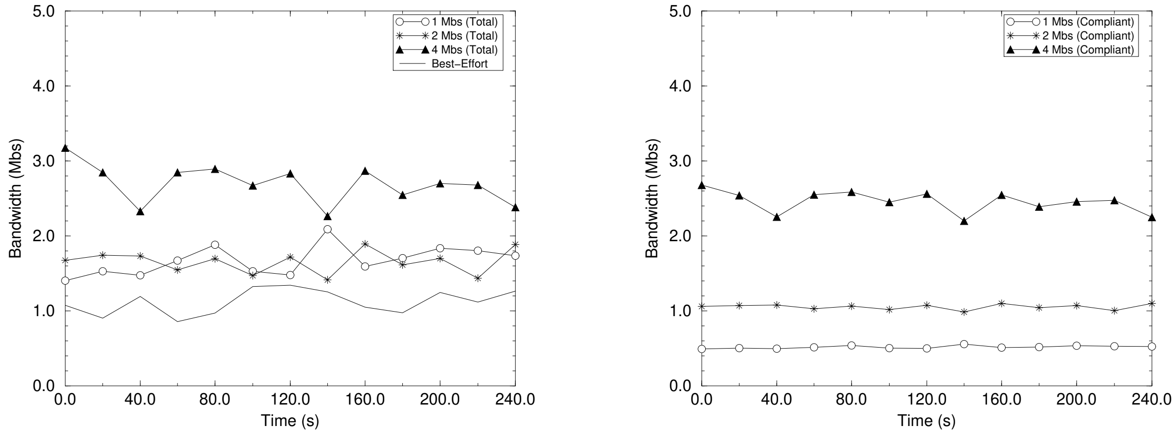
Studies have shown that RED gateways are better than traditional drop-tail routers in terms of fairness to bursty traffic [4]. It can also be parameterized to control the queue size, and hence, the queueing delay. Note that ERED guarantees low loss rate to conformant controlled-load traffic. However, since it uses a common FIFO queue, the queueing delay experienced by the conformant controlled-load traffic and best-effort traffic are the same. We argue that elastic and tolerant playback applications can withstand a reasonable amount of queueing delay and can really exploit the guarantees on throughput to improve performance. Also, note that the ERED queue parameters can be tuned to limit the queueing delay experienced by packets in order to meet the specification of controlled-load service. Of course, there are ways of realizing controlled-load service more accurately. We show, however, that even with these simple enhancements, we can adequately provide a useful service to a large class of existing and emerging applications.

As mentioned earlier, the primary focus of this paper is not to propose packet queueing and scheduling mechanisms to realize an entire suite of integrated services on the Internet. Our objective is to study how we can offer a particularly useful network service with minimal changes to the routers and the end-hosts. For a systematic deployment of the full portfolio of services in the Internet, a number of sophisticated mechanisms, such as class-based queueing [5] and weighted fair queueing [1, 2, 6, 7, 14–16], have been proposed. In Section 6, we discuss how the ERED mechanism can be embedded in a more elaborate packet queueing and scheduling architecture.

3 Understanding TCP Dynamics

This section is devoted to the study of TCP dynamics in an integrated services environment. For the purpose of the experiments, we modified the NS [11] simulator. The NS simulator has been used extensively in a number of studies reported in the literature. While the simulator does not use production TCP code, it implements congestion and error control algorithms used in different implementations of TCP with remarkable accuracy. For most of the experiments reported here, we use a Reno-variant of TCP [9]. We modified the simulator by adding policing and extending the RED queueing discipline.

For the experiments in this section, we consider a simple network topology shown in Figure 1. The capacity of each bi-directional link is labeled and has a transmission delay of `10ms`. Connections requesting a reservation specify a peak and a mean rate of service, and the maximum size of a burst. At the source, tokens are generated at the service rate and are accumulated in a token bucket. The depth of the token bucket is the same as the maximum



(a) Aggregate throughput

(b) Compliant throughput.

Figure 2: Effect of reservation on end-to-end throughput.

burst size specified by the source. Throughout this paper, the token bucket size is measured in units of time. In units of tokens, it is equivalent to token generation rate times the bucket size in units of time. The peak rate is set to the link speed by default. TCP segments belonging to reserved connections are transmitted as marked datagrams if there are sufficient tokens available in the token bucket at the time of transmission. Otherwise, they are sent as unmarked datagrams. TCP segments belonging to best-effort connections are sent as unmarked datagrams. We assume that sources are greedy, that is, they always have data to send.

3.1 Effect of Service Rate

This experiment is designed to investigate the effect of service rate on end-to-end throughput. For the purpose of this study we ran three connections with reservations of `1Mbs`, `2Mbs`, and `4Mbs`, and three best-effort connections from node `n0` to `n5`. We set the max_{th} [1] for the ERED queue to `80KB` and the min_{th} [2] to `20KB`. The drop probability of marked packets was set to zero. The maximum drop probability of the of the unmarked packets for this experiment was `0.02`. Each controlled-load source used a token bucket of depth `50ms`. We also experimented with other ERED parameters and observed similar results.

Figure 2(a) shows the throughput seen by each connection. Throughput is computed by measuring the data received at the receiver over an observation period and dividing it by the observation interval. Figure 2(b) plots the compliant throughput seen by connections with reservations. This is the portion of the throughput that is contributed by marked packets. Ideally, it should be equal to the reserved rate of service. From Figure 2(a), we observe that connections with higher reservations generally see better throughput than connections with lower or no reservations. However, from Figure 2(b) we observe that the compliant portions of the bandwidth received by all reserved connections are less than their respective service rates.

The explanation for the observations from Figure 2 lies in the flow and congestion control mechanisms used by TCP. The TCP sessions with reservations exercise their flow and congestion control mechanisms in the same way as best-effort connections. However, they have a lower probability of losing a packet at the routers since their marked packets have lower (in this case close to zero) probability of getting dropped. Because connections with higher reservations mark their packets at a higher rate, they have a decreased probability of having a packet dropped. This is why connections with higher reservations see higher throughput than connections with lower or no reservations. However, as observed from Figure 2(b), TCP fails to fully exploit the benefits of the reservation. Since marked packets aren't dropped by the network and since the compliant part of the throughput is less than the reservation levels in all cases, it is apparent that not all of the tokens generated are used to mark packets at the source. That is, there are significant token losses due to token bucket overflow. Although the sender is a greedy source, the TCP congestion control mechanism throttles the source to an extent that makes it impossible for it to fill up the reserved portion of the communication pipe. To confirm this speculation, we plotted the packet trace from the connection with a `4Mbs` reservation.

[1] All arriving packets are dropped when the queue size reaches this level.

[2] Random packet drops continue until the queue length falls below this threshold.

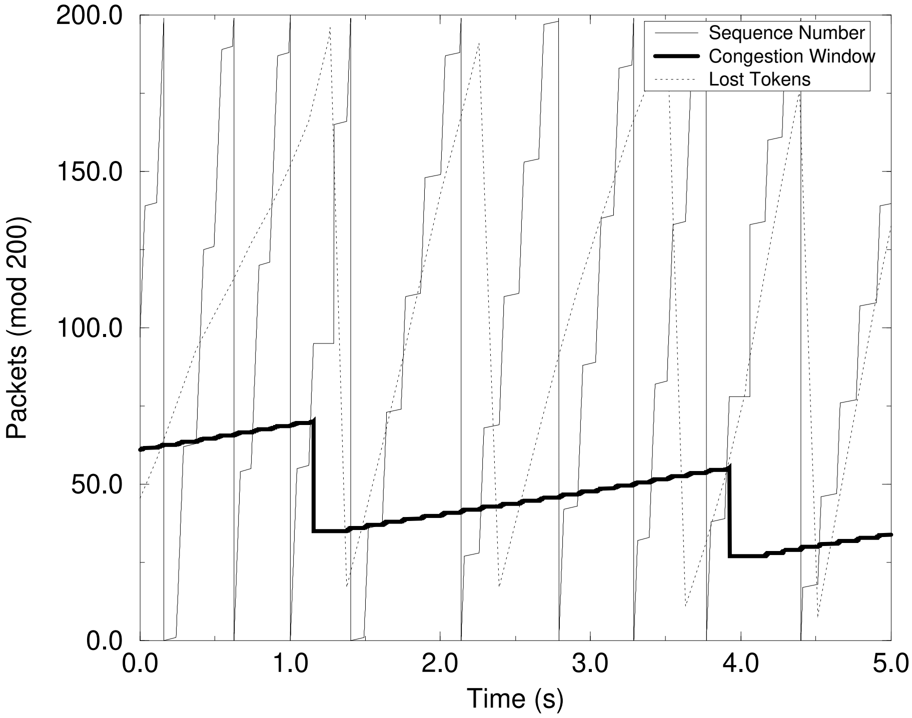


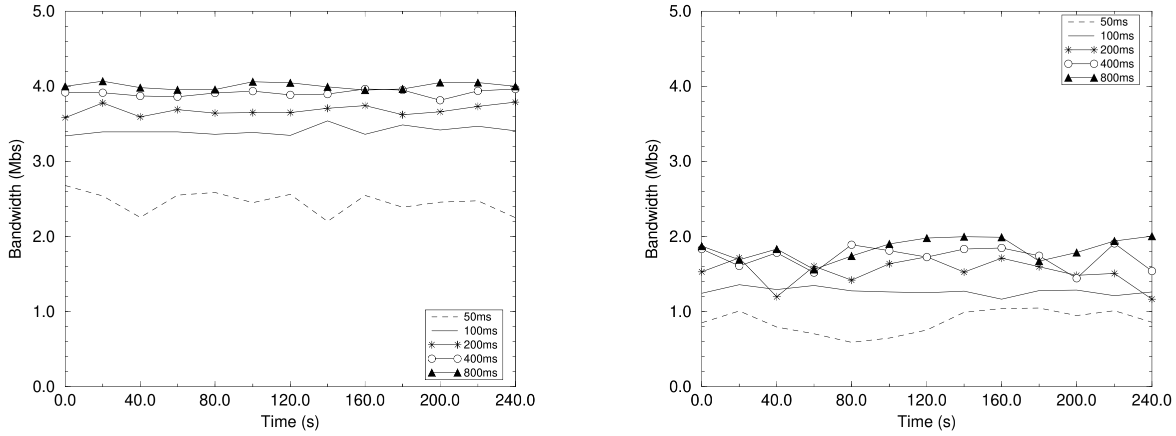
Figure 3: Packet trace of the connection with `4Mbs` reservation.

Figure 3 shows the packet trace of the connection with `4Mbs` reservation over a five-second interval. The plot shows the sequence number [3] (modulo 200) and the congestion window of the sender as well as the number of lost tokens (given in packets modulo 200) for the connection. A close look at the packet trace reveals a steady rate of token loss throughout the observation period. The windowing mechanism used by TCP is partly responsible for this phenomenon.

TCP uses two windows for the purpose of flow and congestion control. The receiver maintains and enforces an advertised window (AWND) as a measure of its buffering capacity. The sender enforces a congestion window (CWND) as a measure of the capacity of the network. The sender is prohibited from sending more than the minimum of AWND and CWND worth of unacknowledged data. When the loss of a segment is detected, TCP reduces the congestion window and initiates a fast recovery or a slow start phase. For the fast recovery phase, the congestion window is cut to half of its original size while in the slow start phase it is set to 1. For connections with reservations, this is an overly conservative behavior since it is insensitive to the reservation that a particular connection may have. Thus, even when tokens are present, and the sender is eligible to transmit a new segment, it may be throttled by the congestion window. As shown in Figure 3, the rate of token loss increases significantly when a packet loss is detected (as indicated by the decrease in congestion window), and slowly decreases as the congestion window opens up.

Another cause for token loss is the presence of persistent gaps in the acknowledgment stream. Such gaps are part of a phenomenon commonly referred to as ack-compression [18]. Since TCP uses acknowledgments to trigger transmissions, any significant time gap between the receipt of successive acknowledgments causes the token bucket to overflow and results in a loss of transmission credits. The effects of these gaps can be seen in many places in the trace where the sequence number is frozen. There are several ways in which these gaps can develop. One is through the recovery process after a loss is detected using TCP's fast recovery and fast retransmit mechanisms. After detecting a loss (by the receipt of a given number of duplicate acknowledgments), TCP cuts its congestion window in half by halting additional transmissions until one half of the original window's packets have cleared the network. Freezing the sender for this

[3] We use segments of size `1KB`. The sequence number is the sender's packet sequence number.



(a) One-way traffic.

(b) Two-way traffic.

Figure 4: Compliant throughput of the connection with `4Mbs` reservation.

period of time causes the token bucket to overflow, but more importantly, puts a gap in the data stream which results in a gap in the acknowledgment stream during the next round-trip interval. Gaps in the acknowledgments cause the token bucket to overflow and cause gaps in the data stream once again. Another way gaps can form is through the normal dynamics of network traffic. Congestion on the forward and/or reverse path, as well as additional queueing delays and jitter experienced as new connections come on-line, can also create significant gaps in the stream.

3.2 Effect of Token Bucket Depth

One way to alleviate the problem of token loss is to use a deeper token bucket. To investigate the impact of the token bucket depth on compliant throughput, we repeated the experiment described in the last section across a range of token bucket depths. Figure 4(a) shows the compliant throughput seen by the connection with a `4Mbs` reservation for token bucket sizes of `50ms`, `100ms`, `200ms`, `400ms`, and `800ms` using the same network topology and traffic. Increasing the token bucket depth improves the compliant throughput seen by a connection. However, it is only when the token buckets are very large (`400ms` and `800ms` in this case) that the compliant throughput seen by a connection remains at the reserved rate. Unfortunately, for a `4Mbs` connection, this bucket depth corresponds to a maximum burst of compliant packets of up to `200KB`. In order for the network to ensure that compliant packets are not dropped, it must have the capacity to buffer such bursts. Without sufficient buffer space, a significant amount of burst losses can occur, causing the performance of the TCP connection to deteriorate.

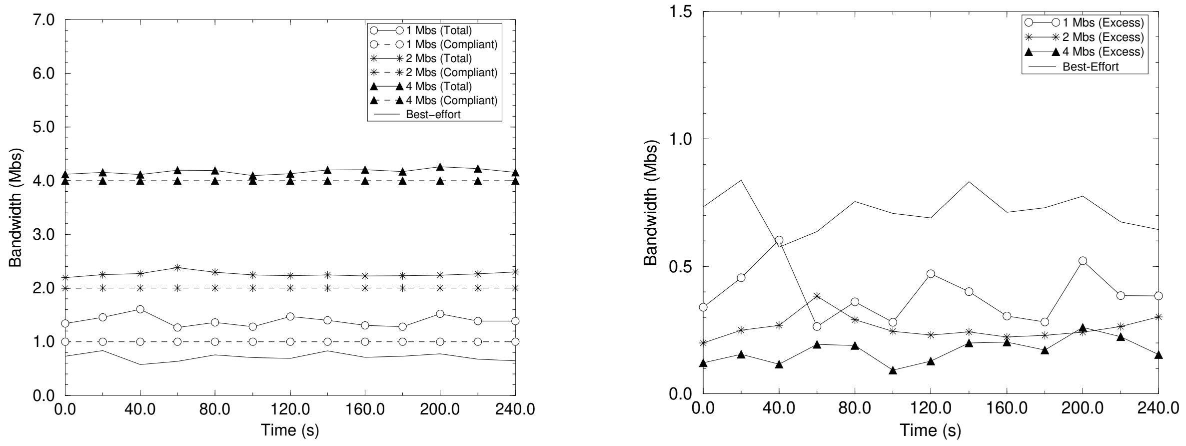
Another problem with simply increasing the size of the token bucket is that it is sensitive to variance in network traffic. Figure 4(b) shows the same experiment with identical traffic going in the reverse direction. That is, all connections are bidirectional. In contrast to Figure 4(a), large token buckets do not give any additional performance improvement. The connection never receives a compliant throughput more than half of its `4Mbs` reservation.

4 TCP Adaptations

In this section we propose and experiment with modifications to TCP's control mechanisms. These refinements can help TCP adapt better in an integrated services environment.

4.1 Timed Transmissions

Since deeper token buckets require larger buffers in routers, it is desirable to keep the size of the token buckets small. To alleviate the effects of persistent gaps in acknowledgment without increasing the token bucket depth significantly, we experimented with two different schemes, delayed and timed transmissions. These mechanisms better adapt the acknowledgment-based transmit triggers to the rate-based reservation paradigm. In the delayed transmission mechanism a segment is held back for a random amount of time when there aren't enough tokens to transmit it as a marked packet. This, in effect, adds randomization to the data stream of the connection which can potentially eliminate persistent gaps in the acknowledgment stream. In addition, this scheme reduces the



(a) Total and compliant throughput.

(b) Share of excess bandwidth.

Figure 5: Throughput with timer-triggered transmissions.

probability of the packets getting dropped inside the network since holding back packets increases the probability that they are sent as marked packets. While the delayed transmissions work reasonably well when the reverse path is lightly loaded [3], additional experiments have shown that it is not very effective in the presence of reverse path congestion.

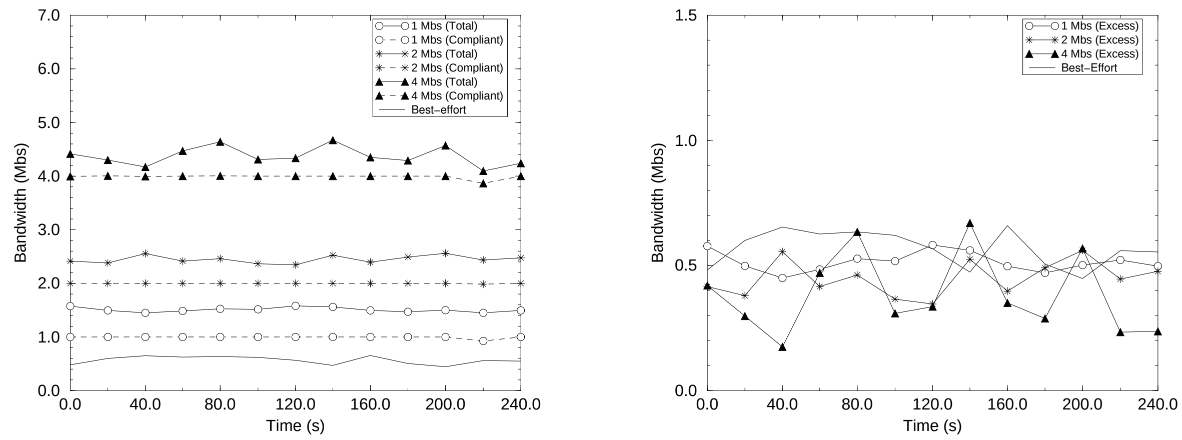
The second mechanism we examine involves the use of a periodic timer. In this scheme, we augment TCP's acknowledgment-triggered transmissions with a timer-triggered transmission mechanism. This timer-based triggering ensures that transmission opportunities are not lost while the connection is waiting for an acknowledgment. In the timed transmission mechanism, each reserved connection uses at most one timer which can have an interval which is customized. Connections can also share a single timer depending on the overhead on the end-host. In the timed transmission scheme, the acknowledgment-clocked transmission algorithm is left unmodified. However, whenever a periodic timer expires, the connection examines the tokens in the token bucket. If there are sufficient tokens available in the token bucket and there is room under the advertised window of the receiver, the sender sends the packet as marked, ignoring the value of the congestion window. The timer is then reset to wake up another timer interval later.

The intuition behind timed transmission is very simple. If there are enough tokens in the bucket, as per the contract with the network, the sender is eligible to inject new data in the network. Hence, we temporarily disregard the congestion window under such circumstances. Note, that the connection still adheres to the advertised window constraint to avoid overflowing the receiver's buffers. In case of network disruption, the sending TCP freezes when the number of unacknowledged packets reaches the advertised window. Thus, the time-triggered sends do not continue to occur in the presence of network failure. Having a timer trigger transmissions alleviates the problem of lost tokens caused by gaps in the acknowledgments. In order to guarantee zero token loss, the timer interval should be equal to [TokenBucketDepth - (PacketSize - 1)]/TokenBucketRate. This takes care of the worst case where there are (PacketSize - 1) tokens in the bucket when a timer interrupt occurs.

Using this timer mechanism, we reran the same experiment described earlier. For the experiment, we used token buckets of depth `50ms`, and a timer granularity of `20ms`. Figure 5(a) plots the total bandwidth received by all connections and the compliant bandwidth received by the connections with reservations. As shown in the figure, each connection gets its reserved rate and a share of the excess bandwidth.

While the timed transmissions allow for temporary violations of the congestion window to occur, non-compliant packets are sent only when there is room under the congestion window. Thus, this mechanism does not alter the base windowing mechanism used by TCP. Using TCP's windowing mechanism can be a problem since upon the detection of a loss, the congestion window is cut in half or reduced to 1 regardless of a connection's reservation. Thus, although the timed transmission mechanism allows the connection to receive its reserved rate, TCP's windowing mechanism can restrict the controlled-load connections from competing for the excess bandwidth [4] in the network. Figure 5(b) plots the throughput seen by a best-effort connection and the non-compliant throughput seen by each of the reserved connections using timed transmis-

[4] Non-conformant controlled-load traffic is treated as best-effort traffic. Hence, residual network capacity should be fairly shared between best-effort traffic and non-conformant controlled-load traffic.



(a) Aggregate and compliant throughput.

(b) Share of Excess Bandwidth.

Figure 6: Throughput after windowing modification.

sions. The plots show that connections with reservations receive a smaller share of the residual capacity when compared to the best-effort connection. The connections with larger reservations are penalized to a greater extent since halving the congestion window of a connection with 4Mbs reservation has a more drastic impact than halving the congestion window of a 1Mbs connection.

4.2 Rate Adaptive Windowing

For reserved connections, the swings in the congestion window should always be above the window guaranteed by the reserved rate. To account for and exploit the reservation, we modified TCP's windowing algorithm. The key idea behind this modification is that for reserved connections, CWND consists of two parts: a reserved part equal to the product of the reserved rate and the estimated round-trip time, and a variable part that tries to estimate the residual capacity and share it with other active connections. Note that the reserved part of CWND is a function of the round-trip time. While we currently use the common TCP round-trip measurements to estimate this, measurements using the proposed TCP timestamps option (RTTM) [10] can provide a more accurate estimate.

Let us assume that the size of the reservation window is RWND. Hence, the size of the variable window is CWND-RWND. In the modified scheme, we adjust the size of the variable window using the traditional TCP windowing mechanism and simply add it to the calculated value of RWND. Specifically, the sender, instead of reducing CWND by half at the beginning of the fast recovery, sets it to RWND + (CWND-RWND)/2. At the beginning of a slow start after detection of a lost segment through the retransmission timeout, it sets CWND to RWND+1 instead of 1. In both cases, SSTHRESH is set to the minimum of RWND+(CWND - RWND)/2 and AWND instead of the minimum of CWND/2 and AWND. Finally, because packets sent under RWND should not clock congestion window increases, we scale all window increases by (CWND-RWND)/CWND. Note that even with these modifications to the windowing algorithm, the sender must still adhere to the AWND restriction. That is, it is prohibited from sending more than the minimum of AWND and CWND worth of unacknowledged data. Because of this, the size of the receiver's buffer must be at least the size of the reservation window in order to sustain the reserved rate using TCP.

We repeated the experiments described in the last section with the windowing modifications in place. Figure 6(a) shows the aggregate and compliant throughput seen by each reserved connection after modifications to the windowing mechanisms. It also shows throughput seen by a best-effort connection between the same source and destination. As seen in the figure, all connections perform as expected. Figure 6(b) plots the amount of excess bandwidth received by each reserved connection, as well as the bandwidth received by the best-effort connection. When compared to Figure 5(b), the reserved connections obtain a fairer share of the excess bandwidth.

A common concern with any modification to TCP's windowing mechanism is that the change may be too aggressive and thus, cause unnecessary congestion. The experiments we have conducted so far, including the ones reported in this paper, show no bias towards connections using the modified windowing mechanism. We experimented with different flavors of the windowing algorithm. They differ in the way RWND is computed and CWND is clocked. We compute RWND by multiplying the reserved

CPU Type	133MHz PowerPC	33MHz POWER
Timer setting	7.4μs	14.0μs
Timer handling	7.1μs	30.1μs
Timer canceling	6.5μs	9.6μs

Table 1: Timer overheads (AIX 4.2 kernel).

rate with the estimated round-trip time. Depending on how conservative we want the windowing mechanism to be, we can use different estimates of round-trip time. We experimented with both best and average estimates of round-trip times. In all the experiments we have conducted, they perform equally well. However, in times of congestion, the estimated round-trip time tends to be large and thus, the rate-based window can also grow large during a period of time when the network needs a respite. Using the best observed round-trip time in this case, allows the connection to be on the conservative side in calculating its rate-based window.

5 Fine-Grained Timers

This section explores the cost associated with deploying fine-grained timers into TCP as well as the benefits of using such a timer for sending data.

5.1 Timer Overheads

In our description of the timed transmission algorithm, we have assumed the existence of connection-specific timers. However, it is possible, and desirable, to use a common timer shared amongst all reserved connections. Such optimizations can be easily incorporated using techniques such as the protocol timer services in the BSD-style TCP/IP stack. One of the common criticisms against the use of timers is the overhead associated with handling timer interrupts. For that reason, TCP uses coarse-grained (typically 200ms and 500ms) timers. However, the state of the art in processor technology has advanced considerably since the first design and implementation of TCP. Processors today are much faster, and consequently the overheads of handling timer interrupts are much lower.

Table 1 shows the overheads of setting, canceling, and handling timers in two IBM RS/6000 machines running AIX 4.2, one equipped with a 33MHz POWER CPU and the other with a 133 MHz PowerPC CPU. We observe that the overheads of timer operations in modern systems (133 MHz PowerPC) are quite small. Even when older systems, such as the 33MHz RS/6000, are considered in this study, the overheads are well within acceptable limits. Note that these measurements were taken without any optimization to the timer data structures in the AIX kernel. In AIX 4.2 timer blocks are arranged in a linear array. The overhead of timer operations are expected to be even lower if the timer blocks are stored as a hash table. However, at this point such an optimization is not deemed necessary.

5.2 Buffer Requirements

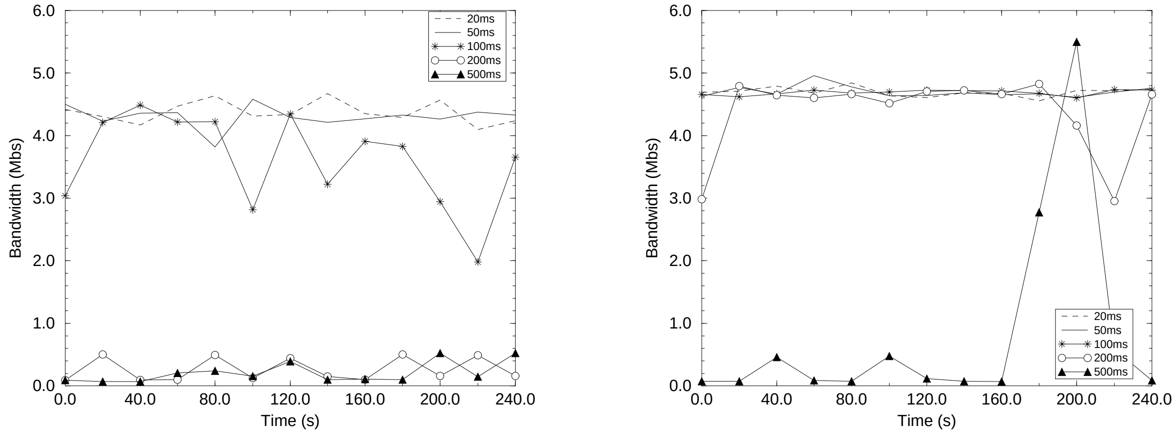
While there are concrete costs associated with using fine-grained timers, there are also significant benefits. One benefit in using these timers, is that it reduces the size of the token buckets used for each application. From the calculations in Section 4, given a certain timer interval, the token bucket depth should be at least TimerInterval $\times$ TokenBucketRate+(PacketSize-1) to prevent token loss. Because the token bucket size grows linearly with the timer interval, using fine-grained timers allow applications to request smaller token buckets. Because each router must be able to buffer a burst the size of the entire token bucket for each application, the size of these buckets has a direct impact on the amount of buffer space required in network routers.

Figure 7 shows the impact that the timer-interval has on the throughput of the `4Mbs` connection using the same multi-hop network setup. The simulations were run using both the timer and windowing modifications as described in Section 4. As Figure 7(a) shows, as the timer interrupt interval increases, the throughput of this connection drops considerably. The reason why this drop is so dramatic is that the lack of buffer space in the network causes a significant amount of burst losses. Burst losses severely limit the throughput of Reno TCP-variants since it takes one round trip time to recover from each loss. This causes the sending TCP to degenerate into a stop-and-wait protocol. Figure 7(b) shows the results of the same experiment using buffers which are twice the size (`160KB`). With significantly larger buffers, the connection is able to get its share of the bandwidth over a larger range of timer interrupt intervals.

6 CBQ and ERED

The experiments in this paper have shown how to effectively provide minimal bandwidth guarantees in the network using ERED gateways. While this service may be useful to a large class of applications, in a fully-evolved integrated services Internet, such a mechanism must coexist with other mechanisms for providing a range of alternative services. This will allow applications written using such a service, to continue to work as the Internet infrastructure is upgraded and more sophisticated packet and link scheduling support is put into place. In this section, we examine the different approaches to traffic management in the Internet and the role the ERED mechanism can play in such an environment.

Class-based queueing (CBQ) [5] is one of the most popular mechanisms proposed for packet and link scheduling



(a) 80KB Buffers.

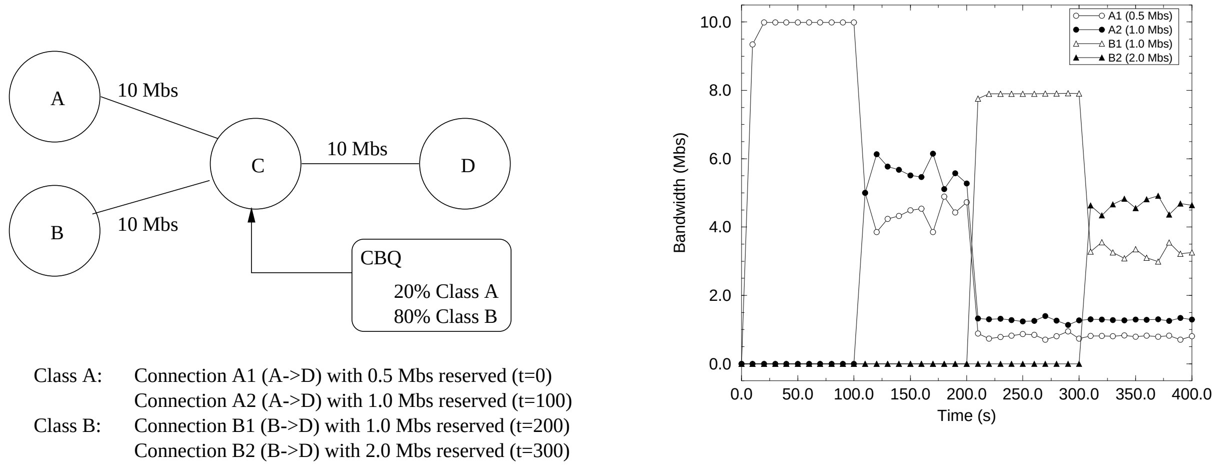
(b) 160KB Buffers.

Figure 7: Aggregate throughput of 4Mbs connection over various timer interrupt granularities.

in an integrated services Internet. In CBQ, datagrams belonging to different classes are put in different queues in the routers. The queues are serviced in different priority order based on the allocation given to the associated traffic class. The use of per-session fair queueing mechanisms, including the weighted fair queueing (WFQ) [1, 2, 6, 7, 14–16], has also been considered in this context. One possible way to realize controlled-load service is to treat each controlled-load connection as a separate session and reserve a rate of service commensurate with its Tspec. Any excess traffic will then automatically receive a fair share of the residual capacity. While per-session fair queueing provides an effective way to implement controlled-load service, the underlying scheduling mechanism may be expensive, especially in the overloaded and under-powered routers in today's Internet.

While the ERED mechanism can be thought of as a simple extension to the queueing structure in the existing routers, it can also be embedded into more sophisticated scheduling disciplines such as the ones described above. For example, the ERED queue can be easily embedded as a separate queue in a WFQ system supporting multiple service classes or as a class queue in CBQ. Embedded as a class in CBQ, this mechanism can be used to provide weighted bandwidth sharing between connections of a class. By aggregating connections with varying bandwidth requirements in one class, we reduce the total number of classes in a class-based queue and thus, the overhead in the link scheduling. To examine this possibility, we embedded the ERED queue into the CBQ implementation of the NS simulator. We then examined its performance in the network shown in Figure 8(a). This network consists of two agencies, A and B, who share a common link (between nodes C and D) to a provider's network. In this setup, agency A is entitled to 20% of the link's bandwidth while agency B is entitled to the remaining 80% of it. Node C uses CBQ with 20% of the link share allocated to traffic from agency A, and 80% allocated to traffic from agency B. Note that while either one of the two agencies is idle, the other, active agency, is entitled to use the entire link for itself. Both queues within the CBQ system use the ERED mechanism to share bandwidth between individual connections.

Figure 8(b) shows the throughput seen by connections originating from A and B and traversing the link between C and D. Connections A1 and A2 originate from agency A and have reserved rates of `0.5Mbs` and `1Mbs`, respectively. They start at times `0s` and `100s`. Connections B1 and B2 originate from agency B and have reserved rates of `1Mbs` and `2Mbs`. These connections start at times `200s` and `300s`, respectively. As the graph shows, between `0s` and `100s`, connection A1 gets all of the bandwidth since it is the only active connection. Between `100s` and `200s` (after connection A2 starts) the link's bandwidth is shared between connections A1 and A2. However, since A2 has a `1Mbs` reservation, it gets slightly more total bandwidth than A1. When B1 starts at `200s`, it is the only active connection from agency B. Hence, it receives the entire 80% of the link's bandwidth (`8Mbs`). The two connections from agency A then share the remaining bandwidth (`2Mbs`) according to their reservations. Finally, at `300s`, connection B2 starts and the `8Mbs` allocated to agency B is split between connection B1 and B2 in accordance with their reservations, that is, B2 gets approximately `1.0Mbs` more than B1. What happens throughout the course of this experiment is that when the class is allowed to be overlimited, the ERED queue is drained at a sufficient rate so as to support higher rates of input data. As soon as the class becomes regulated, the queue builds up, the ERED queue drops unmarked packets and the connections in the



(a) Network Topology. (b) Throughput.

Figure 8: CBQ experiment.

class resumes sending at a lower rate.

7 Conclusions

We have examined ways of providing a large class of overload-sensitive Internet applications with a useful service using minimal enhancements to the network infrastructure. Towards this end, we have proposed a simple extension to the packet queueing and scheduling mechanism at the routers. We have shown how minor modifications to TCP senders, in conjunction with this simple network support, allows connections with reservations to obtain their reserved rates and share excess bandwidth in the network.

The study reported in this paper can be extended in many ways. We are working on implementing and experimenting with the mechanisms proposed here in a network testbed. We are also considering applications of this work in the context of other transport protocols, especially RTP and UDP. Many multimedia applications do not require the reliable delivery that TCP provides. While this study focuses on TCP, implementing a similar scheme using RTP and UDP fitted with TCP's flow-control mechanism is possible.

Another key area of future work is the admission control policies for such a service. While we have not addressed this aspect here, we plan on using our observations on token bucket depths, router buffer sizes, source burstiness, and ERED-parameterization to develop admission control policies for this service.

We also plan on examining the effect of round-trip time on end-to-end throughput. A number of studies have shown that relative throughput seen by best-effort connections sharing a common bottleneck link has a strong dependence on their round-trip times. Preliminary results (not reported here) from our experiments, show that the compliant throughput of connections with reservations is largely independent of their round-trip times. However, the non-compliant part of the throughput and the throughput seen by best-effort connections are sensitive to round-trip time. A thorough investigation is underway to explore this aspect in more detail.

On a final note, any allocation-based sharing of network capacity has to be associated with a policy and/or pricing scheme. We believe that the proposal of prioritizing a part of a traffic stream with marking and competing with best-effort traffic for sharing the residual capacity fits in very well with pricing schemes which are currently being considered for the Internet. Users paying for a certain level of marking see incrementally better performance over those who do not. During times of light loads, when the incremental costs of congestion are low, the user can decrease his/her level of bandwidth reservation and costs until an acceptable level of aggregate throughput is observed.

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