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TCP Friendly Rate Control (TFRC): the Small-Packet (SP) Variant

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Abstract

The proposal in this document is intended to be experimental. More specifically, this proposal is intended for further experimentation, but not for widespread deployment at this time in the global Internet.

TCP-Friendly Rate Control (TFRC) is a congestion control mechanism for unicast flows operating in a best-effort Internet environment [RFC 3448]. TFRC was intended for applications that use a fixed packet size, and was designed to be reasonably fair when competing for bandwidth with TCP connections using the same packet size. This document proposes TFRC-SP, a Small-Packet (SP) variant of TFRC, that is designed for applications that send small packets. The design goal for TFRC-SP is to achieve the same bandwidth in bps as a TCP flow using packets of up to 1500 bytes. TFRC-SP enforces a Min Interval of 10 ms between data packets, to prevent a single flow from sending small packets arbitrarily frequently.

Flows using TFRC-SP compete reasonably fairly with large-packet TCP and TFRC flows in environments where large-packet flows and small-packet flows experience similar packet drop rates. However, in environments where small-packet flows experience lower packet drop rates than large-packet flows (e.g., with Drop-Tail queues in units of bytes), TFRC-SP can receive considerably more than its share of the bandwidth.

TO BE DELETED BY THE RFC EDITOR UPON PUBLICATION:

Changes from draft-ietf-dccp-tfrc-voip-03.txt:
* Added a paragraph saying that this is intended for Experimental, for further experimentation and not for widespread deployment.
* Editing of abstract so that it still fits the 25-line limit.

Changes from draft-ietf-dccp-tfrc-voip-02.txt:
* Changed name from "VoIP variant of TFRC" to "TFRC-SP".
* Added Section 4.5 on "The Nominal Packet Size", discussing possible differences in packet drop rates between small and large packets.
* Added text to Section 5 on "A Comparison with RFC 3714".
* Added text to Section 6 on "TFRC-SP with Applications that Modify the Packet Size"
* Added simulations with small-packet TCP flows.
* Added a Security Considerations section.
* Minor editing.

Changes from draft-ietf-dccp-tfrc-voip-01.txt:
* Added modified algorithm for calculating the loss event rate, for short intervals with multiple packet drops.
* Moved Faster Restart to a separate document.
* Added simulations with a configured byte drop rate.
* Added many more simulations, including Drop-Tail with a queue in bytes.
* Added a discussion of unfairness for Drop-Tail with a queue in bytes.

Changes from draft-ietf-dccp-tfrc-voip-00.txt:
* Added more simulations.
* Added a Related Work section.

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1. Conventions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC 2119].

2. Introduction

This document specifies TFRC-SP, a Small-Packet variant of TCP-friendly rate control (TFRC) [RFC 3448].

TFRC was designed to be reasonably fair when competing for bandwidth with TCP flows, but to avoid the abrupt changes in the sending rate characteristic of TCP's congestion control mechanisms. TFRC is intended for applications such as streaming media applications where a relatively smooth sending rate is of importance. Conventional TFRC measures loss rates by estimating the loss event ratio as described in [RFC 3448], and uses this loss event rate to determine the sending rate in packets per round-trip time. This has consequences for the rate a TFRC flow can achieve when sharing a bottleneck with large-packet TCP flows. In particular, a low-bandwidth, small-packet TFRC flow sharing a bottleneck with high-bandwidth, large-packet TCP flows may be forced to slow down, even though the TFRC application's nominal rate in bytes per second is less than the rate achieved by the TCP flows. Intuitively, this would be "fair" only if the network limitation was in packets per second (such as a routing lookup), rather than bytes per second (such as link bandwidth). Conventional wisdom is that many of the network limitations in today's Internet are in bytes per second, even though the network limitations of the future might move back towards limitations in packets per second.

TFRC-SP is intended for flows that need to send frequent small packets, limited by a minimum interval between packets of 10 ms. (RFC 3448 refers to this variant of TFRC as TFRC-PS, for applications that might vary their packet size in response to congestion.) It will better support applications that do not want their sending rates in bytes per second to suffer from their use of small packets. This variant is restricted to applications that send packets no more than once every 10 ms (the Min Interval). Given this restriction, TFRC-SP effectively calculates the TFRC fair rate as if the bottleneck restriction was in bytes per second. Applications using TFRC-SP could have a fixed packet size, or could vary their packet size in response to congestion.

TFRC-SP is motivated in part by the approach in RFC 3714, which argues that it is acceptable for VoIP flows to assume that the network limitation is in bytes per second (Bps) rather in packets per second (pps), and to have the same sending rate in bytes per second as a TCP flow with 1500-byte packets and the same packet drop rate. RFC 3714 states the following:

> "While the ideal would be to have a transport protocol that is able to detect whether the bottleneck links along the path are limited in Bps or in pps, and to respond appropriately when the limitation is in pps, such an ideal is hard to achieve. We would not want to delay the deployment of congestion control for telephony traffic until such an ideal could be accomplished. In addition, we note that the current TCP congestion control mechanisms are themselves not very effective in an environment where there is a limitation along the reverse path in pps. While the TCP mechanisms do provide an incentive to use large data packets, TCP does not include any effective congestion control mechanisms for the stream of small acknowledgement packets on the reverse path. Given the arguments above, it seems acceptable to us to assume a network limitation in Bps rather than in pps in

considering the minimum sending rate of telephony traffic."

Translating the discussion in [RFC 3714] to the congestion control mechanisms of TFRC, it seems acceptable to standardize a variant of TFRC that allows low-bandwidth flows sending small packets to achieve a rough fairness with TCP flows in terms of the sending rate in Bps, rather than in terms of the sending rate in pps. This is accomplished by TFRC-SP, a small modification to TFRC, as described below.

Maintaining incentives for large packets: Because the bottlenecks in the network in fact can include limitations in pps as well as in Bps, we pay special attention to the potential dangers of encouraging a large deployment of best-effort traffic in the Internet consisting entirely of small packets. This is discussed in more detail in Section 4.3. In addition, as again discussed in Section 4.3, TFRC-SP includes the limitation of the Min Interval between packets of 10 ms.

Packet drop rates as a function of packet size: TFRC-SP essentially assumes that the small-packet TFRC-SP flow receives roughly the same packet drop rate as a large-packet TFRC or TCP flow. As we show, this assumption is not necessarily correct for all environments in the Internet.

Calculating the loss event rate for TFRC-SP: TFRC-SP requires a modification in TFRC's calculation of the loss event rate, because a TFRC-SP connection can send many small packets when a standard TFRC or TCP connection would send a single large packet. It is not possible for a standard TFRC or TCP connection to repeatedly send multiple packets per round-trip time in the face of a high packet drop rate. As a result, TCP and standard TFRC only respond to a single loss event per round-trip time, and are still able to detect and respond to increasingly heavy packet loss rates. However, in a highly-congested environment, when a TCP connection might be sending, on average, one large packet per round-trip time, a corresponding TFRC-SP connection might be sending many small packets per round-trip time. As a result, in order to maintain fairness with TCP, and to be able to detect changes in the degree of congestion, TFRC-SP needs to be sensitive to the actual packet drop rate during periods of high congestion. This is discussed in more detail in the section below.

3. TFRC-SP Congestion Control

TFRC uses the TCP throughput equation given in Section 3.1 of [RFC 3448], which gives the allowed sending rate X in bytes per second as a function of the loss event rate, packet size, and round-trip time. [RFC 3448] specifies that the packet size s used in the throughput equation should be the packet size used by the application, or the estimated mean packet size if there are variations in the packet size depending on the data. This gives rough fairness with TCP flows using the same packet size.

TFRC-SP changes this behavior in the following ways.

- The nominal packet size: The nominal packet size s is set to 1460 bytes. Following [RFC 3714], this provides a goal of fairness, in terms of the sending rate in bytes per second, with a TCP flow with 1460 bytes of application data per packet but with the same packet drop rate.
- Taking packet headers into account: The allowed transmit rate X in bytes per second is reduced by a factor that accounts for packet header size. This gives the application some incentive, beyond the Min Interval, not to use unnecessarily small packets. In particular, we introduce a new parameter H, which represents the expected size in bytes of network and transport headers to be used on the TFRC connection's packets. This is used to reduce the allowed transmit rate X as follows:

X := X * s_true / (s_true + H),

where s_true is the true average data packet size for the connection in bytes, excluding the transport and network headers.

The H parameter is set to the constant 40 bytes. Thus, if the TFRC-SP application used 40-byte data segments, the allowed transmit rate X would be halved to account for the fact that half of the sending rate would be used by the headers. Section 4.2 justifies this definition. However, a connection using TFRC-SP MAY instead use a more precise estimate of H, based on the actual network and transport headers to be used on the connection's packets. For example, a DCCP connection [DCCP] over IPv4, where data packets use the DCCP-Data packet type, and there are no IP or DCCP options, could set H to 20 + 12 = 32 bytes.

- Measuring the loss event rate in times of high loss: During short loss intervals (those at most two round-trip times), the loss rate is computed by counting the actual number of packets lost or marked, not by counting at most one loss event per loss interval. Without this change, TFRC-SP could send multiple packets per round-trip time even in the face of heavy congestion, for a steady-state behavior of multiple packets dropped each round-trip time.

  In standard TFRC, the TFRC receiver estimates the loss event rate by calculating the average loss interval in packets, and inverting to get the loss event rate. Thus, for a short loss interval with N packets and K losses, standard TFRC calculates the size of that loss interval as N packets, contributing to a loss event rate of 1/N. However, for TFRC-SP, for small loss intervals of at most two round-trip times, if the loss interval consists of N packets including K losses, the size of the loss interval is calculated as N/K, contributing to a loss event rate of K/N instead of 1/N.
- A minimum interval between packets: TFRC-SP enforces a Min Interval between packets of 10 ms. A flow that wishes its transport protocol to exceed this Min Interval MUST use the conventional TFRC equations, rather than TFRC-SP. The motivation for this is discussed below.

4. TFRC-SP Discussion

4.1. The TCP Throughput Equation

TFRC-SP uses the TCP throughput equation given in [RFC 3448]. As shown in Table 1 of [RFC 3714], for high packet drop rates, this throughput equation gives rough fairness with the most aggressive possible current TCP: a SACK TCP flow using timestamps and ECN. Because it is not recommended for routers to use ECN-marking in highly-congested environments (e.g., with packet drop rates greater than 10%), we note that it would be useful to have a throughput equation with a somewhat more moderate sending rate for packet drop rates of 40% and above.

4.2. Accounting for Header Size

[RFC 3714] makes the optimistic assumption that the limitation of the network is in bandwidth in bytes per second (Bps), and not in CPU cycles or in packets per second (pps). However, some attention must be paid to the load in pps as well as to the load in Bps. Even aside from the Min Interval, TFRC-SP gives the application some incentive to use fewer but larger packets, when larger packets would suffice, by including the bandwidth used by the packet header in the allowed sending rate.

As an example, a sender using 120-byte packets needs a TCP-friendly rate of 128 Kbps to send 96 Kbps of application data. This is because the TCP-friendly rate is reduced by a factor of s_true/(s_true + H) = 120/160, to account for the effect of packet headers. If the sender suddenly switched to 40-byte data segments, the allowed sending rate would reduce to 64 Kbps of application data; and the use of one-byte data segments would reduce the allowed sending rate to 3.12 Kbps of application data. (In fact, the Min Interval would prevent senders from achieving these rates, since applications using TFRC-SP cannot send more than 100 packets per second.)

Unless it has a more precise estimate of the header size, TFRC-SP assumes 40 bytes for the header size, although the header could be larger (due to IP options, IPv6, IP tunnels, and the like) or smaller (due to header compression, DCCP instead of UDP) on the wire. Requiring the use of the exact header size in bytes would require significant additional complexity, and would have little additional benefit. TFRC-SP's default assumption of a 40-byte header is sufficient to get a rough estimate of the throughput, and to give the application some incentive not to use unnecessarily-many small packets. Because we are only aiming at rough fairness, and at a rough incentive for applications, the default use of a 40-byte header in the calculations of the header bandwidth seems sufficient.

4.3. The TFRC-SP Min Interval

The header size calculation provides an incentive for applications to use fewer, but larger, packets. Another incentive is that when the path limitation is in pps, the application using more small packets is likely to cause higher packet drop rates, and to have to reduce its sending rate accordingly. That is, if the congestion is in terms of pps, then the flow sending more pps will increase the packet drop rate, and have to adjust its sending rate accordingly. However, the increased congestion caused by the use of small packets in an environment limited by pps is experienced not only by the flow using the small packets, but by all of the competing traffic on that congested link. These incentives are therefore insufficient to provide sufficient protection for pps network limitations.

TFRC-SP, then, includes a Min Interval of 10 ms. This provides additional restrictions on the use of unnecessarily many small packets.

One justification for the Min Interval is the practical one that the applications that currently want to send small packets are the VoIP applications that send at most one packet every 10 ms, so this restriction does not affect current traffic. A second justification is that there is no pressing need for best-effort traffic in the current Internet to send small packets more frequently than once every 10 ms (rather than taking the 10 ms delay at the sender, and merging the two small packets into one larger one). This 10 ms delay for merging small packets is likely to be dominated by the network propagation, transmission, and queueing delays of best-effort traffic in the current Internet. As a result, our judgment would be that the benefit to the user of having less than 10 ms between packets is outweighed by the benefit to the network of avoiding unnecessarily many small packets.

The Min Interval causes TFRC-SP not to support applications sending small packets very frequently. Consider a TFRC flow with a fixed packet size of 100 bytes, but with a variable sending rate and a fairly uncongested path. When this flow was sending at most 100 pps, it would be able to use TFRC-SP. If the flow wished to increase its sending rate to more than 100 pps, but to keep the same packet size, it would no longer be able to achieve this with TFRC-SP, and would have to switch to the default TFRC, receiving a dramatic, discontinuous decrease in its allowed sending rate. This seems not only acceptable, but desirable for the global Internet.

What is to prevent flows from opening multiple connections, each with a 10 ms Min Interval, and thereby getting around the limitation of the Min Interval? Obviously, there is nothing to prevent flows from doing this, just as there is currently nothing to prevent flows from using UDP, or from opening multiple parallel TCP connections, or from using their own congestion control mechanism. Of course, implementations or middleboxes are also free to limit the number of parallel TFRC connections opened to the same destination in times of congestion, if that seems desirable. And flows that open multiple parallel connections are subject to the inconveniences of reordering and the like.

4.4. Counting Packet Losses

It is not possible for a TCP connection to persistently send multiple packets per round-trip time in the face of high congestion, with a steady-state with multiple packets dropped per round-trip time. For TCP, when one or more packets are dropped each round-trip time, the sending rate is quickly dropped to less than one packet per round-trip time. In addition, for TCP with Tahoe, NewReno, or SACK congestion control mechanisms, the response to congestion is largely independent of the number of packets dropped per round-trip time.

As a result, standard TFRC can best achieve fairness with TCP, even in highly congested environments, by calculating the loss event rate rather than the packet drop rate, where a loss event is one or more packets dropped or marked from a window of data.

However, with TFRC-SP, it is no longer possible to achieve fairness with TCP or with standard TFRC by counting only the loss event rate [WBL04]. Instead of sending one large packet per round-trip time, TFRC-SP could be sending N small packets (where N small packets equal one large 1500-byte packet). The loss measurement used with TFRC-SP has to be able to detect a connection that is consistently receiving multiple packet losses or marks per round-trip time, to allow TFRC-SP to respond appropriately.

In TFRC-SP, the loss event rate is calculated by counting at most one loss event in loss intervals longer than two round-trip times, and by counting each packet lost or marked in shorter loss intervals. In particular, for a short loss interval with N packets, including K lost or marked packets, the loss interval length is calculated as N/K, instead as N. The average loss interval I_mean is still averaged over the most recent eight loss intervals, as specified in Section 5.4 of RFC 3448. Thus, if eight successive loss intervals are short loss intervals with N packets and K losses, the loss event rate is calculated as K/N, rather than as 1/N.

4.5. The Nominal Packet Size

The guidelines in Section 3 above say that the nominal packet size s is set to 1460 bytes, providing a goal of fairness, in terms of the sending rate in bytes per second, with a TCP flow with 1460 bytes of application data per packet but with the same packet drop rate. This follows the assumption that a TCP flow with 1460-byte packets will have a higher sending rate than a TCP flow with smaller packets. While this assumption holds in an environment where the packet drop rate is independent of packet size, this assumption does not necessarily hold in an environment where larger packets are more likely to be dropped than are small packets.

The table below shows the results of simulations with standard (SACK) TCP flows, where, for each *byte*, the packet containing that byte is dropped with probability p. Consider the approximation for the TCP response function for packet drop rates up to 10% or so; for this environments, the sending rate in bytes per RTT is roughly 1.2 s/sqrt(p), for a packet size of s bytes and packet drop rate p. Given a fixed *byte* drop rate p1, and a TCP packet

size of s bytes, the packet drop rate is roughly s*p1, producing a sending rate in bytes per RTT of roughly 1.2 sqrt(s)/sqrt(p1). Thus, for TCP in an environment with a fixed byte drop rate, the sending rate should grow roughly as sqrt(s), for packet drop rates up to 10% or so.

Each row of Table 1 below shows a separate simulation with ten TCP connection and a fixed byte drop rate of 0.0001, with each simulation using a different segment size. For each simulation, the TCP sending rate and goodput are averaged over the ten flows. As one would expect from the paragraph above, the TCP sending rate grows somewhat less than linearly with an increase in packet size, up to a packet size of 1460 bytes, corresponding to a packet drop rate of 13%. After that, further increases in the packet size result in a *decrease* in the TCP sending rate, as the TCP connection enters the regime of exponential backoff of the retransmit timer.

Segment Size (B)	Packet DropRate	TCP Rates (Kbps) SendRate	Goodput
14	0.005	6.37	6.34
128	0.016	30.78	30.30
256	0.028	46.54	44.96
512	0.053	62.43	58.37
1460	0.134	94.15	80.02
4000	0.324	35.20	21.44
8000	0.531	15.36	5.76

Table 1: TCP Median Send Rate vs. Packet Size I: Byte Drop Rate 0.0001

Table 2 below shows similar results for a byte drop rate of 0.001. In this case, the TCP sending rate grows with increasing packet size up to a packet size of 128 bytes, corresponding to a packet drop rate of 16%. After than, the TCP sending rate decreases and then increases again, as the TCP connection enters the regime of exponential backoff of the retransmit timer. Note that with this byte drop rate, with packet sizes of 4000 and 8000 bytes, the TCP sending rate increases but the TCP goodput rate remains essentially zero. This makes sense, as almost all packets that are sent are dropped.

Segment Size (B)	Packet DropRate	TCP Rates (Kbps) SendRate	Goodput
14	0.053	1.68	1.56
128	0.159	7.66	6.13
256	0.248	6.21	4.32
512	0.402	1.84	1.11
1460	0.712	1.87	0.47
4000	0.870	3.20	0.00
8000	0.890	5.76	0.00

Table 2: TCP Median Send Rate vs. Packet Size II: Byte Drop Rate 0.001

The TCP behavior in the presence of a fixed byte drop rate suggests that instead of the goal of a TFRC-SP flow achieving the same sending rate in bytes per second as a TCP flow using 1500-byte packets, it makes more sense to consider an ideal goal of a TFRC-SP flow

achieving the same sending rate as a TCP flow with the optimal packet size, given that the packet size is at most 1500 bytes. This does not mean that we need to change the TFRC-SP mechanisms for computing the allowed transmit rate; this means simply that in evaluating the fairness of TFRC-SP, we should consider fairness relative to the TCP flow using the optimal packet size (though still at most 1500 bytes) for that environment.

5. A Comparison with RFC 3714

RFC 3714 considers the problems of fairness, potential congestion collapse, and poor user quality that could occur with the deployment of non-congestion-controlled IP telephony over congested best-effort networks. The March 2004 document cites ongoing efforts in the IETF, including work on TFRC and DCCP. RFC 3714 recommends that for best-effort traffic with applications that have a fixed or minimum sending rate, the application or transport protocol should monitor the packet drop rate, and discontinue sending for a period if the steady-state packet drop rate significantly exceeds the allowed threshold for that minimum sending rate.

In determining the allowed packet drop rate for a fixed sending rate, RFC 3714 assumes that an IP telephony flow should be allowed to use the same sending rate in bytes per second as a 1460-byte-packet TCP flow experiencing the same packet drop rate as that of the IP telephony flow. As an example, following this guideline, a VoIP connection with a round-trip time of 0.1 sec and a minimum sending rate of 64 kbps would be required to terminate or suspend when the persistent packet drop rate significantly exceeded 25%.

One limitation of the lack of fine-grained control in the minimal mechanism described in RFC 3714 is that an IP telephony flow would not adapt its sending rate in response to congestion. In contrast, with TFRC-SP, a small-packet flow would reduce its sending rate somewhat in response to moderate packet drop rates, possibly avoiding a period with unnecessarily-heavy packet drop rates in the network.

Because RFC 3714 assumes that the allowed packet drop rate for an IP telephony flow is determined by the sending rate that a TCP would use *with the same packet drop rate*, the minimal mechanism in RFC 3714 would not provide fairness between TCP and IP telephony traffic in an environment where small packets are less likely to be dropped than large packets. In such an environment, the small-packet IP telephony flow would make the optimistic assumption that a large-packet TCP flow would receive the same packet drop rate as the IP telephony flow, and as a result the small-packet IP telephony flow would receive a larger fraction of the link bandwidth than a competing large-packet TCP flow.

6. TFRC-SP with Applications that Modify the Packet Size

One possible use for TFRC-SP would be with applications that maintain a fixed sending rate in packets per second, but modify their packet size in response to congestion. TFRC-SP monitors the connection's packet drop rate, and determines the allowed sending rate in bytes per second. Given an application with a fixed sending rate in packets per second, the TFRC-SP sender could determine the data packet size that would be needed for the sending rate in bytes per second not to exceed the allowed sending rate. In environments where the packet drop rate is affected by the packet size, decreasing the packet size could also result in a decrease in the packet drop rate experienced by the flow.

There are many questions about how an adaptive application would use TFRC-SP that are beyond the scope of this document. In particular, an application might wish to avoid unnecessary reductions in the packet size. In this case, an application might wait for some period of time before reducing the packet size, to avoid an unnecessary reduction in the

packet size. The details of how long an application might wait before reducing the packet size can be addressed in documents that are more application-specific.

Similarly, an application using TFRC-SP might only have a few packet sizes that it is able to use. In this case, the application might not reduce the packet size until the current packet drop rate has significantly exceeded the packet drop rate threshold for the current sending rate, to avoid unnecessary oscillations between two packet sizes and two sending rates. Again, the details will have to be addressed in documents that are more application-specific.

7. Simulation Results

This section explores the performance of TFRC-PS in simulation scenarios with configured packet or byte drop rates, and in scenarios with a range of queue management mechanisms at the congested link. The simulations explore environments where standard TFRC significantly limits the throughput of small-packet flows, and TFRC-SP gives the desired throughput. The simulations also explore environments where standard TFRC allows small-packet flows to receive good performance, while TFRC-SP is overly aggressive.

The general lessons from the simulations are as follows.

- In scenarios where large and small packets receive similar packet drop rates, TFRC-SP gives roughly the desired sending rate (Sections 7.1, 7.3).
- In scenarios where each *byte* is equally likely to be dropped, standard TFRC gives reasonable fairness between small-packet TFRC flows and large-packet TCP flows (Section 7.2).
- In scenarios where small packets are less likely to be dropped than large packets, TFRC-SP does not give reasonable fairness between small-packet TFRC-SP flows and large-packet TCP flows; small-packet TFRC-SP flows can receive considerably more bandwidth than competing large-packet TCP flows (Sections 7.2, 7.3, 7.4).
- Scenarios where small packets are less likely to be dropped than large packets include those with Drop-Tail queues in bytes, and with AQM mechanisms in byte mode (Sections 7.3, 7.4).

Those who are not interested in the details of the simulations could proceed directly to Section 8 on General Discussion.

TFRC-SP has been added to the NS simulator, and is illustrated in the validation test "./test-all-friendly" in the directory tcl/tests. The simulation scripts for the simulations in this document are available at "http://www.icir.org/tfrc/voipsims.html". There is also a pointer to the document "Graphs for draft-ietf-dccp-tfrc-voip-03", which has graphs showing the information in tables in this document.

7.1. Simulations with Configured Packet Drop Rates

In this section we describe simulation results from simulations comparing the throughput of standard (SACK) TCP flows, TCP flows with timestamps and ECN, TFRC-SP flows, and standard TFRC (Stnd TFRC) flows. In these simulations we configure the router to randomly drop or mark packets at a specified rate, independently of the packet size. For each specified packet drop rate, we give a flow's average sending rate in Kbps over the second half of a 100-second simulation, averaged over ten flows.

Packet DropRate	Send Rates (Kbps, 1460B seg) TCP	ECN TCP	TFRC
0.001	2020.85	1904.61	982.09
0.005	811.10	792.11	878.08
0.01	515.45	533.19	598.90
0.02	362.93	382.67	431.41
0.04	250.06	252.64	284.82
0.05	204.48	218.16	268.51
0.1	143.30	148.41	146.03
0.2	78.65	93.23*	55.14
0.3	26.26	59.65*	32.87
0.4	9.87	47.79*	25.45
0.5	3.53	32.01*	18.52

* ECN scenarios marked with an asterisk are not realistic, as routers are not recommended to mark packets when packet drop/mark rates are 20% or higher.

Table 3: Send Rate vs. Packet Drop Rate I:
1460B TFRC Segments
(1.184 Kbps Maximum TFRC Data Sending Rate)

Table 3 shows the sending rate for a TCP and a standard TFRC flow for a range of configured packet drop rates, when both flows have 1460-byte data packets, in order to illustrate the relative fairness of TCP and TFRC when both flows use the same packet size. For example, a packet drop rate of 0.1 means that 10% of the TCP and TFRC packets are dropped. The TFRC flow is configured to send at most 100 packets per second. There is good relative fairness until the packet drop percentages reach 40 and 50%, when the TFRC flow receives three to five times more bandwidth than the standard TCP flow. We note that an ECN TCP flow would receive a higher throughput than the TFRC flow. However, we don't use the ECN TCP sending rate in these high-packet-drop scenarios as the target sending rate for TFRC, as routers are advised to drop rather than mark packets during high levels of congestion.

	< - - - - - - - Send Rates (Kbps) - - - - - - >			
Packet DropRate	TCP (1460B seg)	ECN TCP (1460B seg)	TFRC-SP (14B seg)	Stnd TFRC (14B seg)
0.001	1787.54	1993.03	17.71	17.69
0.005	785.11	823.75	18.11	17.69
0.01	533.38	529.01	17.69	17.80
0.02	317.16	399.62	17.69	13.41
0.04	245.42	260.57	17.69	8.84
0.05	216.38	223.75	17.69	7.63
0.1	142.75	138.36	17.69	4.29
0.2	58.61	91.54*	17.80	1.94
0.3	21.62	63.96*	10.26	1.00
0.4	10.51	41.74*	4.78	0.77
0.5	1.92	19.03*	2.41	0.56

* ECN scenarios marked with an asterisk are not realistic, as routers are not recommended to mark packets when packet drop/mark rates are 20% or higher.

Table 4: Send Rate vs. Packet Drop Rate II:
14B TFRC Segments
(5.6 Kbps Maximum TFRC Data Sending Rate)

Table 4 shows the results of simulations where each TFRC-SP flow has a maximum data sending rate of 5.6 Kbps, with 14-byte data packets and a 32-byte packet header for DCCP and IP. Each TCP flow has a receive window of 100 packets and a data packet size of 1460 bytes, with a 40-byte packet header for TCP and IP. The TCP flow uses SACK TCP with Limited Transmit, but without timestamps or ECN. Each flow has a round-trip time of 240 ms in the absence of queueing delay.

The TFRC sending rate in Table 4 is the sending rate for the 14-byte data packet with the 32-byte packet header. Thus, only 30% of the TFRC sending rate is for data, and with a packet drop rate of p, only a fraction 1-p of that data makes it to the receiver. Thus, the TFRC data receive rate can be considerably less than the TFRC sending rate in the table. Because TCP uses large packets, 97% of the TCP sending rate is for data, and the same fraction 1-p of that data makes it to the receiver.

Table 4 shows that for the 5.6 Kbps data stream with TFRC, Standard TFRC (Stnd TFRC) gives a very poor sending rate in bps, relative to the sending rate for the large-packet TCP connection. In contrast, the sending rate for the TFRC-SP flow is relatively close to the desired goal of fairness in bps with TCP.

Table 4 shows that with TFRC-SP, the 5.6 Kbps data stream doesn't reduce its sending rate until packet drop rates greater than 20%, as desired. With packet drop rates of 30-40%, the sending rate for the TFRC-SP flow is somewhat less than that of the average large-packet TCP flow, while for packet drop rates of 50% the sending rate for the TFRC-SP flow is somewhat greater than that of the average large-packet TCP flow.

	< - - - - - - - Send Rates (Kbps) - - - - - - >			
Packet DropRate	TCP (1460B seg)	ECN TCP (1460B seg)	TFRC-SP (200B seg)	Stnd TFRC (200B seg)
0.001	1908.98	1869.24	183.45	178.35
0.005	854.69	835.10	185.06	138.06
0.01	564.10	531.10	185.33	92.43
0.02	365.38	369.10	185.57	62.18
0.04	220.80	257.81	185.14	45.43
0.05	208.97	219.41	180.08	39.44
0.1	141.67	143.88	127.33	21.96
0.2	62.66	91.87*	54.66	9.40
0.3	16.63	65.52*	24.50	4.73
0.4	6.62	42.00*	13.47	3.35
0.5	1.01	21.34*	10.51	2.92

* ECN scenarios marked with an asterisk are not realistic, as routers are not recommended to mark packets when packet drop/mark rates are 20% or higher.

Table 5: Sending Rate vs. Packet Drop Rate III:
200B TFRC Segments
(160 Kbps Maximum TFRC Data Sending Rate)

Table 5 shows results with configured packet drop rates when the TFRC flow uses 200-byte data packets, with a maximum data sending rate of 160 Kbps. As in Table 4, the performance of Standard TFRC is quite poor, while the performance of TFRC-SP is essentially as desired for packet drop rates up to 30%. Again as expected, with packet drop rates of 40-50% the TFRC-SP sending rate is somewhat higher than desired.

For these simulations, the sending rate of a TCP connection using timestamps is similar to the sending rate shown for a standard TCP connection without timestamps.

7.2. Simulations with Configured Byte Drop Rates

In this section we explore simulations where the router is configured to drop or mark each *byte* at a specified rate. When a byte is chosen to be dropped (or marked), the entire packet containing that byte is dropped (or marked).

		< - - - - - Send Rates (Kbps) - - - - - >			
Byte DropRate	TCP SegSize	TCP	ECN TCP	TFRC-SP (14B seg)	Stnd TFRC (14B seg)
0.00001	1460	423.02	431.26	17.69	17.69
0.0001	1460	117.41	114.34	17.69	17.69
0.001	128	10.78	11.50	17.69	8.37
0.005	14	1.75	2.89	18.39	1.91
0.010	1460	0.31	0.26	7.07	0.84
0.020	1460	0.29	0.26	1.61	0.43
0.040	1460	0.12	0.26*	0.17	0.12
0.050	1460	0.15	0.26*	0.08	0.06

* ECN scenarios marked with an asterisk are not realistic, as routers are not recommended to mark packets when packet drop/mark rates are 20% or higher.

TFRC's maximum data sending rate is 5.6 Kbps.

Table 6: Sending Rate vs. Byte Drop Rate

Table 6 shows the TCP and TFRC send rates for various byte drop rates. For each byte drop rate, Table 6 shows the TCP sending rate, with and without ECN, for the TCP segment size that gives the highest TCP sending rate. Simulations were run with TCP segments of 14, 128, 256, 512, and 1460 bytes. Thus, for a byte drop rate of 0.00001, the table shows the TCP sending rate with 1460-byte data segments, but with a byte drop rate of 0.001, the table shows the TCP sending rate with 128-byte data segments. For each byte drop rate, Table 6 also shows the TFRC-SP and that Standard TFRC sending rates. With configured byte drop rates, TFRC-SP gives an unfair advantage to the TFRC-SP flow, while Standard TFRC gives essentially the desired performance.

Byte DropRate	TCP Pkt DropRate (1460B seg)	TFRC Pkt DropRate (14B seg)	TCP/TFRC Pkt Drop Ratio
0.00001	0.015	0.0006	26.59
0.0001	0.13	0.0056	24.94
0.001	0.77	0.054	14.26
0.005	0.99	0.24	4.08
0.01	1.00	0.43	2.32
0.05	1.00	0.94	1.05

Table 7: Packet Drop Rate Ratio vs. Byte Drop Rate

Table 7 converts the byte drop rate p to packet drop rates for the TCP and TFRC packets, where the packet drop rate for an N-byte packet is 1-(1-p)^N. Thus, a byte drop rate of 0.001, with each byte being dropped with probability 0.001, converts to a packet drop rate of 0.77, or 77%, for the 1500-byte TCP packets, and a packet drop rate of 0.054, or 5.4%, for the 56-byte TFRC packets.

The right column of Table 7 shows the ratio between the TCP packet drop rate and the TFRC packet drop rate. For low byte drop rates, this ratio is close to 26.8, the ratio between the TCP and TFRC packet sizes. For high byte drop rates, where even most small TFRC packets are likely to be dropped, this drop ratio approaches 1. As Table 6 shows, with byte drop rates, the Standard TFRC sending rate is close to optimal, competing fairly with a TCP connection using the optimal packet size within the allowed range. In contrast, the TFRC-SP connection gets more than its share of the bandwidth, though it does reduce its sending rate for a byte drop rate of 0.01 or more (corresponding to a TFRC-SP *packet* drop rate of 0.43.

Table 6 essentially shows three separate regions. In the low-congestion region, with byte drop rates less than 0.0001, the TFRC-SP connection is sending at its maximum sending rate. In this region the optimal TCP connection is the one with 1460-byte packets, and the TCP sending rate goes as 1/sqrt(p), for packet drop rate p. This low-congestion region holds for packet drop rates up to 10-15%.

In the middle region of Table 6, with byte drop rates from 0.0001 to 0.005, the optimal TCP segment size is a function of the byte drop rate. In particular, the optimal TCP segment size seems to be the one that keeps the packet drop rate at most 15%, keeping the TCP connection in the regime controlled by a 1/sqrt(p) sending rate, for packet drop rate p. For a TCP packet size of S bytes (including headers), and a *byte* drop rate of B, the packet drop rate is roughly B*S. For a given byte drop rate in this regime, if the optimal TCP performance occurs with a packet size chosen to give a packet drop rate of at most 15%, keeping the TCP connection out of the regime of exponential backoffs of the retransmit timer, then this requires B*S = 0.15, or S = 0.15/B.

In the high-congestion regime of Table 6, with high congestion and with byte drop rates of 0.01 and higher, the TCP performance is dominated by the exponential backoff of the retransmit timer regardless of the segment size. Even a 40-byte packet with a zero-byte data segment would have a packet drop rate of at least 33%. In this regime, the optimal TCP *sending* rate comes with a large, 1460-byte data segment, but the TCP sending rate is low with any segment size, considerably less than one packet per round-trip time. We note that in this regime, while a larger packet gives a higher TCP *sending* rate, a smaller packet gives a better *goodput* rate.

In general, Tables 4 and 5 show good performance for TFRC-SP in environments with stable packet drop rates, where the decision to drop a packet is independent of the packet size. However, in some environments the packet size might affect the likelihood that a packet is dropped. For example, with heavy congestion and a Drop Tail queue with a fixed number of bytes rather than a fixed number of packet-sized buffers, small packets might be more likely than large packets to find room at the end of an almost-full queue. As a further complication, in a scenario with Active Queue Management, the AQM mechanism could either be in packet mode, dropping each packet with equal probability, or in byte mode, dropping each byte with equal probability. Sections 7.3 and 7.4 show simulations with packets dropped at Drop Tail or AQM queues, rather that from a probabilistic process.

7.3. Packet Dropping Behavior at Routers with Drop-Tail Queues

One of the problems with comparing the throughput of two flows using different packet sizes is that the packet size itself can influence the packet drop rate [V00, WBL04].

The default TFRC was designed for rough fairness with TCP, for TFRC and TCP flows with the same packet size and experiencing the same packet drop rate. When the issue of fairness between flows with different packets sizes is addressed, it matters whether the

packet drop rates experienced by the flows is related to the packet size. That is, are small packets just as likely to be dropped as large TCP packets, or are the smaller packets less likely to be dropped [WBL04]? And what is the relationship between the packet-dropping behavior of the path, and the loss event measurements of TFRC?

	< - - - - - Send Rates in Kbps - - - - - >			
Web	TCP (1460B seg)		TFRC-SP (200B seg)	
Sessions	DropRate	SendRate	DropRate	SendRate
10	0.04	316.18	0.05	183.05
25	0.07	227.47	0.07	181.23
50	0.08	181.10	0.08	178.32
100	0.14	85.97	0.12	151.42
200	0.17	61.20	0.14	73.88
400	0.20	27.79	0.18	36.81
800	0.29	3.50	0.27	16.33
1600	0.37	0.63	0.33	6.29

Table 8: Drop and Send Rates for Drop-Tail Queues in Packets

Table 8 shows the results of the second half of 100-second simulations, with five TCP connections and five TFRC-SP connections competing with web traffic in a topology with a 3 Mbps shared link. The TFRC-SP application generates 200-byte data packets every 10 ms, for a maximum data rate of 160 Kbps. The five long-lived TCP connections use a data packet size of 1460 bytes, and the web traffic uses a data packet size of 512 bytes. The five TCP connections have roundtrip times from 40 to 240 ms, and the five TFRC connections have the same set of round-trip times. The SACK TCP connections in these simulations use the default parameters in the NS simulator, with Limited Transmit, and a minimum RTO of 200 ms. Adding timestamps to the TCP connection didn't change the results appreciably. The simulations include reverse-path traffic, to add some small control packets to the forward path, and some queueing delay to the reverse path. The number of web sessions is varied to create different levels of congestion. The Drop-Tail queue is in units of packets, which each packet arriving to the queue requires a single buffer, regardless of the packet size.

Table 8 shows the average TCP and TFRC sending rates, each averaged over the five flows. As expected, the TFRC-SP flows see similar packet drop rates as the TCP flows, though the TFRC-SP flows receives higher throughput than the TCP flows with packet drop rates of 25% or higher.

	< - - - - - - Send Rates in Kbps - - - - - >			
Web	TCP (1460B seg)		TFRC-SP (200B seg)	
Sessions	DropRate	SendRate	DropRate	SendRate
10	0.06	239.81	0.00	185.19
25	0.09	189.02	0.01	184.95
50	0.14	99.46	0.01	185.07
100	0.20	16.42	0.02	183.77
200	0.26	4.46	0.03	181.98
400	0.29	4.61	0.05	151.88
800	0.49	1.01	0.08	113.10
1600	0.65	0.67	0.12	65.17

Table 9: Drop and Send Rates for Drop-Tail Queues in Bytes I: 1460B TCP Segments

However, the fairness results can change significantly if the Drop-Tail queue at the congested output link is in units of bytes rather than packets. For a queue in packets, the queue has a fixed number of buffers, and each buffer can hold exactly one packet, regardless of its size in bytes. For a queue in bytes, the queue has a fixed number of *bytes*, and an almost-full queue might have room for a small packet but not for a large one. This, for a simulation with a Drop-Tail queue in bytes, large packets are more likely to be dropped than are small ones. The NS simulator doesn't yet have a more realistic intermediate model, where the queue has a fixed number of buffers, each buffer has a fixed number of bytes, and each packet would require one or more free buffers. In this model, a small packet would use one buffer, while a larger packet would require several buffers.

As Table 9 shows, with a Drop-Tail queue in bytes, the TFRC-SP flow sees a much smaller packet drop rate than the TCP flow, and as a consequence receives a much larger sending rate. For the simulations in Table 9, the TFRC-SP flows use 200-byte data segments, while the long-lived TCP flows use 1460-byte data segments. For example, when the five TCP flows and five TFRC-SP flows share the link with 800 web sessions, the five TCP flows see an average drop rate of 49% in the second half of the simulation, while the five TFRC-SP flows receive an average drop rate of 8%, and as a consequence receive more than 100 times the throughput of the TCP flows. This raises serious questions about making the assumption that flows with small packets see the same packet drop rate as flows with larger packets. Further work will have to include an investigation into the range of realistic Internet scenarios, in terms of whether large packets are considerably more likely to be dropped than are small ones.

	< - - - - - - Send Rates in Kbps - - - - - >			
Web	TCP (512B seg)		TFRC-SP (200B seg)	
Sessions	DropRate	SendRate	DropRate	SendRate
10	0.02	335.05	0.00	185.16
25	0.02	289.94	0.00	185.36
50	0.04	139.99	0.01	184.98
100	0.06	53.50	0.01	184.66
200	0.10	16.14	0.04	167.87
400	0.16	6.36	0.07	114.85
800	0.24	0.90	0.11	67.23
1600	0.42	0.35	0.18	39.32

Table 10: Drop and Send Rates for Drop-Tail Queues in Bytes II: 512B TCP Segments

Table 10 shows that in the scenario the long-lived TCP flows receive a higher packet drop rate than the TFRC-SP flows, and receive considerably less throughput, even when the long-lived TCP flows use 512-byte segments.

7.4. Packet Dropping Behavior at Routers with AQM

As expected, the packet dropping behavior also can be varied by varying the Active Queue Management (AQM) mechanism in the router. When the routers use RED (Random Early Detection), there are several parameters than can affect the packet dropping or marking behavior as a function of the packet size.

First, as with Drop-Tail, the RED queue can be either in units of packets or of bytes. This can affect the packet dropping behavior when RED is unable to control the average queue size, and the queue overflows.

Second, and orthogonally, RED can be configured to be either in packet mode or in byte mode. In packet mode, each *packet* has the same probability of being dropped by RED, while in byte mode, each *byte* has the same probability of being dropped. In packet mode, large-packet and small-packet flows receive roughly the same packet drop rate, while in byte mode, large-packet and small-packet flows with the same throughput in bps receive roughly the same *number* of packet drops. The simulations reported in the appendix show that for RED in packet mode, the packet drop rates for the TFRC-SP flows are similar to those for the TCP flows, with a resulting acceptable throughput for the TFRC-SP flows. This is true with the queue in packets or in bytes, and with or without Adaptive RED (discussed below). As we show below, this fairness between TCP and TFRC-SP flows does not hold for RED in byte mode.

The third RED parameter that affects the packet dropping or marking behavior as a function of packet size is whether the RED mechanism is using Standard RED or Adaptive RED; Adaptive RED tries to maintain the same average queue size, regardless of the packet drop rate. The use of Adaptive RED allows the RED mechanism to function more effectively in the presence of high packet drop rates (e.g., greater than 10%). Without Adaptive RED, there is a fixed dropping threshold, and all arriving packets are dropped when the dropping or marking rate exceeds this threshold. In contrast, with Adaptive RED, the dropping function is adapted to accommodate high-drop-rate regimes. One consequence is that when byte mode is used with Adaptive RED, the byte mode extends even to high-drop-rate regimes. When byte mode is used with standard RED, however, the

byte mode is no longer in use when the drop rate exceeds the fixed dropping threshold (set by default to 10% in the NS simulator).

In the simulations in this section, we explore the TFRC-SP behavior over some of this range of scenarios. In this simulations, as in Section 7.3 above, the application for the TFRC-SP flow uses 200-byte data packets, generating 100 packets per second.

	< - - - - - Send Rates in Kbps - - - - >			
Web	TCP (1460B seg)		TFRC-SP (200B seg)	
Sessions	DropRate	SendRate	DropRate	SendRate
10	0.05	305.76	0.04	182.82
25	0.06	224.16	0.06	175.91
50	0.09	159.12	0.08	152.51
100	0.13	90.77	0.11	106.13
200	0.14	48.53	0.14	70.25
400	0.20	22.08	0.19	41.50
800	0.27	3.55	0.25	17.50
1600	0.42	1.87	0.34	8.81

Table 11: Drop and Send Rates for RED Queues in Packet Mode

For the simulations in Table 11, with a congested router with a RED queue in packet mode, the results are similar to those with a Drop-Tail queue in packets, as in Table 8 above. The TFRC-SP flow receives similar packet drop rates as the TCP flow, though it receives higher throughput in the more congested environments. The simulations are similar with a RED queue in packet mode with the queue in bytes, and with or without Adaptive RED. In these simulations, TFRC-SP gives roughly the desired performance.

	< - - - - - Send Rates in Kbps - - - - >			
Web	TCP (1460B seg)		TFRC-SP (200B seg)	
Sessions	DropRate	SendRate	DropRate	SendRate
10	0.06	272.16	0.02	184.37
25	0.07	175.82	0.02	184.06
50	0.10	75.65	0.04	180.56
100	0.14	38.98	0.07	151.65
200	0.19	16.66	0.11	106.80
400	0.26	4.85	0.15	69.41
800	0.35	3.12	0.20	27.07
1600	0.42	0.67	0.29	10.68

Table 12: Drop and Send Rates for RED Queues in Byte Mode

Table 12 shows that with a standard RED queue in byte mode instead of packet mode, there is a somewhat greater different between the packet drop rates between the TCP and TFRC-SP flows, particularly for lower packet drop rates. For the simulation in Table 12, the packet drop rates for the TCP flows can range from 1 1/2 to four times greater than the packet drop rates for the TFRC-SP flows. However, because the TFRC-SP flow has an upper bound on the sending rate, its sending rate is not affected in the lower packet-drop-rate regimes; its sending rate is only affected in the regimes with packet drop rates of 10%

or more. The sending rate for TFRC-SP in the scenarios in Table 12 with higher packet drop rates are greater than desired, e.g., for the scenarios with 400 web sessions or greater. However, the results with TFRC-SP are at least better than that of small-packet flows with no congestion control at all.

	< - - - - - - Send Rates in Kbps - - - - - >			
Web	TCP (512B seg)		TFRC-SP (200B seg)	
Sessions	DropRate	SendRate	DropRate	SendRate
10	0.01	337.86	0.01	184.06
25	0.02	258.71	0.01	184.03
50	0.02	184.71	0.01	183.99
100	0.04	63.63	0.03	184.43
200	0.08	28.95	0.06	149.80
400	0.12	17.03	0.10	88.21
800	0.24	5.94	0.15	36.80
1600	0.32	3.37	0.21	19.45

Table 13: Drop and Send Rates for RED Queues in Byte Mode

Table 13 shows that with a standard RED queue in byte mode and with long-lived TCP flows with 512-byte data segments, there is only a moderate difference between the packet drop rate for the 552-byte TCP packets and the 240-byte TFRC-SP packets. However, the sending rate for TFRC-SP in the scenarios in Table 13 with higher packet drop rates are still greater than desired, even given the goal of fairness with TCP flows with 1500-byte data segments instead of 512-byte data segments.

	< - - - - - - Send Rates in Kbps - - - - - >			
Web	TCP (1460B seg)		TFRC-SP (200B seg)	
Sessions	DropRate	SendRate	DropRate	SendRate
10	0.04	318.10	0.02	185.34
25	0.08	175.34	0.03	184.38
50	0.10	81.60	0.04	181.95
100	0.12	28.51	0.05	178.79
200	0.20	3.65	0.06	173.78
400	0.27	1.44	0.08	161.41
800	0.40	0.58	0.06	159.62
1600	0.55	0.29	0.02	180.92

Table 14: Drop and Send Rates with Adaptive RED Queues in Byte Mode

For the simulations in Table 14, the congested router uses an Adaptive RED queue in byte mode.

For this router, the output queue is in units of bytes rather than of packets, each *byte* is dropped with the same probability, and because of the use of Adaptive RED, this byte-dropping mode extends even for the high-packet-drop-rate regime.

As Table 14 shows, for a scenario with Adaptive RED in byte mode, the packet drop rate for the TFRC-SP traffic is *much* lower than that for the TCP traffic, and as a

consequence, the sending rate for the TFRC-SP traffic in a highly congested environment is *much* higher than that of the TCP traffic. In fact, in these scenarios the TFRC-SP congestion control mechanisms are largely ineffective for the small-packet traffic.

We note that the unfairness in these simulations, in favor of TFRC-SP, is even greater than the unfairness shown in Table 9 for a Drop-Tail queue in bytes. At the same time, it is not known if there is any deployment in the Internet of any routers with Adaptive RED in byte mode, or of any AQM mechanisms with similar behavior; we don't know the extent of the deployment of standard RED, or or any of the proposed AQM mechanisms.

	< - - - - - Send Rates in Kbps - - - - - >			
Web	TCP (512B seg)		TFRC-SP (200B seg)	
Sessions	DropRate	SendRate	DropRate	SendRate
10	0.01	306.56	0.01	185.11
25	0.02	261.41	0.01	184.41
50	0.02	185.07	0.01	184.54
100	0.04	59.25	0.03	181.58
200	0.08	16.32	0.06	150.87
400	0.12	11.53	0.10	98.10
800	0.25	5.85	0.15	46.59
1600	0.32	1.43	0.22	19.40

Table 15: Drop and Send Rates for Adaptive RED Queues in Byte Mode

Table 15 shows that when TFRC-SP with 240-byte packets competes with TCP with 552-byte packets in a scenario with Adaptive RED in byte mode, the TFRC-SP flows still receive more bandwidth that the TCP flows, but the level of unfairness is less severe, and the packet drop rates of the TCP flows is at most twice that of the TFRC-SP flows. That is, the TFRC-SP flows still receive more than their share of the bandwidth, but the TFRC-SP congestion control is more effective that than in Table 14 above.

8. General Discussion

Dropping rates for small packets: The goal of TFRC-SP is for small-packet TFRC-SP flows to have rough fairness with large-packet TCP flows in the sending rate in bps, in a scenario where each packet receives roughly the same probability of being dropped. In a scenario where large packets are more likely to be dropped than small packets, or where flows with a bursty sending rate are more likely to have packets dropped than are flows with a smooth sending rate, small-packet TFRC-SP flows can receive significantly more bandwidth than competing large-packet TCP flows.

The accuracy of the TCP response function used in TFRC: For applications with a maximum sending rate of 96 Kbps or less (i.e., packets of at most 120 bytes) TFRC-SP only restricts the sending rate when the packet drop rate is fairly high, e.g., greater than 10%. [Derivation: A TFRC-SP flow with a 200 ms round-trip time and a maximum sending rate with packet headers of 128 Kbps would have a sending rate in bytes per second equivalent to a TCP flow with 1460-byte packets sending 2.2 packets per round-trip time. From Table 1 of RFC 3714, this sending rate can be sustained with a packet drop rate slightly greater than 10%.] In this high-packet-drop regime, the performance of TFRC-SP is determined in part by the accuracy of the TCP response function in representing the actual sending rate of a TCP connection.

In this regime of high packet drop rates, TCP performance is also affected by the TCP algorithm (e.g., SACK or not), by the minimum RTO, by the use or not of Limited Transmit, by the use of timestamps and/or of ECN, and the like. It is good to insure that simulations or experiments exploring fairness include the exploration of fairness with the most aggressive TCP mechanisms conformance with the current standards. Our simulations use SACK TCP with Limited Transmit and with a minimum RTO of 200 ms. Adding the use of timestamps has not made a big difference. We haven't used TCP with ECN, because our judgment is that in high packet drop regimes, it is preferable for AQM mechanisms to drop rather than mark packets.

General issues for TFRC: The congestion control mechanisms in TFRC and TFRC-SP limit a flow's sending rate in packets per second. Simulations by Tom Phelan [P04] explore how such a limitation in sending rate can lead to unfairness for the TFRC flow in some scenarios, e.g., when competing with bursty TCP web traffic, in scenarios with low levels of statistical multiplexing at the congested link.

9. Security Considerations

There are no security considerations introduced in this document.

General security considerations for TFRC are discussed in RFC 3448. The security considerations for TFRC include the need to protect against spoofed feedback, and the need for protection mechanisms to protect the congestion control mechanisms against incorrect information from the receiver.

Security considerations for DCCP's Congestion Control ID 3, TFRC Congestion Control, are discussed in [CCID 3 PROFILE]. That document extensively discussed the mechanisms the sender can use to verify the information sent by the receiver.

10. IANA Considerations

There are no IANA considerations in this document.

11. Thanks

We thank Tom Phelan for discussions of TFRC-SP and for his paper exploring the fairness between TCP and TFRC-SP flows. We thank Joerg Widmer for feedback on earlier versions of this draft. We also thank the DCCP Working Group for feedback and discussions.

A. Appendix: Related Work on Small-Packet Variants of TFRC

Other proposals for variants of TFRC for applications with variable packet sizes include [WBL04] and [V00]. [V00] proposed that TFRC should be modified so that flows are not penalized by sending smaller packets. In particular, [V00] proposes using the path MTU in the TCP-friendly equation, instead of the actual packet size used by TFRC, and counting the packet drop rate by using an estimation algorithm that aggregates both packet drops and received packets into MTU-sized units.

[WBL04] also argues that adapting TFRC for variable packet sizes by just using the packet size of a reference TCP flow in the TFRC equation would not suffice in the high-packet-loss regime; such a modified TFRC would have a strong bias in favor of smaller packets, because multiple lost packets in a single round-trip time would be aggregated into a single packet loss. [WBL04] proposes modifying the loss measurement process to account for the bias in favor of smaller packets.

The TFRC-SP variant of TFRC proposed in our document differs from [WBL04] in restricting its attention to flows that send at most 100 packets per second. The TFRC-SP variant proposed in our document also differs from the straw proposal discussed in [WBL04] in that the allowed bandwidth includes the bandwidth used by packet headers.

[WBL04] discusses four methods for modifying the loss measurement process, "unbiasing", "virtual packets", "random sampling", and "Loss Insensitive Period (LIP) scaling". [WBL04] finds only the second and third methods sufficiently robust when the network drops packets independently of packet size. They find only the second method sufficiently robust when the network is more likely to drop large packets than small packets. Our method for calculating the loss event rate is somewhat similar to the random sampling method proposed in [WBL04], except that randomization is not used; instead of randomization, the exact packet loss rate is computed for short loss intervals, and the standard loss event rate calculation is used for longer loss intervals. [WBL04] includes simulations with a Bernoulli loss model, a Bernoulli loss model with a drop rate varying over time, and a Gilbert loss model, as well as more realistic simulations with a range of TCP and TFRC flows.

[WBL04] produces both a byte-mode and a packet-mode variant of the TFRC transport protocol, for connections over paths with per-byte and per-packet dropping respectively. We would argue that in the absence of transport-level mechanisms for determining whether the packet dropping in the network is per-packet, per-byte, or somewhere in between, a single TFRC implementation is needed, independently of the packet-dropping behaviors of the routers along the path. It would of course be preferable to have a mechanism that gives roughly acceptable behavior, to the connection and to the network as a whole, on paths with both per-byte and per-packet dropping (and on paths with multiple congested routers, some with per-byte dropping mechanisms, some with per-packet dropping mechanisms, and some with dropping mechanisms that lie somewhere between per-byte and per-packet).

A first step will be to investigate the range of behaviors actually present in today's networks, in terms of packet-dropping as a function of packet size. We will report on these investigations in a separate document.

B. Appendix: A Discussion of Packet Size and Packet Dropping

The table below gives a general summary of the relative advantages of packet-dropping behavior at routers independent of packet size, versus packet dropping behavior where large packets are more likely to be dropped than small ones.

```
Advantages of Packet Dropping Independent of Packet Size:
---------------------------------------------------------
1.  Adds another incentive for end nodes to use large packets.

2.  Matches an environment with a limitation in pps rather than
    bps.
---------------------------------------------------------

Advantages of Packet Dropping as a Function of Packet Size:
---------------------------------------------------------
1.  Small control packets are less likely to be dropped than are
    large data packets, improving TCP performance.

2.  Matches an environment with a limitation in bps rather than
    pps.

3.  Reduces the penalty of TCP and other transport protocols
    against flows with small packets (where the allowed sending
    rate is roughly a linear function of packet size).

4.  A queue limited in bytes is better than a queue limited in
    packets for matching the worst-case queue size to the
    worst-case queueing delay in seconds.
---------------------------------------------------------
```

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