A rate controller for long-lived TCP flows*

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Abstract. In this paper a new mechanism for providing an assured rate to a long-lived TCP flow is proposed. The mechanism is called TCP rate controller (TRC) and operates as a traffic conditioner at the edge of a network. The TRC seeks to achieve the requested rate by imposing well directed drops and (artificial) delays on the flow's packets. The choice of drop probability and delay is based on an analytical model of TCP sending behavior. It is shown in a simulation study that the TRC performs well over a broad range of requested rates and network RTTs.
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1 Introduction

In the recent years there has been a growing interest in IP Quality of Service (QoS). New applications that have high requirements on network performance are being developed. Some of those applications, for example video conferencing or IP telephony, require some minimum network quality to be useful at all. From the operator's point of view it is hoped that services that deliver high QoS can be profitable.

Currently, a lot of research work is based on the paradigm of Differentiated Services (DiffServ) [1, 2]. DiffServ seeks to provide QoS in IP networks in a simple and scalable fashion. It is tried to remove complex tasks from the core and shift them to the edges of the network instead. As an example, traffic controllers that operate on single flows are only acceptable at the ingress/egress of a DiffServ domain.

The focus of this paper is on the topic of rate assurance for TCP flows. Given that TCP is the number one transport protocol [3] in today's Internet, we believe that such a service could be of interest. The topic of assuring TCP rates has been investigated in several other publications, e.g. [4–6].

The goal of this work is to develop a traffic conditioning mechanism that can be used to assure a certain level of goodput to long-lived TCP flows. The proposed conditioner could be used in a DiffServ network to enable such a service class. We propose a TCP rate controller (TRC) that regulates the goodput of a TCP flow by controlling packet drops and the round trip time (RTT) of the flow's packets. The TRC is based on a model of TCP sending behavior.

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The remainder of this paper is organized as follows: Section 2 discusses related work. Section 3 summarizes the essential network environment for the TRC. The TRC is described in Section 4. Exemplary simulation results for a variety of network parameter settings are presented in Section 5. Section 6 concludes the paper.

2 Related Work

The Capped Leaky Bucket (CLB) as proposed in [6] is an improved Leaky Bucket traffic conditioner. To take the behavior of TCP into account it is tried to estimate the RTT by measuring the time between two bursts. If the input rate is higher than the target rate one packet each two RTTs is marked as out-profile. Simulations in [6] show performance that is not appropriate to give assurances and a bias against big reservations.

In [4] equations how to set the parameters of a token bucket marker for achieving a requested rate are proposed. The parameter setting depends on the requested rate (R_{req}), drop probability of out-profile packets (p_2) and the RTT. With the equations in [4] it is possible to make correct goodput assumptions for a known value of RTT and p_2. But the crucial aspect in the application of this model is that the RTT and p_2 are not constant for different connections and also strongly vary over time due to changes in the level of congestion.

We therefore believe that it is very difficult, if not impossible, to assure TCP rates by employing a token bucket marker that is configured with a *static* parameter set. A very interesting adaptive marking algorithm has been proposed recently in [5].

3 Network Environment

The TCP rate controller essentially requires some conditions from the network. The first aspect is that at the network edge the TRC must exclusively control single long-lived TCP flows that are not multiplexed with a different kind of traffic (e.g. short-lived TCP flows or UDP flows) [7]. Second, some kind of admission control framework must ensure that the sum of requested rates over all accepted reservations is in fact available. This condition provides an over-provisioned network. Further the used code-point can be different in different domains. In the core netowkr the traffic can be multiplexed with any kind of traffic, but it has to be ensured that drop probability (p) is zero and the network RTT (RTT_{net}) is known and nearly constant.

The TRC does not need any special queue management mechanism, because no packet should be dropped in the network. It has to be established that a few packets can be buffered in the queue.

The receiver window has to be larger than the maximum congestion window (W_{max}) otherwise the achieved rate will be controlled by the receiver and not by the TRC. Further also TCP's Slow Start threshold ($ssthresh$) should be larger than W_{max} otherwise the performance during Slow Start will be worse.

The TRC has to be placed at the ingress point of the network. Rate controlling has to be done on a per-flow basis, therefore every flow has to be extracted from an aggregate. Only one TRC can be applied to one flow, because if two TRCs are working on the same flow two times the packets needed to control the flow are dropped.

It has to be ensured that from the receiver to the sender there is no congestion, because RTT_{net} is assumed to be small and constant. Consequently also ACKs have to be marked with the same code-point than packets from the sender to the receiver.

4 TCP rate controller

4.1 Goal of the TRC

The goal of the TRC is it to provide the TCP sender with a goodput that was requested (by some means of QoS request) in a prior step. The TRC tries to achieve that goal by imposing well directed drops and delays on the flow's packets. The choice of drop probability *p* and (artificial) delay RTT_{TRC} is based on an analytical model of TCP sending behavior. We performed our study using the well-known `ns-2` simulator, version 2.1b6, and realized that existing TCP models [8, 9] do not accurately predict the sending behavior of the TCP SACK implementation we used. In order to exclude errors in the TCP model and to focus on the feasibility of the TRC approach itself, we derived our own model which describes the TCP sending behavior by equations 1-4.

$$time\ per\ cycle = \frac{5}{2} + \frac{W_{max}}{2} \tag{1}$$

$$data\ per\ cycle = \frac{1}{p} = \frac{3}{8} * W_{max}{}^2 + \frac{5}{4} * W_{max} - 2 \tag{2}$$

$$W_{avg} = \frac{data\ per\ cycle}{time\ per\ cycle} == \frac{\frac{3}{8} * W_{max}{}^2 + \frac{5}{4} * W_{max} - 2}{\frac{5}{2} + \frac{W_{max}}{2}} \tag{3}$$

$$BW = \frac{W_{avg} * MSS}{RTT} \tag{4}$$

Thus the sending rate (BW) of the TCP sender depends on the average congestion window (W_{avg}) multiplied with the maximum segment size (MSS) divided by the RTT.

The TCP model does not need to take into account timeouts, because losses are exclusively controlled by the TRC and do not force any timeout. From Equation 2 and 3 it is obvious that W_{max} and thus W_{avg} only depend on p and that RTT does not influence W_{max}.

4.2 Principal idea of the TCP rate controller

Now, the basic idea of the TRC is that by controlling the amount of dropped packets and the RTT, the rate of the TCP flow can be pruned to the requested rate. The simple algorithm that has to be executed upon each packet arrival is shown in pseudo code in Figure 1.

The TRC drops packets at the network ingress and thereby enforces the rate of the TCP flow to oscillate around the requested level. Consequently, assuming proper functioning of the resource control framework, TRC-controlled flows experience no

```
for each packet arrival:
    if (packets since drop >= 1/p + E)
        drop packet
    else
        delay packet for RTT_TRC
```

Fig. 1. Pseudo-code for each packet arrival

sustained congestion but merely small and transient queuing delays inside the network. Therefore, the RTT is mainly comprised by RTT_{net} and can thus be well estimated.

Besides dropping, the TRC can add an (artificial) delay RTT_{TRC} in order to control the achieved rate of the TCP flow. The total RTT can then be approximated as the sum of RTT_{net} and RTT_{TRC}.

Consequently for a known value of RTT_{net} the TRC exclusively controls p and RTT. Based on the underlying TCP model the TRC is thus able to make correct assumptions of the achieved rate of a TCP connection.

For a given requested rate there exist several combinations of p and RTT which achieve the same rate. The tradeoffs in the choice of the two parameters are discussed in Section 4.3.

The term E in Figure 1 is used to compensate the drop in the last cycle and has a value of 1. The TRC can however be equally operated in ECN mode which means that packets are not dropped but marked instead. In that case, the term E has a value of 0.

4.3 Tradeoffs in Parameter setting for each request

It has to be ensured that after a drop there are enough packets in the network to receive three duplicate acknowledgements and trigger further packet transmissions during fast retransmit. Therefore the maximum window W_{max} has to be at least 5 packets (1 loss, 3 duplicate ACK, 1 to trigger further transmissions).

Clearly, the accuracy of the TRC is mainly influenced by the ability of the TCP model to accurately predict the flow's sending behavior. The deviation between the model's prediction and the real sending rate increases with the drop probability. A W_{max} of 5 corresponds to a drop probability of 0.0735; we have seen that any p greater than this value results in unacceptably large deviations from the model. Even for some values smaller than 0.0735 (and thus W_{max} larger than 5) we noticed significant deviations in the `ns-2` simulations. We tried to find values for p such that the corresponding W_{max} achieves at least the rate that's estimated by the model. We found that if W_{max} is set as an even number plus 0.5 this condition is fulfilled and that the achieved rate is at most 3% higher than estimated.

The second parameter which can be tuned is the RTT_{TRC} and consequently the RTT. As explained above there exist a lot of combinations of setting p and RTT to achieve a requested rate. One choice would be to fix p so that W_{max} is 6.5 and enable different requested rates by imposing different packet delays inside the TRC. This would mean that the greater the requested rate the smaller the RTT would be. For big requests this could lead to the problem that i) small deviations in RTT_{net} have a significant influence on the achieved goodput see Section 5.4 for details or ii) that RTT_{net} is even greater than the total RTT should be. To avoid this, a lower bound for the RTT, called RTT_{min}, has to be fixed. If the required RTT (RTT_{req}) is smaller than RTT_{min} the next greater value of W_{max}, i.e. $W_{max}+2$ as discussed above, has to be used.

To impose a delay of RTT_{TRC}, packets have to be stored in a buffer. The greater the delay for the same rate the greater must the buffer be. Thus on the one hand the delay should be kept high to keep the influence of a deviation the RTT small; on the other hand the delay should be kept small to keep buffer requirements low. This is a tradeoff that an operator must take into account when choosing RTT and p for a requested rate. It will be further discussed in the next section. Figure 2 shows the pseudo code of the algorithm that computes p and RTT_{TRC} upon each request.

```
for each request:
    set RTT_min min(RTT*_min, RTT_net)
    calculate RTT_req for W_max = 6.5
    if RTT_req > RTT_min
        set drop probability to achieve W_max of 6.5
        set RTT_TRC to (RTT - RTT_net)
    else
        set RTT to RTT_min
        calculate an appropriate W_max
        recalculate RTT based on appropriate W_max
        set drop probability to achieve appropriate W_max
        set RTT_TRC to (RTT - RTT_net)
```

Fig. 2. Pseudo-code for parameter computation

4.4 Tradeoffs in network parameter configuration

An operator has to provide two parameters for the initial configuration of the TRC. One is called RTT_{dev} and denotes an operator's estimate on the maximum deviation between RTT_{net} and the real RTT. The second one is called $gput_{error}$ which is the error in the achieved goodput that should be compensated by the TRC. In order to be on the safe side, the rate that is requested by the user is increased by $gput_{error}+1$ percent.

The smaller RTT_{dev}, the smaller is RTT_{TRC} and thus the smaller the buffer can be. On the other hand, if RTT_{dev} is high, this requires a large buffer space due to the high RTT_{TRC}.

Due to the bursty sending behavior of TCP sources a few packets will be queued at the bottleneck leading to a slight variance in RTT. This occurs especially in high load scenarios. Consequently, RTT_{dev} can generally not be zero.

Equation 5 can be used for determining RTT^*_{min}. As an example assume that a 5% error in the achieved rate is taken into account for TRC configuration and the error in the RTT estimation is not greater than 10ms. In that case RTT^*_{min} would be 200ms. Throughout the rest of the paper this value is used for RTT^*_{min}.

$$RTT^*_{min} = \frac{RTT_{dev}}{gput_{error}} \tag{5}$$

Thus lower bounds for drop probability and for delay are fixed. The other aspect which has to be taken under consideration is the buffer size needed by each connection. The higher the requested rate and the higher the delay the higher the demand of buffer is. The buffer has to be able to store $ceil(W_{max})$ packets from each flow.

To keep RTT_{dev} small not the whole available bandwidth can be sold. We propose that the network is over-provisioned by some amount. In our case using `ns-2` it should be over-provisioned by at least 6%. Further the maximum error of the TCP model which is 3% has to be taken into account. To compensate a RTT_{dev} of 5% the achieved goodput is increased by 6%. The sum of requested rates has to be smaller than $\rho * BW$. Where ρ for the above case must not be greater than 0.85 (6% over-provisioned, 3% TCP model error and 6% compensate RTT_{dev}).

In the core network the queue has to be able to store at least one half of the maximum window of the largest possible request. Because during Slow Start packets at the queue are arriving in a burst with a rate that is twice the bandwidth of the bottleneck. Further a few packets have to be stored when a burst of packets from two or more connections arrives at the same time. It is proposed that the buffer size is 50 packets plus one half of the largest W_{max}. This is a topic of further research.

```
for each network:
    set RTT*_min to  RTT_dev/gput_error
    set rate_increase to gput_error + 1%
```

Fig. 3. Pseudo-code for initial configuration

Figure 3 shows the pseudo code of the initial parameterization of the TRC done by each operator. Figure 4 shows examples how to set p and RTT_{TRC} for R_{req} between 50kbps and 600kbps. RTT_{net} is assumed to be zero and RTT_{dev} is assumed to be 10ms. RTT_{min} is set to 200ms.

4.5 Control of Slow Start

Despite the TRC is designed for long-lived TCP flows the Slow Start phase can not be neglected in general. We try to control the Slow Start phase such that the achieved goodput is not significantly lower or higher than during congestion avoidance. At the beginning of Slow Start the window is small. This results in low goodput which has influence on the overall achieved goodput. If the connection stays too long in Slow Start the window will increase significantly above W_{max}. This will have the effect that there is more traffic in the network than estimated and packets of all other TRC-controlled connections might be dropped. Dropping packets inside the network would destroy the

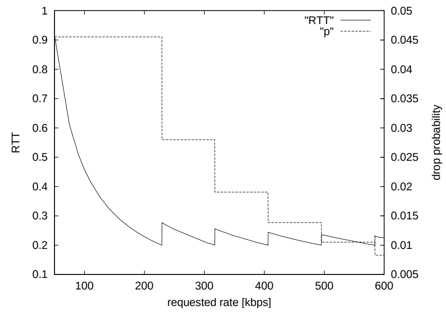


Fig. 4. Exemplary setting of p and RTT_{TRC} based on R_{req}

whole concept because drops would then be no longer exclusively controlled by the TRC.

Consequently an appropriate parameter setting for Slow Start would be to control packet losses so that the window does not exceed W_{max}. The RTT has to be controlled such that as long as the congestion window is smaller than $W_{max}/2$ the TCP sender sends with a constant rate which equals the requested rate. Analyzing the window behavior during Slow Start shows that when $floor(W_{max})$ packets are forwarded and then one packet is dropped the window will be $ceil(W_{max})$ when the loss is detected. Therefore the buffer for delaying packets has to be able to store $ceil(W_{max})$ packets.

If both algorithms work adequately then the Slow Start phase does not significantly influence a connection's performance. This theoretical approach has some restrictions: first, it is not possible to reduce the RTT below RTT_{net} and for small windows it may thus be impossible to achieve the requested rate. Nevertheless this approach of controlling the Slow Start phase is superior to an approach where the Slow Start phase is not particularly taken care of. Second, the influence of the retransmission timeout (RTO) during Slow Start has to be evaluated, because the RTT_{TRC} is increased over time. And thus during Slow Start phase a RTO may occur yielding to a retransmit and a congestion window of one segment. Consequently the rate during Slow Start will be smaller than estimated and thus may influence the overall goodput. This is left for further study. When Slow Start is taken into account the code executed for each packet arrival is slightly modified and can be found in Figure 5.

5 Simulation study

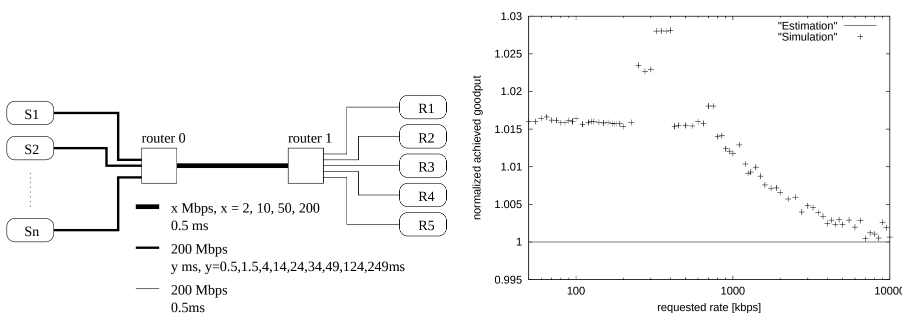
5.1 Simulation Topology

The behavior of the TCP rate controller is studied by means of network simulations using `ns-2` [10]. The simulations were run on a simple dump-bell topology shown in Figure 6(a). FTP senders start sending data from hosts *S1-Sn* at a random point of time within the first 10 seconds to hosts *R1-R5*. The bottleneck bandwidth, access delay and requested rate of the TCP senders are varied for the different scenarios to evaluate different effects. Both routers are using a DropTail queue. Traffic conditioning is done by

```
init_slowstart=1;
for each packet arrival
    if init_slowstart //connection is in early Slow Start phase
        if packets since drop >= W_max
            drop packet
            init_slowstart=0
        else
            queue packet to achieve requested rate
    else
        code like during congestion avoidance
```

Fig. 5. Pseudo-code of TRC with Slow Start

the TRC on a per-flow basis at each sending host. Unless noted otherwise, simulations last for 1100 seconds where the first 100 seconds are not taken into consideration for the statistics. The purpose of the simulation study is it to investigate over a broad range of scenarios to what extent the TRC is able to provide TCP rate assurances. Especially the effect of a mixture of different RTTs and requested rates under maximum load is analyzed. All simulations are run with and without ECN showing equal results.



(a) Simulation topology

(b) Simulation results for exact parameter setting

Fig. 6. Simulation topology and first results

5.2 Performance evaluation for $RTT_{dev} = 0$

In this section we demonstrate that for an exact knowledge of RTT_{net} the TRC is able to give goodput assurances for a wide range of requested rates. Simulations are run on a 200Mbps bottleneck link so that queuing delay can be neglected. All links have a delay of 0.5ms consequently RTT_{net} is 3ms. The requested rates vary between 50 – 10^4 kbps.

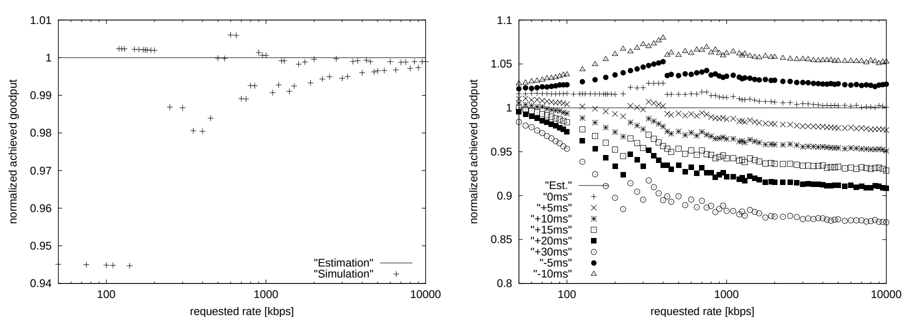
Figure 6(b) shows the normalized achieved goodput over the requested rate. Each flow achieves at least the requested rate. The maximum deviation between simulation and estimation is about 3%.

5.3 Slow Start behavior

This section provides simulation results for evaluation of the mechanism introduced to control the Slow Start phase. Simulations are run on the topology shown in Figure 6(a) where all links have a capacity of 200Mbps and a delay of 0.5ms.

Simulations are run until the second drop is detected by the sending host. This is chosen for two main reasons. On the one hand the transition from control of Slow Start to Congestion Avoidance should be shown. On the other hand if simulations were stopped after Slow Start only a few packets were transmitted. There are three packets (triggering 3 dupACKs) that have already arrived at the receiver but were not taken into account for statistics, because they are not acknowledged. These packets have great influence on the achieved goodput. The more packets are transmitted the smaller the influence will be.

Figure 7(a) shows normalized achieved goodput for several requested rates. A few earlier discussed effects can be seen in this graphs. The marks in the lower left corner are flows that had a RTO during Slow Start because increasing the RTT_{TRC} during Slow Start does not take into account the value of the RTO. The marks above the estimated goodput come from the effect that for moderate loss rates the achieved goodput is underestimated by about 3%. Simulation results providing values between 0.98 and 1 show the influence of mainly two aspects. The one is that the three packets arrived at the receiver but not yet acknowledged ones, have influence on the achieved goodput. The second aspect is that for a large W in the early phase of Slow Start with small congestion window it is not possible to reduce the RTT so far that the requested rate is achieved. If RTT_{net} increases the achieved rate during Slow Start will decrease.



(a) Simulation Results during Slow Start

(b) Simulation results for deviations in RTT

Fig. 7. Simulation results

5.4 Simulations with RTT_{dev} greater than 0

Simulation results in this section should show the influence of a deviation in the RTT on the achieved goodput. Simulations were run on a 200Mbps bottleneck link consequently the queuing delay can be neglected. The access delay is varied to achieve different deviations. The TRC is configured with an RTT_{min} of 200ms. The parameters of the TRC are not adapted to compensate the RTT_{dev}.

Figure 7(b) shows the normalized achieved goodput over different requested rates for a few RTT_{dev}. RTT_{min} has direct influence on the goodput error for big requests. For an error of 10ms and a RTT_{min} of 200ms the deviation for big requests is 5%. For small requests the influence is smaller, because small requests have a much higher RTT than RTT_{min}. The maximum deviation between simulation results with no RTT error and with RTT error can be approximated by RTT_{dev}/RTT_{min}.

5.5 Simulations under maximum load

This section provides simulation results for maximum load scenarios. The network parameters were set according to Section 4.3 which means that ρ is set to 0.85. The bottleneck bandwidth is set to 2Mbps, 10Mbps and 50Mbps respectively. The access delay is varied to achieve the desired network behavior. Simulations are run 100 times.

The first simulation study should analyze if the TRC has some bias against R_{req}. For evaluating the influence of requested rates the RTT is homogeneous and varied for different scenarios. Simulations are run for an access delay of 0.5, 1.5, 4, 24, 49, 124, 249ms. The requested rates are varied from 50kbps up to 10Mbps within one scenario.

Table 1 provides simulation results for several selected scenarios. Each line shows results from one scenario. It can be seen that there is no bias against big requests. All flows achieve the requested rate.

The rate of a TCP sender is in general heavily influenced by the RTT. Flows with lower RTT are more aggressive. For evaluating the influence of different RTTs the access delays for link S1 router0 is set to 0.5ms up to 249ms for link S7 router0.

Table 2 shows simulation results for different RTTs. It can be seen that the TCP flows achieve nearly the same rate. Thus, the achieved rate of a TRC policed TCP flow has no bias against any RTT and the TRC is able to provide goodput assurances for a wide range of RTTs.

Further simulations are run to evaluate the influence of mixed RTTs and mixed R_{req}. Therefore the bottleneck bandwidth is set to 50Mbps. Table 3 shows simulation results where the parameters are set so that the higher the target rate the higher the RTT and Table 4 for a parameter setting where the higher the target rate the lower the RTT. Each combination of requested rate and RTT achieves the requested rate.

6 Conclusion

In this paper a TCP rate controller (TRC) is proposed. The TRC seeks to achieve goodput assurances for long-lived TCP flows.

Table 1. Simulation results for different requested rates

BW [Mbps]	delay [ms]	50 [kbps]	100 [kbps]	250 [kbps]	500 [kbps]	750 [kbps]	1000 [kbps]	2500 [kbps]	5000 [kbps]	10000 [kbps]
2	1.5	53.229 53.222-53.237	105.64 105.633-105.647	X X	517.38 517.348-517.412	X X	1024.87 1024.81-1024.93	X X	X X	X X
10	1.5	53.518 53.516-53.519	106.353 106.345-106.361	265.035 265.019-265.051	526.374 526.343-526.405	786.818 786.756-786.88	1043.61 1043.53-1043.69	X X	5156.98 5156.4-5157.56	X X
10	124	53.475 53.472-53.478	106.238 106.23-106.246	266.73 266.716-266.744	525.35 525.316-525.384	787.662 787.6-787.724	1047.22 1047.14-1047.3	X X	5179.03 5178.28-5179.78	X X
50	1.5	53.746 53.743-53.75	107.234 107.217-107.251	268.824 268.753-268.895	533.844 533.690-533.998	799.138 798.9-799.376	1063.5 1063.18-1063.82	2638.91 2638.08-2639.74	5261.22 5259.42-5263.02	10495.9 1049.24-10499.4
50	124	53.718 53.713-53723	108.055 108.037-108.073	270.198 270.133-270.263	532.422 532.195-532.649	798.072 797.839-798.305	1061.36 1061.07-1061.65	2639 2638.23-2639.77	5263.32 5261.77-5264.87	10504.5 10500.8-10508.2

Table 2. Simulation results for different RTTs

BW [Mbps]	R_req [kbps]	0.5ms [kbps]	1.5ms [kbps]	4ms [kbps]	24ms [kbps]	49ms [kbps]	124ms [kbps]	249ms [kbps]
2	100	105.231 105.179-105.283	105.225 105.172-105.278	105.225 105.172-105.278	105.225 105.172-105.278	105.222 105.169-105275	105.217 105.165-105.269	106.601 106.566-106.636
10	100	107.263 107.176-107.350	107.256 107.170-107.342	107.257 107.171-107.343	107.256 107-170-107.342	107.254 107.168-107.340	107.249 107.163-107.335	108.101 107.998-108.204
10	1000	1048.22 1047.7-1048.74	1048.22 1047.7-1048.74	1048.21 1047.69-1048.73	1048.19 1047.68-1048.7	1048.21 1047.7-1048.72	1049.13 1048.52-1049.74	1052.94 1052.06-1053.82
50	100	105.55 99.901-111.198	105.546 99.898-111-194	105.547 99.899-111.195	105.546 99.898-111.194	105.546 99.898-111.194	105.538 99.890-111.186	106.301 100.612-111.190

Table 3. Simulation results (The higher the requested rate, the higher the RTT)

scenario	50k 3ms	100k 5ms	250k 10ms	500k 30ms	750k 50ms	1M 70ms	2.5M 100ms	5M 250ms	10M 500ms
achieved rate [kbps]	53.719 53.712-53.725	107.115 107.086-107.144	268.21 268.078-268.342	532.809 532.531-533.087	797.508 797.078-797.938	1061.19 1060.62-1061.76	2631.05 2629.49-2632.61	5256.47 5252.28-5260.66	10337.9 10270.3-10405.5

Table 4. Simulation results (The higher the requested rate, the lower the RTT)

scenario	50k 500ms	100k 250ms	250k 100ms	500k 70ms	750k 50ms	1M 30ms	2.5M 10ms	5M 5ms	10M 3ms
achieved rate [kbps]	53.715 53.711-53.720	107.22 107.2-107-24	268.794 268.705-268.883	533.786 533.592-533.98	799.047 798.751-799.343	1063.37 1062.97-1063.77	2638.58 2637.53-2639.63	5260.43 5258.21-5262.65	10494.9 10491.2-10498.6

The task of the TRC is it to control the achieved goodput of a TCP connection by controlling a connections RTT and p. The TRC is based on a TCP model which predicts the sending behavior for known values of RTT and p. The idea of the TRC is it to fix RTT and p of a connection. Therefore the TRC drops packets to control the window size of the TCP connection and delays packets to increase the RTT. Consequently the achieved rate of the connection is controlled. The TRC is a traffic conditioner which has to be placed at the ingress point of the network. A TCP model for `ns-2` TCP SACK implementation was derived. Based on this model the TRC is constructed.

The TRC is evaluated by simulations using `ns-2`. It is shown that the TRC has no bias against any requested rate or RTT. The requested rate is achieved with a very high probability and confidence intervals for the achieved goodput are small.

Overall concluding from the simulation results it seems promising to drop packets already at the ingress point of the network.

The whole concept of the TRC is based on simulations in `ns-2`. So the TRC has to be evaluated by real measurements in TCP/IP networks, because the accuracy of the TRC depends on the TCP model. In real TCP/IP networks there exist a lot of slightly different TCP implementations [11]. Consequently the applicability of the TRC in such an environment has to be evaluated.

The TRC rate controller does not need more suppositions as needed in QoS networks. The diversity of TCP implementations is a general problem of all attempts that try to control or estimate TCP rates based on a TCP model.

There are still open issues which have to be investigated in more detail. For example the dependence of RTT_{dev} on link bandwidth and requested rates has to be analyzed.

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