Linux TCP

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ABSTRACT

The TCP protocol is used by the majority of the network applications on the Internet. TCP performance is strongly influenced by its congestion control algorithms that limit the amount of transmitted traffic based on the estimated network capacity and utilization. Because the freely available Linux operating system has gained popularity especially in the network servers, its TCP implementation affects many of the network interactions carried out today. This paper describes the fundamentals of the Linux TCP design, concentrating on the congestion control algorithms. The Linux TCP implementation supports SACK, TCP timestamps, Explicit Congestion Notification, and techniques to undo congestion window adjustments after incorrect congestion notifications.

This paper describes the basic concepts in Linux TCP and its congestion control engine. In addition to features specified by IETF, Linux has implementation details beyond the specifications aimed to further improve its performance. The paper presents how Linux TCP differs from the traditional TCP. Finally, we discuss whether it would be reasonable to implement TCP as a kernel module or as a user library.

1. INTRODUCTION

The *Transmission Control Protocol (TCP)* [18 21] has evolved for over 20 years, being the most commonly used transport protocol on the Internet today. An important characteristic feature of TCP are its congestion control algorithms, which are essential for preserving network stability when the network load increases. The TCP congestion control principles require that if the TCP sender detects a packet loss, it should reduce its transmission rate, because the packet was probably dropped by a congested router.

Linux is a freely available Unix-like operating system that has gained popularity in the last years. The Linux source code is publicly available[1], which makes Linux an attractive tool for the computer science researchers in various research areas. Therefore, a large number of people have contributed to Linux development during its lifetime. However, many people find it tedious to study the different aspects of the Linux behavior just by reading the source code. Therefore, this paper describes the design solutions selected in the TCP implementation of the Linux kernel version 2.4. The Linux TCP implementation contains features that differ from the other TCP implementations used today, and I believe that the protocol designers working with TCP find these features interesting considering their work.

[1]The Linux kernel source can be obtained from `http://www.kernel.org/`.

The Internet protocols are standardized by the *Internet Engineering Task Force (IETF)* in documents called *Request For Comments (RFC)*. Currently there are thousands of RFCs, of which tens are related to the TCP protocol. In addition to the mandatory TCP specifications, there are a number of experimental and informational specifications of TCP enhancements for improving the performance under certain conditions, which can be implemented optionally.

Building up a single consistent protocol implementation which conforms to the different RFCs is not a straightforward task. For example, the TCP congestion control specification [2] gives a detailed description of the basic congestion control algorithm, making it easier for the implementor to apply it. However, if the TCP implementation supports SACK TCP [15], it needs to follow congestion control specifications that use a partially different set of concepts and variables than those given in the standard congestion control RFC [6 3]. Therefore, strictly following the algorithms used in the specifications makes an implementation unnecessarily complicated, as usually both algorithms need to be included in the TCP implementation at the same time.

This work describes the approach taken in Linux TCP for implementing the congestion control algorithms. Linux TCP implements many of the RFC specifications in a single congestion control engine, using the common code for supporting both SACK TCP and NewReno TCP without SACK information. In addition, Linux TCP refines many of the specifications in order to improve the TCP efficiency. We describe the Linux-specific protocol enhancements in this paper. Additionally, our goal is to point out the details where Linux TCP behavior differs from the conventional TCP implementations or the RFC specifications.

This paper is organized as follows. Section 2 describes the TCP protocol and its congestion control algorithms in more detail. Section 3 introduces the basics of the Linux TCP and describe the main algorithms governing the packet retransmission logic. Additionally, the section describe some features specific to the Linux TCP

implementation. Section 4 discusses how the current Linux TCP implementation relates to the IETF specifications, and whether Linux TCP is compliant to IETF specifications. Section 5 discusses whether it would be feasible to implement Linux TCP as a kernel module or as a user-space library, and considers the benefits and drawbacks of doing so. Section 6 concludes this work.

2. TCP BASICS

We now briefly describe the TCP congestion control algorithms that are referred to throughout this paper. Because the congestion control algorithms play an important role in TCP performance, a number of further enhancements for the TCP algorithms have been suggested. We describe here the ones considered most important. Finally, we point out a few details considered problematic in the current TCP specifications by IETF as a motivation for the Linux TCP approach.

2.1 Congestion control

The TCP protocol basics are specified in RFC 793 [18]. In order to avoid the network congestion that became a serious problem as the number of network hosts increased dramatically, the basic algorithms for performing congestion control were given by Jacobson [11]. Later, the congestion control algorithms have been included in the standards track TCP specification by the IETF [2].

The TCP sender uses a *congestion window (cwnd)* in regulating its transmission rate based on the feedback it gets from the network. The congestion window is the TCP sender's estimate of how much data can be outstanding in the network without packets being lost. After initializing `cwnd` to one or two segments, the TCP sender is allowed to increase the congestion window either according to a *slow start* algorithm, that is, by one segment for each incoming acknowledgement (ACK), or according to *congestion avoidance*, at a rate of one segment in a round-trip time. The *slow start threshold (ssthresh)* is used to determine whether to use slow start or congestion avoidance algorithm. The TCP sender starts with the slow start algorithm and moves to congestion avoidance when `cwnd` reaches the `ssthresh`.

The TCP sender detects packet losses from incoming duplicate acknowledgements, which are generated by the receiver when it receives out-of-order segments. After three successive duplicate ACKs, the sender retransmits a segment and sets `ssthresh` to half of the amount of currently outstanding data. `cwnd` is set to the value of `ssthresh` plus three segments, accounting for the segments that have already left the network according to the arrived duplicate ACKs. In effect the sender halves its transmission rate from what it was before the loss event. This is done because the packet loss is taken as an indication of congestion, and the sender needs to reduce its transmission rate to alleviate the network congestion.

The retransmission due to incoming duplicate ACKs is called *fast retransmit*. After fast retransmit the TCP sender follows the *fast recovery* algorithm until all segments in the last window have been acknowledged. During fast recovery the TCP sender maintains the number of outstanding segments by sending a new segment for each incoming acknowledgement, if the congestion window allows. The TCP congestion control specification temporarily increases the congestion window for each incoming duplicate ACK to allow forward transmission of a segment, and deflates it back to the value at the beginning of the fast recovery when the fast recovery is over.

Two variants of the fast recovery algorithm have been suggested by the IETF. The standard variant exits the fast recovery when the first acknowledgement advancing the window arrives at the sender. However, if there is more than one segment dropped in the same window, the standard fast retransmit does not perform efficiently. Therefore, an alternative called *NewReno* was suggested [7] to improve the TCP performance. NewReno TCP exits the fast recovery only after all segments in the last window have been successfully acknowledged.

Retransmissions may also be triggered by the retransmission timer, which expires at the TCP sender when no new data is acknowledged for a while. *Retransmission timeout (RTO)* is taken as a loss indication, and it triggers retransmission of the unacknowledged segments. In addition, when RTO occurs, the sender resets the congestion window to one segment, since the RTO may indicate that the network load has changed dramatically.

The TCP sender estimates packet round-trip times (RTT) and uses the estimator in determining the RTO value. When a segment arrives at the TCP sender, the IETF specifications instruct it to adjust the RTO value as follows [16]:

$$\texttt{RTTVAR} \leftarrow \tfrac{3}{4} * RTTVAR + \tfrac{1}{4} * |SRTT - R|$$
$$\texttt{SRTT} \leftarrow \tfrac{7}{8} * SRTT + \tfrac{1}{8} * R$$
$$\texttt{RTO} \leftarrow max(SRTT + 4 * RTTVAR, 1s.)$$

where `R` is the measured round-trip time, `RTTVAR` is variation of the recent round-trip times, and `SRTT` is the smoothed mean round-trip time based on the recent measurements.

2.2 Enhancements

Recovery from the packet losses is inefficient in the standard TCP because the cumulative acknowledgements allow only one retransmission in a round-trip time. Therefore, *Selective Acknowledgements (SACK)* [15] were suggested to make it possible for the receiver to acknowledge scattered blocks of incoming data instead of a single cumulative acknowledgement, allowing the TCP sender to make more than one retransmission in a round-trip time. SACK can be used only if both ends of the TCP connection support it.

Availability of the SACK information allows the TCP sender to perform congestion control more accurately. Instead of temporarily adjusting the congestion window, the sender can keep track of the amount of outstanding data and compare it against the congestion window when deciding whether it can transmit new segments [3]. However, the unacknowledged segments can be treated in different ways when accounting for outstanding data. The conservative approach promoted by IETF is to consider all unacknowledged data to be outstanding in the network. The *Forward Acknowledgements (FACK)* algorithm [14] takes a more aggressive approach and considers the unacknowledged holes between the SACK blocks as lost packets. Although this approach often results in better TCP performance than the conservative approach, it is overly aggressive if packets have been reordered in the network, because the holes between SACK blocks do not indicate lost packets in this case.

The SACK blocks can also be used for reporting spurious retransmissions. The *Duplicate-SACK (D-SACK)* enhancement [8] allows the TCP receiver to report any duplicate segments it gets by using the SACK blocks. Having this information the TCP sender

can conclude in certain circumstances whether it has unnecessarily reduced its congestion control parameters, and thus revert the parameters to the values preceding the retransmission. For example, packet reordering is a potential reason for unnecessary retransmissions, because out-of-order segments trigger duplicate ACKs at the receiver.

The *TCP Timestamp option* [4] was suggested to allow more accurate round-trip time measurements, especially on network paths with high bandwidth-delay product. A timestamp is attached to each TCP segment, which is then echoed back in the acknowledgement for the segment. From the echoed timestamp the TCP sender can measure exact round-trip times for the segments and use the measurement for deriving the retransmission timeout estimator. In addition to the more exact round-trip time measurement, use of TCP timestamps allows algorithms for protecting against old segments from the previous incarnations of the TCP connection.

The timestamp option also allows detection of unnecessary retransmissions. The *Eifel Algorithm* [12] suggests that if an acknowledgement for a retransmitted segment echoes a timestamp earlier than the timestamp of the retransmission stored at the sender, the original segment has arrived at the receiver, and the retransmission was unnecessarily made. In such a case, the TCP sender can continue by sending new data and revert the recent changes made to the congestion control parameters.

Instead of inferring congestion from the lost packets, *Explicit Congestion Notification (ECN)* [19] was suggested for routers to explicitly mark packets when they arrive to a congested point in the network. When the TCP sender receives an echoed ECN notification from the receiver, it should reduce its transmission rate to mitigate the congestion in the network. ECN allows the TCP senders to be congestion-aware without necessarily suffering from packet losses.

2.3 Criticism

Some details in IETF specifications are problematic in practice. Although many of the RFCs suggest a general algorithm that could be applied to an implementation, combining the algorithms from several RFCs may be inconvenient. For example, combining the congestion control requirements for SACK TCP and NewReno TCP turns out to be problematic due to different variables and algorithms used in the specifications.

The TCP congestion control specifications artificially increase the congestion window during the fast recovery in order to allow forward transmissions to keep the number of outstanding segments stable. Therefore, the congestion window size does not actually reflect the number of segments allowed to be outstanding during the fast recovery. When fast recovery is over, the congestion window is deflated back to a proper size. This procedure is needed because the congestion window is traditionally evaluated against the difference of the highest data segment transmitted (`SND.NXT`) and the first unacknowledged segment (`SND.UNA`). By taking a more flexible method for evaluating the number of outstanding segments, the congestion window size can be constantly maintained at a proper level corresponding to the network capacity.

Adjusting the congestion window consistently becomes an issue especially when SACK information can be used by the TCP sender. By using the selective acknowledgements, the sender can derive the number of packets with a better accuracy than by just using the cumulative acknowledgements. In order to make a coherent implementation of the congestion control algorithms, it is desirable to have common variables and routines both for SACK TCP and for the TCP variant to use when the other end does not support SACK.

Finally, the details of the RTO algorithm presented above have been questioned. Since many networks have round-trip delays of a few tens of milliseconds or less, the RTO algorithm details may not have a significant effect on TCP performance, since the minimum RTO value is limited to one second. However, there are high-delay networks for which the effectiveness of the RTO calculation is important. It has been pointed out that the RTO estimator results in overly large values due to the weight given for the variance of the round-trip time, when the round-trip time suddenly drops for some reason. On the other hand, when the congestion window size increases at a steady pace during the slow start, it is possible that the RTO estimator is not increased fast enough due to small variance in the round-trip times. This may result in spurious retransmission timeouts. Alternative RTO estimators, such as the *Eifel Retransmission Timer* [13], have been suggested to overcome the potential problems in the standard RTO algorithm. However, although the Eifel Retransmission Timer is efficient in avoiding the problems of the standard RTO algorithm, it introduces a rather complex set of equations compared to the standard RTO. Therefore, evaluating the possible side effects of different network scenarios on Eifel RTT dynamics is difficult.

3. LINUX APPROACH

This section describes the Linux TCP implementation, starting from basic data structures and the fundamentals of the congestion control engine. Finally, some specific features of Linux TCP are described.

3.1 Basic data structures

The state of the TCP connection is stored in a `sock` structure. `sock` structure is a protocol-independent entity that stores information about source and destination addresses to be used for the connection, a pointer to the destination cache that is used for finding packet destinations efficiently, and other miscellaneous data about the connection state.

The `sock` structure has also a protocol specific portion of data, in which the TCP state is stored. This area holds information about the current congestion window size, maximum segment size, round-trip time measurements, retransmission counters, and various other statistics. In fact, most of the variables referred to in the rest of this section are stored in the TCP-specific part of the `sock` structure.

Each packet belonging to a socket is held in a structure called `sk_buff`. Each socket has a dedicated queues for incoming and outgoing `sk_buffs`. In addition to the packet header and payload, `sk_buff` has a small control portion, that includes various pointers and protocol identifiers. `sk_buffs` are linked to each other as a circular bidirectional linked list, which allows flexible iteration through packets. There is no physical socket buffers for incoming and outgoing data, but the socket buffer sizes that are set, for example, by using socket options only set an upper limit to the number of `sk_buffs` that may be linked to the socket. By taking this approach, the number of expensive memory copy operations can be minimized.

Figure 1 illustrates the relation ship of `sock` structures and

sk_buffs. There are three different queues for storing sk_buffs[2]. *Write queue* stores the data written by the application not yet acknowledged by the receiver, *receive queue* stores the packets received from the network not yet read by the application, and *out-of-order queue* (not shown in the figure) stores packets that have arrived from the network, but cannot be delivered to the application because some data is missing before the segments in the out-of-order queue. For example, when data is fetched from application, and a new sk_buff is created, the application data is filled to the area reserved for payload. At this point, a placeholder is left for TCP, IP and lower layer headers. The necessary pointers are filled to the control portion in the beginning of sk_buff. The protocol headers are filled based on the data in the sock struct only before the packet is passed to the lower layers to be transmitted. If the payload was not at its maximum size, it may be filled later when new data is received from the application, as long as the TCP sender has not transmitted the segment.

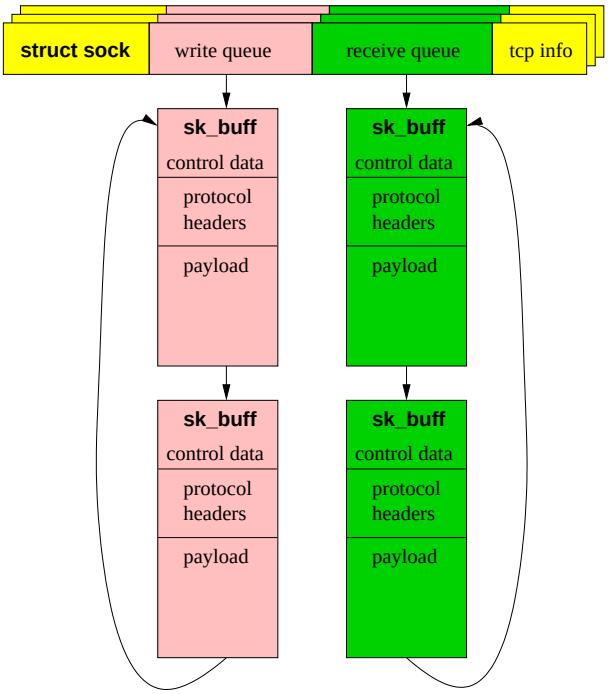

Figure 1: Socket state and sk_buffs for storing packets.

3.2 Congestion control engine

Although Linux conforms to the TCP congestion control principles, it takes a different approach in carrying out the congestion control. Instead of comparing the congestion window to the difference of SND.NXT and SND.UNA, the Linux TCP sender determines the number of packets currently outstanding in the network. The Linux TCP sender then compares the number of outstanding segments to the congestion window when making decisions on how much to transmit. Linux tracks the number of outstanding segments in units of full-sized packets, whereas the TCP specifications and some implementations compare cwnd to the number of transmitted octets. This results in different behavior if small segments are used: if the implementation uses a byte-based congestion window, it allows several small segments to be injected in the network for each full-sized segment in the congestion window. Linux, on the other hand, allows only one packet to be transmitted for each segment in the congestion window, regardless of its size. Therefore, Linux is more conservative compared to the byte-based approach when the TCP payload consists of small segments.

The Linux TCP sender uses the same set of concepts and functions for determining the number of outstanding packets with the NewReno recovery and with the two flavors of SACK recovery supported. When the SACK information can be used, the sender can either follow the *Forward Acknowledgements (FACK)* [14] approach considering the holes between the SACK blocks as lost segments, or a more conservative approach similar to the ongoing work under IETF [3]. In the latter alternative the unacknowledged segments are considered outstanding in the network. As a basis for all recovery methods the Linux TCP sender uses the equations:

```
left_out <- sacked_out + lost_out
in_flight <- packets_out -
             left_out + retrans_out
```

in defining the number of segments outstanding in the network. In the equation above, packets_out is the number of originally transmitted segments above SND.UNA, sacked_out is the number of segments acknowledged by SACK blocks, lost_out is an estimation of the number of segments lost in the network, and retrans_out is the number of retransmitted segments. Determining the lost_out parameter depends on the selected recovery method. For example, when FACK is in use, all unacknowledged segments between the highest SACK block and the cumulative acknowledgement are counted in lost_out. The selected approach makes it easy to add new heuristics for evaluating which segments are lost.

In the absence of SACK information, the Linux TCP sender increases sacked_out by one for each incoming duplicate acknowledgement. This is in conformance with the TCP congestion control specification, and the resulting behavior is similar to the *NewReno* algorithm with its forward transmissions. The design chosen in Linux does not require arbitrary adjusting of the congestion window, but cwnd holds the valid number of segments allowed to be outstanding in the network throughout the fast recovery.

The counters used for tracking the number of outstanding, acknowledged, lost, or retransmitted packets require additional data structures for supporting them. The Linux sender maintains the state of each outstanding segment in a scoreboard, where it marks the known state of the segment. The segment can be marked as outstanding, acknowledged, retransmitted, or lost. Combinations of these bits are also possible. For example, a segment can be declared lost and retransmitted, in which case the sender is expecting to get an acknowledgement for the retransmission. Using this information the Linux sender knows which segments need to be retransmitted, and how to adjust the counters used for determining in_flight when a new acknowledgement arrives. The scoreboard also plays an important role when determining whether a segment has been incorrectly assumed lost, for example due to packet reordering.

The scoreboard markings and the counters used for determining the in_flight variable should be in consistent state at all times.

[2]TCP receiver also maintains *prequeue* and *backlog queue* for directly copying data from lower layers to application, but these features are not discussed in this paper.

Markings for outstanding, acknowledged and retransmitted segments are straightforward to maintain, but when to place a *lost* mark depends on the recovery method used. With NewReno recovery, the first unacknowledged packet is marked lost when the sender enters the fast recovery. In effect, this corresponds to the fast retransmit of the IETF congestion control specifications. Furthermore, when a partial ACK not acknowledging all the data outstanding at the beginning of the fast recovery arrives, the first unacknowledged segment is marked lost. This results in retransmission of the next unacknowledged segment, as the NewReno specification requires.

When SACK is used, more than one segment can be marked lost at a time. With the conservative approach, the TCP sender does not count the holes between the acknowledged blocks in `lost_out`, but when FACK is enabled, the sender marks the holes between the SACK blocks lost as soon as they appear. The `lost_out` counter is adjusted appropriately.

The Linux TCP sender is governed by a state machine that determines the sender actions when acknowledgements arrive. Figure 2 illustrates how the states are related with each other. The congestion control states are as follows:

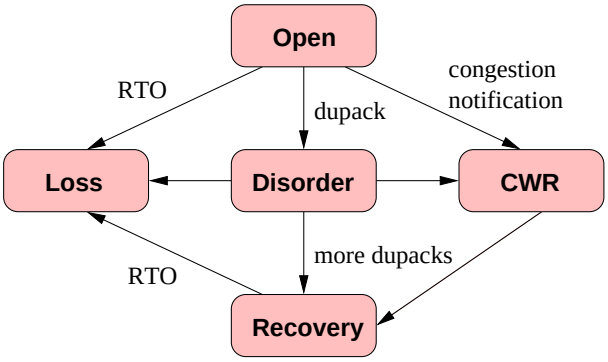

Figure 2: Congestion control state machine.

- **Open**. This is the normal state in which the TCP sender follows the fast path of execution optimized for the common case in processing incoming acknowledgements. When an acknowledgement arrives, the sender increases the congestion window according to either slow start or congestion avoidance, depending on whether the congestion window is smaller or larger than the slow start threshold, respectively.
- **Disorder**. When the sender detects duplicate ACKs or selective acknowledgements, it moves to the *Disorder* state. In this state the congestion window is not adjusted, but each incoming packet triggers transmission of a new segment. Therefore, the TCP sender follows the packet conservation principle [11], which states that a new packet is not sent out until an old packet has left the network. In practice the behavior in this state is similar to the *limited transmit* proposal by IETF [1], which was suggested to allow more efficient recovery by using fast retransmit when congestion window is small, or when a large number of segments are lost in the last window of transmission.
- **CWR**. The TCP sender may receive congestion notifications either by Explicit Congestion Notification, *ICMP source quench* [17], or from a local device. When receiving a congestion notification, the Linux sender does not reduce the congestion window at once, but by one segment for every second incoming ACK until the window size is halved. When the sender is in process of reducing the congestion window size and it does not have outstanding retransmissions, it is in *CWR (Congestion Window Reduced)* state. CWR state can be interrupted by *Recovery* or *Loss* states described below.
- **Recovery**. After a sufficient amount of successive duplicate ACKs arrive at the sender, it retransmits the first unacknowledged segment and enters the *Recovery* state. By default, the threshold for entering *Recovery* is three successive duplicate ACKs, a value recommended by the TCP congestion control specification. During the *Recovery* state, the congestion window size is reduced by one segment for every second incoming acknowledgement, similar to the *CWR* state. The window reduction ends when the congestion window size is equal to ssthresh, i.e. half of the window size when entering the *Recovery* state. The congestion window is not increased during the recovery state, and the sender either retransmits the segments marked lost, or makes forward transmissions on new data according to the packet conservation principle. The sender stays in the *Recovery* state until all of the segments outstanding when the *Recovery* state was entered are successfully acknowledged. After this the sender goes back to the *Open* state. A retransmission timeout can also interrupt the *Recovery* state.
- **Loss**. When an RTO expires, the sender enters the *Loss* state. All outstanding segments are marked lost, and the congestion window is set to one segment, hence the sender starts increasing the congestion window using the slow start algorithm. A major difference between the *Loss* and *Recovery* states is that in the *Loss* state the congestion window is increased after the sender has reset it to one segment, but in the *Recovery* state the congestion window size can only be reduced. The *Loss* state cannot be interrupted by any other state, thus the sender exits to the *Open* state only after all data outstanding when the *Loss* state began have successfully been acknowledged. For example, fast retransmit cannot be triggered during the *Loss* state, which is in conformance with the NewReno specification.

Linux TCP avoids explicit calls to transmit a packet in any of the above mentioned states, for example, regarding the fast retransmit. The current congestion control state determines how the congestion window is adjusted, and whether the sender considers the unacknowledged segments lost. After the TCP sender has processed an incoming acknowledgement according to the state it is in presently, it transmits segments while `in_flight` is smaller than `cwnd`. The sender either retransmits earlier segments marked lost and not yet retransmitted, or new data segments if there are no lost segments waiting for retransmission.

There are occasions where the number of outstanding packets decreases suddenly by several segments. For example, a retransmitted segment and the following forward transmissions can be acknowledged with a single cumulative ACK. These situations would cause bursts of data to be transmitted into the network, unless they are taken into account in the TCP sender implementation. The Linux TCP sender avoids the bursts by limiting the congestion window to allow at most three segments to be transmitted for an incoming

ACK. Since burst avoidance may reduce the congestion window size below the slow start threshold, it is possible for the sender to enter slow start after several segments have been acknowledged by a single ACK.

When a TCP connection is established, many of the TCP variables need to be initialized with some fixed values. However, in order to improve the communication efficiency at the beginning of the connection, the Linux TCP sender stores in its destination cache the slow start threshold, the variables used for the RTO estimator, and an estimator measuring the likeliness of reordering after each TCP connection. If another connection is established to the same destination IP address that is found in the cache, the cached values can be used to get adequate initial values for the new TCP connection. If the network conditions between the sender and the receiver change for some reason, the values in the destination cache could get momentarily outdated. However, this is considered a minor disadvantage.

3.3 Specific features

We now list selected Linux TCP features that differ from a typical TCP implementation. Linux implements a number of TCP enhancements proposed recently by IETF, such as *Explicit Congestion Notification* [19] and *D-SACK* [8]. These features are not yet widely deployed in TCP implementations, but are likely to be in the future because they are promoted by the IETF.

3.3.1 Retransmission timer calculation

Some TCP implementations use a coarse-grained retransmission timer, having granularities up to 500 ms. The round-trip time samples are often measured once in a round-trip time. In addition, the present retransmission timer specification requires that the RTO timer should not be less than one second. Considering that most of the present networks provide round-trip times of less than 500 ms, studying the feasibility of the traditional retransmission timer algorithm standardized by IETF has not excited much interest.

Linux TCP has a retransmission timer granularity of 10 ms and the sender takes a round-trip time sample for each segment. Therefore it is capable of achieving more accurate estimations for the retransmission timer, if the assumptions in the timer algorithm are correct. Linux TCP deviates from the IETF specification by allowing a minimum limit of 200 ms for the RTO. Because of the finer timer granularity and the smaller minimum limit for the RTO timer, the correctness of the algorithm for determining the RTO is more important than with a coarse-grain timer. The traditional algorithm for retransmission timeout computation has been found to be problematic in some networking environments [13]. This is especially true if a fine-grained timer is used and the round-trip time samples are taken for each segment.

Section 2 described two problems regarding the standard RTO algorithm. First, when the round-trip time decreases suddenly, RTT variance increases momentarily and causes the RTO value to be overestimated. Second, the RTT variance can decay to a small value when RTT samples are taken for every segment while the window is large. This increases the risk for spurious RTOs that result in unnecessary retransmissions.

The Linux RTO estimator attacks the first problem by giving less weight for the measured mean deviance (MDEV) when the measured RTT decreases significantly below the smoothed average. The reduced weight given for the MDEV sample is based on the multipliers used in the standard RTO algorithm. First, the MDEV sample is weighed by $\frac{1}{8}$, corresponding to the multiplier used for the recent RTT measurement in the SRTT equation given in Section 2. Second, MDEV is further multiplied by $\frac{1}{4}$ corresponding to the weight of 4 given for the RTTVAR in the standard RTO algorithm. Therefore, choosing the weight of $\frac{1}{32}$ for the current MDEV neutralizes the effect of the sudden change of the measured RTT on the RTO estimator, and assures that RTO holds a steady value when the measured RTT drops suddenly. This avoids the unwanted peak in the RTO estimator value, while maintaining a conservative behavior. If the round-trip times stay at the reduced level for the next measurements, the RTO estimator starts to decrease slowly to a lower value. In summary, the equation for calculating the MDEV is as follows:

```
if (R < SRTT and |SRTT - R| > MDEV) {
      MDEV <- 31/32 * MDEV + 1/32 * |SRTT - R|
} else {
      MDEV <- 3/4 * MDEV + 1/4 * |SRTT - R|
}
```

where R is the recent round-trip time measurement, and SRTT is the smoothed average round-trip time. Linux does not directly modify the RTTVAR variable, but makes the adjustments first on the MDEV variable which is used in adjusting the RTTVAR which determines the RTO. The SRTT and RTO estimator variables are set according to the standard specification.

A separate MDEV variable is needed, because the Linux TCP sender allows decreasing the RTTVAR variable only once in a round-trip time. However, RTTVAR is increased immediately when MDEV gives a higher estimate, thus RTTVAR is the maximum of the MDEV estimates during the last round-trip time. The purpose of this solution is to avoid the problem of underestimated RTOs due to low round-trip time variance, which was the second of the problems described earlier.

Figure 3 compares the result of RTO calculation of the standard RTO estimator algorithm (RFC 2988) and the Linux RTO algorithm when given round-trip times are measured (mrtt). In this graph, the RTT measurement granularity is 10 ms for both algorithms, and the samples are taken for each packet. The figure shows that when RTT measurements tend to stay in level, the standard RTO estimator decays to the level of measured RTT, making the TCP sender vulnerable to spurious RTOs. The Linux RTO estimator, on the other hand, is reduced only once in a round-trip time and the RTT variance used in calculation is 50 ms at its minimum. Furthermore, when RTT suddenly decreases, the standard RTO estimator is momentarily increased, whereas the Linux RTO estimator is steadily decreased.

Linux TCP supports the *TCP Timestamp option* that allows accurate round-trip time measurement also for retransmitted segments, which is not possible without using timestamps. Having a proper algorithm for RTO calculation is even more important with the timestamp option. According to our experiments, the algorithm proposed above gives reasonable RTO estimates also with TCP timestamps, and avoids the pitfalls of the standard algorithm.

The RTO timer is reset every time an acknowledgement advancing the window arrives at the sender. The RTO timer is also reset when the sender enters the *Recovery* state and retransmits the first seg-

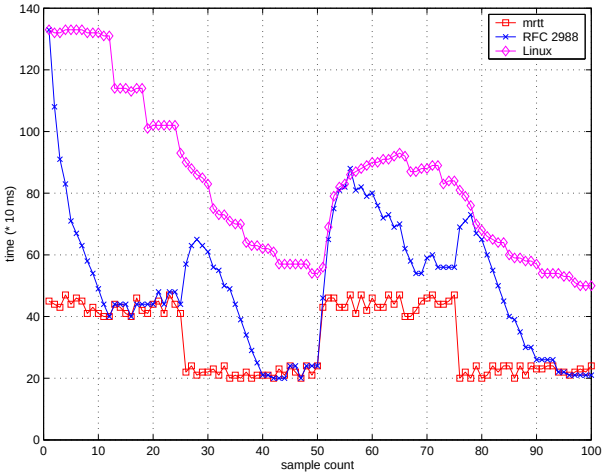

Figure 3: Linux RTO calculation with generated round-trip times.

ment. During the rest of the Recovery state the RTO timer is not reset, but a packet is marked lost, if more than RTO's worth of time has passed from the first transmission of the same segment. This allows more efficient retransmission of packets during the *Recovery* state even though the information from acknowledgements is not sufficient enough to declare the packet lost. However, this method can only be used for segments not yet retransmitted.

3.3.2 *Undoing congestion window adjustments*

Because the currently used mechanisms on the Internet do not provide explicit loss information to the TCP sender, it needs to speculate when keeping track of which packets are lost in the network. For example, reordering is often a problem for the TCP sender because it cannot distinguish whether the missing ACKs are caused by a packet loss or by a delayed packet that will arrive later. The Linux TCP sender can, however, detect unnecessary congestion window adjustments afterwards, and do the necessary corrections in the congestion control parameters. For this purpose, when entering the *Recovery* or *Loss* states, the Linux TCP sender stores the old `ssthresh` value prior to adjusting it.

A delayed segment can trigger an unnecessary retransmission, either due to spurious retransmission timeout or due to packet reordering. The Linux TCP sender has mainly two methods for detecting afterwards that it unnecessarily retransmitted the segment. Firstly, the receiver can inform by a *Duplicate-SACK (D-SACK)* that the incoming segment was already received. If all segments retransmitted during the last recovery period are acknowledged by D-SACK, the sender knows that the recovery period was unnecessarily triggered. Secondly, the Linux TCP sender can detect unnecessary retransmissions by using the TCP timestamp option attached to each TCP header. When this option is in use, the TCP receiver echoes the timestamp of the segment that triggered the acknowledgement back to the sender, allowing the TCP sender to conclude whether the ACK was triggered by the original or by the retransmitted segment. The *Eifel* algorithm uses a similar method for detecting spurious retransmissions.

When an unnecessary retransmission is detected by using TCP timestamps, the logic for undoing the congestion window adjustments is simple. If the sender is in the *Loss* state, i.e. it is retransmitting after an RTO which was triggered unnecessarily, the *lost* mark is removed from all segments in the scoreboard, causing the sender to continue with transmitting new data instead of retransmissions. In addition, `cwnd` is set to the maximum of its present value and `ssthresh` * 2, and the `ssthresh` is set to its prior value stored earlier. Since `ssthresh` was set to the half of the number of outstanding segments when the packet loss is detected, the effect is to continue in congestion avoidance at a similar rate as when the *Loss* state was entered.

Unnecessary retransmission can also be detected by the TCP timestamps while the sender is in the *Recovery* state. In this case the Recovery state is finished normally, with the exception that the congestion window is increased to the maximum of its present value and `ssthresh` * 2, and `ssthresh` is set to its prior value. In addition, when a partial ACK for the unnecessary retransmission arrives, the sender does not mark the next unacknowledged segment lost, but continues according to present scoreboard markings, possibly transmitting new data.

In order to use D-SACK for undoing the congestion control parameters, the TCP sender tracks the number of retransmissions that have to be declared unnecessary before reverting the congestion control parameters. When the sender detects a D-SACK block, it reduces the number of revertable outstanding retransmissions by one. If the D-SACK blocks eventually acknowledge every retransmission in the last window as unnecessarily made and the retransmission counter falls to zero due to D-SACKs, the sender increases the congestion window and reverts the last modification to `ssthresh` similarly to what was described above.

While handling the unnecessary retransmissions, the Linux TCP sender maintains a metric measuring the observed reordering in the network in variable `reordering`. This variable is also stored in the destination cache after the connection is finished. `reordering` is updated when the Linux sender detects unnecessary retransmission during the *Recovery* state by TCP timestamps or D-SACK, or when an incoming acknowledgement is for an unacknowledged hole in the sequence number space below selectively acknowledged sequence numbers. In these cases `reordering` is set to the number of segments between the highest segment acknowledged and the currently acknowledged segment, in other words, it corresponds to the maximum distance of reordering in segments detected in the network. Additionally, if FACK was in use when reordering was detected, the sender switches to use the conservative variant of SACK, which is not too aggressive in a network involving reordering.

3.3.3 *Delayed acknowledgements*

The TCP specifications state that the TCP receiver should delay the acknowledgements for a maximum time of 500 ms in order to avoid the *Silly Window Syndrome* [5]. The specifications do not mandate any specific delay time, but many implementations use a static delay of 200 ms for this purpose. However, a fixed delay time may not be adequate in all networking environments with different properties. Thus, the Linux TCP receiver adjusts the timer for delaying acknowledgements dynamically to estimate the doubled packet interarrival time, while sending acknowledgements for at least every second incoming segment. A similar approach was also suggested in an early RFC by Clark [5]. However, the maximum delay for sending an acknowledgement is limited to 200 ms.

Using delayed ACKs slows down the TCP sender, because it increases the congestion window size based on the rate of incoming acknowledgements. In order to speed up the transmission in the beginning of the slow start, the Linux TCP receiver refrains from delaying the acknowledgements for the first incoming segments at the beginning of the connection. This is called *quick acknowledgements.*

The number of quick acknowledgements sent by the Linux TCP receiver is at most half of the number of segments required to reach the receiver's advertised window limit. Therefore, using quick acknowledgements does not open the opportunity for the Silly Window Syndrome to occur. In addition, the Linux receiver monitors whether the traffic appears to be bidirectional, in which case it disables the quick acknowledgements mechanism. This is done to avoid transmitting pure acknowledgements unnecessarily when they can be piggybacked with data segments.

3.3.4 Congestion Window Validation

The Linux sender reduces the congestion window size if it has not been fully used for one RTO estimate's worth of time. This scheme is similar to the *Congestion Window Validation* suggested by the IETF [9]. The motivation for Congestion Window Validation is that if the congestion window is not fully used, the TCP sender may have an invalid estimate of the present network conditions. Therefore, a network-friendly sender should reduce the congestion window as a precaution.

When the Congestion Window Validation is triggered, the TCP sender decreases the congestion window to half way between the actually used window and the present congestion window. Before doing this, `ssthresh` is set to the maximum of its current value and $\frac{3}{4}$ of the congestion window, as suggested in RFC 2861.

3.3.5 Explicit Congestion Notification

Linux implements *Explicit Congestion Notification (ECN)* to allow the ECN-capable congested routers to report congestion before dropping packets. A congested router can mark a bit in the IP header, which is then echoed to the TCP sender by an ECN-capable receiver. When the TCP sender gets the congestion signal, it enters the *CWR* state, in which it gradually decreases the congestion window to half of its current size at the rate of one segment for two incoming acknowledgements. Besides making it possible for the TCP sender to avoid some of the congestion losses, ECN is expected to improve the network performance when it is more widely deployed to the Internet routers.

4. LINUX AND IETF SPECIFICATIONS

Since Linux combines the features specified in different IETF specifications following certain design principles described earlier, some IETF specifications are not fully implemented according to the algorithms given in the RFCs. Table 1 shows which RFC specifications related to TCP congestion control are implemented in Linux. Some of the features shown in the table can be found in Linux, but they do not fully follow the given specification in all details. These features are marked with an asterisk in the table, and we will explain the differences between Linux and the corresponding RFC in more detail below.

Linux fast recovery does not fully follow the behavior given in RFC 2582. First, the sender adjusts the threshold for triggering fast retransmit dynamically, based on the observed reordering in the network. Therefore, it is possible that the third duplicate ACK does not trigger a fast retransmit in all situations. Second, the Linux sender does not artificially adjust the congestion window during fast recovery, but maintains its size while adjusting the `in_flight` estimator based on incoming acknowledgements. The different approach alone would not cause significant effect on TCP performance, but when entering the fast recovery, the Linux sender does not reduce the congestion window size at once, as RFC 2582 suggests. Instead, the sender decreases the congestion window size gradually, by one segment per two incoming acknowledgements, until the congestion window meets half of its original value. This approach was originally suggested by Hoe [10], and later it was named *Rate-halving* according to an expired Internet Draft by Mathis, et. al. Rate-halving avoids pauses in transmission, but is slightly too aggressive after the congestion notification, until the congestion window has reached a proper size.

Table 1: TCP congestion control related IETF specifications implemented in Linux. + = implemented, * = implemented, but details differ from specification.

Specification	Status
RFC 1323 (Perf. Extensions)	+
RFC 2018 (SACK)	+
RFC 2140 (Ctrl block sharing)	+
RFC 2581 (Congestion control)	*
RFC 2582 (NewReno)	*
RFC 2861 (Cwnd validation)	+
RFC 2883 (D-SACK)	+
RFC 2988 (RTO)	*
RFC 3042 (Lim. xmit)	+
RFC 3168 (ECN)	*

As described in Section 3.3, the round-trip time estimator and RTO calculation in Linux differs from the Proposed Standard specification by the IETF. Linux follows the basic patterns given in RFC 2988, but the implementation differs from the specification in adjusting the RTTVAR. A significant difference between RFC 2988 and Linux implementation is that Linux uses the minimum RTO limit of 200 ms instead of 1000 ms given in RFC 2988.

RFC 2018 defines the format and basic usage of the SACK blocks, but does not give detailed specification of the congestion control algorithm that should be used with SACK. Therefore, applying the FACK congestion control algorithm, as Linux does by default, does not violate the current IETF specifications. However, since FACK results in overly aggressive behavior when packets have been reordered in the network, the Linux sender changes from FACK to a more conservative congestion control algorithm when it detects reordering. The IETF currently has a work in progress draft defining a congestion control algorithm to be used with SACK [3], which is similar to the conservative SACK alternative in Linux. Furthermore, Linux follows the D-SACK basics given in RFC 2883.

Linux implements RFC 1323, which defines the TCP timestamp and window scaling options, and the limited transmit enhancement defined in RFC 3042, which is taken care of by the *Disorder* state of the Linux TCP state machine. However, if the `reordering` estimator has been increased from the default of three segments, the Linux TCP sender transmits a new segment for each incoming acknowledgement, not only for the two first ACKs. Finally, the Linux destination cache provides functionality similar to the

RFC 2140 that proposes Control Block Interdependence between the TCP connections.

5. WHAT COULD BE DIFFERENT?

Currently the Linux TCP implementation is a fixed part of the kernel core, and it cannot be compiled as a kernel module. This means, for example, that if one does not use methods like *User-Mode Linux*, the machine needs to be rebooted in order to activate experimental modifications compiled in the TCP implementation.

This section discusses the possibility of either compiling the TCP as a kernel module, thus allowing loading of alternative TCP implementations without booting the machine, or as a user-space library that would be even easier to be loaded dynamically and debugged if so needed. Neither of these alternatives are impossible by definition, since there currently are SCTP [22] implementations for Linux both as a kernel module, and as a user library.

5.1 TCP as a kernel module

Traditionally, the Linux TCP and IPv4 implementations have been a part of the monolithic kernel core and they cannot be detached, even though for example IPv6 code can be compiled as a dynamically loadable kernel module. Although some reports of successful efforts of making modularized TCP implementations are posted to the mailing lists, none of the code patches have ever made their way to the main kernel implementation.

Having the TCP implementation as a kernel module would be desirable in order to allow including TCP modifications in the kernel without having to reboot the machine. As of today, several TCP enhancements have been presented, and the TCP implementor has a number of design choices to select from. Although many of the TCP parameters can be adjusted in Linux by setting *sysctl* parameters, changing the basic implementation would still benefit people who want to experiment with new TCP modifications.

The presumed main reason for not having a TCP module in the main kernel have been that the developers deciding on the networking features have not seen it important enough to go for the relatively large effort of converting the current TCP implementation in to a kernel module. The TCP implementation has a number of shared data structures with the IP code, causing the required modifications to be a notable effort. Moreover, since TCP and UDP protocols are in use most of the time in a modern Unix system, the capability of dynamically loading and unloading the TCP implementation is not very useful for a common user.

5.2 TCP as a user-space implementation

To take the idea of modular TCP even further, it may be interesting to discuss whether it would be feasible to take the TCP code completely out of the kernel and implement TCP as a part of C library. Currently, the kernel provides mechanisms that almost allow this kind of approach, such as *raw sockets*, *packet sockets*, or the *netfilter interface*, which can be used to handle network packets in user-space hooks. However, these mechanisms are not designed for to be used in replacing the TCP or UDP protocol handling, so they would need to be slightly modified to better support this kind of use.

Benefits of an user-space TCP would be quite obvious. if the protocol would be included as a part of the user applications, different users at the same host could use their own TCP preferences. Debugging the TCP code would be as easy as running `gdb`, and fatal errors in the code could cause a segment violation at their worst, instead of crashing the machine.

There are also a number of drawbacks in having TCP as a user-space library. Firstly, the performance on highly loaded systems, such as busy web servers, can be questioned. Context switches between kernel and user space may become a burden that seriously deflates the overall system performance. Moreover, considering security in multi-user systems, it might not be wise to give the users a chance to introduce their own TCP variants, even though it might require root privileges. In addition, there are system-wide data, such as the results of Path MTU discovery, that would need to be shared between all TCP connections in the same system, and would probably be better located in the kernel. However, it could be possible to provide a common interface that would optionally allow use of user-space TCP instead of the kernel TCP implementation when system administrator has specifically enabled it. In this model, experimental TCP modification could be first tested as a user-space code, after which it could be moved to kernel with little effort. This kind of development model has been used in some Real-Time Linux systems, such as RTAI.

6. CONCLUDING REMARKS

This paper presented the basic ideas of the Linux TCP implementation, and gave a description of the details in Linux TCP that differ from a conventional TCP implementation. Linux implements many of the recent TCP enhancements suggested by the IETF, some of which are still at a draft state. Therefore Linux makes it possible to test the interoperability of the recent enhancements in an actual network. For example, Linux had the first implementation of the *Explicit Congestion Notification* enhancement, which revealed bugs in several firewall implementations. The current design of Linux retransmission engine makes it easy to implement and study new alternative congestion control policies also in future.

The Linux TCP behavior is strongly governed by the packet conservation principle and the sender's estimate of which packets are still in the network, which are acknowledged, and which are declared lost. Whether to retransmit or transmit new data depends on the markings made in the TCP sender's scoreboard. In most of the cases none of the requirements given by the IETF are violated, although in marginal scenarios the detailed behavior may be different from what is given in the IETF specifications. However, the TCP essentials, in particular the congestion control principles and conservation of packets, are maintained in all cases.

The selected approach can also be problematic when implementing some features. Because Linux combines the features in different IETF specifications under the same congestion control engine, an uncareful implementation may break some parts of the retransmission logic. For example, if the balance between congestion window and `in_flight` variable is broken, fast recovery algorithm may not work correctly in all situations.

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