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Y. Cheng
N. Cardwell
Google, Inc

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RACK: a time-based fast loss detection algorithm for TCP
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Abstract

This document presents a new TCP loss detection algorithm called RACK ("Recent ACKnowledgment"). RACK uses the notion of time, instead of packet or sequence counts, to detect losses, for modern TCP implementations that can support per-packet timestamps and the selective acknowledgment (SACK) option. It is intended to replace the conventional DUPACK threshold approach and its variants, as well as other nonstandard approaches.

Status of This Memo

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1. Introduction

This document presents a new loss detection algorithm called RACK ("Recent ACKnowledgment"). RACK uses the notion of time instead of the conventional packet or sequence counting approaches for detecting losses. RACK deems a packet lost if some packet sent sufficiently later has been delivered. It does this by recording packet transmission times and inferring losses using cumulative acknowledgments or selective acknowledgment (SACK) TCP options.

In the last couple of years we have been observing several increasingly common loss and reordering patterns in the Internet:

1. Lost retransmissions. Traffic policers [POLICER16] and burst losses often cause retransmissions to be lost again, severely increasing TCP latency.

2. Tail drops. Structured request-response traffic turns more losses into tail drops. In such cases, TCP is application-limited, so it cannot send new data to probe losses and has to rely on retransmission timeouts (RTOs).

3. Reordering. Link layer protocols (e.g., 802.11 block ACK) or routers' internal load-balancing can deliver TCP packets out of order. The degree of such reordering is usually within the order of the path round trip time.

Despite TCP stacks (e.g. Linux) that implement many of the standard and proposed loss detection algorithms [RFC3517][RFC4653][RFC5827][RFC5681][RFC6675][RFC7765][FACK][THIN-STREAM][TLP], we've found that together they do not perform well. The main reason is that many of them are based on the classic rule of counting duplicate acknowledgments [RFC5681]. They can either detect loss quickly or accurately, but not both, especially when the sender is application-limited or under reordering that is unpredictable. And under these conditions none of them can detect lost retransmissions well.

Also, these algorithms, including RFCs, rarely address the interactions with other algorithms. For example, FACK may consider a

packet is lost while RFC3517 may not. Implementing N algorithms while dealing with N^2 interactions is a daunting task and error-prone.

The goal of RACK is to solve all the problems above by replacing many of the loss detection algorithms above with one simpler, and also more effective, algorithm.

2. Overview

The main idea behind RACK is that if a packet has been delivered out of order, then the packets sent chronologically before that were either lost or reordered. This concept is not fundamentally different from [RFC5681][RFC3517][FACK]. But the key innovation in RACK is to use a per-packet transmission timestamp and widely deployed SACK options to conduct time-based inferences instead of inferring losses with packet or sequence counting approaches.

Using a threshold for counting duplicate acknowledgments (i.e., dupthresh) is no longer reliable because of today's prevalent reordering patterns. A common type of reordering is that the last "runt" packet of a window's worth of packet bursts gets delivered first, then the rest arrive shortly after in order. To handle this effectively, a sender would need to constantly adjust the dupthresh to the burst size; but this would risk increasing the frequency of RTOs on real losses.

Today's prevalent lost retransmissions also cause problems with packet-counting approaches [RFC5681][RFC3517][FACK], since those approaches depend on reasoning in sequence number space. Retransmissions break the direct correspondence between ordering in sequence space and ordering in time. So when retransmissions are lost, sequence-based approaches are often unable to infer and quickly repair losses that can be deduced with time-based approaches.

Instead of counting packets, RACK uses the most recently delivered packet's transmission time to judge if some packets sent previous to that time have "expired" by passing a certain reordering settling window. On each ACK, RACK marks any already-expired packets lost, and for any packets that have not yet expired it waits until the

reordering window passes and then marks those lost as well. In either case, RACK can repair the loss without waiting for a (long) RTO. RACK can be applied to both fast recovery and timeout recovery, and can detect losses on both originally transmitted and retransmitted packets, making it a great all-weather recovery mechanism.

3. Requirements

The reader is expected to be familiar with the definitions given in the TCP congestion control [RFC5681] and selective acknowledgment

[RFC2018] RFCs. Familiarity with the conservative SACK-based recovery for TCP [RFC6675] is not expected but helps.

RACK has three requirements:

1. The connection MUST use selective acknowledgment (SACK) options [RFC2018].

2. For each packet sent, the sender MUST store its most recent transmission time with (at least) millisecond granularity. For round-trip times lower than a millisecond (e.g., intra-datacenter communications) microsecond granularity would significantly help the detection latency but is not required.

3. For each packet sent, the sender MUST store whether the packet has been retransmitted or not.

We assume that requirement 1 implies the sender keeps a SACK scoreboard, which is a data structure to store selective acknowledgment information on a per-connection basis. For the ease of explaining the algorithm, we use a pseudo-scoreboard that manages the data in sequence number ranges. But the specifics of the data structure are left to the implementor.

RACK does not need any change on the receiver.

4. Definitions of variables

A sender needs to store these new RACK variables:

"Packet.xmit_ts" is the time of the last transmission of a data packet, including any retransmissions, if any. The sender needs to record the transmission time for each packet sent and not yet acknowledged. The time MUST be stored at millisecond granularity or finer.

"RACK.xmit_ts" is the most recent Packet.xmit_ts among all the packets that were delivered (either cumulatively acknowledged or selectively acknowledged) on the connection.

"RACK.end_seq" is the ending TCP sequence number of the packet that was used to record the RACK.xmit_ts above.

"RACK.RTT" is the associated RTT measured when RACK.xmit_ts, above, was changed. It is the RTT of the most recently transmitted packet that has been delivered (either cumulatively acknowledged or selectively acknowledged) on the connection.

"RACK.reo_wnd" is a reordering window for the connection, computed in the unit of time used for recording packet transmission times. It is used to defer the moment at which RACK marks a packet lost.

"RACK.min_RTT" is the estimated minimum round-trip time (RTT) of the connection.

Note that the Packet.xmit_ts variable is per packet in flight. The RACK.xmit_ts, RACK.RTT, RACK.reo_wnd, and RACK.min_RTT variables are per connection.

5. Algorithm Details

5.1. Transmitting a data packet

Upon transmitting a new packet or retransmitting an old packet, record the time in Packet.xmit_ts. RACK does not care if the retransmission is triggered by an ACK, new application data, an RTO, or any other means.

5.2. Upon receiving an ACK

Step 1: Update RACK.min_RTT.

Use the RTT measurements obtained in [RFC6298] or [RFC7323] to update the estimated minimum RTT in RACK.min_RTT. The sender can track a simple global minimum of all RTT measurements from the connection, or a windowed min-filtered value of recent RTT measurements. This document does not specify an exact approach.

Step 2: Update RACK.reo_wnd.

To handle the prevalent small degree of reordering, RACK.reo_wnd serves as an allowance for settling time before marking a packet lost. By default it is 1 millisecond. We RECOMMEND implementing the reordering detection in [REORDER-DETECT][RFC4737] to dynamically adjust the reordering window. When the sender detects packet reordering RACK.reo_wnd MAY be changed to RACK.min_RTT/4. We discuss more about the reordering window in the next section.

Step 3: Advance RACK.xmit_ts and update RACK.RTT and RACK.end_seq

Given the information provided in an ACK, each packet cumulatively ACKed or SACKed is marked as delivered in the scoreboard. Among all the packets newly ACKed or SACKed in the connection, record the most recent Packet.xmit_ts in RACK.xmit_ts if it is ahead of RACK.xmit_ts. Ignore the packet if any of its TCP sequences has been retransmitted before and either of two condition is true:

1. The Timestamp Echo Reply field (TSecr) of the ACK's timestamp option [RFC7323], if available, indicates the ACK was not acknowledging the last retransmission of the packet.

2. The packet was last retransmitted less than RACK.min_rtt ago. While it is still possible the packet is spuriously retransmitted because of a recent RTT decrease, we believe that our experience suggests this is a reasonable heuristic.

If this ACK causes a change to RACK.xmit_ts then record the RTT and sequence implied by this ACK:

```
RACK.RTT = Now() - RACK.xmit_ts
RACK.end_seq = Packet.end_seq
```

Exit here and omit the following steps if RACK.xmit_ts has not changed.

Step 4: Detect losses.

For each packet that has not been fully SACKed, if RACK.xmit_ts is after Packet.xmit_ts + RACK.reo_wnd, then mark the packet (or its corresponding sequence range) lost in the scoreboard. The rationale is that if another packet that was sent later has been delivered, and the reordering window or "reordering settling time" has already passed, the packet was likely lost.

If a packet that was sent later has been delivered, but the reordering window has not passed, then it is not yet safe to deem the given packet lost. Using the basic algorithm above, the sender would wait for the next ACK to further advance RACK.xmit_ts; but this risks a timeout (RTO) if no more ACKs come back (e.g, due to losses or application limit). For timely loss detection, the sender MAY install a "reordering settling" timer set to fire at the earliest moment at which it is safe to conclude that some packet is lost. The earliest moment is the time it takes to expire the reordering window of the earliest unacked packet in flight.

This timer expiration value can be derived as follows. As a starting point, we consider that the reordering window has passed if the RACK packet was sent sufficiently after the packet in question, or a sufficient time has elapsed since the RACK packet was S/ACKed, or some combination of the two. More precisely, RACK marks a packet as lost if the reordering window for a packet has elapsed through the sum of:

1. delta in transmit time between a packet and the RACK packet

2. delta in time between the S/ACK of the RACK packet (RACK.ack_ts) and now

So we mark a packet as lost if:

```
RACK.xmit_ts > Packet.xmit_ts   AND
(RACK.xmit_ts - Packet.xmit_ts) + (now - RACK.ack_ts) > RACK.reo_wnd
```

If we solve this second condition for "now", the moment at which we can declare a packet lost, then we get:

now > Packet.xmit_ts + RACK.reo_wnd + (RACK.ack_ts - RACK.xmit_ts)

Then (RACK.ack_ts - RACK.xmit_ts) is just the RTT of the packet we used to set RACK.xmit_ts, so this reduces to:

now > Packet.xmit_ts + RACK.RTT + RACK.reo_wnd

The following pseudocode implements the algorithm above. When an ACK is received or the RACK timer expires, call RACK_detect_loss(). The algorithm includes an additional optimization to break timestamp ties by using the TCP sequence space. The optimization is particularly useful to detect losses in a timely manner with TCP Segmentation Offload, where multiple packets in one TSO blob have identical timestamps. It is also useful when the timestamp clock granularity is close to or longer than the actual round trip time.

```
RACK_detect_loss():
min_timeout = 0

For each packet, Packet, in the scoreboard:
    If Packet is already SACKed, ACKed,
       or marked lost and not yet retransmitted:
        Skip to the next packet

    If Packet.xmit_ts > RACK.xmit_ts:
        Skip to the next packet
    If Packet.xmit_ts == RACK.xmit_ts AND // Timestamp tie breaker
        Skip to the next packet

    timeout = Packet.xmit_ts + RACK.RTT + RACK.reo_wnd + 1
    If Now() >= timeout
        Mark Packet lost
    Else If (min_timeout == 0) or (timeout is before min_timeout):
        min_timeout = timeout

If min_timeout != 0
    Arm a timer to call RACK_detect_loss() after min_timeout
```


6. Analysis and Discussion

6.1. Advantages

The biggest advantage of RACK is that every data packet, whether it is an original data transmission or a retransmission, can be used to detect losses of the packets sent prior to it.

Example: tail drop. Consider a sender that transmits a window of three data packets (P1, P2, P3), and P1 and P3 are lost. Suppose the transmission of each packet is at least RACK.reo_wnd (1 millisecond by default) after the transmission of the previous packet. RACK will mark P1 as lost when the SACK of P2 is received, and this will trigger the retransmission of P1 as R1. When R1 is cumulatively acknowledged, RACK will mark P3 as lost and the sender will retransmit P3 as R3. This example illustrates how RACK is able to repair certain drops at the tail of a transaction without any timer. Notice that neither the conventional duplicate ACK threshold [RFC5681], nor [RFC6675], nor the Forward Acknowledgment [FACK] algorithm can detect such losses, because of the required packet or sequence count.

Example: lost retransmit. Consider a window of three data packets (P1, P2, P3) that are sent; P1 and P2 are dropped. Suppose the transmission of each packet is at least RACK.reo_wnd (1 millisecond by default) after the transmission of the previous packet. When P3 is SACKed, RACK will mark P1 and P2 lost and they will be retransmitted as R1 and R2. Suppose R1 is lost again (as a tail drop) but R2 is SACKed; RACK will mark R1 lost for retransmission again. Again, neither the conventional three duplicate ACK threshold approach, nor [RFC6675], nor the Forward Acknowledgment [FACK] algorithm can detect such losses. And such a lost retransmission is very common when TCP is being rate-limited, particularly by token bucket policers with large bucket depth and low rate limit. Retransmissions are often lost repeatedly because standard congestion control requires multiple round trips to reduce the rate below the policed rate.

Example: (small) degree of reordering. Consider a common reordering event: a window of packets are sent as (P1, P2, P3). P1 and P2 carry a full payload of MSS octets, but P3 has only a 1-octet payload due to application-limited behavior. Suppose the sender has detected reordering previously (e.g., by implementing the algorithm in [REORDER-DETECT]) and thus RACK.reo_wnd is min_RTT/4. Now P3 is reordered and delivered first, before P1 and P2. As long as P1 and P2 are delivered within min_RTT/4, RACK will not consider P1 and P2 lost. But if P1 and P2 are delivered outside the reordering window,

then RACK will still falsely mark P1 and P2 lost. We discuss how to reduce the false positives in the end of this section.

The examples above show that RACK is particularly useful when the sender is limited by the application, which is common for interactive, request/response traffic. Similarly, RACK still works when the sender is limited by the receive window, which is common for applications that use the receive window to throttle the sender.

For some implementations (e.g., Linux), RACK works quite efficiently with TCP Segmentation Offload (TSO). RACK always marks the entire TSO blob lost because the packets in the same TSO blob have the same transmission timestamp. By contrast, the counting based algorithms (e.g., [RFC3517][RFC5681]) may mark only a subset of packets in the TSO blob lost, forcing the stack to perform expensive fragmentation of the TSO blob, or to selectively tag individual packets lost in the scoreboard.

6.2. Disadvantages

RACK requires the sender to record the transmission time of each packet sent at a clock granularity of one millisecond or finer. TCP implementations that record this already for RTT estimation do not require any new per-packet state. But implementations that are not yet recording packet transmission times will need to add per-packet internal state (commonly either 4 or 8 octets per packet) to track transmission times. In contrast, the conventional approach requires one variable to track number of duplicate ACK threshold.

6.3. Adjusting the reordering window

RACK uses a reordering window of min_rtt / 4. It uses the minimum RTT to accommodate reordering introduced by packets traversing slightly different paths (e.g., router-based parallelism schemes) or out-of-order deliveries in the lower link layer (e.g., wireless links using link-layer retransmission). Alternatively, RACK can use the smoothed RTT used in RTT estimation [RFC6298]. However, smoothed RTT can be significantly inflated by orders of magnitude due to congestion and buffer-bloat, which would result in an overly conservative reordering window and slow loss detection. Furthermore, RACK uses a quarter of minimum RTT because Linux TCP uses the same factor in its implementation to delay Early Retransmit [RFC5827] to reduce spurious loss detections in the presence of reordering, and experience shows that this seems to work reasonably well.

One potential improvement is to further adapt the reordering window

by measuring the degree of reordering in time, instead of packet distances. But that requires storing the delivery timestamp of each

packet. Some scoreboard implementations currently merge SACKed packets together to support TSO (TCP Segmentation Offload) for faster scoreboard indexing. Supporting per-packet delivery timestamps is difficult in such implementations. However, we acknowledge that the current metric can be improved by further research.

6.4. Relationships with other loss recovery algorithms

The primary motivation of RACK is to ultimately provide a simple and general replacement for some of the standard loss recovery algorithms [RFC5681][RFC6675][RFC5827][RFC4653] and nonstandard ones [FACK][THIN-STREAM]. While RACK can be a supplemental loss detection on top of these algorithms, this is not necessary, because the RACK implicitly subsumes most of them.

[RFC5827][RFC4653][THIN-STREAM] dynamically adjusts the duplicate ACK threshold based on the current or previous flight sizes. RACK takes a different approach, by using only one ACK event and a reordering window. RACK can be seen as an extended Early Retransmit [RFC5827] without a FlightSize limit but with an additional reordering window. [FACK] considers an original packet to be lost when its sequence range is sufficiently far below the highest SACKed sequence. In some sense RACK can be seen as a generalized form of FACK that operates in time space instead of sequence space, enabling it to better handle reordering, application-limited traffic, and lost retransmissions.

Nevertheless RACK is still an experimental algorithm. Since the oldest loss detection algorithm, the 3 duplicate ACK threshold [RFC5681], has been standardized and widely deployed, we RECOMMEND TCP implementations use both RACK and the algorithm specified in Section 3.2 in [RFC5681] for compatibility.

RACK is compatible with and does not interfere with the the standard RTO [RFC6298], RTO-restart [RFC7765], F-RTO [RFC5682] and Eifel algorithms [RFC3522]. This is because RACK only detects loss by using ACK events. It neither changes the timer calculation nor detects spurious timeouts.

Furthermore, RACK naturally works well with Tail Loss Probe [TLP]

because a tail loss probe solicit seither an ACK or SACK, which can be used by RACK to detect more losses. RACK can be used to relax TLP's requirement for using FACK and retransmitting the the highest-sequenced packet, because RACK is agnostic to packet sequence numbers, and uses transmission time instead. Thus TLP can be modified to retransmit the first unacknowledged packet, which can improve application latency.

6.5. Interaction with congestion control

RACK intentionally decouples loss detection from congestion control. RACK only detects losses; it does not modify the congestion control algorithm [RFC5681][RFC6937]. However, RACK may detect losses earlier or later than the conventional duplicate ACK threshold approach does. A packet marked lost by RACK SHOULD NOT be retransmitted until congestion control deems this appropriate (e.g. using [RFC6937]).

RACK is applicable for both fast recovery and recovery after a retransmission timeout (RTO) in [RFC5681]. The distinction between fast recovery or RTO recovery is not necessary because RACK is purely based on the transmission time order of packets. When a packet retransmitted by RTO is acknowledged, RACK will mark any unacked packet sent sufficiently prior to the RTO as lost, because at least one RTT has elapsed since these packets were sent.

6.6. RACK for other transport protocols

RACK can be implemented in other transport protocols. The algorithm can skip step 3 and simplify if the protocol can support unique transmission or packet identifier (e.g. TCP echo options). For example, the QUIC protocol implements RACK [QUIC-LR] .

7. Security Considerations

RACK does not change the risk profile for TCP.

An interesting scenario is ACK-splitting attacks [SCWA99]: for an MSS-size packet sent, the receiver or the attacker might send MSS ACKs that SACK or acknowledge one additional byte per ACK. This

would not fool RACK. RACK.xmit_ts would not advance because all the sequences of the packet are transmitted at the same time (carry the same transmission timestamp). In other words, SACKing only one byte of a packet or SACKing the packet in entirety have the same effect on RACK.

8. IANA Considerations

This document makes no request of IANA.

Note to RFC Editor: this section may be removed on publication as an RFC.

9. Acknowledgments

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10. References

10.1. Normative References

[RFC793] Postel, J., "Transmission Control Protocol", September 1981.

[RFC2018] Mathis, M. and J. Mahdavi, "TCP Selective Acknowledgment Options", RFC 2018, October 1996.

[RFC6937] Mathis, M., Dukkipati, N., and Y. Cheng, "Proportional Rate Reduction for TCP", May 2013.

[RFC4737] Morton, A., Ciavattone, L., Ramachandran, G., Shalunov, S., and J. Perser, "Packet Reordering Metrics", RFC 4737, November 2006.

[RFC6675] Blanton, E., Allman, M., Wang, L., Jarvinen, I., Kojo, M., and Y. Nishida, "A Conservative Loss Recovery Algorithm Based on Selective Acknowledgment (SACK) for TCP", RFC 6675, August 2012.

[RFC6298] Paxson, V., Allman, M., Chu, J., and M. Sargent, "Computing TCP's Retransmission Timer", RFC 6298, June 2011.

[RFC5827] Allman, M., Ayesta, U., Wang, L., Blanton, J., and P. Hurtig, "Early Retransmit for TCP and Stream Control Transmission Protocol (SCTP)", RFC 5827, April 2010.

[RFC5682] Sarolahti, P., Kojo, M., Yamamoto, K., and M. Hata, "Forward RTO-Recovery (F-RTO): An Algorithm for Detecting Spurious Retransmission Timeouts with TCP", RFC 5682, September 2009.

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", RFC 2119, March 1997.

[RFC5681] Allman, M., Paxson, V., and E. Blanton, "TCP Congestion Control", RFC 5681, September 2009.

[RFC2883] Floyd, S., Mahdavi, J., Mathis, M., and M. Podolsky, "An Extension to the Selective Acknowledgement (SACK) Option for TCP", RFC 2883, July 2000.

[RFC7323] Borman, D., Braden, B., Jacobson, V., and R. Scheffenegger, "TCP Extensions for High Performance", September 2014.

10.2. Informative References

[FACK] Mathis, M. and M. Jamshid, "Forward acknowledgement: refining TCP congestion control", ACM SIGCOMM Computer Communication Review, Volume 26, Issue 4, Oct. 1996. , 1996.

[TLP] Dukkipati, N., Cardwell, N., Cheng, Y., and M. Mathis, "Tail Loss Probe (TLP): An Algorithm for Fast Recovery of Tail Drops", draft-dukkipati-tcpm-tcp-loss-probe-01 (work

in progress), August 2013.

[RFC7765] Hurtig, P., Brunstrom, A., Petlund, A., and M. Welzl, "TCP and SCTP RTO Restart", February 2016.

[REORDER-DETECT] Zimmermann, A., Schulte, L., Wolff, C., and A. Hannemann, "Detection and Quantification of Packet Reordering with TCP", draft-zimmermann-tcpm-reordering-detection-02 (work in progress), November 2014.

[QUIC-LR] Iyengar, J. and I. Swett, "QUIC Loss Recovery And Congestion Control", draft-tsvwg-quic-loss-recovery-01 (work in progress), June 2016.

[THIN-STREAM] Petlund, A., Evensen, K., Griwodz, C., and P. Halvorsen, "TCP enhancements for interactive thin-stream applications", NOSSDAV , 2008.

[SCWA99] Savage, S., Cardwell, N., Wetherall, D., and T. Anderson, "TCP Congestion Control With a Misbehaving Receiver", ACM Computer Communication Review, 29(5) , 1999.

[POLICER16] Flach, T., Papageorge, P., Terzis, A., Pedrosa, L., Cheng, Y., Karim, T., Katz-Bassett, E., and R. Govindan, "An Analysis of Traffic Policing in the Web", ACM SIGCOMM , 2016.

Authors' Addresses

Yuchung Cheng
Google, Inc
1600 Amphitheater Parkway
Mountain View, California 94043
USA

Email: ycheng@google.com

Neal Cardwell
Google, Inc
76 Ninth Avenue
New York, NY 10011
USA

Email: ncardwell@google.com