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## **QUIC Loss Detection and Congestion Control draft-ietf-quic-recovery-08**

### Abstract

This document describes loss detection and congestion control mechanisms for QUIC.

### Note to Readers

Discussion of this draft takes place on the QUIC working group mailing list ([quic@ietf.org](mailto:quic@ietf.org)), which is archived at [https://mailarchive.ietf.org/arch/search/?email\\_list=quic](https://mailarchive.ietf.org/arch/search/?email_list=quic) [1].

Working Group information can be found at <https://github.com/quicwg> [2]; source code and issues list for this draft can be found at <https://github.com/quicwg/base-drafts/labels/-recovery> [3].

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## 1. Introduction

QUIC is a new multiplexed and secure transport atop UDP. QUIC builds on decades of transport and security experience, and implements mechanisms that make it attractive as a modern general-purpose transport. The QUIC protocol is described in [[QUIC-TRANSPORT](#)].

QUIC implements the spirit of known TCP loss recovery mechanisms, described in RFCs, various Internet-drafts, and also those prevalent in the Linux TCP implementation. This document describes QUIC congestion control and loss recovery, and where applicable, attributes the TCP equivalent in RFCs, Internet-drafts, academic papers, and/or TCP implementations.

### 1.1. Notational Conventions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [BCP 14](#) [[RFC2119](#)] [[RFC8174](#)] when, and only when, they appear in all capitals, as shown here.



## **2. Design of the QUIC Transmission Machinery**

All transmissions in QUIC are sent with a packet-level header, which includes a packet sequence number (referred to below as a packet number). These packet numbers never repeat in the lifetime of a connection, and are monotonically increasing, which prevents ambiguity. This fundamental design decision obviates the need for disambiguating between transmissions and retransmissions and eliminates significant complexity from QUIC's interpretation of TCP loss detection mechanisms.

Every packet may contain several frames. We outline the frames that are important to the loss detection and congestion control machinery below.

- o Retransmittable frames are frames requiring reliable delivery. The most common are STREAM frames, which typically contain application data.
- o Crypto handshake data is sent on stream 0, and uses the reliability machinery of QUIC underneath.
- o ACK frames contain acknowledgment information. ACK frames contain one or more ranges of acknowledged packets.

### **2.1. Relevant Differences Between QUIC and TCP**

Readers familiar with TCP's loss detection and congestion control will find algorithms here that parallel well-known TCP ones. Protocol differences between QUIC and TCP however contribute to algorithmic differences. We briefly describe these protocol differences below.

#### **2.1.1. Monotonically Increasing Packet Numbers**

TCP conflates transmission sequence number at the sender with delivery sequence number at the receiver, which results in retransmissions of the same data carrying the same sequence number, and consequently to problems caused by "retransmission ambiguity". QUIC separates the two: QUIC uses a packet number for transmissions, and any data that is to be delivered to the receiving application(s) is sent in one or more streams, with delivery order determined by stream offsets encoded within STREAM frames.

QUIC's packet number is strictly increasing, and directly encodes transmission order. A higher QUIC packet number signifies that the packet was sent later, and a lower QUIC packet number signifies that the packet was sent earlier. When a packet containing frames is



deemed lost, QUIC rebundles necessary frames in a new packet with a new packet number, removing ambiguity about which packet is acknowledged when an ACK is received. Consequently, more accurate RTT measurements can be made, spurious retransmissions are trivially detected, and mechanisms such as Fast Retransmit can be applied universally, based only on packet number.

This design point significantly simplifies loss detection mechanisms for QUIC. Most TCP mechanisms implicitly attempt to infer transmission ordering based on TCP sequence numbers - a non-trivial task, especially when TCP timestamps are not available.

#### **2.1.2. No Reneging**

QUIC ACKs contain information that is similar to TCP SACK, but QUIC does not allow any acked packet to be reneged, greatly simplifying implementations on both sides and reducing memory pressure on the sender.

#### **2.1.3. More ACK Ranges**

QUIC supports many ACK ranges, opposed to TCP's 3 SACK ranges. In high loss environments, this speeds recovery, reduces spurious retransmits, and ensures forward progress without relying on timeouts.

#### **2.1.4. Explicit Correction For Delayed Acks**

QUIC ACKs explicitly encode the delay incurred at the receiver between when a packet is received and when the corresponding ACK is sent. This allows the receiver of the ACK to adjust for receiver delays, specifically the delayed ack timer, when estimating the path RTT. This mechanism also allows a receiver to measure and report the delay from when a packet was received by the OS kernel, which is useful in receivers which may incur delays such as context-switch latency before a userspace QUIC receiver processes a received packet.

### **3. Loss Detection**

QUIC senders use both ack information and timeouts to detect lost packets, and this section provides a description of these algorithms. Estimating the network round-trip time (RTT) is critical to these algorithms and is described first.





### **3.1. Computing the RTT estimate**

RTT is calculated when an ACK frame arrives by computing the difference between the current time and the time the largest newly acked packet was sent. If no packets are newly acknowledged, RTT cannot be calculated. When RTT is calculated, the ack delay field from the ACK frame SHOULD be subtracted from the RTT as long as the result is larger than the Min RTT. If the result is smaller than the min\_rtt, the RTT should be used, but the ack delay field should be ignored.

Like TCP, QUIC calculates both smoothed RTT and RTT variance as specified in [\[RFC6298\]](#).

Min RTT is the minimum RTT measured over the connection, prior to adjusting by ack delay. Ignoring ack delay for min RTT prevents intentional or unintentional underestimation of min RTT, which in turn prevents underestimating smoothed RTT.

### **3.2. Ack-based Detection**

Ack-based loss detection implements the spirit of TCP's Fast Retransmit [\[RFC5681\]](#), Early Retransmit [\[RFC5827\]](#), FACK, and SACK loss recovery [\[RFC6675\]](#). This section provides an overview of how these algorithms are implemented in QUIC.

(TODO: Define unacknowledged packet, ackable packet, outstanding bytes.)

#### **3.2.1. Fast Retransmit**

An unacknowledged packet is marked as lost when an acknowledgment is received for a packet that was sent a threshold number of packets (kReorderingThreshold) after the unacknowledged packet. Receipt of the ack indicates that a later packet was received, while kReorderingThreshold provides some tolerance for reordering of packets in the network.

The RECOMMENDED initial value for kReorderingThreshold is 3.

We derive this default from recommendations for TCP loss recovery [\[RFC5681\]](#) [\[RFC6675\]](#). It is possible for networks to exhibit higher degrees of reordering, causing a sender to detect spurious losses. Detecting spurious losses leads to unnecessary retransmissions and may result in degraded performance due to the actions of the congestion controller upon detecting loss. Implementers MAY use algorithms developed for TCP, such as TCP-NCR [\[RFC4653\]](#), to improve QUIC's reordering resilience, though care should be taken to map TCP



specifics to QUIC correctly. Similarly, using time-based loss detection to deal with reordering, such as in PR-TCP, should be more readily usable in QUIC. Making QUIC deal with such networks is important open research, and implementers are encouraged to explore this space.

### **3.2.2. Early Retransmit**

Unacknowledged packets close to the tail may have fewer than `kReorderingThreshold` number of ackable packets sent after them. Loss of such packets cannot be detected via Fast Retransmit. To enable ack-based loss detection of such packets, receipt of an acknowledgment for the last outstanding ackable packet triggers the Early Retransmit process, as follows.

If there are unacknowledged ackable packets still pending, they ought to be marked as lost. To compensate for the reduced reordering resilience, the sender SHOULD set an alarm for a small period of time. If the unacknowledged ackable packets are not acknowledged during this time, then these packets MUST be marked as lost.

An endpoint SHOULD set the alarm such that a packet is marked as lost no earlier than  $1.25 * \max(\text{SRTT}, \text{latest\_RTT})$  since when it was sent.

Using  $\max(\text{SRTT}, \text{latest\_RTT})$  protects from the two following cases:

- o the latest RTT sample is lower than the SRTT, perhaps due to reordering where packet whose ack triggered the Early Retransmit process encountered a shorter path;
- o the latest RTT sample is higher than the SRTT, perhaps due to a sustained increase in the actual RTT, but the smoothed SRTT has not yet caught up.

The 1.25 multiplier increases reordering resilience. Implementers MAY experiment with using other multipliers, bearing in mind that a lower multiplier reduces reordering resilience and increases spurious retransmissions, and a higher multiplier increases loss recovery delay.

This mechanism is based on Early Retransmit for TCP [[RFC5827](#)]. However, [[RFC5827](#)] does not include the alarm described above. Early Retransmit is prone to spurious retransmissions due to its reduced reordering resilience without the alarm. This observation led Linux TCP implementers to implement an alarm for TCP as well, and this document incorporates this advancement.



### 3.3. Timer-based Detection

Timer-based loss detection implements the spirit of TCP's Tail Loss Probe and Retransmission Timeout mechanisms.

#### 3.3.1. Tail Loss Probe

The algorithm described in this section is an adaptation of the Tail Loss Probe algorithm proposed for TCP [[TLP](#)].

A packet sent at the tail is particularly vulnerable to slow loss detection, since acks of subsequent packets are needed to trigger ack-based detection. To ameliorate this weakness of tail packets, the sender schedules an alarm when the last ackable packet before quiescence is transmitted. When this alarm fires, a Tail Loss Probe (TLP) packet is sent to evoke an acknowledgement from the receiver.

The alarm duration, or Probe Timeout (PTO), is set based on the following conditions:

- o PTO SHOULD be scheduled for  $\max(1.5 \cdot \text{SRTT} + \text{MaxAckDelay}, 10\text{ms})$
- o If RTO ([Section 3.3.2](#)) is earlier, schedule a TLP alarm in its place. That is, PTO SHOULD be scheduled for  $\min(\text{RTO}, \text{PTO})$ .

MaxAckDelay is the maximum ack delay supplied in an incoming ACK frame. MaxAckDelay excludes ack delays that aren't included in an RTT sample because they're too large and excludes those which reference an ack-only packet.

QUIC diverges from TCP by calculating MaxAckDelay dynamically, instead of assuming a constant delayed ack timeout for all connections. QUIC includes this in all probe timeouts, because it assume the ack delay may come into play, regardless of the number of packets outstanding. TCP's TLP assumes if at least 2 packets are outstanding, acks will not be delayed.

A PTO value of at least  $1.5 \cdot \text{SRTT}$  ensures that the ACK is overdue. The 1.5 is based on [[LOSS-PROBE](#)], but implementations MAY experiment with other constants.

To reduce latency, it is RECOMMENDED that the sender set and allow the TLP alarm to fire twice before setting an RTO alarm. In other words, when the TLP alarm fires the first time, a TLP packet is sent, and it is RECOMMENDED that the TLP alarm be scheduled for a second time. When the TLP alarm fires the second time, a second TLP packet is sent, and an RTO alarm SHOULD be scheduled [Section 3.3.2](#).



A TLP packet SHOULD carry new data when possible. If new data is unavailable or new data cannot be sent due to flow control, a TLP packet MAY retransmit unacknowledged data to potentially reduce recovery time. Since a TLP alarm is used to send a probe into the network prior to establishing any packet loss, prior unacknowledged packets SHOULD NOT be marked as lost when a TLP alarm fires.

A TLP packet MUST NOT be blocked by the sender's congestion controller. The sender MUST however count these bytes as additional bytes in flight, since a TLP adds network load without establishing packet loss.

A sender may not know that a packet being sent is a tail packet. Consequently, a sender may have to arm or adjust the TLP alarm on every sent ackable packet.

### **3.3.2. Retransmission Timeout**

A Retransmission Timeout (RTO) alarm is the final backstop for loss detection. The algorithm used in QUIC is based on the RTO algorithm for TCP [[RFC5681](#)] and is additionally resilient to spurious RTO events [[RFC5682](#)].

When the last TLP packet is sent, an alarm is scheduled for the RTO period. When this alarm fires, the sender sends two packets, to evoke acknowledgements from the receiver, and restarts the RTO alarm.

Similar to TCP [[RFC6298](#)], the RTO period is set based on the following conditions:

- o When the final TLP packet is sent, the RTO period is set to  $\max(\text{SRTT} + 4 \cdot \text{RTTVAR} + \text{MaxAckDelay}, \text{minRTO})$
- o When an RTO alarm fires, the RTO period is doubled.

The sender typically has incurred a high latency penalty by the time an RTO alarm fires, and this penalty increases exponentially in subsequent consecutive RTO events. Sending a single packet on an RTO event therefore makes the connection very sensitive to single packet loss. Sending two packets instead of one significantly increases resilience to packet drop in both directions, thus reducing the probability of consecutive RTO events.

QUIC's RTO algorithm differs from TCP in that the firing of an RTO alarm is not considered a strong enough signal of packet loss, so does not result in an immediate change to congestion window or recovery state. An RTO alarm fires only when there's a prolonged





period of network silence, which could be caused by a change in the underlying network RTT.

QUIC also diverges from TCP by including MaxAckDelay in the RTO period. QUIC is able to explicitly model delay at the receiver via the ack delay field in the ACK frame. Since QUIC corrects for this delay in its SRTT and RTTVAR computations, it is necessary to add this delay explicitly in the TLP and RTO computation.

When an acknowledgment is received for a packet sent on an RTO event, any unacknowledged packets with lower packet numbers than those acknowledged MUST be marked as lost.

A packet sent when an RTO alarm fires MAY carry new data if available or unacknowledged data to potentially reduce recovery time. Since this packet is sent as a probe into the network prior to establishing any packet loss, prior unacknowledged packets SHOULD NOT be marked as lost.

A packet sent on an RTO alarm MUST NOT be blocked by the sender's congestion controller. A sender MUST however count these bytes as additional bytes in flight, since this packet adds network load without establishing packet loss.

### **3.3.3. Handshake Timeout**

Handshake packets, which contain STREAM frames for stream 0, are critical to QUIC transport and crypto negotiation, so a separate alarm is used for them.

The initial handshake timeout SHOULD be set to twice the initial RTT.

At the beginning, there are no prior RTT samples within a connection. Resumed connections over the same network SHOULD use the previous connection's final smoothed RTT value as the resumed connection's initial RTT.

If no previous RTT is available, or if the network changes, the initial RTT SHOULD be set to 100ms.

When the first handshake packet is sent, the sender SHOULD set an alarm for the handshake timeout period.

When the alarm fires, the sender MUST retransmit all unacknowledged handshake data. On each consecutive firing of the handshake alarm, the sender SHOULD double the handshake timeout and set an alarm for this period.



When an acknowledgement is received for a handshake packet, the new RTT is computed and the alarm SHOULD be set for twice the newly computed smoothed RTT.

Handshake data may be cancelled by handshake state transitions. In particular, all non-protected data SHOULD no longer be transmitted once packet protection is available.

(TODO: Work this section some more. Add text on client vs. server, and on stateless retry.)

### **3.4. Pseudocode**

#### **3.4.1. Constants of interest**

Constants used in loss recovery are based on a combination of RFCs, papers, and common practice. Some may need to be changed or negotiated in order to better suit a variety of environments.

kMaxTLPs (default 2): Maximum number of tail loss probes before an RTO fires.

kReorderingThreshold (default 3): Maximum reordering in packet number space before FACK style loss detection considers a packet lost.

kTimeReorderingFraction (default 1/8): Maximum reordering in time space before time based loss detection considers a packet lost. In fraction of an RTT.

kUsingTimeLossDetection (default false): Whether time based loss detection is in use. If false, uses FACK style loss detection.

kMinTLPTimeout (default 10ms): Minimum time in the future a tail loss probe alarm may be set for.

kMinRTOTimeout (default 200ms): Minimum time in the future an RTO alarm may be set for.

kDelayedAckTimeout (default 25ms): The length of the peer's delayed ack timer.

kDefaultInitialRtt (default 100ms): The default RTT used before an RTT sample is taken.



### **3.4.2. Variables of interest**

Variables required to implement the congestion control mechanisms are described in this section.

`loss_detection_alarm`: Multi-modal alarm used for loss detection.

`handshake_count`: The number of times the handshake packets have been retransmitted without receiving an ack.

`tlp_count`: The number of times a tail loss probe has been sent without receiving an ack.

`rto_count`: The number of times an rto has been sent without receiving an ack.

`largest_sent_before_rto`: The last packet number sent prior to the first retransmission timeout.

`time_of_last_sent_packet`: The time the most recent packet was sent.

`largest_sent_packet`: The packet number of the most recently sent packet.

`largest_acked_packet`: The largest packet number acknowledged in an ACK frame.

`latest_rtt`: The most recent RTT measurement made when receiving an ack for a previously unacked packet.

`smoothed_rtt`: The smoothed RTT of the connection, computed as described in [[RFC6298](#)]

`rttvar`: The RTT variance, computed as described in [[RFC6298](#)]

`min_rtt`: The minimum RTT seen in the connection, ignoring ack delay.

`max_ack_delay`: The maximum ack delay in an incoming ACK frame for this connection. Excludes ack delays for ack only packets and those that create an RTT sample less than `min_rtt`.

`reordering_threshold`: The largest delta between the largest acked retransmittable packet and a packet containing retransmittable frames before it's declared lost.

`time_reordering_fraction`: The reordering window as a fraction of `max(smoothed_rtt, latest_rtt)`.



**loss\_time:** The time at which the next packet will be considered lost based on early transmit or exceeding the reordering window in time.

**sent\_packets:** An association of packet numbers to information about them, including a number field indicating the packet number, a time field indicating the time a packet was sent, a boolean indicating whether the packet is ack only, and a bytes field indicating the packet's size. `sent_packets` is ordered by packet number, and packets remain in `sent_packets` until acknowledged or lost.

#### **3.4.3. Initialization**

At the beginning of the connection, initialize the loss detection variables as follows:

```
loss_detection_alarm.reset()
handshake_count = 0
tlp_count = 0
rto_count = 0
if (kUsingTimeLossDetection)
    reordering_threshold = infinite
    time_reordering_fraction = kTimeReorderingFraction
else:
    reordering_threshold = kReorderingThreshold
    time_reordering_fraction = infinite
loss_time = 0
smoothed_rtt = 0
rttvar = 0
min_rtt = 0
max_ack_delay = 0
largest_sent_before_rto = 0
time_of_last_sent_packet = 0
largest_sent_packet = 0
```

#### **3.4.4. On Sending a Packet**

After any packet is sent, be it a new transmission or a rebundled transmission, the following `OnPacketSent` function is called. The parameters to `OnPacketSent` are as follows:

- o `packet_number`: The packet number of the sent packet.
- o `is_ack_only`: A boolean that indicates whether a packet only contains an ACK frame. If true, it is still expected an ack will be received for this packet, but it is not congestion controlled.





- o `sent_bytes`: The number of bytes sent in the packet, not including UDP or IP overhead, but including QUIC framing overhead.

Pseudocode for `OnPacketSent` follows:

```
OnPacketSent(packet_number, is_ack_only, sent_bytes):  
    time_of_last_sent_packet = now  
    largest_sent_packet = packet_number  
    sent_packets[packet_number].packet_number = packet_number  
    sent_packets[packet_number].time = now  
    sent_packets[packet_number].ack_only = is_ack_only  
    if !is_ack_only:  
        OnPacketSentCC(sent_bytes)  
        sent_packets[packet_number].bytes = sent_bytes  
        SetLossDetectionAlarm()
```

#### **3.4.5. On Ack Receipt**

When an ack is received, it may acknowledge 0 or more packets.

Pseudocode for `OnAckReceived` and `UpdateRtt` follow:



```
OnAckReceived(ack):
    largest_acked_packet = ack.largest_acked
    // If the largest acked is newly acked, update the RTT.
    if (sent_packets[ack.largest_acked]):
        latest_rtt = now - sent_packets[ack.largest_acked].time
        UpdateRtt(latest_rtt, ack.ack_delay)
    // Find all newly acked packets.
    for acked_packet in DetermineNewlyAckedPackets():
        OnPacketAked(acked_packet.packet_number)

    DetectLostPackets(ack.largest_acked_packet)
    SetLossDetectionAlarm()

UpdateRtt(latest_rtt, ack_delay):
    // min_rtt ignores ack delay.
    min_rtt = min(min_rtt, latest_rtt)
    // Adjust for ack delay if it's plausible.
    if (latest_rtt - min_rtt > ack_delay):
        latest_rtt -= ack_delay
        // Only save into max ack delay if it's used
        // for rtt calculation and is not ack only.
        if (!sent_packets[ack.largest_acked].ack_only)
            max_ack_delay = max(max_ack_delay, ack_delay)
    // Based on {{RFC6298}}.
    if (smoothed_rtt == 0):
        smoothed_rtt = latest_rtt
        rttvar = latest_rtt / 2
    else:
        rttvar = 3/4 * rttvar + 1/4 * abs(smoothed_rtt - latest_rtt)
        smoothed_rtt = 7/8 * smoothed_rtt + 1/8 * latest_rtt
```

#### **3.4.6. On Packet Acknowledgment**

When a packet is acked for the first time, the following OnPacketAked function is called. Note that a single ACK frame may newly acknowledge several packets. OnPacketAked must be called once for each of these newly acked packets.

OnPacketAked takes one parameter, acked\_packet, which is the packet number of the newly acked packet, and returns a list of packet numbers that are detected as lost.

If this is the first acknowledgement following RT0, check if the smallest newly acknowledged packet is one sent by the RT0, and if so, inform congestion control of a verified RT0, similar to F-RT0 [[RFC5682](#)]



Pseudocode for OnPacketAacked follows:

```
OnPacketAacked(acked_packet_number):
  OnPacketAackedCC(acked_packet_number)
  // If a packet sent prior to RTO was acked, then the RTO
  // was spurious. Otherwise, inform congestion control.
  if (rto_count > 0 &&
      acked_packet_number > largest_sent_before_rto)
    OnRetransmissionTimeoutVerified()
  handshake_count = 0
  tlp_count = 0
  rto_count = 0
  sent_packets.remove(acked_packet_number)
```

#### **3.4.7. Setting the Loss Detection Alarm**

QUIC loss detection uses a single alarm for all timer-based loss detection. The duration of the alarm is based on the alarm's mode, which is set in the packet and timer events further below. The function SetLossDetectionAlarm defined below shows how the single timer is set based on the alarm mode.

##### **3.4.7.1. Handshake Alarm**

When a connection has unacknowledged handshake data, the handshake alarm is set and when it expires, all unacknowledged handshake data is retransmitted.

When stateless rejects are in use, the connection is considered immediately closed once a reject is sent, so no timer is set to retransmit the reject.

Version negotiation packets are always stateless, and MUST be sent once per handshake packet that uses an unsupported QUIC version, and MAY be sent in response to 0RTT packets.

##### **3.4.7.2. Tail Loss Probe and Retransmission Alarm**

Tail loss probes [[LOSS-PROBE](#)] and retransmission timeouts [[RFC6298](#)] are an alarm based mechanism to recover from cases when there are outstanding retransmittable packets, but an acknowledgement has not been received in a timely manner.

The TLP and RTO timers are armed when there is not unacknowledged handshake data. The TLP alarm is set until the max number of TLP packets have been sent, and then the RTO timer is set.



#### **3.4.7.3. Early Retransmit Alarm**

Early retransmit [[RFC5827](#)] is implemented with a 1/4 RTT timer. It is part of QUIC's time based loss detection, but is always enabled, even when only packet reordering loss detection is enabled.

#### **3.4.7.4. Pseudocode**

Pseudocode for SetLossDetectionAlarm follows:

```
SetLossDetectionAlarm():
    // Don't arm the alarm if there are no packets with
    // retransmittable data in flight.
    if (num_retransmittable_packets_outstanding == 0):
        loss_detection_alarm.cancel()
        return

    if (handshake_packets_are_outstanding):
        // Handshake retransmission alarm.
        if (smoothed_rtt == 0):
            alarm_duration = 2 * kDefaultInitialRtt
        else:
            alarm_duration = 2 * smoothed_rtt
            alarm_duration = max(alarm_duration, kMinTLPTimeout)
            alarm_duration = alarm_duration * (2 ^ handshake_count)
    else if (loss_time != 0):
        // Early retransmit timer or time loss detection.
        alarm_duration = loss_time - time_of_last_sent_packet
    else if (tlp_count < kMaxTLPS):
        // Tail Loss Probe
        alarm_duration = max(1.5 * smoothed_rtt + max_ack_delay,
                            kMinTLPTimeout)
    else:
        // RTO alarm
        alarm_duration = smoothed_rtt + 4 * rttvar
        alarm_duration = max(alarm_duration, kMinRTOTimeout)
        alarm_duration = alarm_duration * (2 ^ rto_count)

    loss_detection_alarm.set(time_of_last_sent_packet
                            + alarm_duration)
```

#### **3.4.8. On Alarm Firing**

QUIC uses one loss recovery alarm, which when set, can be in one of several modes. When the alarm fires, the mode determines the action to be performed.

Pseudocode for OnLossDetectionAlarm follows:





```
OnLossDetectionAlarm():
  if (handshake packets are outstanding):
    // Handshake retransmission alarm.
    RetransmitAllHandshakePackets()
    handshake_count++
  else if (loss_time != 0):
    // Early retransmit or Time Loss Detection
    DetectLostPackets(largest_acked_packet)
  else if (tlp_count < kMaxTLPs):
    // Tail Loss Probe.
    SendOnePacket()
    tlp_count++
  else:
    // RTO.
    if (rto_count == 0)
      largest_sent_before_rto = largest_sent_packet
    SendTwoPackets()
    rto_count++

  SetLossDetectionAlarm()
```

#### **[3.4.9.](#) Detecting Lost Packets**

Packets in QUIC are only considered lost once a larger packet number is acknowledged. `DetectLostPackets` is called every time an ack is received. If the loss detection alarm fires and the `loss_time` is set, the previous largest acked packet is supplied.

##### **[3.4.9.1.](#) Handshake Packets**

The receiver MUST close the connection with an error of type `OPTIMISTIC_ACK` when receiving an unprotected packet that acks protected packets. The receiver MUST trust protected acks for unprotected packets, however. Aside from this, loss detection for handshake packets when an ack is processed is identical to other packets.

##### **[3.4.9.2.](#) Pseudocode**

`DetectLostPackets` takes one parameter, `acked`, which is the largest acked packet.

Pseudocode for `DetectLostPackets` follows:



```
DetectLostPackets(largest_acked):
    loss_time = 0
    lost_packets = {}
    delay_until_lost = infinite
    if (kUsingTimeLossDetection):
        delay_until_lost =
            (1 + time_reordering_fraction) * max(latest_rtt, smoothed_rtt)
    else if (largest_acked.packet_number == largest_sent_packet):
        // Early retransmit alarm.
        delay_until_lost = 5/4 * max(latest_rtt, smoothed_rtt)
    foreach (unacked < largest_acked.packet_number):
        time_since_sent = now() - unacked.time_sent
        delta = largest_acked.packet_number - unacked.packet_number
        if (time_since_sent > delay_until_lost):
            lost_packets.insert(unacked)
        else if (delta > reordering_threshold)
            lost_packets.insert(unacked)
        else if (loss_time == 0 && delay_until_lost != infinite):
            loss_time = now() + delay_until_lost - time_since_sent

    // Inform the congestion controller of lost packets and
    // lets it decide whether to retransmit immediately.
    if (!lost_packets.empty())
        OnPacketsLost(lost_packets)
        foreach (packet in lost_packets)
            sent_packets.remove(packet.packet_number)
```

### 3.5. Discussion

The majority of constants were derived from best common practices among widely deployed TCP implementations on the internet. Exceptions follow.

A shorter delayed ack time of 25ms was chosen because longer delayed acks can delay loss recovery and for the small number of connections where less than packet per 25ms is delivered, acking every packet is beneficial to congestion control and loss recovery.

The default initial RTT of 100ms was chosen because it is slightly higher than both the median and mean min\_rtt typically observed on the public internet.

## 4. Congestion Control

QUIC's congestion control is based on TCP NewReno[RFC6582] congestion control to determine the congestion window and pacing rate. QUIC congestion control is specified in bytes due to finer control and the ease of appropriate byte counting[RFC3465].



#### **4.1. Slow Start**

QUIC begins every connection in slow start and exits slow start upon loss. QUIC re-enters slow start anytime the congestion window is less than `sssthresh`, which typically only occurs after an RTO. While in slow start, QUIC increases the congestion window by the number of acknowledged bytes when each ack is processed.

#### **4.2. Congestion Avoidance**

Slow start exits to congestion avoidance. Congestion avoidance in NewReno uses an additive increase multiplicative decrease (AIMD) approach that increases the congestion window by one MSS of bytes per congestion window acknowledged. When a loss is detected, NewReno halves the congestion window and sets the slow start threshold to the new congestion window.

#### **4.3. Recovery Period**

Recovery is a period of time beginning with detection of a lost packet. Because QUIC retransmits stream data and control frames, not packets, it defines the end of recovery as a packet sent after the start of recovery being acknowledged. This is slightly different from TCP's definition of recovery ending when the lost packet that started recovery is acknowledged.

During recovery, the congestion window is not increased or decreased. As such, multiple lost packets only decrease the congestion window once as long as they're lost before exiting recovery. This causes QUIC to decrease the congestion window multiple times if retransmissions are lost, but limits the reduction to once per round trip.

#### **4.4. Tail Loss Probe**

If recovery sends a tail loss probe, no change is made to the congestion window or pacing rate. Acknowledgement or loss of tail loss probes are treated like any other packet.

#### **4.5. Retransmission Timeout**

When retransmissions are sent due to a retransmission timeout alarm, no change is made to the congestion window or pacing rate until the next acknowledgement arrives. When an ack arrives, if packets prior to the first retransmission timeout are acknowledged, then the congestion window remains the same. If no packets prior to the first retransmission timeout are acknowledged, the retransmission timeout



has been validated and the congestion window must be reduced to the minimum congestion window and slow start is begun.

#### **4.6. Pacing Rate**

The pacing rate is a function of the mode, the congestion window, and the smoothed rtt. Specifically, the pacing rate is 2 times the congestion window divided by the smoothed RTT during slow start and 1.25 times the congestion window divided by the smoothed RTT during congestion avoidance. In order to fairly compete with flows that are not pacing, it is recommended to not pace the first 10 sent packets when exiting quiescence.

#### **4.7. Pseudocode**

##### **4.7.1. Constants of interest**

Constants used in congestion control are based on a combination of RFCs, papers, and common practice. Some may need to be changed or negotiated in order to better suit a variety of environments.

kDefaultMss (default 1460 bytes): The default max packet size used for calculating default and minimum congestion windows.

kInitialWindow (default  $10 * kDefaultMss$ ): Default limit on the amount of outstanding data in bytes.

kMinimumWindow (default  $2 * kDefaultMss$ ): Default minimum congestion window.

kLossReductionFactor (default 0.5): Reduction in congestion window when a new loss event is detected.

##### **4.7.2. Variables of interest**

Variables required to implement the congestion control mechanisms are described in this section.

bytes\_in\_flight: The sum of the size in bytes of all sent packets that contain at least one retransmittable or PADDING frame, and have not been acked or declared lost. The size does not include IP or UDP overhead. Packets only containing ACK frames do not count towards byte\_in\_flight to ensure congestion control does not impede congestion feedback.

congestion\_window: Maximum number of bytes in flight that may be sent.





`end_of_recovery`: The largest packet number sent when QUIC detects a loss. When a larger packet is acknowledged, QUIC exits recovery.

`ssthresh`: Slow start threshold in bytes. When the congestion window is below `ssthresh`, the mode is slow start and the window grows by the number of bytes acknowledged.

#### [4.7.3.](#) Initialization

At the beginning of the connection, initialize the congestion control variables as follows:

```
congestion_window = kInitialWindow
bytes_in_flight = 0
end_of_recovery = 0
ssthresh = infinite
```

#### [4.7.4.](#) On Packet Sent

Whenever a packet is sent, and it contains non-ACK frames, the packet increases `bytes_in_flight`.

```
OnPacketSentCC(bytes_sent):
    bytes_in_flight += bytes_sent
```

#### [4.7.5.](#) On Packet Acknowledgement

Invoked from loss detection's `OnPacketAked` and is supplied with `acked_packet` from `sent_packets`.

```
OnPacketAkedCC(acked_packet):
    // Remove from bytes_in_flight.
    bytes_in_flight -= acked_packet.bytes
    if (acked_packet.packet_number < end_of_recovery):
        // Do not increase congestion window in recovery period.
        return
    if (congestion_window < ssthresh):
        // Slow start.
        congestion_window += acked_packets.bytes
    else:
        // Congestion avoidance.
        congestion_window +=
            kDefaultMss * acked_packets.bytes / congestion_window
```



#### **4.7.6. On Packets Lost**

Invoked by loss detection from DetectLostPackets when new packets are detected lost.

```
OnPacketsLost(lost_packets):  
    // Remove lost packets from bytes_in_flight.  
    for (lost_packet : lost_packets):  
        bytes_in_flight -= lost_packet.bytes  
        largest_lost_packet = lost_packets.last()  
        // Start a new recovery epoch if the lost packet is larger  
        // than the end of the previous recovery epoch.  
        if (end_of_recovery < largest_lost_packet.packet_number):  
            end_of_recovery = largest_sent_packet  
            congestion_window *= kLossReductionFactor  
            congestion_window = max(congestion_window, kMinimumWindow)  
            ssthresh = congestion_window
```

#### **4.7.7. On Retransmission Timeout Verified**

QUIC decreases the congestion window to the minimum value once the retransmission timeout has been verified.

```
OnRetransmissionTimeoutVerified()  
    congestion_window = kMinimumWindow
```

### **5. IANA Considerations**

This document has no IANA actions. Yet.

### **6. References**

#### **6.1. Normative References**

[QUIC-TRANSPORT]

Iyengar, J., Ed. and M. Thomson, Ed., "QUIC: A UDP-Based Multiplexed and Secure Transport", [draft-ietf-quic-transport-00](#) (work in progress), December 2017.

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[RFC4653] Bhandarkar, S., Reddy, A., Allman, M., and E. Blanton, "Improving the Robustness of TCP to Non-Congestion Events", [RFC 4653](#), DOI 10.17487/RFC4653, August 2006, <<https://www.rfc-editor.org/info/rfc4653>>.



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- [RFC6298] Paxson, V., Allman, M., Chu, J., and M. Sargent, "Computing TCP's Retransmission Timer", [RFC 6298](#), DOI 10.17487/RFC6298, June 2011, <<https://www.rfc-editor.org/info/rfc6298>>.
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- [RFC8174] Leiba, B., "Ambiguity of Uppercase vs Lowercase in [RFC 2119](#) Key Words", [BCP 14](#), [RFC 8174](#), DOI 10.17487/RFC8174, May 2017, <<https://www.rfc-editor.org/info/rfc8174>>.

## **6.2. Informative References**

- [LOSS-PROBE] Dukkkipati, N., Cardwell, N., Cheng, Y., and M. Mathis, "Tail Loss Probe (TLP): An Algorithm for Fast Recovery of Tail Losses", [draft-dukkkipati-tcpm-tcp-loss-probe-01](#) (work in progress), February 2013.
- [RFC3465] Allman, M., "TCP Congestion Control with Appropriate Byte Counting (ABC)", [RFC 3465](#), DOI 10.17487/RFC3465, February 2003, <<https://www.rfc-editor.org/info/rfc3465>>.
- [RFC6582] Henderson, T., Floyd, S., Gurtov, A., and Y. Nishida, "The NewReno Modification to TCP's Fast Recovery Algorithm", [RFC 6582](#), DOI 10.17487/RFC6582, April 2012, <<https://www.rfc-editor.org/info/rfc6582>>.



- [TLP] Dukkupati, N., Cardwell, N., Cheng, Y., and M. Mathis, "Tail Loss Probe (TLP): An Algorithm for Fast Recovery of Tail Losses", [draft-dukkupati-tcpm-tcp-loss-probe-01](#) (work in progress), February 2013.

### **6.3. URIs**

- [1] [https://mailarchive.ietf.org/arch/search/?email\\_list=quic](https://mailarchive.ietf.org/arch/search/?email_list=quic)
- [2] <https://github.com/quicwg>
- [3] <https://github.com/quicwg/base-drafts/labels/-recovery>

### **Appendix A. Acknowledgments**

### **Appendix B. Change Log**

\*RFC Editor's Note:\* Please remove this section prior to publication of a final version of this document.

#### **B.1. Since [draft-ietf-quic-recovery-06](#)**

Nothing yet.

#### **B.2. Since [draft-ietf-quic-recovery-05](#)**

- o Add more congestion control text (#776)

#### **B.3. Since [draft-ietf-quic-recovery-04](#)**

No significant changes.

#### **B.4. Since [draft-ietf-quic-recovery-03](#)**

No significant changes.

#### **B.5. Since [draft-ietf-quic-recovery-02](#)**

- o Integrate F-RTT (#544, #409)
- o Add congestion control (#545, #395)
- o Require connection abort if a skipped packet was acknowledged (#415)
- o Simplify RTT calculations (#142, #417)





**B.6.** Since [draft-ietf-quic-recovery-01](#)

- o Overview added to loss detection
- o Changes initial default RTT to 100ms
- o Added time-based loss detection and fixes early retransmit
- o Clarified loss recovery for handshake packets
- o Fixed references and made TCP references informative

**B.7.** Since [draft-ietf-quic-recovery-00](#)

- o Improved description of constants and ACK behavior

**B.8.** Since [draft-iyengar-quic-loss-recovery-01](#)

- o Adopted as base for [draft-ietf-quic-recovery](#)
- o Updated authors/editors list
- o Added table of contents

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