       rLEDBAT: receiver-driven Low Extra Delay Background Transport for TCP
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Abstract

   This document specifies the rLEDBAT, a set of mechanisms that enable
   the execution of a less-than-best-effort congestion control algorithm
   for TCP at the receiver end.

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1. Introduction

   LEDBAT (Low Extra Delay Background Transport) [RFC6817] is a
   congestion-control algorithm that implements a less-than-best-effort
   (LBE) traffic class.

   When LEDBAT traffic shares a bottleneck with one or more TCP
   connections using standard congestion control algorithms such as
   Cubic [RFC8312] (hereafter standard-TCP for short), it reduces its
   sending rate earlier and more aggressively than standard-TCP
   congestion control, allowing standard-TCP traffic to use more of the
   available capacity. In the absence of competing standard-TCP
   traffic, LEDBAT aims to make an efficient use of the available
   capacity, while keeping the queuing delay within predefined bounds.

   LEDBAT reacts both to packet loss and to variations in delay.
   Regarding to packet loss, LEDBAT reacts with a multiplicative
   decrease, similar to most TCP congestion controllers. Regarding
   delay, LEDBAT aims for a target queueing delay. When the measured
   current queueing delay is below the target, LEDBAT increases the
   sending rate and when the delay is above the target, it reduces the
   sending rate. LEDBAT estimates the queuing delay by subtracting the
   measured current one-way delay from the estimated base one-way delay
   (i.e. the one-way delay in the absence of queues).

The LEDBAT specification [RFC6817] defines the LEDBAT congestion-
control algorithm, implemented in the sender to control its sending
rate. LEDBAT is specified in a protocol and layer agnostic manner.

LEDBAT++ [I-D.balasubramanian-iccrg-ledbatplusplus] is also an LBE
congestion control algorithm which is inspired in LEDBAT while
addressing several problems identified with the original LEDBAT
specification. In particular the differences between LEDBAT and
LEDBAT++ include: i) LEDBAT++ uses the round-trip-time (RTT) (as
opposed to the one way delay used in LEDBAT) to estimate the queuing
delay; ii) LEDBAT++ uses an Additive Increase/Multiplicative Decrease
algorithm to achieve inter-LEDBAT++ fairness and avoid the late-comer
advantage observed in LEDBAT; iii) LEDBAT++ performs periodic
slowdowns to improve the measurement of the base delay; iv) LEDBAT++
is defined for TCP.

In this note, we describe rLEDBAT, a set of mechanisms that enable
the execution of an LBE delay-based congestion control algorithm such
as LEDBAT or LEDBAT++ in the receiver end of a TCP connection.

2. Motivations for rLEDBAT

rLEDBAT enables new use cases and new deployment models, fostering
the use of LBE traffic and benefitting the global Internet by
improving overall allocation of resources. The following scenarios
are enabled by rLEDBAT:

   Content Delivery Networks and more sophisticated file distribution
   scenarios: Consider the case where the source of a file to be
   distributed (e.g., a software developer that wishes to distribute
   a software update) would prefer to use LBE and it enables LEDBAT/
   LEDBAT++ in the servers containing the source file. However,
   because the file is being distributed through a CDN which
   surrogates do not support LBE congestion control, the result is
   that the file transfers, originated from CDN surrogates will not
   be using LBE. Interestingly enough, in the case of the software
   update, the developer may also control the software performing the
   download in the client, the receiver of the file, but because
   current LEDBAT/LEDBAT++ are sender-based algorithms, controlling
   the client is not enough to enable LBE congestion control in the
   communication. rLEDBAT would enable the use of LBE traffic class
   for file distribution in this setup.

   Interference from proxies and other middleboxes: Proxies and other
   middleboxes are a commonplace in the Internet. For instance, in
   the case of mobile networks, proxies are frequently used. In the
   case of enterprise networks, it is common to deploy corporate
   proxies for filtering and firewalling. In the case of satellite

links, Performance Enhancement Proxies (PEPs) are deployed to
mitigate the effect of the long delay in TCP connection. These
proxies terminate the TCP connection on both ends and prevent the
use of LBE congestion control in the segment between the proxy and
the sink of the content, the client. By enabling rLEDBAT, clients
would be able to enable LBE traffic between them and the proxy.

Receiver-defined preferences. It is frequent that the bottleneck
of the communication is the access link. This is particularly
true in the case of mobile devices. It is then especially
relevant for mobile devices to properly manage the capacity of the
access link. With current technologies, it is possible for the
mobile device to use different congestion control algorithms
expressing different preferences for the traffic. For instance, a
device can choose to use standard-TCP for some traffic and to use
LEDBAT/LEDBAT++ for other traffic. However, this would only
affect the outgoing traffic since both standard-TCP and LEDBAT/
LEDBAT++ are sender-driven. The mobile device has no means to
manage the traffic in the down-link, which is in most cases, the
communication bottleneck for a typical eye-ball end-user. rLEDBAT
enables the mobile device to selectively use LBE traffic class for
some of the incoming traffic. For instance, by using rLEDBAT, a
user can use regular standard-TCP/UDP for video stream (e.g.,
Youtube) and use rLEDBAT for other background file download.

3. rLEDBAT mechanisms

rLEDBAT provides the mechanisms to implement an LBE congestion
control algorithm at the receiver-end of a TCP connection. The
rLEDBAT receiver controls the sender's rate through the Receive
Window announced to the receiver in the TCP header.

rLEDBAT assumes that the sender is a standard TCP sender. rLEDBAT
does not require any rLEDBAT-specific modifications to the TCP
sender. The envisioned deployment model for rLEDBAT is that the
clients implement rLEDBAT and this enable rLEDBAT in communications
with existent standard TCP senders. In particular, the sender MUST
implement [I-D.ietf-tcpm-rfc793bis] and it also MUST implement the
Time Stamp Option as defined in [RFC7323]. Also, the sender SHOULD
implement some of the standard congestion control mechanisms, such as
Cubic [RFC8312] or New Reno [RFC5681].

rLEDBAT does not defines a new congestion control algorithm. The LBE
congestion control algorithm executed in the rLEDBAT receiver is
defined in other documents. The rLEDBAT receiver MUST use an LBE
congestion control algorithm. Because rLEDBAT assumes a standard TCP
sender, the sender will be using a "best effort" congestion control
algorithm (such as Cubic or New Reno). Since rLEDBAT uses the

Receive Window to control the sender's rate and the sender calculates
the sender's window as the minimum of the Receive window and the
congestion window, rLEDBAT will only be effective as long as the
congestion control algorithm executed in the receiver yields a
smaller window than the one calculated by the sender. This is
normally the case when the receiver is using an LBE congestion
control algorithm. The rLEDBAT receiver SHOULD use the LEDBAT
congestion control algorithm [RFC6817] or the LEDBAT++ congestion
control algorithm [I-D.balasubramanian-iccrg-ledbatplusplus]. The
rLEDBAT MAY use other LBE congestion control algorithms defined
elsewhere. Irrespectively of which congestion control algorithm is
executed in the receiver, an rLEDBAT connection will never be more
aggressive than standard TCP since it is always bounded by the
congestion control algorithm executed at the sender.

rLEDBAT is essentially composed of three types of mechanisms, namely,
those that provide the means to measure the packet delay (either the
round trip time or the one way delay, depending on the selected
algorithm), mechanisms to detect packet loss and the means to
manipulate the Receive Window to control the sender's rate. We
describe them next.

3.1. Controlling the receive window

rLEDBAT uses the Receive Window (RCV.WND) of TCP to enable the
receiver to control the sender's rate. [I-D.ietf-tcpm-rfc793bis]
defines that the RCV.WND is used to announce the available receive
buffer to the sender for flow control purposes. In order to avoid
confusion, we will call fc.WND the value that a standard RFC793bis
TCP receiver calculates to set in the receive window for flow control
purposes. We call rl.WND the window value calculated by rLEDBAT
algorithm and we call RCV.WND the value actually included in the
Receive Window field of the TCP header. For a RFC793bis receiver,
RCV.WND == fc.WND.

In the case of rLEDBAT receiver, the rLEDBAT receiver MUST NOT set
the RCV.WND to a value larger than fc.WND and it SHOULD set the
RCV.WND to the minimum of rl.WND and fc.WND, honoring both.

When using rLEDBAT, two congestion controllers are in action in the
flow of data from the sender to the receiver, namely, the congestion
control algorithm of TCP in the sender side and the LBE congestion
control algorithm executed in the receiver and conveyed to the sender
through the RCV.WND. In the normal TCP operation, the sender uses
the minimum of the congestion window cwnd and the receiver window
RCV.WND to calculate the sender's window SND.WND. This is also true
for rLEDBAT, as the sender is a regular TCP sender. This guarantees
that the rLEDBAT flow will never transmit more aggressively than a

TCP flow, as the sender's congestion window limits the sending rate.
Moreover, because a LBE congestion control algorithm such as LEDBAT/
LEDBAT++ is designed to react earlier and more aggressively to
congestion than regular TCP congestion control, the rl.WND contained
in the RCV.WND field of TCP will be in general smaller than the
congestion window calculated by the TCP sender, implying that the
rLEDBAT congestion control algorithm will be effectively controlling
the sender's window.

In summary, the sender's window is: SND.WND = min(cwnd, rl.WND,
fc.WND)

3.1.1. Avoiding window shrinking

The LEDBAT/LEDBAT++ algorithm executed in a rLEDBAT receiver
increases or decreases the rl.WND according to congestion signals
(variations on the estimations of the queueing delay and packet
loss). If the new congestion window is smaller than the current one
then directly announcing it in the RCV.WND may result in shrinking
the window, i.e., moving the right window edge to the left.
Shrinking the window is discouraged as per [I-D.ietf-tcpm-rfc793bis],
as it may cause unnecessary packet loss and performance penalty. To
be consistent with [I-D.ietf-tcpm-rfc793bis], the rLEDBAT receiver
SHOULD NOT shrink the receive window.

In order to avoid window shrinking, upon the reception of a data
packet, the announced window can be reduced in the number of bytes
contained in the packet at most. This may fall short to honor the
new calculated value of the rl.WND. So, in order to reduce the
window as dictated by the rLEDBAT algorithm, the receiver will
progressively reduce the advertised RCV.WND, always honoring that the
reduction is less or equal than the received bytes, until the target
window determined by the rLEDBAT algorithm is reached. This implies
that it may take up to one RTT for the rLEDBAT receiver to drain
enough in-flight bytes to completely close its receive window without
shrinking it. This is more than sufficient to honor the window
output from the LEDBAT/LEDBAT++ algorithms since they only allows to
perform at most one multiplicative decrease per RTT.

3.1.2. Window Scale Option

The Window Scale (WS) option [RFC7323] is a mean to increase the
maximum window size permitted by the Receive Window. The use of the
WS option implies that the changes in the window are expressed in the
units resulting of the WS option used in the TCP connection. This
means that the rLEDBAT client will have to accumulate the increases
resulting from the different received packets, and only convey a
change in the window when the accumulated sum of increases is equal

or higher than one unit used to express the receive window according
to the WS option in place for the TCP connection.

Changes in the receive window that are smaller than 1 MSS are
unlikely to have any immediate impact on the sender's rate, as usual
TCP segmentation practice results in sending full segments (i.e.,
segments of size equal to the MSS). So, accumulating changes in the
receive window until completing a full MSS in the sender or in the
receiver makes little difference.

Current WS option specification [RFC7323] defines that allowed values
for the WS option are between 0 and 14. Assuming a MSS around 1500
bytes, WS option values between 0 and 11 result in the receive window
being expressed in units that are about 1 MSS or smaller. So, WS
option values between 0 and 11 have no impact in rLEDBAT.

WS option values higher than 11 can affect the dynamics of rLEDBAT,
since control may become too coarse (e.g., with WS of 14, a change in
one unit of the receive window implies a change of 10 MSS in the
effective window).

For the above reasons, the rLEDBAT client SHOULD set WS option values
lower than 12. Additional experimentation is required to explore the
impact of larger WS values in rLEDBAT dynamics.

Note that the recommendation for rLEDBAT to set the WS option value
to lower values does not precludes the communication with servers
that set the WS option values to larger values, since the WS option
value used is set independently for each direction of the TCP
connection.

3.2. Measuring delays

Both LEDBAT and LEDBAT++ measure base and current delays to estimate
the queueing delay. LEDBAT uses the one way delay while LEDBAT++
uses the round trip time. In the next sections we describe how
rLEDBAt mechanisms enable the receiver to measure the one way delay
or the round trip time, whatever needed depending on the congestion
control algorithm used.

3.2.1. Measuring the RTT to estimate the queueing delay

LEDBAT++ uses the round trip time (RTT) to estimate the queueing
delay. In order to estimate the queueing delay using the RTT, the
rLEDBAT receiver estimates the base RTT (i.e., the constant
components of the RTT) and also measures the current RTT. By
subtracting these two values, we obtain the queuing delay to be used
by the rLEDBAT controller.

LEDBAT++ discovers the base RTT (RTTb) by taking the minimum value of
the measured RTTs over a period of time. The current RTT (RTTc) is
estimated using a number of recent samples and applying a filter,
such as the minimum (or the mean) of the last k samples. Using the
RTT to estimate the queueing delay has a number of shortcomings and
difficulties that we discuss next.

The queuing delay measured using the RTT includes also the queueing
delay experienced by the return packets in the direction from the
rLEDBAT receiver to the sender. This is a fundamental limitation of
this approach. The impact of this error is that the rLEDBAT
controller will also react to congestion in the reverse path
direction which results in an even more conservative mechanism.

In order to measure the RTT, the rLEDBAT client MUST enable the Time
Stamp (TS) option [RFC7323]. By matching the TSVal value carried in
outgoing packets with the TSecr value observed in incoming packets,
it is possible to measure the RTT. This allows the rLEDBAT receiver
to measure the RTT even if it is acting as a pure receiver. In a
pure receiver there is no data flowing from the rLEDBAT receiver to
the sender, making impossible to match data packets with
acknowledgements packets to measure the RTT, as it is usually done in
TCP for other purposes.

Depending on the frequency of the local clock used to generate the
values included in the TS option, several packets may carry the same
TSVal value. If that happens, the rLEDBAT receiver will be unable to
match the different outgoing packets carrying the same TSVal value
with the different incoming packets carrying also the same TSecr
value. However, it is not necessary for rLEDBAT to use all packets
to estimate the RTT and sampling a subset of in-flight packets per
RTT is enough to properly assess the queueing delay. The RTT MUST
then be calculated as the time since the first packet with a given
TSVal was sent and the first packet that was received with the same
value contained in the TSecr. Other packets with repeated TS values
SHOULD NOT be used for the RTT calculation.

Several issues must be addressed in order to avoid an artificial
increase of the observed RTT. Different issues emerge depending
whether the rLEDBAT capable host is sending data packets or pure ACKs
to measure the RTT. We next consider the issues separately.

3.2.1.1. Measuring RTT sending pure ACKs

In this scenario, the rLEDBAT node (node A) sends a pure ACK to the
other endpoint of the TCP connection (node B), including the TS
option. Upon the reception of the TS Option, host B will copy the
value of the TSVal into the TSecr field of the TS option and include

that option into the next data packet towards host A. However, there
are two reasons why B may not send a packet immediately back to A,
artificially increasing the measured RTT. The first reason is when A
has no data to send. The second is when A has no available window to
put more packets in-flight. We describe next how each of these cases
is addressed.

The case where the host B has no data to send when it receives the
pure Acknowledgement is expected to be rare in the rLEDBAT use cases.
rLEDBAT will be used mostly for background file transfers so the
expected common case is that the sender will have data to send
throughout the lifetime of the communication. However, if, for
example, the file is structured in blocks of data, it may be the case
that seldom, the sender will have to wait until the next block is
available to proceed with the data transfer and momentarily lack of
data to send. To address this situation, the filter used by the
congestion control algorithm executed in the receiver SHOULD discard
the larger samples (e.g. a min filter would achieve this) when
measuring the RTT using pure ACK packets.

The limitation of available sender's window to send more packets can
come either from the TCP congestion window in host B or from the
announced receive window from the rLEDBAT in host A. Normally, the
receive window will be the one to limit the sender's transmission
rate, since the LBE congestion control algorithm used by the rLEDBAT
node is designed to be more restrictive on the sender's rate than
standard-TCP. If the limiting factor is the congestion window in the
sender, it is less relevant if rLEDBAT further reduces the receive
window due to a bloated RTT measurement, since the rLEDBAT is not
actively controlling the sender's rate. Nevertheless, the proposed
approach to discard larger samples would also address this issue.

To address the case in which the limiting factor is the receive
window announced by rLEDBAT, the congestion control algorithm at the
receiver SHOULD discard the RTT measurements done using pure ACK
packets while reducing the window and avoid including bloated samples
in the queueing delay estimation. The rLEDBAT receiver is aware
whether a given TSVal value was sent in a pure ACK packet where the
window was reduced, and if so, it can discard the corresponding RTT
measurement.

3.2.1.2. Measuring the RTT sending data packets

In the case that the rLEDBAT node is sending data packets and
matching them with pure ACKs to measure the RTT, a factor that can
artificially increase the RTT measured is the presence of delayed
Acknowledgements. According to the TS option generation rules
[RFC7323], the value included in the TSecr for a delayed ACK is the

one in the TSVal field of the earliest unacknowledged segment. This
may artificially increase the measured RTT.

If both endpoints of the connection are sending data packets,
Acknowledgments are piggybacked into the data packets and they are
not delayed. Delayed ACKs only increase the RTT measurement in the
case that the sender has no data to send. Since the expected use
case for rLEDBAT is that the sender will be sending background
traffic to the rLEDBAT receiver, the cases where delayed ACKs
increase the measured RTT are expected to be rare.

Nevertheless, for those measurements done using data packets sent by
the rLEDBAT node matching pure ACKs sent from the other endpoint of
the connection, they will result in an increased RTT. The additional
increase in the measured RTT will range between the transmission
delay of on packet and 500 ms. The reason for this is that delayed
ACKs are generated every second data packet received and not delayed
more than 500 ms according to [I-D.ietf-tcpm-rfc793bis]. The rLEDBAT
receiver MAY discard the RTT measurements done using data packets
from the rLEBDAT receiver and matching pure ACKs, especially if it
has recent measurements done using other packet combinations.Also,
applying a filter that discard larger samples would also address this
issue (e.g. a min filter).

3.2.2. Measuring one way delay to estimate the queueing delay

The LEDBAT algorithm uses the one-way delay of packets as input. A
TCP receiver can measure the delay of incoming packets directly (as
opposed to the sender-based LEDBAT, where the receiver measures the
one-way delay and needs to convey it to the sender).

In the case of TCP, the receiver can use the Time Stamp option to
measure the one way delay by subtracting the time stamp contained in
the incoming packet from the local time at which the packet has
arrived. As noted in [RFC6817] the clock offset between the clock of
the sender and the clock in the receiver does not affect the LEDBAT
operation, since LEDBAT uses the difference between the base one way
delay and the current one way delay to estimate the queuing delay,
effectively canceling the clock offset error in the queueing delay
estimation. There are however two other issues that the rLEDBAT
receiver needs to take into account in order to properly estimate the
one way delay, namely, the units in which the received timestamps are
expressed and the clock skew. We address them next.

In order to measure the one way delay using TCP timestamps, the
rLEDBAT receiver needs to discover the units in which the values of
the TS option are expressed and second, to account for the skew
between the two clocks of the endpoints of the TCP connection. Note

that a mismatch of 100 ppm (parts per million) in the estimation at
the receiver of the clock rate of the sender accounts for 6 ms of
variation per minute in the measured delay for a communication, just
one order of magnitude below the target set for controlling the rate
by rLEDBAT. Typical skew for untrained clocks is reported to be
around 100-200 ppm [RFC6817].

In order to learn both the TS units and the clock skew, the rLEDBAT
receiver compares how much local time has elapsed between the sender
has issued two packets with different TS values. By comparing the
local time difference and the TS value difference, the receiver can
assess the TS units and relative clock skews. In order for this to
be accurate, the packets carrying the different TS values should
experience equal (or at least similar delay) when traveling from the
sender to the receiver, as any difference in the experienced delays
would introduce error in the unit/skew estimation. One possible
approach is to select packets that experienced the minimum delay
(i.e. close to zero queueing delay) to make the estimations.

An additional difficulty regarding the estimation of the TS units and
clock skew in the context of (r)LEDBAT is that the LEDBAT congestion
controller actions directly affect the (queueing) delay experienced
by packets. In particular, if there is an error in the estimation of
the TS units/skew, the LEDBAT controller will attempt to compensate
it by reducing/increasing the load. The result is that the LEDBAT
operation interferes with the TS units/clock skew measurements.
Because of this, measurements are more accurate when there is no
traffic in the connection (in addition to the packets used for the
measurements). The problem is that the receiver is unaware if the
sender is injecting traffic at any point in time, and
opportunistically seize quiet intervals to preform measurements. The
receiver can however, force periodic slowdowns, reducing the
announced receive window to a few packets and perform the
measurements then.

It is possible for the rLEDBAT receiver to perform multiple
measurements to assess both the TS units and the relative clock skew
during the lifetime of the connection, in order to obtain more
accurate results. Clock skew measurements are more accurate if the
time period used to discover the skew is larger, as the impact of the
skew becomes more apparent. Due to the same logic, accurately
learning the clock skew is more pressing as the time separating the
two delays to compare increases. It is a reasonable approach for the
rLEDBAT receiver to perform an early discovery of the TS units (and
the clock skew) using the first few packets of the TCP connection and
then improve the accuracy of the TS units/clock skew estimation using
periodic measurements later in the lifetime of the connection.

3.3. Detecting packet losses and retransmissions

   The rLEDBAT receiver is capable of detecting retransmitted packets in
   the following way. We call RCV.HGH the highest sequence number
   correspondent to a received byte of data (not assuming that all bytes
   with smaller sequence numbers have been received already, there may
   be holes) and we call TSV.HGH the TSVal value corresponding to the
   segment in which that byte was carried. SEG.SEQ stands for the
   sequence number of a newly received segment and we call TSV.SEQ the
   TSVal value of the newly received segment.

   If SEG.SEQ < RCV.HGH and TSV.SEQ > TSV.HGH then the newly received
   segment is a retransmission. This is so because the newly received
   segment was generated later than another already received segment
   which contained data with a larger sequence number. This means that
   this segment was lost and was retransmitted.

   The proposed mechanism to detect retransmissions at the receiver
   fails when there are window tail drops. If all packets in the tail
   of the window are lost, the receiver will not be able to detect a
   mismatch between the sequence numbers of the packets and the order of
   the timestamps. In this case, rLEDBAT will not react to losses but
   the TCP congestion controller at the sender will, most likely
   reducing its window to 1MSS and take over the control of the sending
   rate, until slow start ramps up and catches the current value of the
   rLEDBAT window.

4. Security Considerations

5. IANA Considerations

6. Acknowledgements

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