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TCP Extensions for Long-Delay Paths

Status of This Memo

This memo proposes a set of extensions to the TCP protocol to provide efficient operation over a path with a high bandwidth*delay product. These extensions are not proposed as an Internet standard at this time. Instead, they are intended as a basis for further experimentation and research on transport protocol performance. Distribution of this memo is unlimited.

1. INTRODUCTION

Recent work on TCP performance has shown that TCP can work well over a variety of Internet paths, ranging from 800 Mbit/sec I/O channels to 300 bit/sec dial-up modems [Jacobson88]. However, there is still a fundamental TCP performance bottleneck for one transmission regime: paths with high bandwidth and long round-trip delays. The significant parameter is the product of bandwidth (bits per second) and round-trip delay (RTT in seconds); this product is the number of bits it takes to "fill the pipe", i.e., the amount of unacknowledged data that TCP must handle in order to keep the pipeline full. TCP performance problems arise when this product is large, e.g., significantly exceeds 10**5 bits. We will refer to an Internet path operating in this region as a "long, fat pipe", and a network containing this path as an "LFN" (pronounced "elephan(t)").

High-capacity packet satellite channels (e.g., DARPA’s Wideband Net) are LFN’s. For example, a T1-speed satellite channel has a bandwidth*delay product of 10**6 bits or more; this corresponds to 100 outstanding TCP segments of 1200 bytes each! Proposed future terrestrial fiber-optical paths will also fall into the LFN class; for example, a cross-country delay of 30 ms at a DS3 bandwidth (45Mbps) also exceeds 10**6 bits.

Clever algorithms alone will not give us good TCP performance over LFN’s; it will be necessary to actually extend the protocol. This RFC proposes a set of TCP extensions for this purpose.

There are three fundamental problems with the current TCP over LFN

paths:

(1) Window Size Limitation

The TCP header uses a 16 bit field to report the receive window size to the sender. Therefore, the largest window that can be used is 2**16 = 65K bytes. (In practice, some TCP implementations will "break" for windows exceeding 2**15, because of their failure to do unsigned arithmetic).

To circumvent this problem, we propose a new TCP option to allow windows larger than 2**16. This option will define an implicit scale factor, to be used to multiply the window size value found in a TCP header to obtain the true window size.

(2) Cumulative Acknowledgments

Any packet losses in an LFN can have a catastrophic effect on throughput. This effect is exaggerated by the simple cumulative acknowledgment of TCP. Whenever a segment is lost, the transmitting TCP will (eventually) time out and retransmit the missing segment. However, the sending TCP has no information about segments that may have reached the receiver and been queued because they were not at the left window edge, so it may be forced to retransmit these segments unnecessarily.

We propose a TCP extension to implement selective acknowledgements. By sending selective acknowledgments, the receiver of data can inform the sender about all segments that have arrived successfully, so the sender need retransmit only the segments that have actually been lost.

Selective acknowledgments have been included in a number of experimental Internet protocols -- VMTP [Cheriton88], NETBLT [Clark87], and RDP [Velten84]. There is some empirical evidence in favor of selective acknowledgments -- simple experiments with RDP have shown that disabling the selective acknowlegment facility greatly increases the number of retransmitted segments over a lossy, high-delay Internet path [Partridge87]. A simulation study of a simple form of selective acknowledgments added to the ISO transport protocol TP4 also showed promise of performance improvement [NBS85].

(3) Round Trip Timing

TCP implements reliable data delivery by measuring the RTT, i.e., the time interval between sending a segment and receiving an acknowledgment for it, and retransmitting any segments that are not acknowledged within some small multiple of the average RTT. Experience has shown that accurate, current RTT estimates are necessary to adapt to changing traffic conditions and, without them, a busy network is subject to an instability known as "congestion collapse" [Nagle84].

In part because TCP segments may be repacketized upon retransmission, and in part because of complications due to the cumulative TCP acknowledgement, measuring a segments's RTT may involve a non-trivial amount of computation in some implementations. To minimize this computation, some implementations time only one segment per window. While this yields an adequate approximation to the RTT for small windows (e.g., a 4 to 8 segment Arpanet window), for an LFN (e.g., 100 segment Wideband Network windows) it results in an unacceptably poor RTT estimate.

In the presence of errors, the problem becomes worse. Zhang [Zhang86], Jain [Jain86] and Karn [Karn87] have shown that it is not possible to accumulate reliable RTT estimates if retransmitted segments are included in the estimate. Since a full window of data will have been transmitted prior to a retransmission, all of the segments in that window will have to be ACKed before the next RTT sample can be taken. This means at least an additional window's worth of time between RTT measurements and, as the error rate approaches one per window of data (e.g., 10**-6 errors per bit for the Wideband Net), it becomes effectively impossible to obtain an RTT measurement.

We propose a TCP "echo" option that allows each segment to carry its own timestamp. This will allow every segment, including retransmissions, to be timed at negligible computational cost.

In designing new TCP options, we must pay careful attention to interoperability with existing implementations. The only TCP option defined to date is an "initial option", i.e., it may appear only on a SYN segment. It is likely that most implementations will properly ignore any options in the SYN segment that they do not understand, so new initial options should not cause a problem. On the other hand, we fear that receiving unexpected non-initial options may cause some TCP's to crash.

Therefore, in each of the extensions we propose, non-initial options may be sent only if an exchange of initial options has indicated that both sides understand the extension. This approach will also allow a TCP to determine when the connection opens how big a TCP header it will be sending.

2. TCP WINDOW SCALE OPTION

The obvious way to implement a window scale factor would be to define a new TCP option that could be included in any segment specifying a window. The receiver would include it in every acknowledgment segment, and the sender would interpret it. Unfortunately, this simple approach would not work. The sender must reliably know the receiver's current scale factor, but a TCP option in an acknowledgement segment will not be delivered reliably (unless the ACK happens to be piggy-backed on data).

However, SYN segments are always sent reliably, suggesting that each side may communicate its window scale factor in an initial TCP option. This approach has a disadvantage: the scale must be established when the connection is opened, and cannot be changed thereafter. However, other alternatives would be much more complicated, and we therefore propose a new initial option called Window Scale.

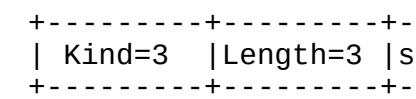
2.1 Window Scale Option

This three-byte option may be sent in a SYN segment by a TCP (1) to indicate that it is prepared to do both send and receive window scaling, and (2) to communicate a scale factor to be applied to its receive window. The scale factor is encoded logarithmically, as a power of 2 (presumably to be implemented by binary shifts).

Note: the window in the SYN segment itself is never scaled.

TCP Window Scale Option:

Kind: 3



Here shift.cnt is the number of bits by which the receiver right-shifts the true receive-window value, to scale it into a 16-bit value to be sent in TCP header (this scaling is explained below). The value shift.cnt may be zero (offering to scale, while applying a scale factor of 1 to the receive window).

This option is an offer, not a promise; both sides must send Window Scale options in their SYN segments to enable window scaling in either direction.

2.2 Using the Window Scale Option

A model implementation of window scaling is as follows, using the notation of RFC-793 [Postel81]:

* The send-window (SND.WND) and receive-window (RCV.WND) sizes in the connection state block and in all sequence space calculations are expanded from 16 to 32 bits.

* Two window shift counts are added to the connection state: snd.scale and rcv.scale. These are shift counts to be applied to the incoming and outgoing windows, respectively. The precise algorithm is shown below.

* All outgoing SYN segments are sent with the Window Scale option, containing a value shift.cnt = R that the TCP would like to use for its receive window.

* Snd.scale and rcv.scale are initialized to zero, and are changed only during processing of a received SYN segment. If the SYN segment contains a Window Scale option with shift.cnt = S, set snd.scale to S and set rcv.scale to R; otherwise, both snd.scale and rcv.scale are left at zero.

* The window field (SEG.WND) in the header of every incoming segment, with the exception of SYN segments, will be left-shifted by snd.scale bits before updating SND.WND:

  SND.WND = SEG.WND << snd.scale

  (assuming the other conditions of RFC793 are met, and using the "C" notation "<<" for left-shift).

* The window field (SEG.WND) of every outgoing segment, with the exception of SYN segments, will have been right-shifted by rcv.scale bits:

  SEG.WND = RCV.WND >> rcv.scale.

TCP determines if a data segment is "old" or "new" by testing if its sequence number is within 2**31 bytes of the left edge of the window. If not, the data is "old" and discarded. To insure that new data is never mistakenly considered old and vice-versa, the

left edge of the sender's window has to be at least 2**31 away from the right edge of the receiver's window. Similarly with the sender's right edge and receiver's left edge. Since the right and left edges of either the sender's or receiver's window differ by the window size, and since the sender and receiver windows can be out of phase by at most the window size, the above constraints imply that 2 * the max window size must be less than 2**31, or

max window < 2**30

Since the max window is 2**S (where S is the scaling shift count) times at most 2**16 - 1 (the maximum unscaled window), the maximum window is guaranteed to be < 2*30 if S <= 14. Thus, the shift count must be limited to 14. (This allows windows of 2**30 = 1 Gbyte.) If a Window Scale option is received with a shift.cnt value exceeding 14, the TCP should log the error but use 14 instead of the specified value.

3. TCP SELECTIVE ACKNOWLEDGMENT OPTIONS

To minimize the impact on the TCP protocol, the selective acknowledgment extension uses the form of two new TCP options. The first is an enabling option, "SACK-permitted", that may be sent in a SYN segment to indicate that the the SACK option may be used once the connection is established. The other is the SACK option itself, which may be sent over an established connection once permission has been given by SACK-permitted.

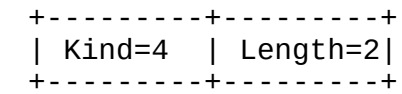
The SACK option is to be included in a segment sent from a TCP that is receiving data to the TCP that is sending that data; we will refer to these TCP's as the data receiver and the data sender, respectively. We will consider a particular simplex data flow; any data flowing in the reverse direction over the same connection can be treated independently.

3.1 SACK-Permitted Option

This two-byte option may be sent in a SYN by a TCP that has been extended to receive (and presumably process) the SACK option once the connection has opened.

TCP Sack-Permitted Option:

Kind: 4



3.2 SACK Option

The SACK option is to be used to convey extended acknowledgment information over an established connection. Specifically, it is to be sent by a data receiver to inform the data transmitter of non-contiguous blocks of data that have been received and queued. The data receiver is awaiting the receipt of data in later retransmissions to fill the gaps in sequence space between these blocks. At that time, the data receiver will acknowledge the data normally by advancing the left window edge in the Acknowledgment Number field of the TCP header.

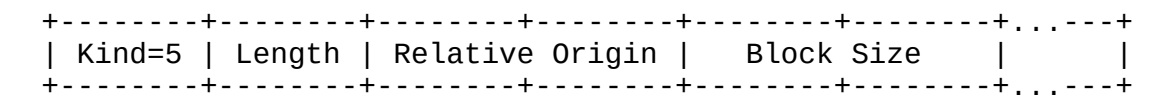
It is important to understand that the SACK option will not change the meaning of the Acknowledgment Number field, whose value will still specify the left window edge, i.e., one byte beyond the last sequence number of fully-received data. The SACK option is advisory; if it is ignored, TCP acknowledgments will continue to function as specified in the protocol.

However, SACK will provide additional information that the data transmitter can use to optimize retransmissions. The TCP data receiver may include the SACK option in an acknowledgment segment whenever it has data that is queued and unacknowledged. Of course, the SACK option may be sent only when the TCP has received the SACK-permitted option in the SYN segment for that connection.

TCP SACK Option:

Kind: 5

Length: Variable



This option contains a list of the blocks of contiguous sequence space occupied by data that has been received and queued within

the window. Each block is contiguous and isolated; that is, the octets just below the block,

Acknowledgment Number + Relative Origin -1,

and just above the block,

Acknowledgment Number + Relative Origin + Block Size,

have not been received.

Each contiguous block of data queued at the receiver is defined in the SACK option by two 16-bit integers:

* Relative Origin

  This is the first sequence number of this block, relative to the Acknowledgment Number field in the TCP header (i.e., relative to the data receiver's left window edge).

* Block Size

  This is the size in octets of this block of contiguous data.

A SACK option that specifies n blocks will have a length of 4*n+2 octets, so the 44 bytes available for TCP options can specify a maximum of 10 blocks. Of course, if other TCP options are introduced, they will compete for the 44 bytes, and the limit of 10 may be reduced in particular segments.

There is no requirement on the order in which blocks can appear in a single SACK option.

> Note: requiring that the blocks be ordered would allow a slightly more efficient algorithm in the transmitter; however, this does not seem to be an important optimization.

3.3 SACK with Window Scaling

If window scaling is in effect, then 16 bits may not be sufficient for the SACK option fields that define the origin and length of a block. There are two possible ways to handle this:

(1) Expand the SACK origin and length fields to 24 or 32 bits.

(2) Scale the SACK fields by the same factor as the window.

The first alternative would significantly reduce the number of blocks possible in a SACK option; therefore, we have chosen the second alternative, scaling the SACK information as well as the window.

Scaling the SACK information introduces some loss of precision, since a SACK option must report queued data blocks whose origins and lengths are multiples of the window scale factor rcv.scale. These reported blocks must be equal to or smaller than the actual blocks of queued data.

Specifically, suppose that the receiver has a contiguous block of queued data that occupies sequence numbers L, L+1, ... L+N-1, and that the window scale factor is S = rcv.scale. Then the corresponding block that will be reported in a SACK option will be:

```
Relative Origin = int((L+S-1)/S)

Block Size = int((L+N)/S) - (Relative Origin)
```

where the function int(x) returns the greatest integer contained in x.

The resulting loss of precision is not a serious problem for the sender. If the data-sending TCP keeps track of the boundaries of all segments in its retransmission queue, it will generally be able to infer from the imprecise SACK data which full segments don't need to be retransmitted. This will fail only if S is larger than the maximum segment size, in which case some segments may be retransmitted unnecessarily. If the sending TCP does not keep track of transmitted segment boundaries, the imprecision of the scaled SACK quantities will only result in retransmitting a small amount of unneeded sequence space. On the average, the data sender will unnecessarily retransmit J*S bytes of the sequence space for each SACK received; here J is the number of blocks reported in the SACK, and S = snd.scale.

3.4 SACK Option Examples

Assume the left window edge is 5000 and that the data transmitter sends a burst of 8 segments, each containing 500 data bytes. Unless specified otherwise, we assume that the scale factor S = 1.

Case 1: The first 4 segments are received but the last 4 are dropped.

The data receiver will return a normal TCP ACK segment acknowledging sequence number 7000, with no SACK option.

Case 2: The first segment is dropped but the remaining 7 are received.

The data receiver will return a TCP ACK segment that acknowledges sequence number 5000 and contains a SACK option specifying one block of queued data:

```
Relative Origin = 500;  Block Size = 3500
```

Case 3: The 2nd, 4th, 6th, and 8th (last) segments are dropped.

The data receiver will return a TCP ACK segment that acknowledges sequence number 5500 and contains a SACK option specifying the 3 blocks:

```
Relative Origin =  500;  Block Size = 500
Relative Origin = 1500;  Block Size = 500
Relative Origin = 2500;  Block Size = 500
```

Case 4: Same as Case 3, except Scale Factor S = 16.

The SACK option would specify the 3 scaled blocks:

```
Relative Origin =   32;  Block Size = 30
Relative Origin =   94;  Block Size = 31
Relative Origin =  157;  Block Size = 30
```

These three reported blocks have sequence numbers 512 through 991, 1504 through 1999, and 2512 through 2992, respectively.

3.5 Generating the SACK Option

Let us assume that the data receiver maintains a queue of valid segments that it has neither passed to the user nor acknowledged because of earlier missing data, and that this queue is ordered by starting sequence number. Computation of the SACK option can be done with one pass down this queue. Segments that occupy

contiguous sequence space are aggregated into a single SACK block, and each gap in the sequence space (except a gap that is terminated by the right window edge) triggers the start of a new SACK block. If this algorithm defines more than 10 blocks, only the first 10 can be included in the option.

3.6 Interpreting the SACK Option

The data transmitter is assumed to have a retransmission queue that contains the segments that have been transmitted but not yet acknowledged, in sequence-number order. If the data transmitter performs re-packetization before retransmission, the block boundaries in a SACK option that it receives may not fall on boundaries of segments in the retransmission queue; however, this does not pose a serious difficulty for the transmitter.

Let us suppose that for each segment in the retransmission queue there is a (new) flag bit "ACK'd", to be used to indicate that this particular segment has been entirely acknowledged. When a segment is first transmitted, it will be entered into the retransmission queue with its ACK'd bit off. If the ACK'd bit is subsequently turned on (as the result of processing a received SACK option), the data transmitter will skip this segment during any later retransmission. However, the segment will not be dequeued and its buffer freed until the left window edge is advanced over it.

When an acknowledgment segment arrives containing a SACK option, the data transmitter will turn on the ACK'd bits for segments that have been selectively acknowleged. More specifically, for each block in the SACK option, the data transmitter will turn on the ACK'd flags for all segments in the retransmission queue that are wholly contained within that block. This requires straightforward sequence number comparisons.

4. TCP ECHO OPTIONS

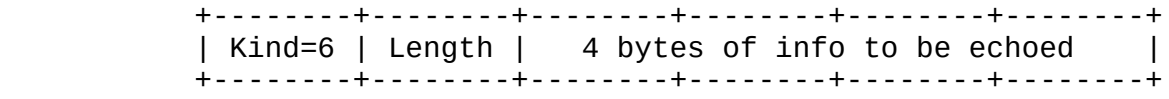
A simple method for measuring the RTT of a segment would be: the sender places a timestamp in the segment and the receiver returns that timestamp in the corresponding ACK segment. When the ACK segment arrives at the sender, the difference between the current time and the timestamp is the RTT. To implement this timing method, the receiver must simply reflect or echo selected data (the timestamp) from the sender's segments. This idea is the basis of the "TCP Echo" and "TCP Echo Reply" options.

4.1 TCP Echo and TCP Echo Reply Options

TCP Echo Option:

Kind: 6

Length: 6



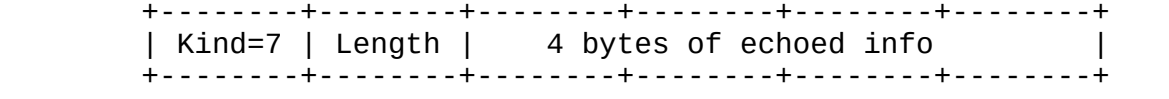
This option carries four bytes of information that the receiving TCP may send back in a subsequent TCP Echo Reply option (see below). A TCP may send the TCP Echo option in any segment, but only if a TCP Echo option was received in a SYN segment for the connection.

When the TCP echo option is used for RTT measurement, it will be included in data segments, and the four information bytes will define the time at which the data segment was transmitted in any format convenient to the sender.

TCP Echo Reply Option:

Kind: 7

Length: 6



A TCP that receives a TCP Echo option containing four information bytes will return these same bytes in a TCP Echo Reply option.

This TCP Echo Reply option must be returned in the next segment (e.g., an ACK segment) that is sent. If more than one Echo option is received before a reply segment is sent, the TCP must choose only one of the options to echo, ignoring the others; specifically, it must choose the newest segment with the oldest sequence number (see next section.)

To use the TCP Echo and Echo Reply options, a TCP must send a TCP Echo option in its own SYN segment and receive a TCP Echo option in a SYN segment from the other TCP. A TCP that does not implement the TCP Echo or Echo Reply options must simply ignore any TCP Echo options it receives. However, a TCP should not receive one of these

options in a non-SYN segment unless it included a TCP Echo option in its own SYN segment.

4.2 Using the Echo Options

If we wish to use the Echo/Echo Reply options for RTT measurement, we have to define what the receiver does when there is not a one-to-one correspondence between data and ACK segments. Assuming that we want to minimize the state kept in the receiver (i.e., the number of unprocessed Echo options), we can plan on a receiver remembering the information value from at most one Echo between ACKs. There are three situations to consider:

(A) Delayed ACKs.

Many TCP's acknowledge only every Kth segment out of a group of segments arriving within a short time interval; this policy is known generally as "delayed ACK's". The data-sender TCP must measure the effective RTT, including the additional time due to delayed ACK's, or else it will retransmit unnecessarily. Thus, when delayed ACK's are in use, the receiver should reply with the Echo option information from the earliest unacknowledged segment.

(B) A hole in the sequence space (segment(s) have been lost).

The sender will continue sending until the window is filled, and we may be generating ACKs as these out-of-order segments arrive (e.g., for the SACK information or to aid "fast retransmit"). An Echo Reply option will tell the sender the RTT of some recently sent segment (since the ACK can only contain the sequence number of the hole, the sender may not be able to determine which segment, but that doesn't matter). If the loss was due to congestion, these RTTs may be particularly valuable to the sender since they reflect the network characteristics immediately after the congestion.

(C) A filled hole in the sequence space.

The segment that fills the hole represents the most recent measurement of the network characteristics. On the other hand, an RTT computed from an earlier segment would probably include the sender's retransmit time-out, badly biasing the sender's average RTT estimate.

Case (A) suggests the receiver should remember and return the Echo option information from the oldest unacknowledged segment. Cases (B)

and (C) suggest that the option should come from the most recent unacknowledged segment. An algorithm that covers all three cases is for the receiver to return the Echo option information from the newest segment with the oldest sequence number, as specified earlier.

A model implementation of these options is as follows.

(1) Receiver Implementation

A 32-bit slot for Echo option data, rcv.echodata, is added to the receiver connection state, together with a flag, rcv.echopresent, that indicates whether there is anything in the slot. When the receiver generates a segment, it checks rcv.echopresent and, if it is set, adds an echo-reply option containing rcv.echodata to the outgoing segment then clears rcv.echopresent.

If an incoming segment is in the window and contains an echo option, the receiver checks rcv.echopresent. If it isn't set, the value of the echo option is copied to rcv.echodata and rcv.echopresent is set. If rcv.echopresent is already set, the receiver checks whether the segment is at the left edge of the window. If so, the segment's echo option value is copied to rcv.echodata (this is situation (C) above). Otherwise, the segment's echo option is ignored.

(2) Sender Implementation

The sender's connection state has a single flag bit, snd.echoallowed, added. If snd.echoallowed is set or if the segment contains a SYN, the sender is free to add a TCP Echo option (presumably containing the current time in some units convenient to the sender) to every outgoing segment.

Snd.echoallowed should be set if a SYN is received with a TCP Echo option (presumably, a host that implements the option will attempt to use it to time the SYN segment).

5. CONCLUSIONS AND ACKNOWLEDGMENTS

We have proposed five new TCP options for scaled windows, selective acknowledgments, and round-trip timing, in order to provide efficient operation over large-bandwidth*delay-product paths. These extensions are designed to provide compatible interworking with TCP's that do not implement the extensions.

The Window Scale option was originally suggested by Mike St. Johns of USAF/DCA. The present form of the option was suggested by Mike Karels of UC Berkeley in response to a more cumbersome scheme proposed by Van Jacobson. Gerd Beling of FGAN (West Germany) contributed the initial definition of the SACK option.

All three options have evolved through discussion with the End-to-End Task Force, and the authors are grateful to the other members of the Task Force for their advice and encouragement.

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