Silly Window Syndrome

Another problem with tinygrams is manifested as *silly window syndrome*

TCP Strategies Silly Window Syndrome



data

Silly Window Syndrome

A is sending data to B, but B's buffer is nearly full and B is reading only one byte at a time;

Silly Window Syndrome



B's buffer fills, and B ACKs with a window of 0;

Silly Window Syndrome

Silly Window Syndrome



A holds off sending more data;

Silly Window Syndrome

Silly Window Syndrome



B reads a byte;

Silly Window Syndrome



Silly Window Syndrome

B sends a window update segment, size 1;

TCP Strategies Silly Window Syndrome



A get this and sends as much data as possible, i.e., 1 byte;

Silly Window Syndrome

Silly Window Syndrome



Silly Window Syndrome

B ACKs with window 0;

Silly Window Syndrome



B reads a byte;

Silly Window Syndrome



B sends an update, size 1;

Silly Window Syndrome

Silly Window Syndrome





Silly Window Syndrome



And so on

Silly Window Syndrome



We are back to the two segment per byte high overhead: this is silly window syndrome



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Better is for B not to send an update of 1, but wait until there is more space



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Better is for B not to send an update of 1, but wait until there is more space

Clarke's algorithm to avoid SWS is in the server

never send an update for a window of 1; only advertise a new window when either (a) there is enough space for a full segment, or (b) the buffer is half empty



Nagle (in the client) and SWS (in the server) fit together naturally



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Note that TCP code doesn't have to implement Nagle or SWS or delayed ACKs or any of these strategies: it's just a good idea if it does!



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We need to look at the case of sending large amounts of data



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We need to look at the case of sending large amounts of data

We want the data to get to the destination as fast as possible, but we now have to consider not just the ability of the destination to cope, but also the capacity of the network itself



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There are several strategies in TCP to help deal with and avoid congestion

The first issue is how to spot congestion, given that it might be happening in a part of the network many hops away from both source and destination

Congestion

We watch for segment loss



Segments can be lost though errors in transmission or being dropped at a congested router (or at the destination)



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Thus TCP treats missing or duplicate ACKs as a sign of congestion

Exercise A missing ACK is understandable as a sign of congestion: reflect briefly on why *duplicate* ACKs can be caused by congestion



Congestion somewhere on the path

Congestion can happen in a router due to lack of capacity in an onward link; a router will drop a packet if it can't cope



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So how do we determine the congestion window? It's not a thing the source or destination can know directly

We do this by sending segments and watching what ACKs we get



If we have a lot of data to send we do not want to wait for each ACK before sending the next segment



If we have a lot of data to send we do not want to wait for each ACK before sending the next segment

Better is to send several segments and then wait to see from the ACKs which were safely received



But sending too many segments at once is bad when the network is congested: our segments will be dropped. We'll just be making things worse for everyone, including ourselves



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If we get it right, we will have a continual stream of segments going out and ACKs coming back

Slow Start & Congestion Avoidance

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TCP Strategies Slow Start & Congestion Avoidance

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This is an another constraint on sending additional to the advertised window: it's a bad idea to send more data than indicated by the either window

Slow Start & Congestion Avoidance

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Every time a timely ACK is received, the congestion window is increased by one segment



data ACK

Poor use of bandwidth

Slow Start (no delayed ACKs)

Slow Start with no delayed ACKs



Poor use of bandwidth

Slow Start

Slow Start with delayed ACKs

Slow Start & Congestion Avoidance

So initially we send one segment

Slow Start & Congestion Avoidance

So initially we send one segment

Then two at a time

Slow Start & Congestion Avoidance

So initially we send one segment

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Then four...

Slow Start & Congestion Avoidance

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This is called *slow start*

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In slow start, the increase continues until we reach the current threshold *ssthresh* or returning ACKs are duplicated or timed out

TCP Strategies Slow Start & Congestion Avoidance

Of course, the rate is also limited by the advertised window of the destination: we can only send the minimum of the current congestion window and the advertised window

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Note that the congestion window is a limit set by the sender, while the advertised window is a limit set by the receiver

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This is now a linear increase over time

Slow Start & Congestion Avoidance



Slow start and congestion avoidance regions

Slow Start & Congestion Avoidance

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Note we might be in either of the slow start or the congestion avoidance phases when congestion occurs: particularly if ssthresh was initially set very large, as its often done these days

Slow Start & Congestion Avoidance

When congestion is detected

Slow Start & Congestion Avoidance

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Slow Start & Congestion Avoidance

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- if it was a timeout, the congestion window cwnd is set back to one segment, and go back into slow start
- when ACKs start coming through, we resume increasing the congestion window again, according to whether we were in slow start or congestion avoidance (i.e., whether cwnd is less than ssthresh or not)



Converging on the optimum rate

The sender eventually converges on a rate that is neither too fast, nor too slow

Slow Start & Congestion Avoidance

And it is dynamic

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Slow Start & Congestion Avoidance

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If conditions on the network change, it will soon adapt to the new rate, be it faster or slower

If there is no congestion on the network, the rate increases until it reaches the advertised window: the limiting factor is then the destination, not the network

This strategy is very effective: get the flow up quickly, but don't overshoot network capacity. Also, back off quickly and try again when a loss happens




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Jacobson's Fast Retransmit strategy builds on the idea that the receipt of several duplicated ACKs is indicative of a lost segment



Recall that the argument is that one or two duplicate ACKs might simply be due to out-of-order delivery, as IP is unreliable



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If this happens, the sender should retransmit the indicated segment immediately: fast retransmit





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Exercise Read RFC2001 for the details we have not mentioned here

TCP Strategies

Congestion

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Congestion

There have been many tweaks to this basic flow control strategy

- Larger initial ssthresh
- Larger initial cwnd
- Slow start counting number of segments ACKed, not just the number of ACKs
- Treating duplicate ACKs like a timeout
- On timeout, setting cwnd to half ssthresh, not just 1 segment
- Fast recovery: wait for the ACK of the entire transmit window before entering congestion avoidance
- Many more



Exercise Read about other strategies, such as TCP Reno, TCP Vegas, TCP New Reno, TCP Hybla, BIC, CUBIC, etc.





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Exercise Read about this



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Particularly *Explicit Congestion Notification* (ECN), which aims to indicate congestion *before* it happens by routers setting flags in the IP TOS/DS header when they think congestion is imminent, so that the hosts get forewarning and can slow down

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Exercise Read about ECN and its use of flags in both the IP header and the TCP header



Exercise Read about Random Early Detection/Drop (RED), which is also used in routers



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Exercise We use ICMP to indicate other kinds of errors, but why is it not a good idea to use ICMP when a router drops a packet due to congestion?



tcpdump

 $\mbox{Exercise}$ Use tcpdump to watch these strategies in operation. The netcat program is an easy way to set up connections and send data

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IP layer fragmentation is expensive, so we employ path MTU discovery: but now we need to look at it from a TCP perspective

TCP has (potentially) more information: namely the optional MSS header sent in the setup handshake

We can send segments of decreasing size, starting with the minimum of the MSS of the sending interface and the MSS announced by the other end, or 536 if the other end did not give an MSS

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Note the cross-layer activity here!



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It is recommended you try a larger MTU once in a while, e.g., every 10 minutes, as routes can vary dynamically