

TCP Timers

Next: TCP has several timers. We have seen

- 2MSL
- Delayed ACK

These are just the start!

TCP Timers

Retransmission Timer

We now consider the timer that determines when to resend in the absence of an ACK: a *retransmission timeout* (RTO)

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And we want a dynamic behaviour that adapts to changing conditions rather than a simple fixed timeout

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If we haven't received an ACK in approximately this time, deem it lost

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α is a smoothing factor, usually 7/8 for easy arithmetic

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True standard deviations are tricky to compute quickly (square roots), so Jacobson suggested using the *mean deviation*

TCP Timers

Retransmission Timer

Mean deviation:

$$D = \beta D + (1 - \beta)|RTT - M|$$

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A typical value for β is $3/4$

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When sending a segment (or, in practice, a burst of segments) set the timer to expire after time T

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- but we also need to update RTT somehow

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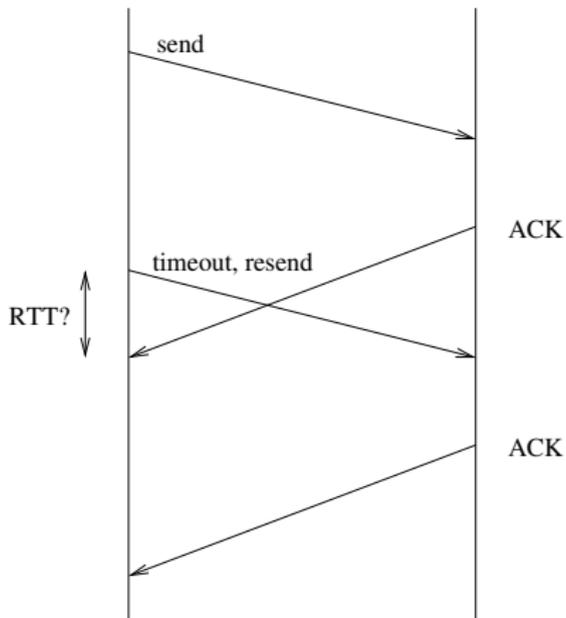
What if the timer expires before the ACK is received?

- we resend the segment, of course
- but we also need to update RTT somehow

But we can't use RTT of the resent segment as we might get the somewhat delayed ACK of the original segment, not of the resent segment

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Retransmission Ambiguity

This is the *retransmission ambiguity problem*

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Alternatively, as is common these days, we have the option header timestamp and this solves the retransmission ambiguity directly

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Persist Timer

The next timer in TCP is the *persist timer*, sometimes called the *persistence timer*

TCP Timers

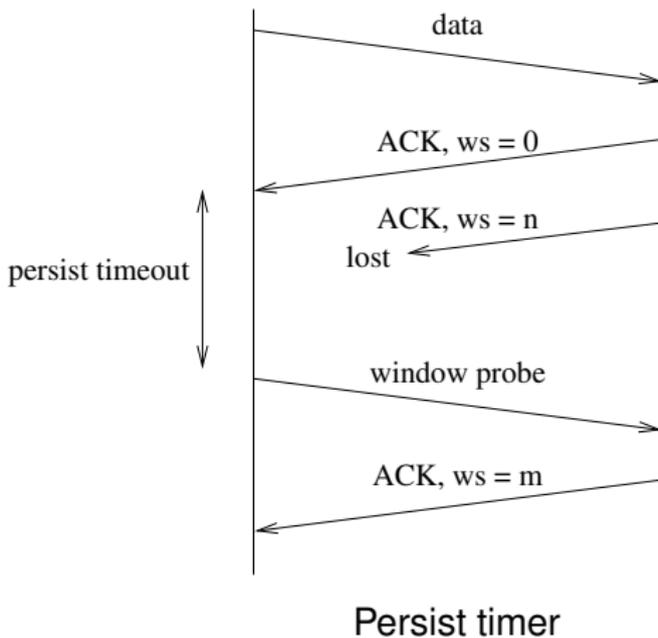
Persist Timer

The next timer in TCP is the *persist timer*, sometimes called the *persistence timer*

Its role is to prevent deadlock through the loss of window update segments

TCP Timers

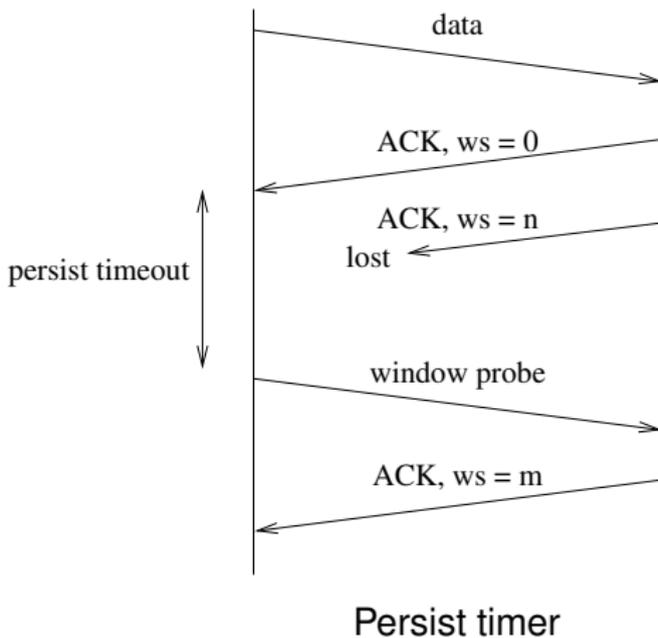
Persist Timer



A sends to B;

TCP Timers

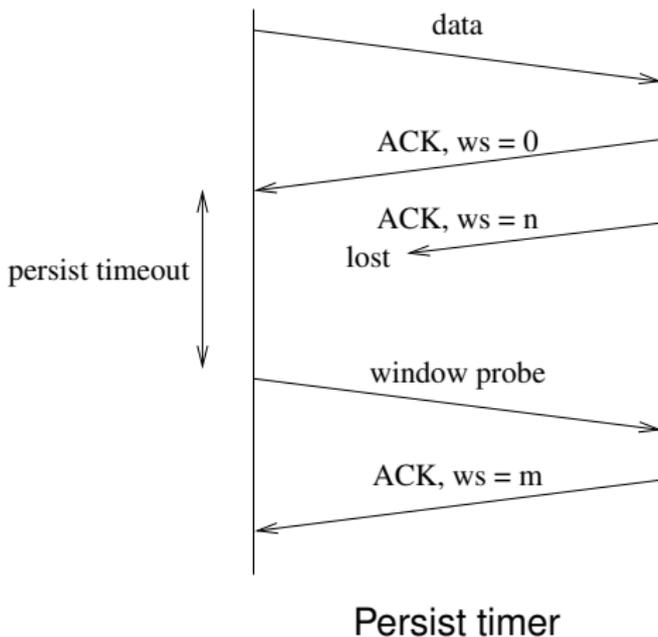
Persist Timer



B replies with an ACK and a window size of 0;

TCP Timers

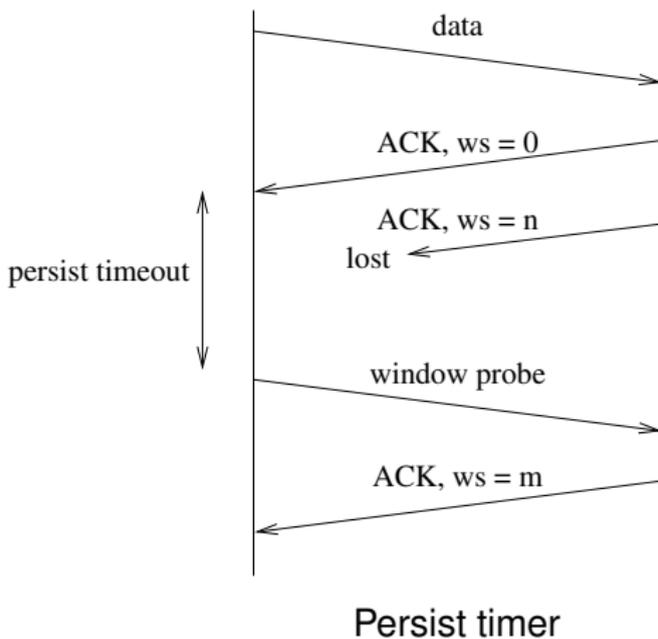
Persist Timer



A gets the ACK and holds off sending to B;

TCP Timers

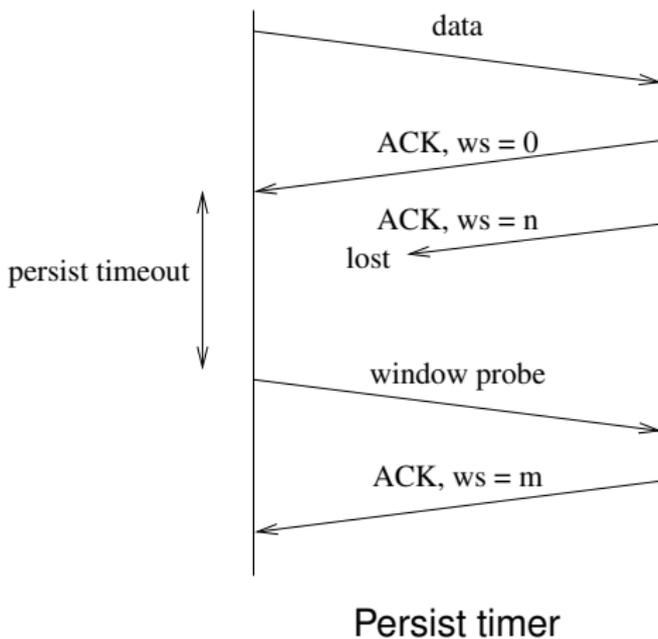
Persist Timer



B frees up some buffer space and sends a window update to A;

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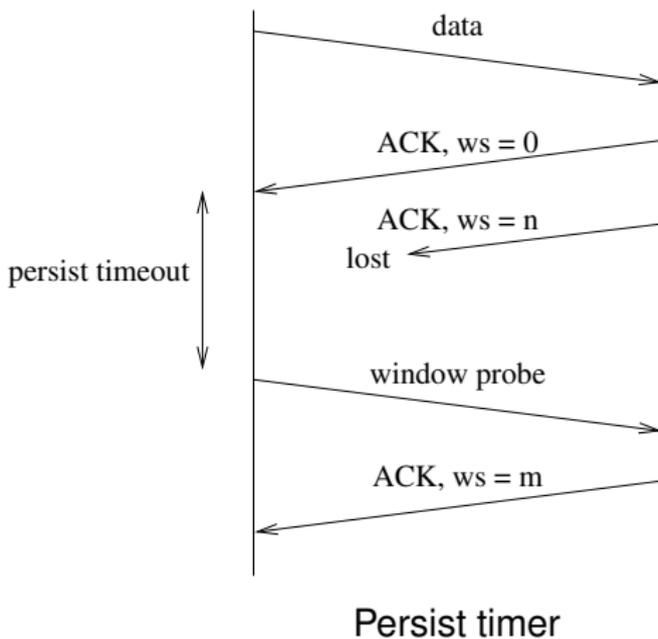
Persist Timer



This is lost;

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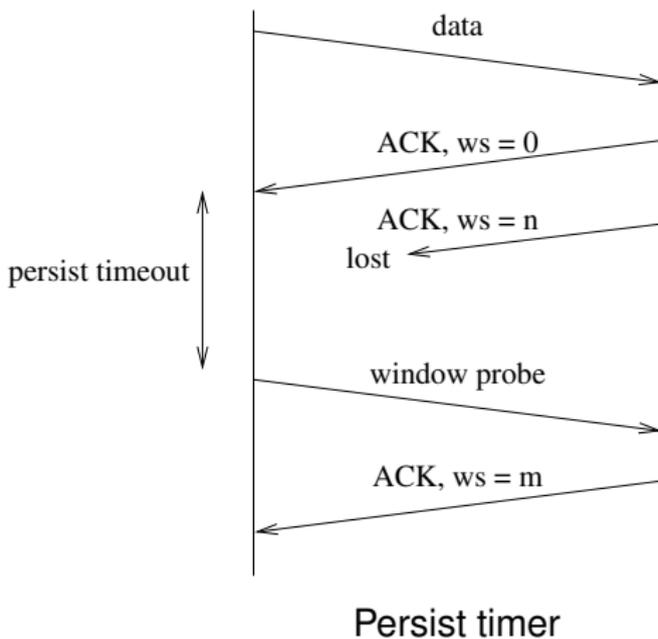
Persist Timer



Now A is waiting for the window update from B and B is waiting for more data from A: deadlock;

TCP Timers

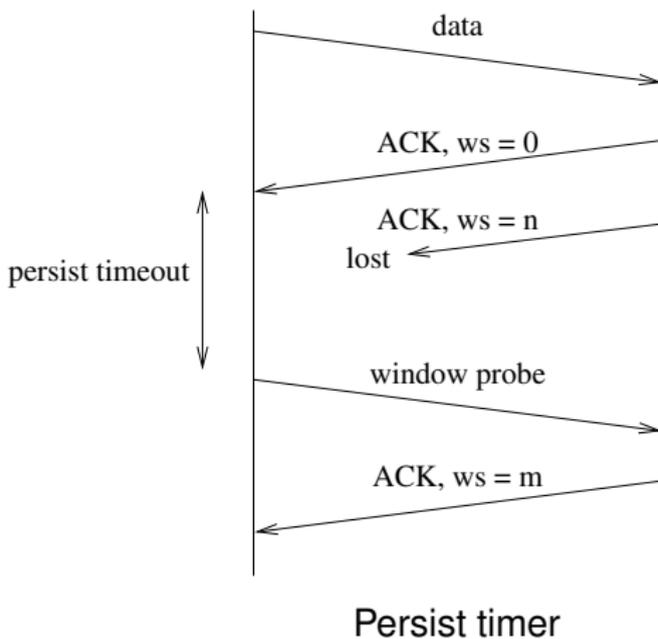
Persist Timer



To prevent this, A starts the persist timer when it gets the 0 window from B;

TCP Timers

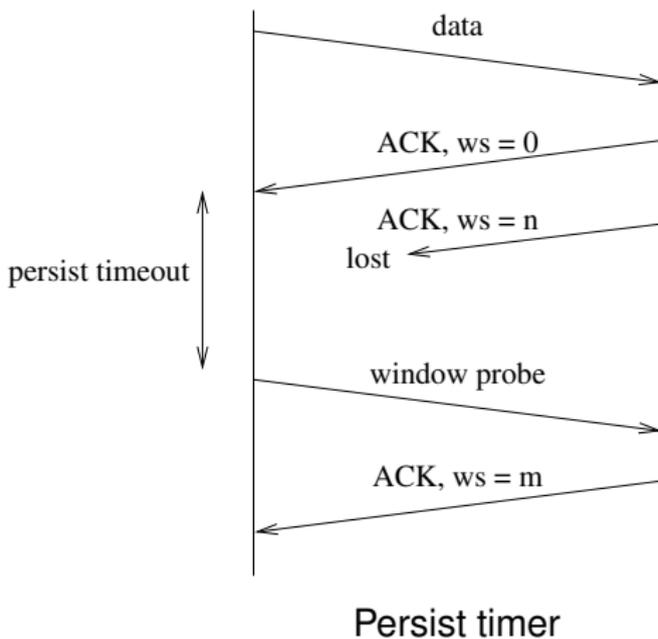
Persist Timer



If the timer expires, A prods B by sending a 1 byte segment: a *window probe*;

TCP Timers

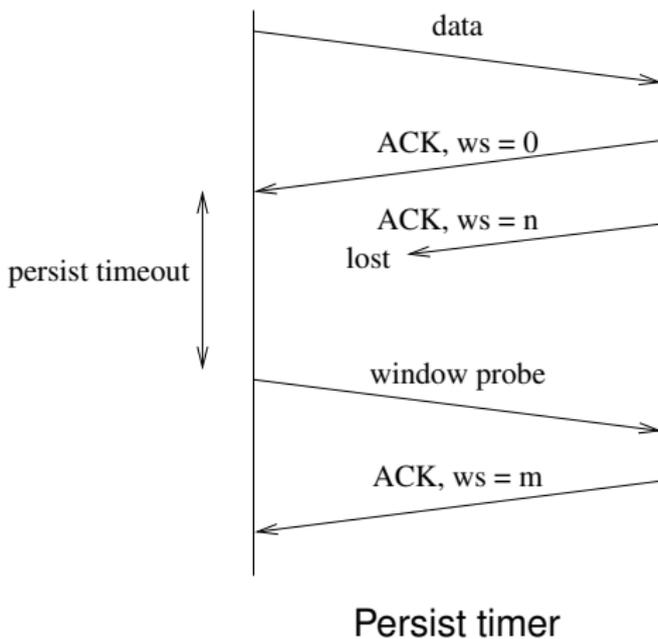
Persist Timer



If B gets this, the ACK will contain B's current window size;

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Persist Timer



If the window is still 0, A resets the timer and tries again later

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The persist timer is unset when a non-zero window is received

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This gives us a bit of resilience against flaky networks

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A typical value is 2 hours

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If not, the server might conclude the client is no longer active

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1. the client is up and running: the keepalive probe is ACKed and everybody is happy. The keepalive timer is reset to 2 hours
2. the client has crashed or is otherwise not responding to TCP: the server gets no ACK and resends after 75 seconds. After 10 probes, 75 seconds apart, if there is no response, the server terminates the connection with “connection timed out” sent to the server application

TCP Timers

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3. the client has crashed and rebooted. The client gets the probe and responds with a RST. The server gets the RST and terminates the connection with “connection reset by peer” sent to the application

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3. the client has crashed and rebooted. The client gets the probe and responds with a RST. The server gets the RST and terminates the connection with “connection reset by peer” sent to the application
4. the client is up and running, but is unreachable, e.g., broken routing. This is indistinguishable from case 2, so the same events ensue

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- some network operators charge per packet

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It is usually possible to disable keepalive in the application: some people think that keepalive should not be in the TCP layer, but should be handled by the application layer (i.e., the non-existent session layer)

TCP Strategies

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Exercise Read about Protect Against Wrapped Sequence numbers (PAWS), Selective Acknowledgement (SACK)

TCP Extensions

Exercise Multipath TCP (MPTCP) has been suggested both for extra performance, failover and for mobile hosts that roam between, say, cellular and Wi-Fi (used in iOS7). It layers one MPTCP connection over one or more TCP connections, e.g., using both the cellular and Wi-Fi links simultaneously for one MPTCP connection

Exercise And potential alternatives to TCP. Read about TCP for Transactions (TTCP), Stream Control Transmission Protocol (SCTP), Datagram Congestion Control Protocol (DCCP)

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Occasionally it's called “TCP/2”, but it's not TCP

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In fact, the QUIC header is encrypted (inside the UDP packet) to prevent routers inspecting or trying to modify it

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This saves time over the current schemes that open TCP and then establishes encryption (see TLS, later)

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For example, a Web page might fetch dozens of items (text, images, JavaScript, . . .) from the same server

These could all be sent within a *single* QUIC connection

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Current browsers do try to multiplex multiple streams over a single TCP connection, but this causes problems as an error in one stream causes TCP's error mechanisms to kick in, affecting *all streams* in the connection, even if the other streams had no error in themselves

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QUIC manages errors at the stream level, not the connection level

TCP Alternatives

And:

- more sophisticated ACK mechanisms
- connection migration, e.g., WiFi to cellular
- sophisticated flow control (still under development)
- and lots of other stuff building on the knowledge gained since TCP was first invented

TCP Alternatives

QUIC is growing, but it will be a long time before it replaces TCP (lots of code to rewrite!)

Exercise Read about how QUIC reduces connection overheads and about the *head-of-line blocking* problem

Exercise Read about SPDY, the predecessor to QUIC, and its relationship to HTTP/2

Exercise Read about the *middlebox* (router) problem and why it means that new protocols will have a hard time on the Internet

UDP Alternatives

Exercise And don't forget UDP: UDPLite, RUDP, UDT, etc.

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It took a lot of work, though!

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(which are 80, 108, 97, 110 in decimal), what did I mean?

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Least significant byte first or most significant byte first?

Presentation

Or is it floating point number?

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Or is it a string of four characters?

Presentation

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Or is it a string of four characters?

In ASCII? Or some other encoding like EBCDIC, or UTF-8?

Presentation

From another point of view:

I want to send "Plan" to you. What do I send?

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If we use some other encoding, it might need more than four bytes

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What do I do if we use different encodings? Perhaps my machine uses ASCII while yours uses EBCDIC

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What do I do if I don't know what encoding you use?

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This is the problem of *presentation*

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Bits are just bits unless they have some agreed-on meaning

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Particularly as some people forget that not everyone uses the same representations for everything