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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- too long a time is poor for the data rate

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

TCP Timers

Retransmission Timer

The timeout value is set to

$$T = RTT + 4D$$

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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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What if the timer expires before the ACK is received?

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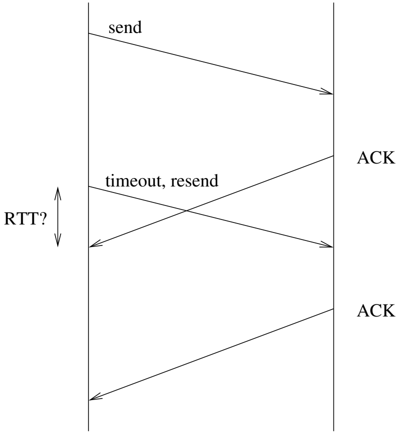
- but we also need to update RTT somehow

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

TCP Timers

Retransmission Timer



Retransmission Ambiguity

This is the *retransmission ambiguity problem*

TCP Timers

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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

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

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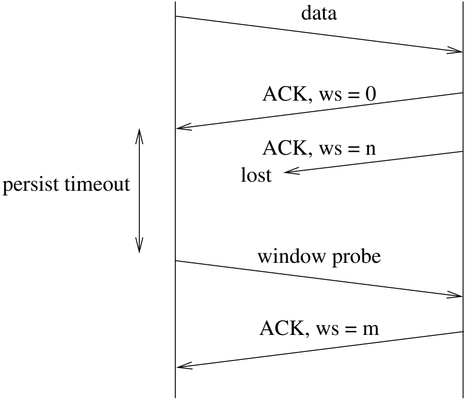
Persist Timer

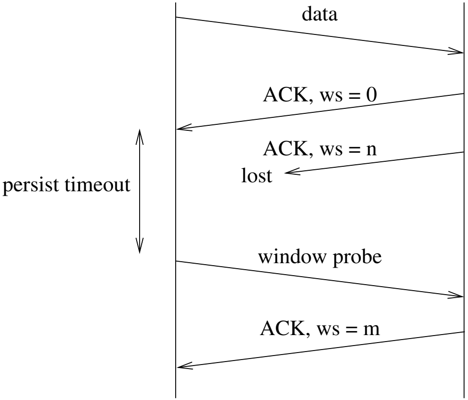


Persist timer

A sends to B;

TCP Timers

Persist Timer

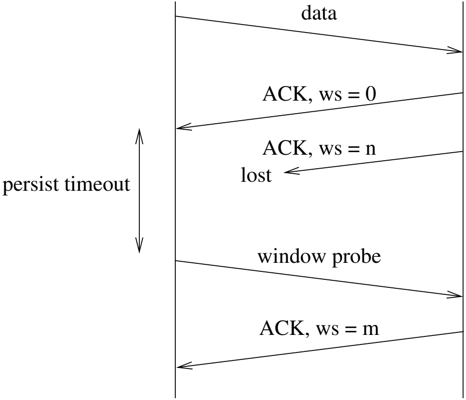


Persist timer

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

TCP Timers

Persist Timer

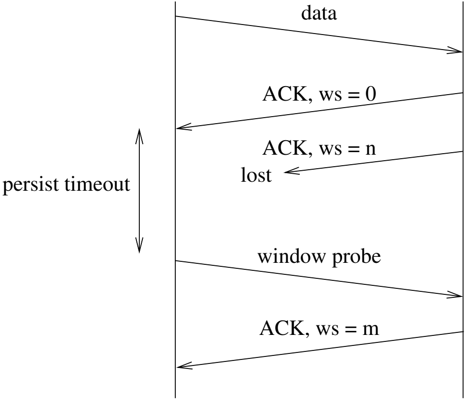


Persist timer

A gets the ACK and holds off sending to B;

TCP Timers

Persist Timer

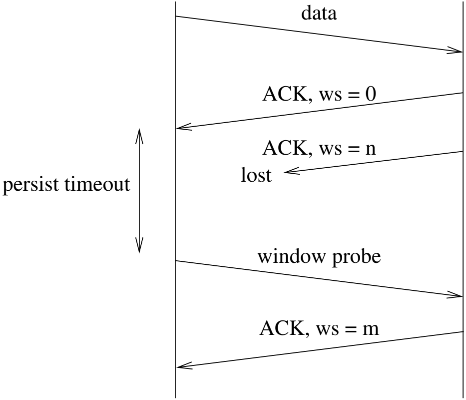


Persist timer

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

TCP Timers

Persist Timer

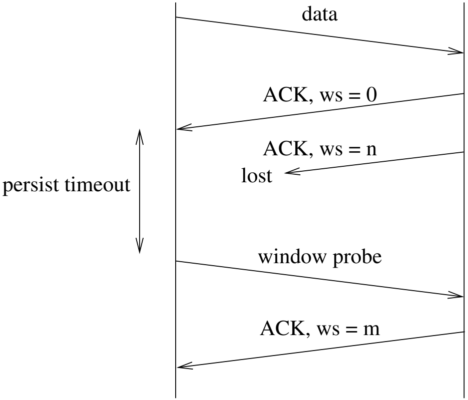


Persist timer

This is lost;

TCP Timers

Persist Timer

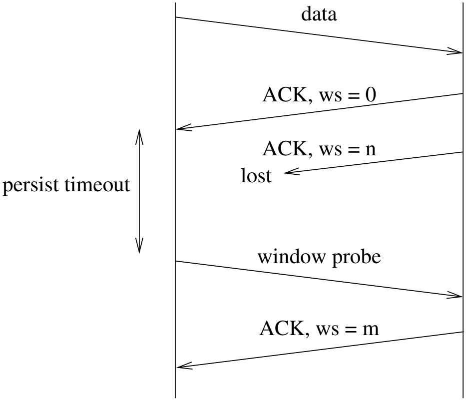


Persist timer

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

TCP Timers

Persist Timer

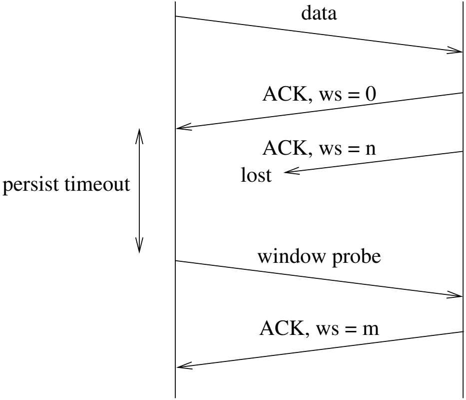


Persist timer

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

TCP Timers

Persist Timer

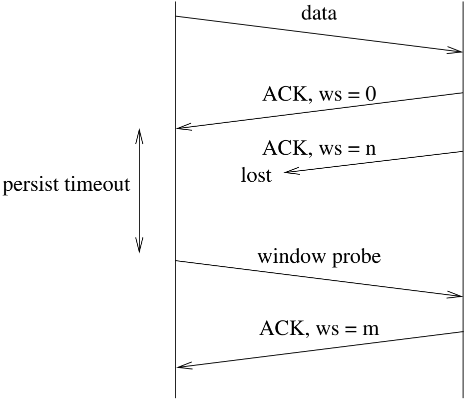


Persist timer

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

TCP Timers

Persist Timer

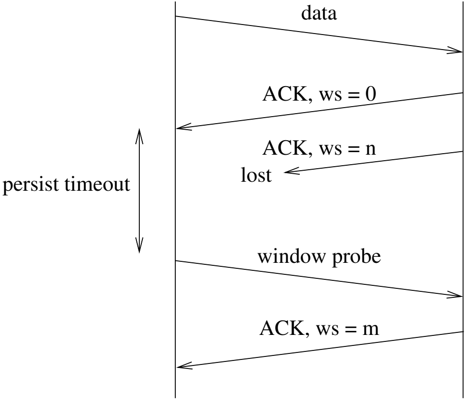


Persist timer

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

TCP Timers

Persist Timer



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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Keepalive Timer

Yet another timer in TCP is the *keepalive*

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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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Keepalive Timer

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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

Keepalive Timer

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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- they use bandwidth
- some network operators charge per packet

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Keepalive Timer

The latter two are not particularly good arguments as the cost is just a couple of packets every 2 hours

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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

Many other strategies to improve throughput have been proposed

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Exercise Read about the problems of *long fat pipes*

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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)

TCP Alternatives

QUIC (originally “quick UDP Internet connection”, now just a name, not an acronym) is a Google-originated alternative to TCP (RFC9000)

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Originally designed as a transport layer for HTTP/3 (the next version of HTTP), QUIC can be used as a general transport protocol

It is reliable, connection oriented, has congestion control, is encrypted and authenticated and is transmitted within UDP datagrams (port 443, mostly)

TCP Alternatives

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There have been several new protocols in the past that have failed to gain popular use as routers would not recognise them

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

TCP Alternatives

Note: QUIC uses UDP purely to avoid router problems: it would be better to layer directly over IP, but history won't let us do that

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QUIC is *not* a lightweight protocol: it is as heavyweight as TCP+TLS

It is "quick" in the sense of "fast", not "simple"

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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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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 does this multiplexing more efficiently, never stopping a good stream within a connection

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

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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.

TCP

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TCP is a huge success: from 1200 bits/sec telephone lines to gigabit networks and beyond it has turned out to be massively flexible and scalable

It took a lot of work, though!

TCP

Here is a small part of the output from `ss -io` (socket statistics) on a Linux machine:

```
tcp   ESTAB 0 0 172.16.2.1:34956  34.117.14.220:https
timer:(keepalive,31sec,0)
ts sack cubic wscale:7,7 rto:220 rtt:18.341/0.5 ato:40 mss:1368
pmtu:1420 rcvmss:647 advmss:1368 cwnd:2 ssthresh:7
bytes_sent:7179 bytes_retrans:240 bytes_acked:6939
bytes_received:6747 segs_out:515 segs_in:508 data_segs_out:198
data_segs_in:188 send 1.19Mbps lastsnd:28652 lastrcv:29228
lastack:28632 pacing_rate 2.39Mbps delivery_rate 634kbps
delivered:191 app_limited busy:32268ms retrans:0/8
rcv_space:13800 rcv_ssthresh:64156 minrtt:17.318
```