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A lot of state about each connection needs to be stored by the OS





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But ACKs on their own do not solve all the problem

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So B sends an acknowledgement to A: "OK"

But the ACK might be intercepted and A might not get the ACK





So A should send an ACK for the ACK back to B



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But this might not get through...



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For full reliability it looks like we need an infinite regress!

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Repeat until A gets an ACK (or A gives up)

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- Flow control: how to increase the rate of sending packets when things are going well, and decrease the rate when they are not



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A TCP header is complicated as it must address many complex issues

16 bit source port 16 bit destination port	Î
32 bit sequence number	
32 bit acknowledgement number	20 bytes
$\begin{array}{c} 4 \text{ bit header length} \\ \begin{array}{c} 4 \text{ bits } \\ \text{reserved} \end{array} \xrightarrow{\mathbb{Q}} \mathbb{P} \xrightarrow{\mathbb{Q}} \mathbb{Q} \\ \xrightarrow{\mathbb{Q}} \mathbb{P} \xrightarrow{\mathbb{Q}} \mathbb{Q} \\ \xrightarrow{\mathbb{Q}} \mathbb{Q} \end{array} \xrightarrow{\mathbb{Q}} \mathbb{Q} \xrightarrow{\mathbb{Q}} \xrightarrow{\mathbb{Q}} \xrightarrow{\mathbb{Q}} \xrightarrow{\mathbb{Q}} \xrightarrow{\mathbb{Q}} \\ \xrightarrow{\mathbb{Q}} \mathbb{Q} \xrightarrow{\mathbb{Q}} \xrightarrow{\mathbb{Q}} \xrightarrow{\mathbb{Q}} \xrightarrow{\mathbb{Q}} \end{array} $ $16 \text{ bit window size}$	
16 bit checksum 16 bit urgent pointer	
Options	

TCP header



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- Two 32 bit values: sequence and acknowledgement

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So if a segment contains 10 bytes of data, the sequence number on the next segment sent will be 10 greater

The sequence number in the header is the number of the first byte of data in the segment

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The destination acknowledges those bytes it has received by filling in the ACK field with the appropriate byte number and setting the ACK flag

The reverse connection from destination to source has its own sequence number as TCP is fully duplex

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Everything we say here is true for data travelling in the reverse direction: the reverse traffic has its own independent sequence numbers and flow control

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Another reason to avoid fragmentation

Sequence numbers

The returning ACK field contains the sequence number of the next byte the destination expects to receive, e.g., if the sequence number is 20001 and 14 bytes are received it returns 20015 in the ACK field

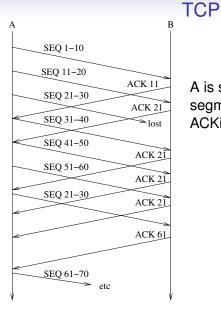
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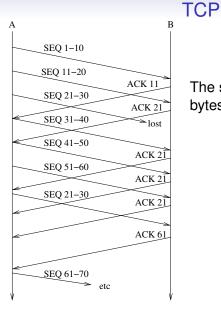
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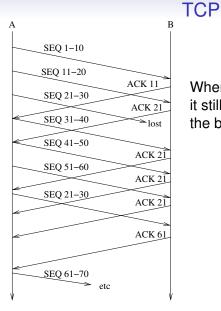
This helps reduce the amount of network traffic



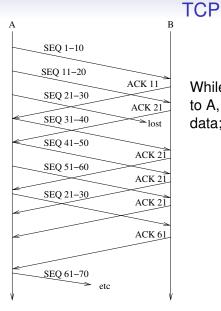
A is sending 10 byte segments to B, and B is ACKing them;



The segment containing bytes 21-30 is lost;

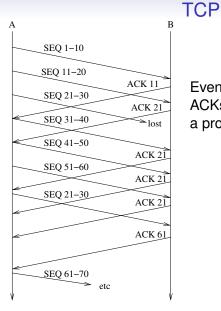


When B next gets a segment it still ACKS with 21: that's the byte it wants next;

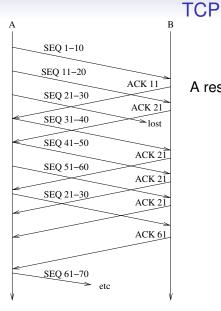


While the ACK travels back to A, A is still sending new data;

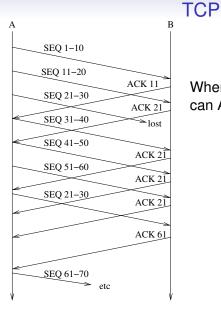
ACKing lost segments



Eventually A gets duplicate ACKs from B: this is a sign of a problem;



A resends bytes 21-30;



When B gets these bytes it can ACK all the way up to 60

Sequence numbers

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TCP specifies that A should continue until it get gets *three duplicate ACKs* (i.e., four ACKs with the same sequence number, not piggybacked on data and not changing the advertised window) before resending

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Exercise When might we receive many ACKs with the same sequence number, but nothing is in error?

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Much more on SEQ and ACKing later, but note that sequence numbers solve the segment ordering problem, too



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Many flags performing various functions

Most of these will be described in more detail as we go along:

- URG: urgent data
- ACK: the acknowledgement field is active
- PSH: push this data to the application as fast as possible
- RST: reset (break) the connection
- SYN: synchronise a new connection
- FIN: finish a connection
- ECE: congestion notification
- CWR: congestion window reduced
- 4 reserved bits, set to 0