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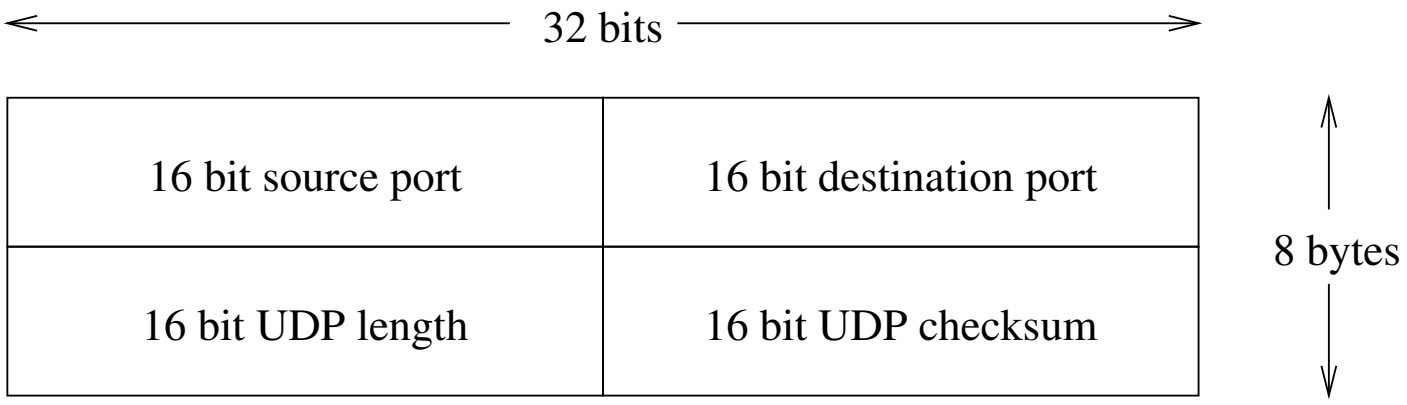
UDP packets are typically called *datagrams* (like telegrams: simple individual messages)

UDP

Header



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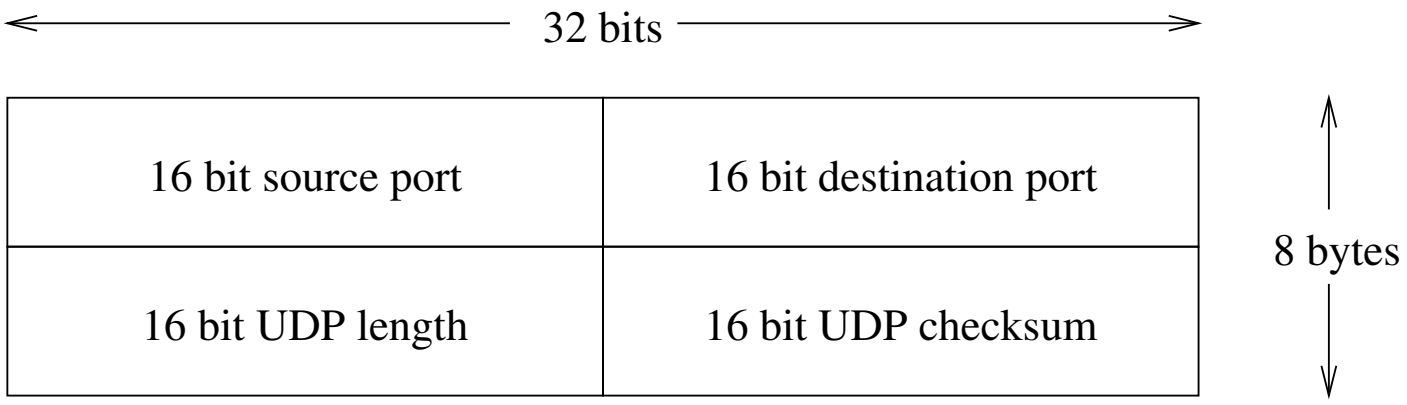

UDP header

- Ports: as described

UDP

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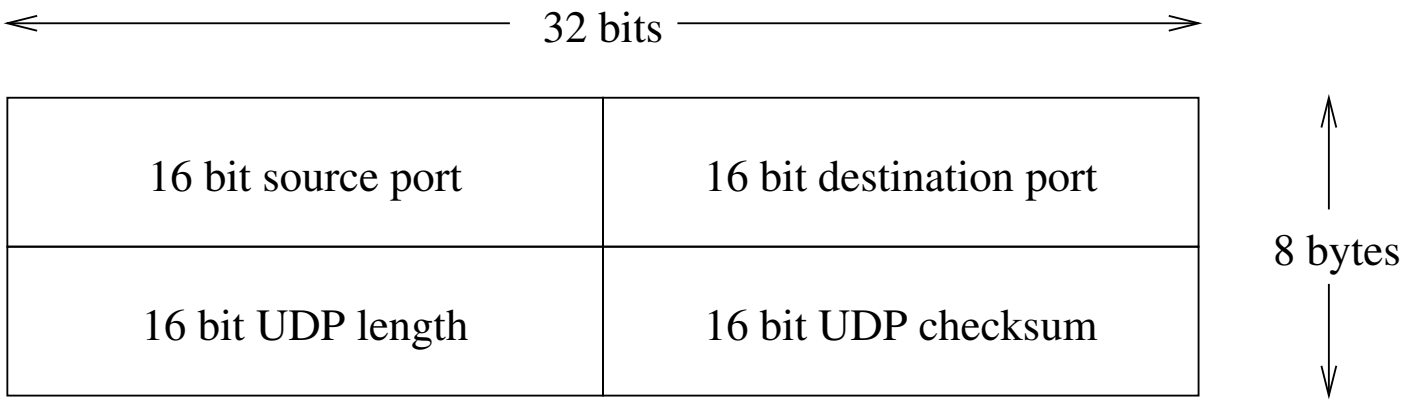

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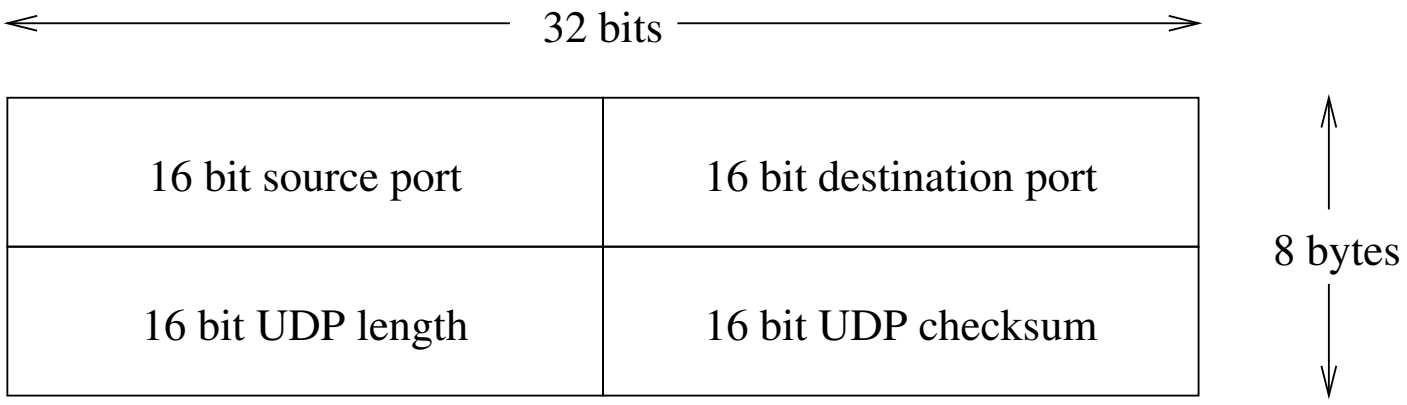

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- Length: of the entire packet, including the 8 bytes of the header: this could be deduced from the IP layer, but this keeps layer independence
- Checksum: of the UDP header, the data *and some fields from the IP header*

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The checksum is optional: put 0 in this field if you want to save a little time: recall UDP is unreliable!

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It is just about as fast and efficient as IP, with only a small overhead (8 bytes)

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- One shot applications. Where we have a single request and reply. For example, DNS
- Where a fast response is required. We have no overhead in setting up a connection before data can be exchanged (see TCP). E.g., DNS
- Where speed is more important than accuracy. For example, media streaming, where the occasional lost packet is not a problem, but a slow packet is

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A video streamer might just patch over a lost packet with a copy of a previous packet; and so on

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Thus the need for TCP

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

TCP

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But ACKs on their own do not solve all the problem

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

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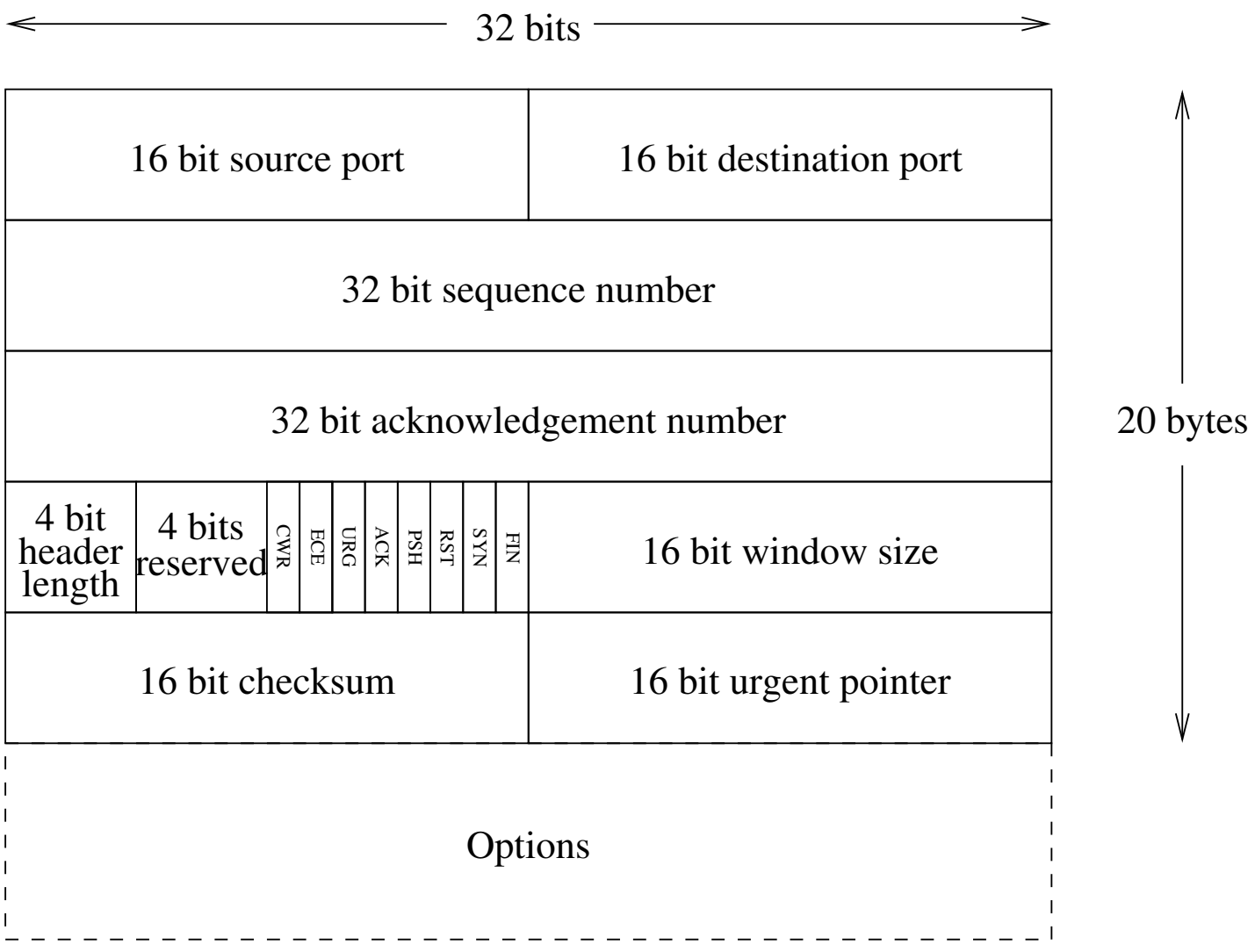
A TCP header is complicated as it must address many complex issues

TCP



TCP header

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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

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Sequence numbers

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 and setting the ACK flag

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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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This helps reduce the amount of network traffic

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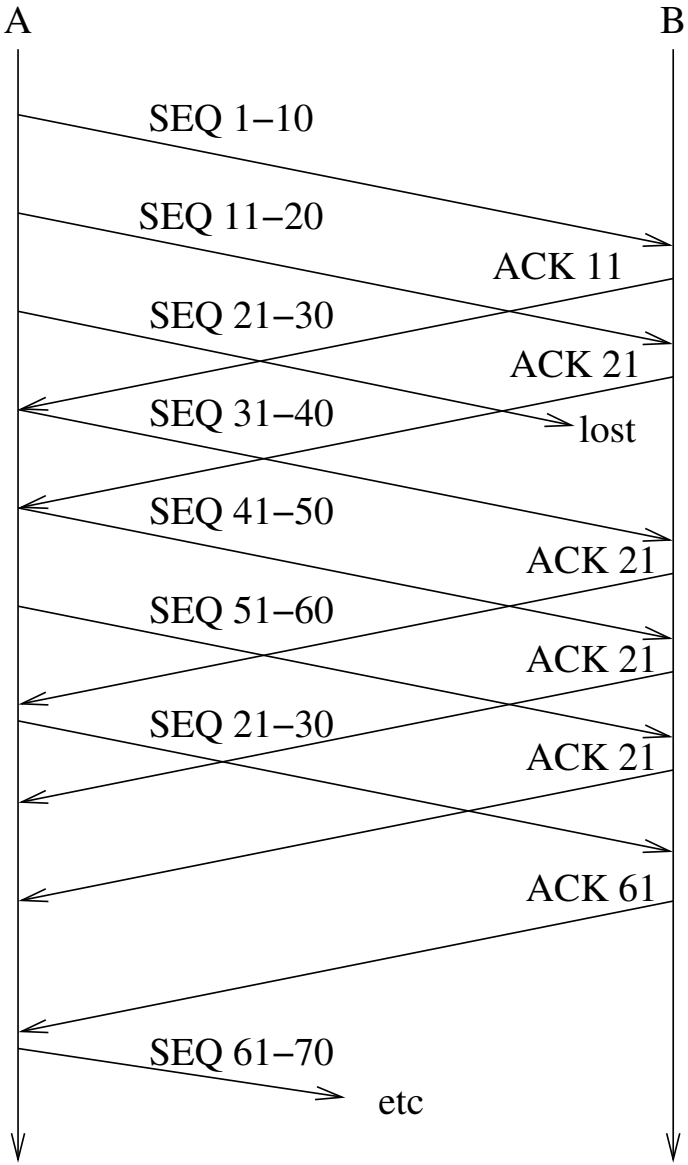
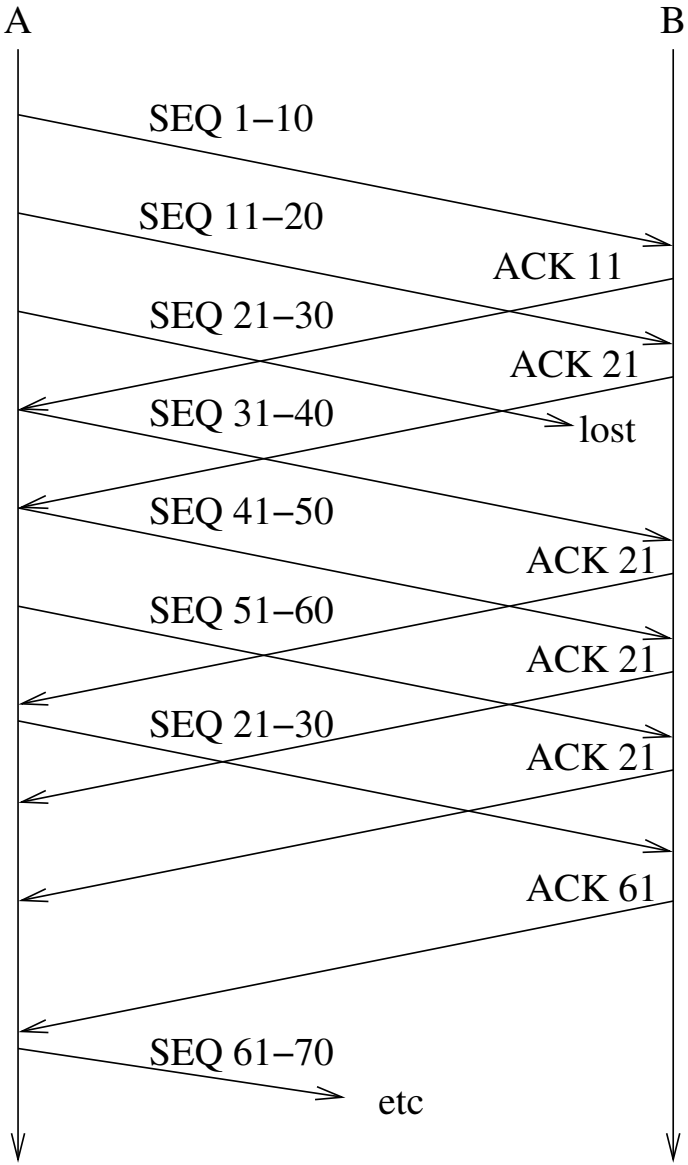


ACKing lost segments

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

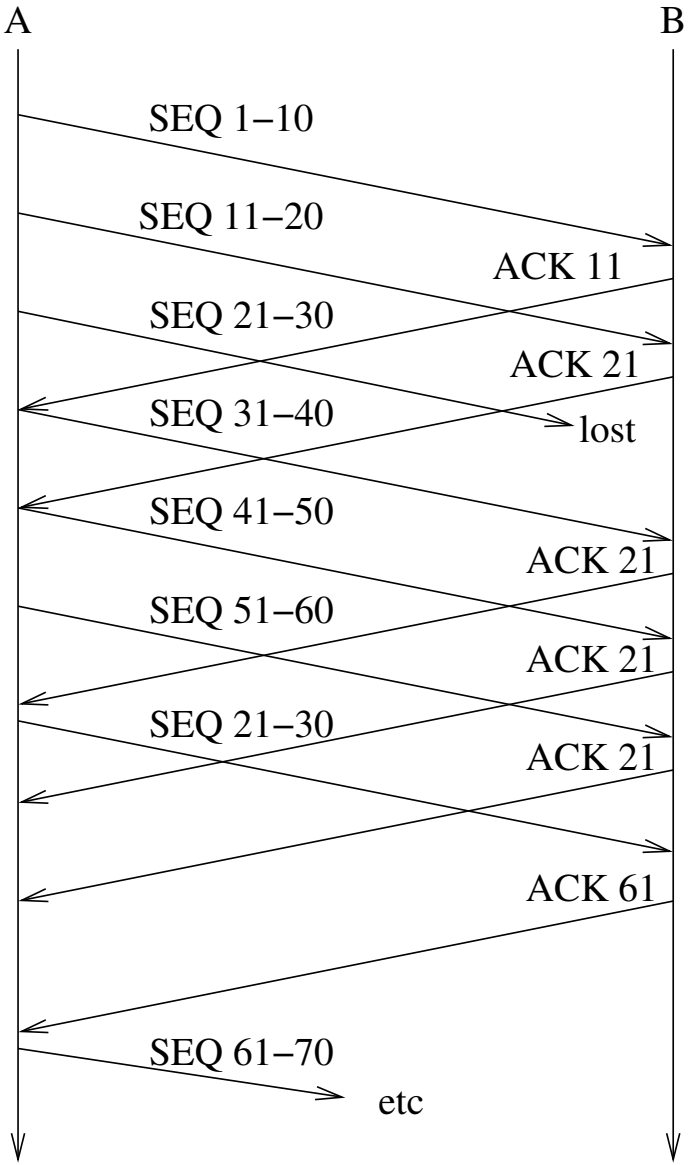
TCP



ACKing lost segments

The segment containing bytes 21-30 is lost;

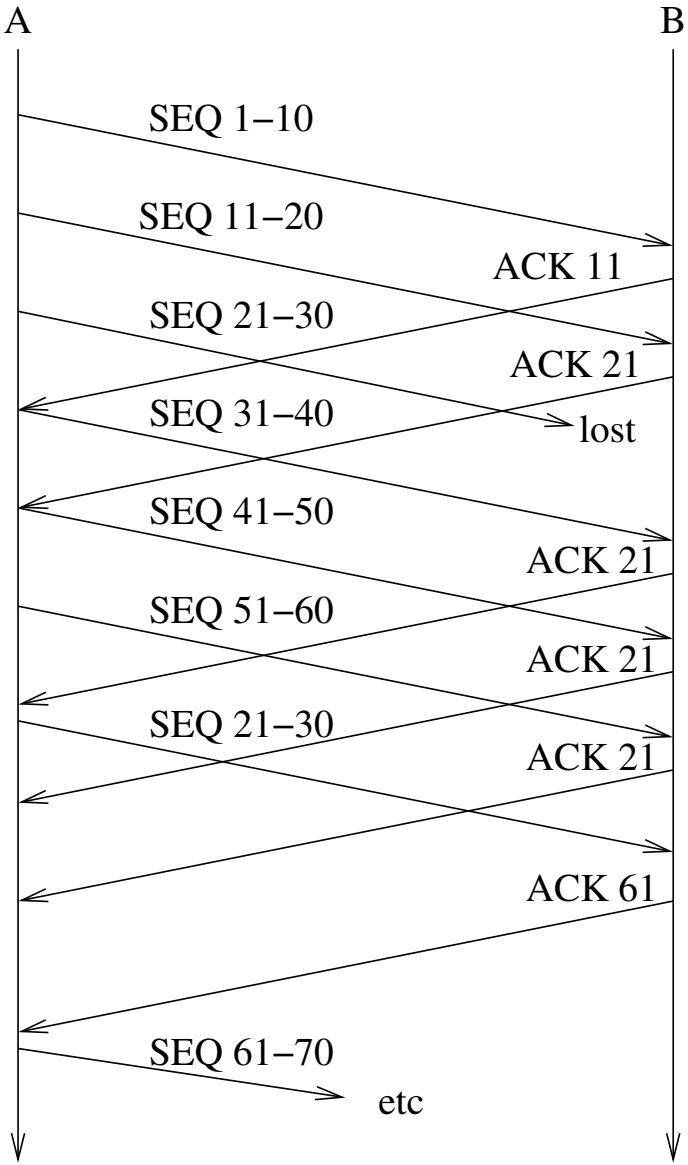
TCP



ACKing lost segments

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

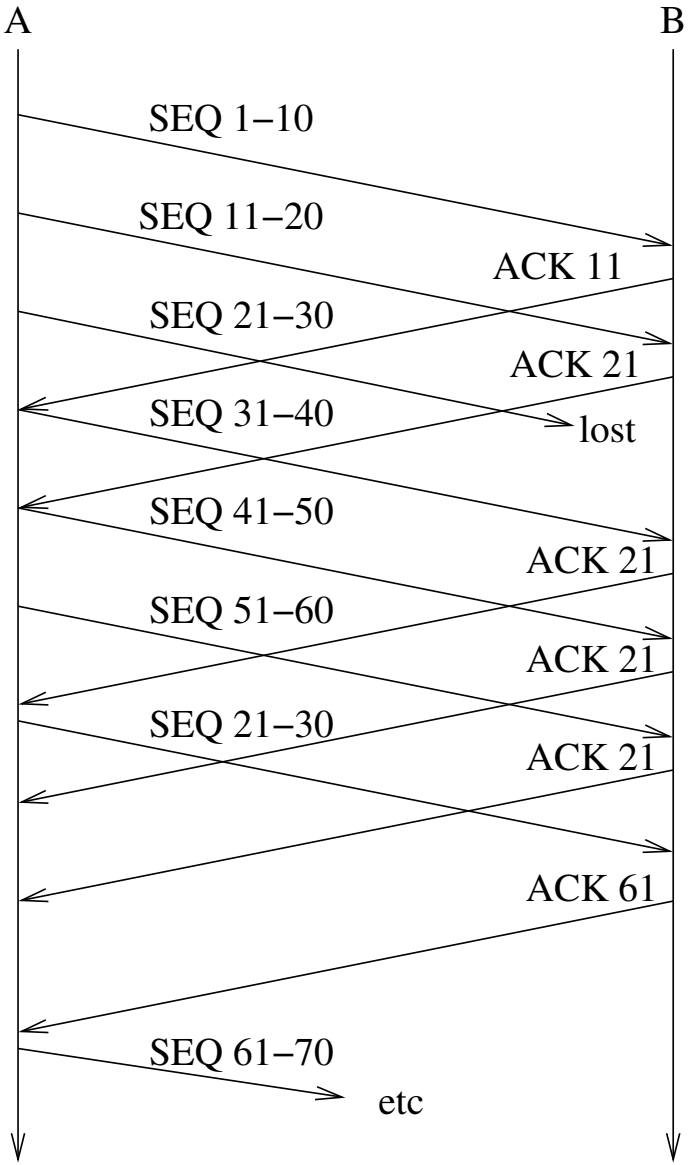
TCP



ACKing lost segments

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

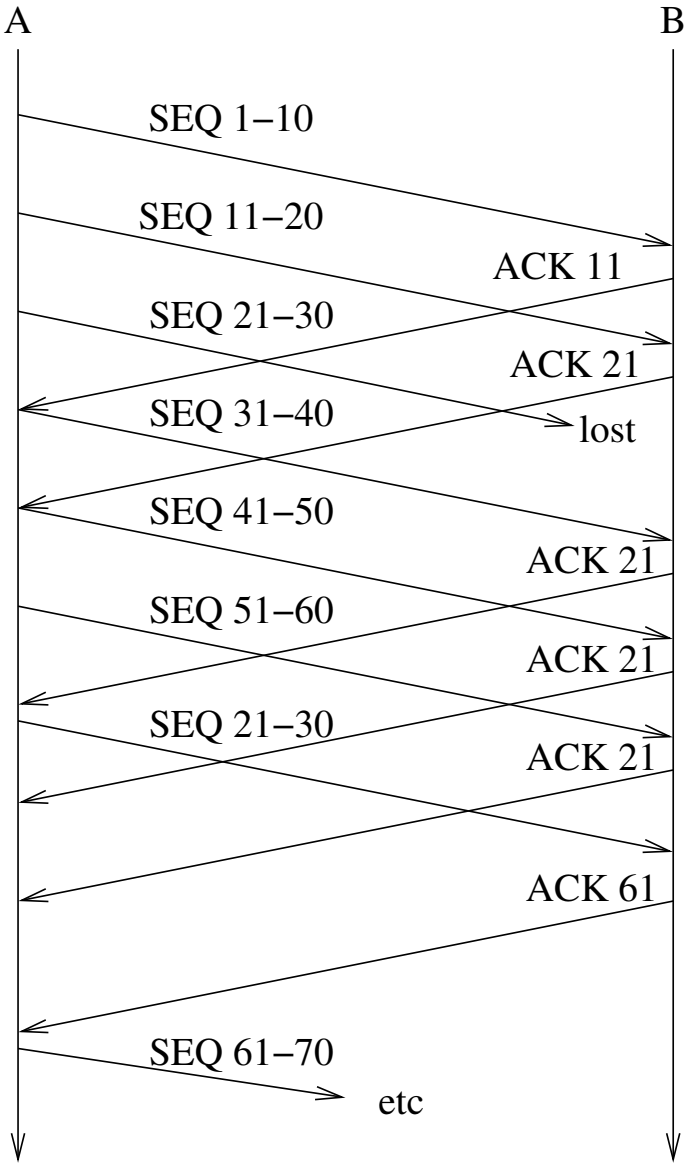
TCP



ACKing lost segments

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

TCP



A resends bytes 21-30;

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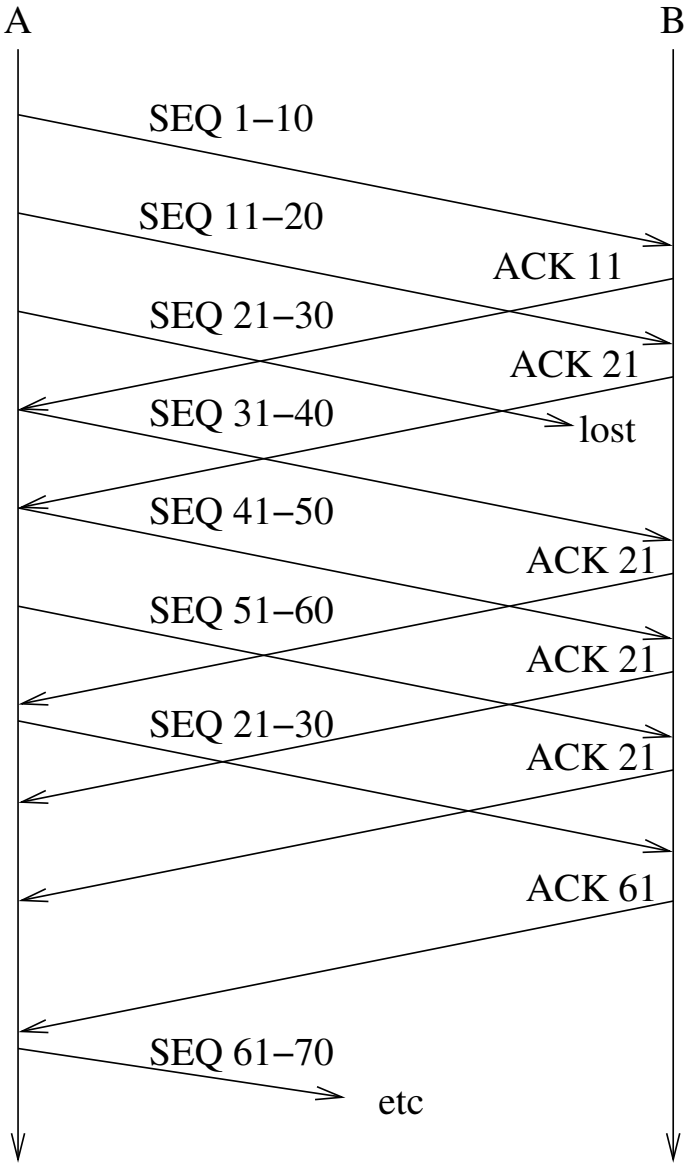

ACKing lost segments

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

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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

TCP

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