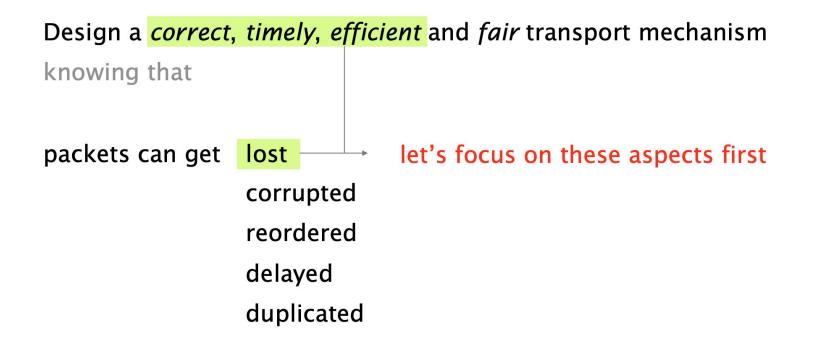
How do we achieve correctness and with what tradeoffs?



How do we achieve correctness and with what tradeoffs?

Alice

Bob

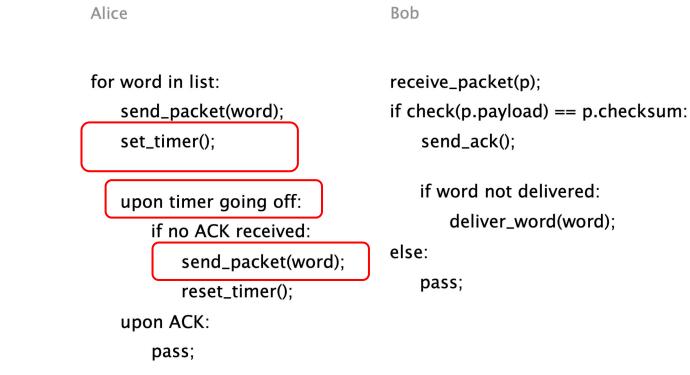
for word in list: send_packet(word); set_timer();

> upon timer going off: if no ACK received: send_packet(word); reset_timer(); upon ACK: pass;

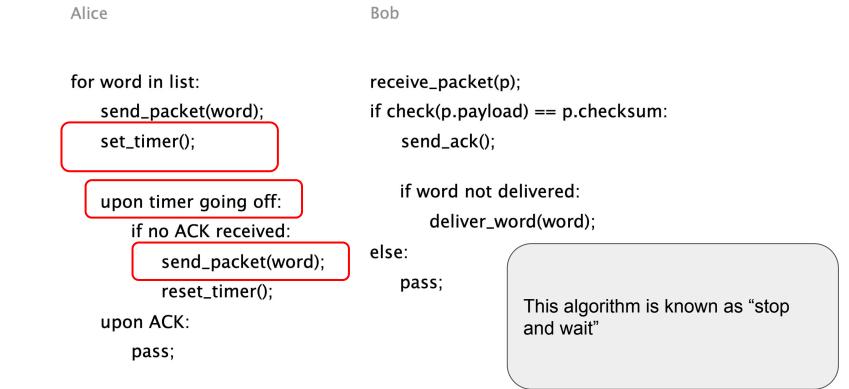
receive_packet(p);
if check(p.payload) == p.checksum:
 send_ack();

if word not delivered: deliver_word(word); else: pass;

There is a clear tradeoff between timeliness and efficiency in the selection of the timeout value



There is a clear tradeoff between timeliness and efficiency in the selection of the timeout value

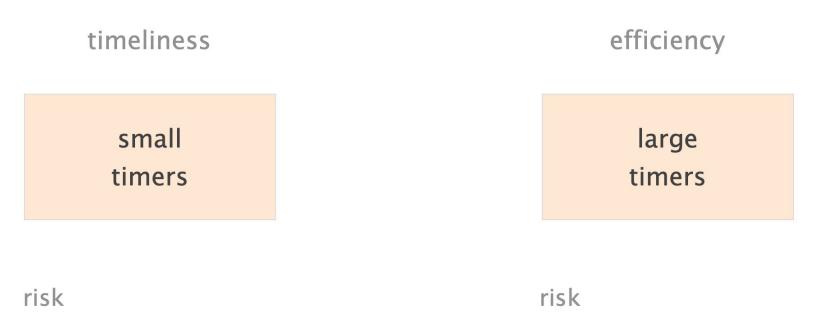


Stop and Wait demo

https://www2.tkn.tu-berlin.de/teaching/rn/animations/gbn_sr/

(for stop and wait, choose go back N and set the window size to 1)

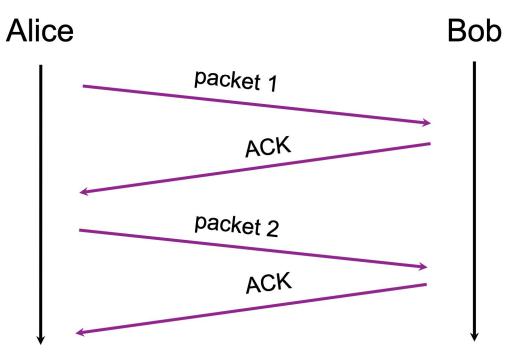
Timeliness argues for small timers, efficiency for large timers



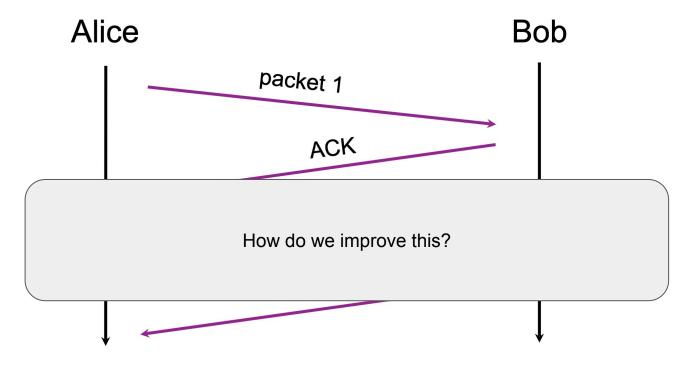
unnecessary retransmissions

slow transmission

Even with small timers, stop and wait has terrible timeliness - one packet per round trip time (RTT)



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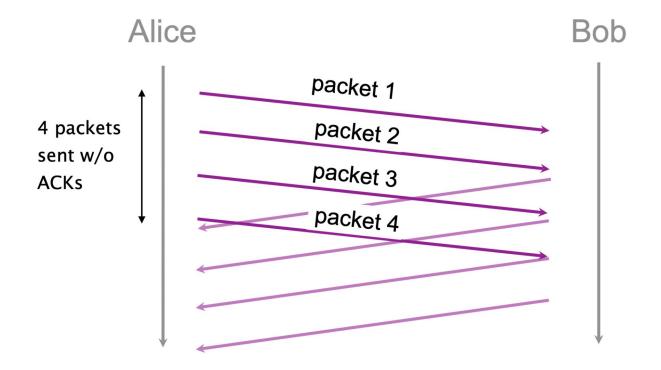
An obvious solution to improve timeliness is to send multiple packets at the same time

approach add sequence number inside each packet

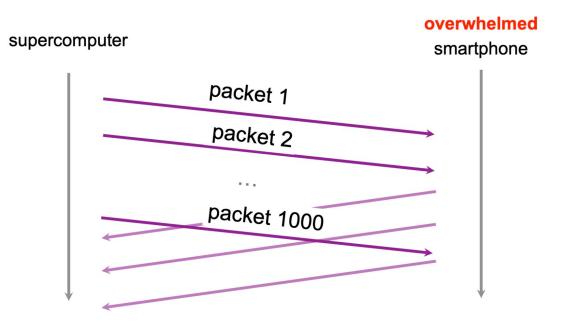
add buffers to the sender and receiver

senderstore packets sent & not acknowledgedreceiverstore out-of-sequence packets received

An obvious solution to improve timeliness is to send multiple packets at the same time



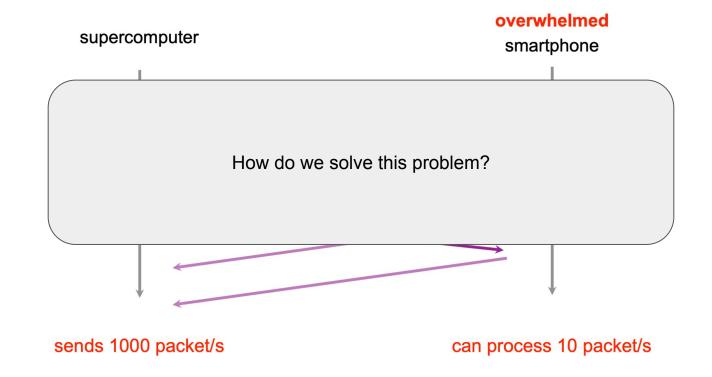
Sending multiple packets improves timeliness, but it can also overwhelm the receiver



sends 1000 packet/s

can process 10 packet/s

Sending multiple packets improves timeliness, but it can also overwhelm the receiver



Sender keeps a list of the sequence # it can send

known as the *sending window*

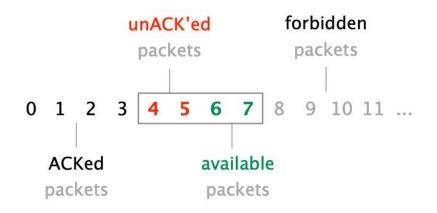
Receiver also keeps a list of the acceptable sequence

known as the *receiving window*

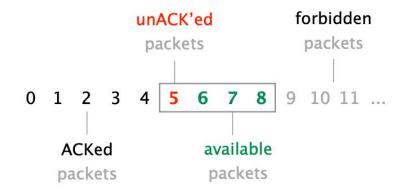
Sender and receiver negotiate the window size

sending window <= receiving window</pre>

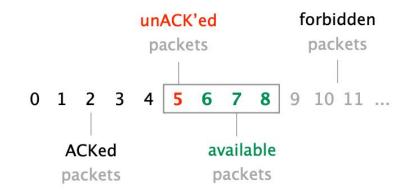
Example with a window composed of 4 packets



Window after sender receives ACK 4



Window after sender receives ACK 4



Timeliness of the window protocol depends on the size of the sending window

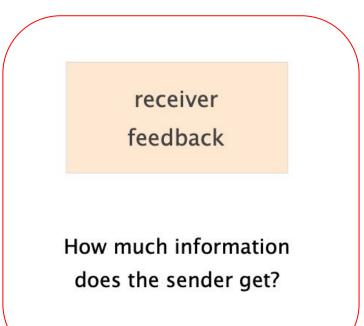
Efficiency of the protocol depends on two factors

receiver feedback behavior upon losses

How much information does the sender get?

How does the sender detect and react to losses?

Efficiency of the protocol depends on two factors



behavior upon losses

How does the sender detect and react to losses?

ACKing individual packets provides detailed feedback, but triggers unnecessary retransmission upon losses

advantages

disadvantages

know fate of each packet

simple window algorithm

W single-packet algorithms

loss of an ACK packet requires a retransmission

causes unnecessary retransmission

not sensitive to reordering

Cumulative ACKs enables to recover from lost ACKs, but provides coarse-grained information to the sender

approach ACK the highest sequence number for which all the previous packets have been received

advantages recover from lost ACKs

disadvantages confused by reordering incomplete information about which packets have arrived causes unnecessary retransmission

Full Information Feedback prevents unnecessary retransmission, but can induce a sizable overhead

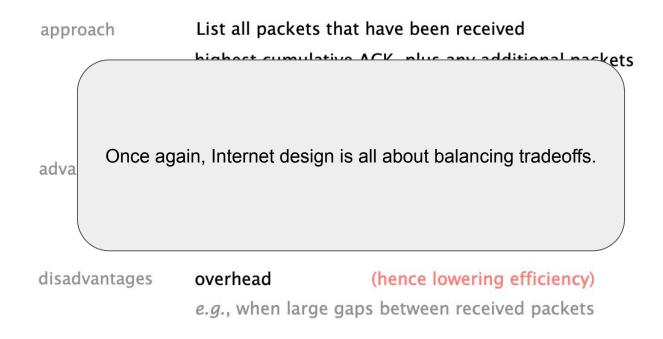
approachList all packets that have been receivedhighest cumulative ACK, plus any additional packets

advantages complete information resilient form of individual ACKs

disadvantages

overhead (hence lowering efficiency) e.g., when large gaps between received packets

Full Information Feedback prevents unnecessary retransmission, but can induce a sizable overhead



Efficiency of the protocol depends on two factors

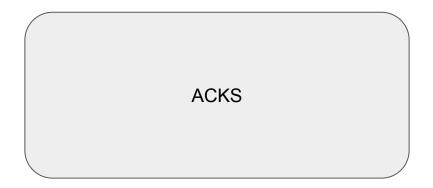
receiver feedback

How much information does the sender get?

behavior upon losses

How does the sender detect and react to losses?

We've been talking about detecting loss using timeouts. That's not the only way



With individual ACKs, missing packets (gaps) are implicit

Assume packet 5 is lost but no other



With full information, missing packets (gaps) are explicit

Assume packet 5 is lost

but no other



With cumulative ACKs, missing packets are harder to know

Assume packet 5 is lost

but no other

1

. . .

ACK stream

2 3 4 4 sent when 6 arrives 4 sent when 7 arrives

Duplicate ACKs are a sign of isolated losses. Dealing with them is trickier though.

situation Lack of ACK progress means that 5 hasn't made it

Stream of ACKs means that (some) packets are delivered

Sender could trigger resend upon receiving *k* duplicates ACKs

but what do you resend?

only 5 or 5 and everything after?

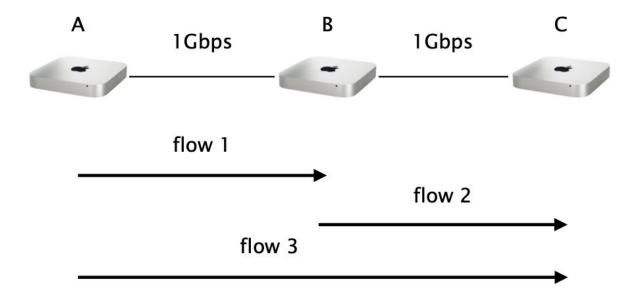
What about fairness?

Design a *correct*, *timely*, *efficient* and *fair* transport mechanism knowing that

packets can get lost corrupted reordered delayed duplicated

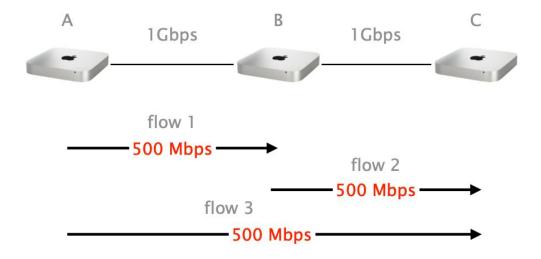
When *n* entities are using our transport mechanism, we want a fair allocation of the available bandwidth

Consider this simple network in which three hosts are sharing two links



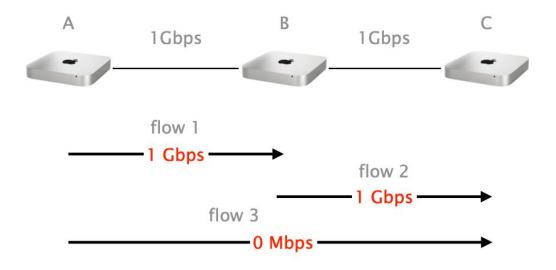
What is a fair allocation for the 3 flows?

An equal allocation is certainly "fair", but what about the efficiency of the network?



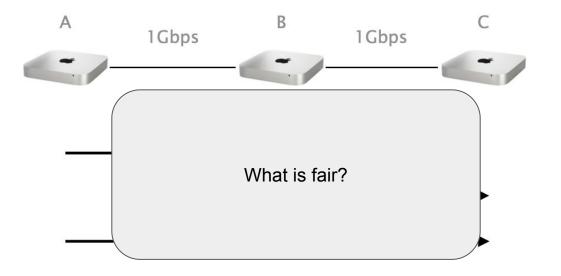
Total traffic is 1.5 Gbps

Fairness and efficiency don't always play along, here an unfair allocation ends up more *efficient*



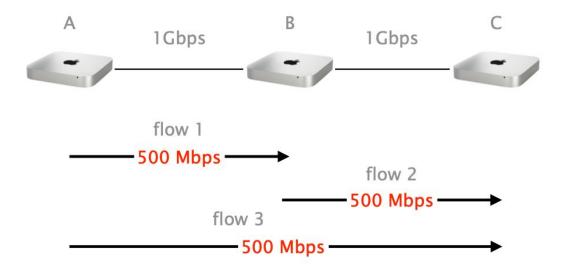
Total traffic is 2 Gbps!

Fairness and efficiency don't always play along, here an unfair allocation ends up more *efficient*



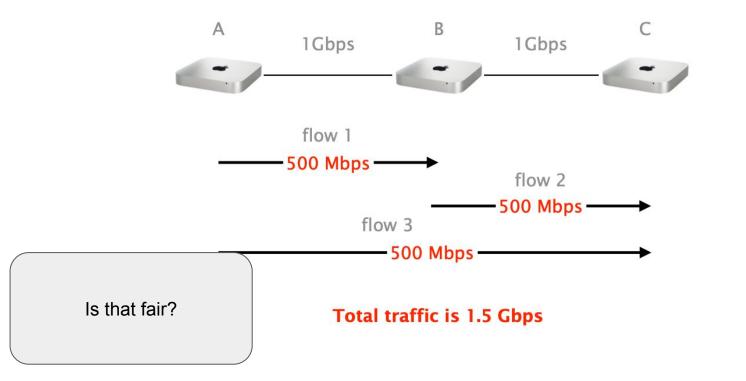
Total traffic is 2 Gbps!

Equal-per-flow isn't really fair as (A,C) crosses two links: it uses more resources

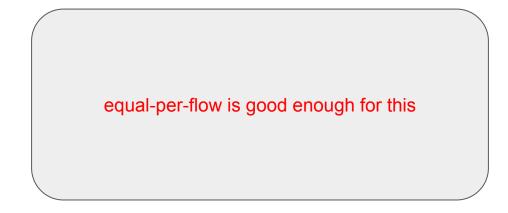


Total traffic is 1.5 Gbps

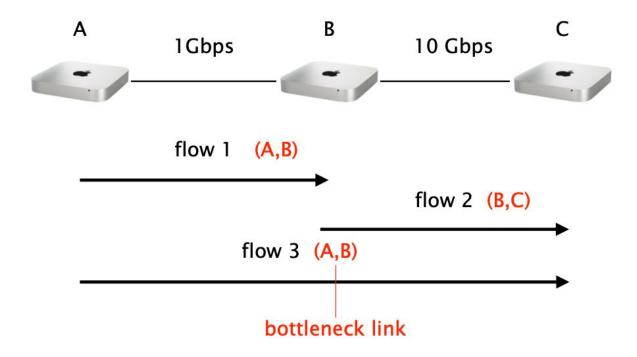
With equal-per-flow, A ends up with 1 Gbps because it sends 2 flows, while B ends up with 500 Mbps



Seeking an exact notion of fairness is not productive. What matters is to avoid starvation.



Simply dividing the available bandwidth doesn't work in practice since flows can see different bottlenecks



Intuitively, we want to give users with "small" demands what they want, and evenly distribute the rest

Max-min fair allocation is such that

the lowest demand is maximized

after the lowest demand has been satisfied, the second lowest demand is maximized

after the second lowest demand has been satisfied, the third lowest demand is maximized

and so on...

Max-min fair allocation can easily be computed

- step 1 Start with all flows at rate 0
- step 2 Increase the flows until there is a new bottleneck in the network
- step 3 Hold the fixed rate of the flows that are bottlenecked
- step 4 Go to step 2 for the remaining flows

Done!