SOLUTION NOTES

Digital Communication II 2002 Paper 9 Question 3 (IAP)

(a) TCP transmitters maintain a window of outstanding (un-ACKed) data. By increasing the size of the window TCP increases its sending rate, hoping to exploit more link bandwidth.

TCP attempts to share the bandwidth available on bottleneck links between competing connections in some "fair" manner.

After leaving slow start, probing for more bandwidth is done by increasing the congestion window in an "additive" fashion, by up to one segment (packet) per RTT. In practice, a little is added for every ACK received.

If a receiver observes loss (through a sequence gap), rather than waiting for a retransmission it may trigger a fast retransmit by issuing a duplicate ACK. Upon receiving the DUP ACK, the transmitter reduces its load on the network by halving its congestion window.

The sequence of sending at a linearly increasing rate then halving upon loss results in the saw tooth behaviour.

(b) Assume every packet is acked, hence congestion window increases by 1 segment every RTT i.e. $\frac{W}{2}$, $1+\frac{W}{2}$, $2+\frac{W}{2}$, $3+\frac{W}{2}\ldots(W)$ then loss occurs.

time between min and max $= R.\frac{W}{2}$

packets between each loss $= \frac{WW}{4} + \frac{W}{8} = \frac{3W^2}{8}$

(c) fraction of packets lost $= p = \frac{8}{3W^2}$

Thus $W = \sqrt{\frac{8}{3p}}$

avg throughput = total packets sent * pkt size / time to send them

$T = \frac{3W^2 B.2}{8.WR}$

subst. for W

$T = \frac{\sqrt{\frac{3}{2}}.B}{R.\sqrt{p}}$

(d) The above model assumes every segment is ACK'ed and that the window increases by 1 segment per RTT. Only one packet is lost during each loss event, and that fast retransmit always occurs (there is no waiting for timeouts). For this to be true the window must be >= 4 packets.

The model assumes that increased RTT due to increased queueing is small wrt the connection's RTT. For a slow link the queueing delay may be significant such that the self-clocking behaviour of TCP will drastically slow down the window increase rate.

(*e*) ECN capable routers set a bit in the packet header if they are "congested". A number of different marking algorithms have been tried such as RED.

ECN enables routers to signal congestion w/o dropping packets (which is inefficient). Furthermore, signalling congestion in this manner enables the router to signal it to all flows rather than just the few which actually incur loss by misfortune of having a packet arrive when the queue is full.

ECN marks received by the TCP receiver are reflected back to the TCP transmitter in the packet header.

A number of different schemes have been tried for responding to ECN. Most employ AIMD techniques but with different constants. The intention is to damp the oscillatory behaviour of normal TCP.