TCP
Congestion Control

Lecture material taken from
"Computer Networks *A Systems Approach*",
Third Ed.,Peterson and Davie,
Morgan Kaufmann, 2003.

TCP Congestion Control

- Essential strategy :: The TCP host sends packets into the network without a reservation and then the host reacts to observable events.
- Originally TCP assumed FIFO queuing.
- Basic idea :: each source determines how much capacity is available to a given flow in the network.
- ACKs are used to *'pace'* the transmission of packets such that TCP is "self-clocking".

AIMD
(Additive Increase / Multiplicative Decrease)

- CongestionWindow (cwnd) is a variable held by the TCP source for each connection.

MaxWindow :: min (**CongestionWindow** , AdvertisedWindow)

EffectiveWindow = MaxWindow – (LastByteSent -LastByteAcked)

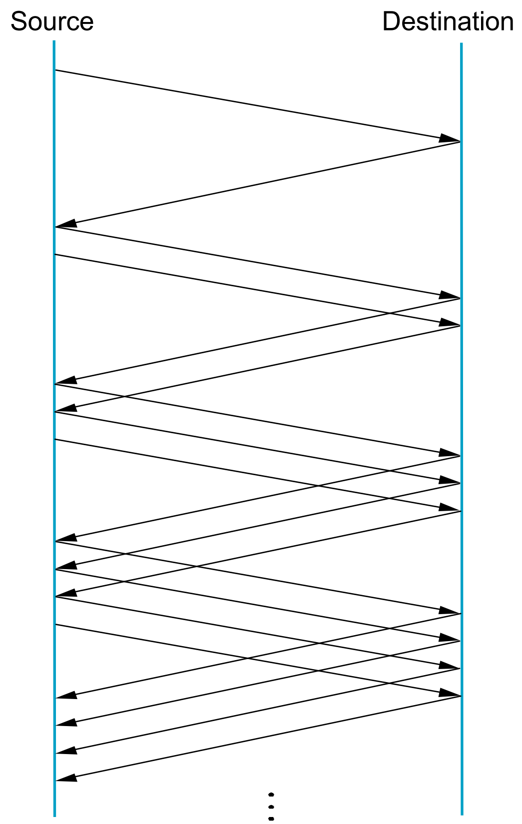
- **cwnd** is set based on the perceived level of congestion. The Host receives *implicit* (packet drop) or *explicit* (packet mark) indications of internal congestion.

Additive Increase

- Additive Increase is a reaction to perceived available capacity.
- Linear Increase basic idea:: For each "cwnd's worth" of packets sent, increase cwnd by 1 packet.
- In practice, **cwnd** is incremented fractionally for each arriving ACK.

increment = MSS x (MSS /cwnd)

cwnd = cwnd + increment



Add one packet each RTT

Figure 6.8 Additive Increase

Multiplicative Decrease

* The key assumption is that a dropped packet and the resultant timeout are due to congestion at a router or a switch.

  Multiplicate Decrease:: TCP reacts to a timeout by halving cwnd.
- Although cwnd is defined in bytes, the literature often discusses congestion control in terms of packets (or more formally in MSS == Maximum Segment Size).
- cwnd is not allowed below the size of a single packet.

AIMD (Additive Increase / Multiplicative Decrease)

- It has been shown that AIMD is a <u>necessary</u> condition for TCP congestion control to be stable.
- Because the simple CC mechanism involves timeouts that cause retransmissions, it is important that hosts have an accurate timeout mechanism.
- Timeouts set as a function of average RTT and standard deviation of RTT.
- However, TCP hosts only sample round-trip time once per RTT using coarse-grained clock.

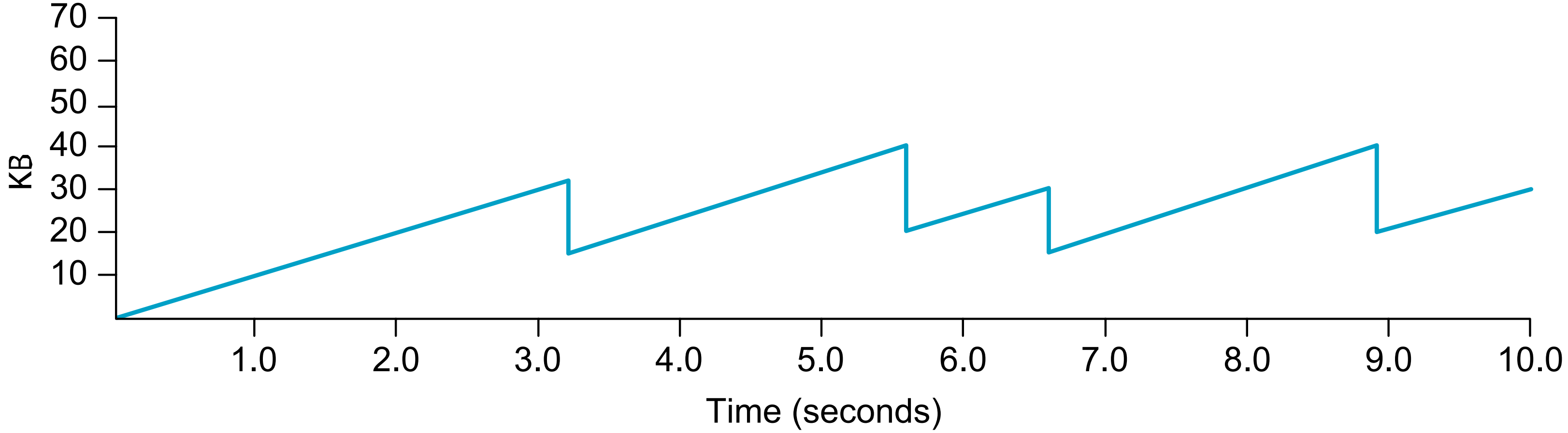


Figure 6.9 Typical TCP Sawtooth Pattern

Slow Start

- Linear additive increase takes too long to ramp up a new TCP connection from cold start.
- Beginning with TCP Tahoe, the slow start mechanism was added to provide an initial exponential increase in the size of cwnd.

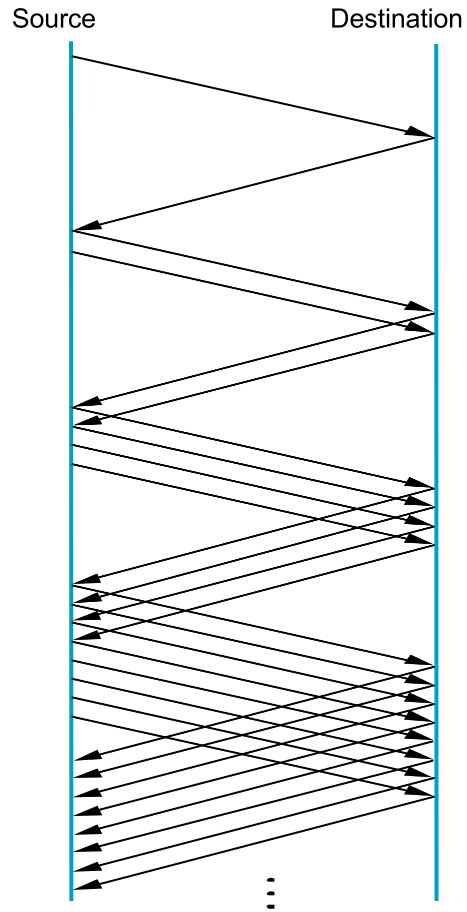
Remember mechanism by: ***slow start prevents a slow start. Moreover, slow start is slower than sending a full advertised window's worth of packets all at once.***

Slow Start

- The source starts with cwnd = 1.
- Every time an ACK arrives, cwnd is incremented.

➔cwnd is effectively doubled per RTT "epoch".

- Two slow start situations:
  - At the very beginning of a connection **{cold start}**.
  - When the connection goes dead waiting for a timeout to occur (i.e, the advertized window goes to zero!)



Slow Start

Add one packet per ACK

Figure 6.10 Slow Start

Slow Start

- However, in the second case the source has more information. The current value of cwnd can be saved as a **congestion threshold**.
- This is also known as the "slow start threshold" **ssthresh**.

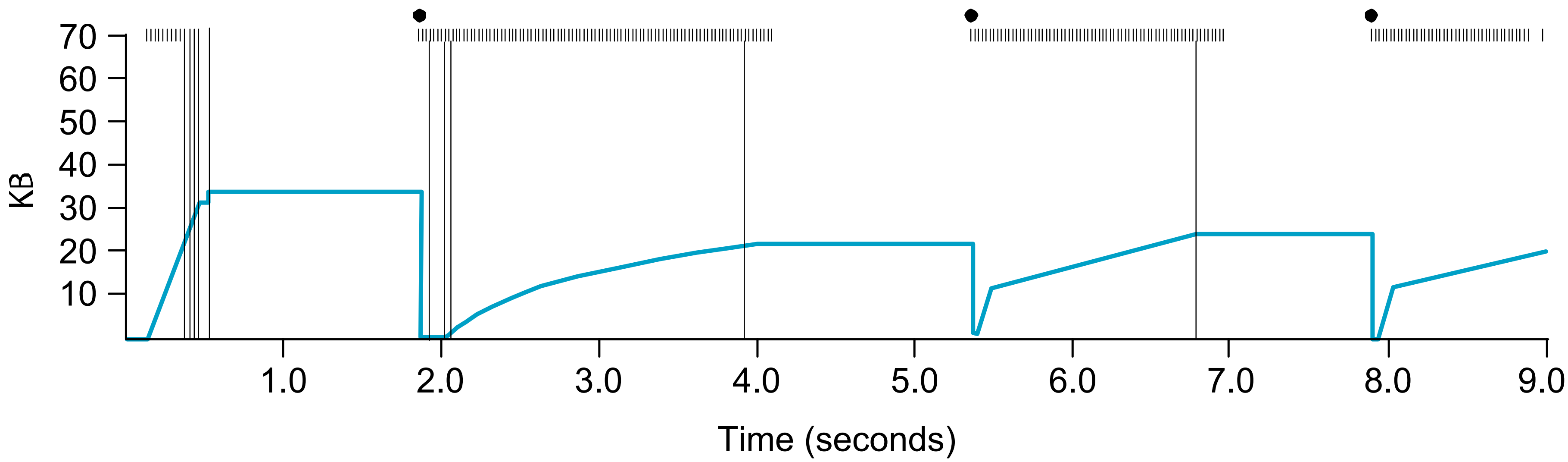


Figure 6.11 Behavior of TCP Congestion Control

Fast Retransmit

- Coarse timeouts remained a problem, and Fast retransmit was added with TCP Tahoe.
- Since the receiver responds every time a packet arrives, this implies the sender will see duplicate ACKs.

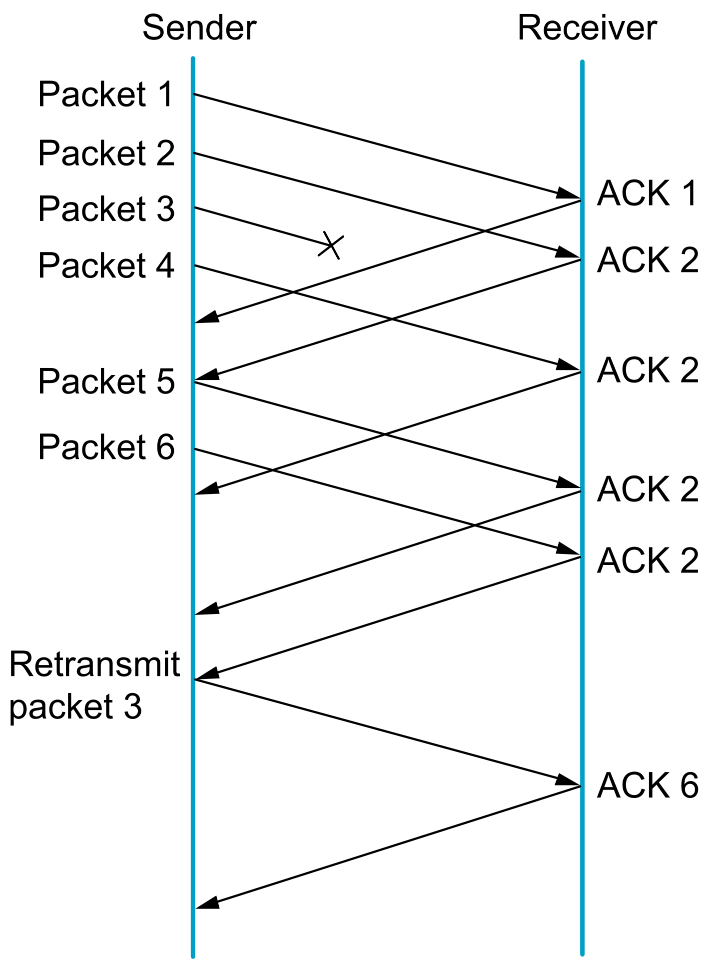
Basic Idea:: *use **duplicate ACKs** to signal lost packet.*

Fast Retransmit

Upon receipt of *three* duplicate ACKs, the TCP Sender retransmits the lost packet.

Fast Retransmit

- Generally, fast retransmit eliminates about half the coarse-grain timeouts.
- This yields roughly a 20% improvement in throughput.
- Note – fast retransmit does not eliminate all the timeouts due to small window sizes at the source.



Fast Retransmit

Based on three duplicate ACKs

Figure 6.12 Fast Retransmit

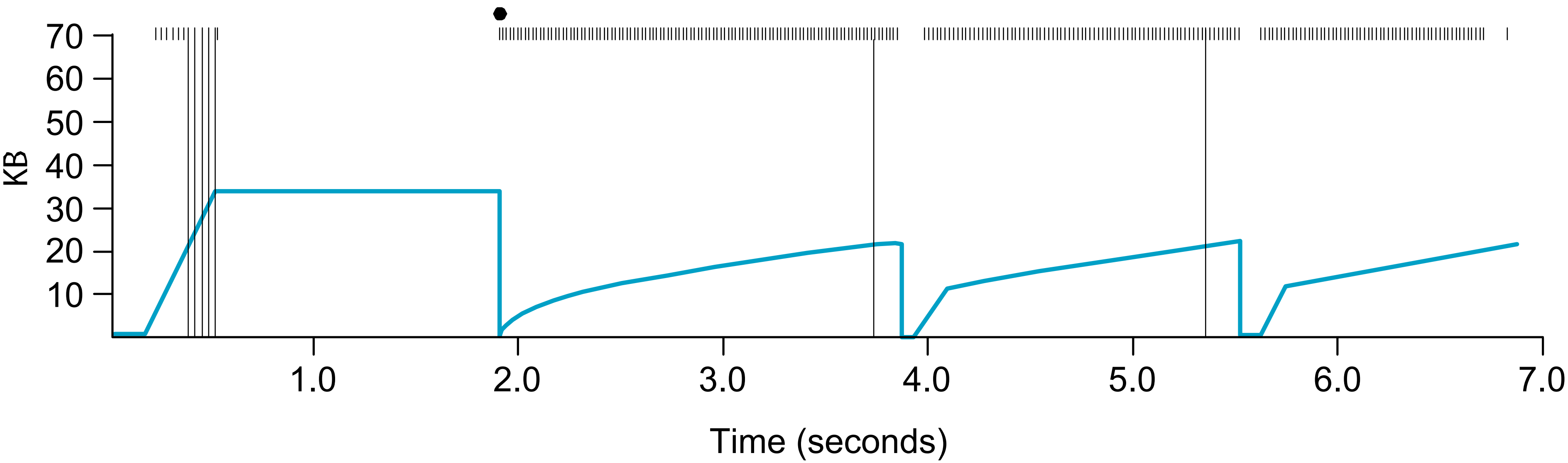
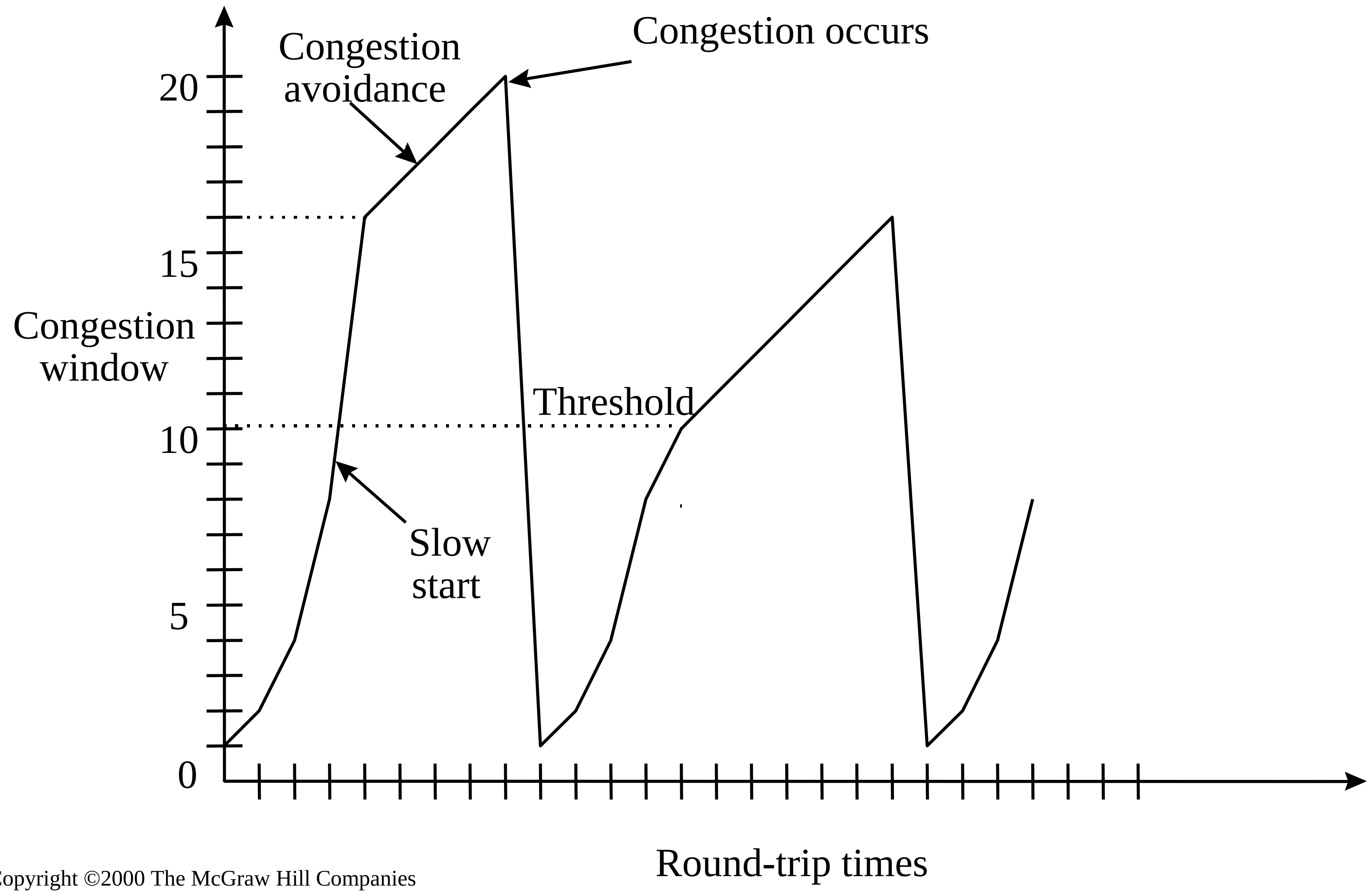


Figure 6.13 TCP Fast Retransmit Trace

TCP Congestion Control



Leon-Garcia & Widjaja: *Communication Networks*

Figure 7.63

Fast Recovery

- Fast recovery was added with TCP Reno.
- Basic idea:: When fast retransmit detects three duplicate ACKs, start the recovery process from congestion avoidance region and use ACKs in the pipe to pace the sending of packets.

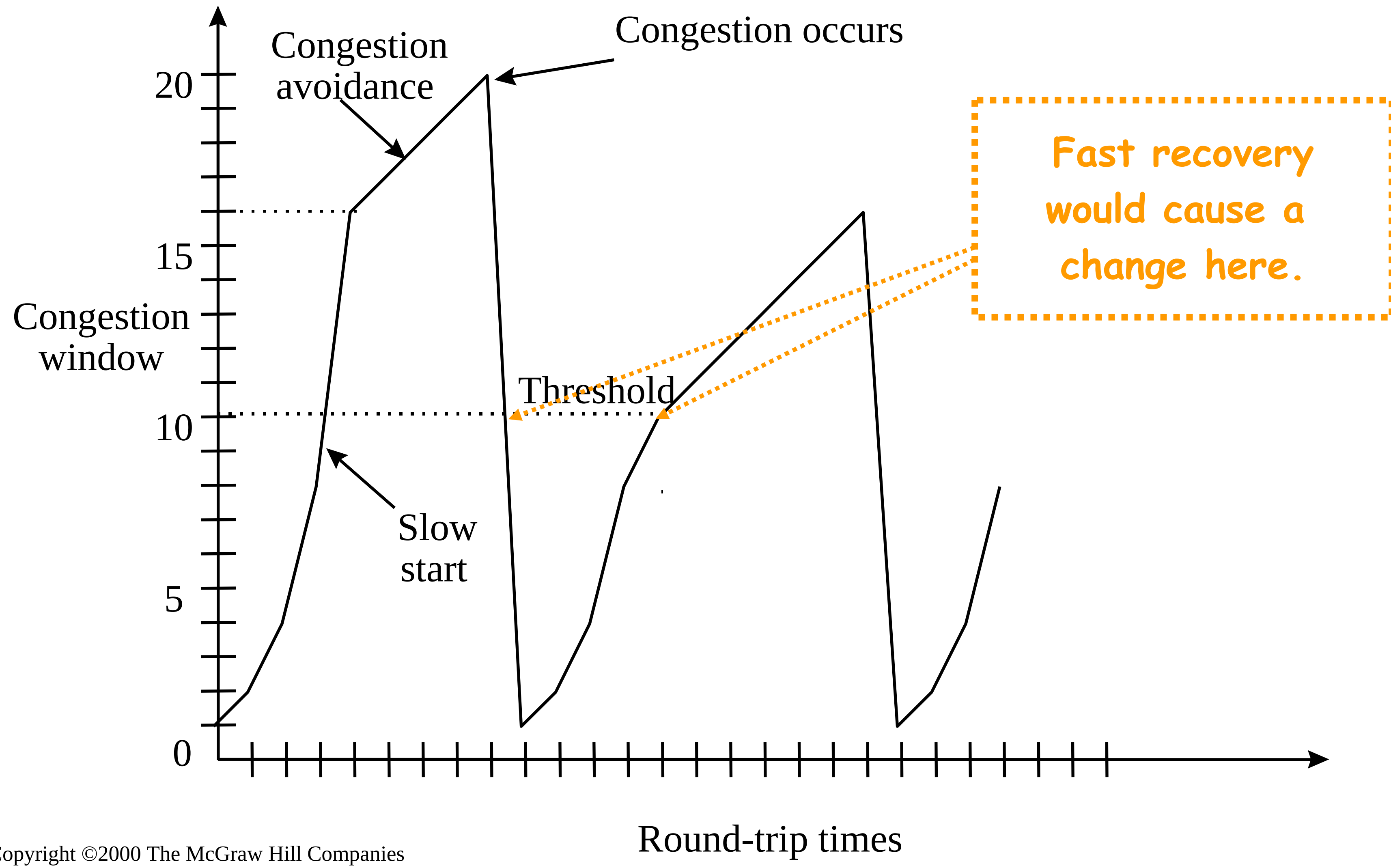
Fast Recovery

After Fast Retransmit, half cwnd and commence recovery from this point using <u>linear</u> additive increase 'primed' by left over ACKs in pipe.

Modified Slow Start

- With fast recovery, **slow start** only occurs:
  - At cold start
  - After a coarse-grain timeout
- *This is the difference between TCP Tahoe and TCP Reno!!*

TCP Congestion Control



Leon-Garcia & Widjaja: *Communication Networks*

Figure 7.63

TCP Congestion Control

Adaptive Retransmissions

RTT:: Round Trip Time between a pair of hosts on the Internet.

- How to set the TimeOut value?
  - The timeout value is set as a function of the expected RTT.
  - Consequences of a bad choice?

Original Algorithm

- Keep a running average of RTT and compute TimeOut as a function of this RTT.
  - Send packet and keep timestamp t_s .
  - When ACK arrives, record timestamp t_a .

$$\text{SampleRTT} = t_a - t_s$$

Original Algorithm

Compute a weighted average:

$$\text{EstimatedRTT} = \alpha \times \text{EstimatedRTT} + (1-\alpha) \times \text{SampleRTT}$$

Original TCP spec: α in range (0.8,0.9)

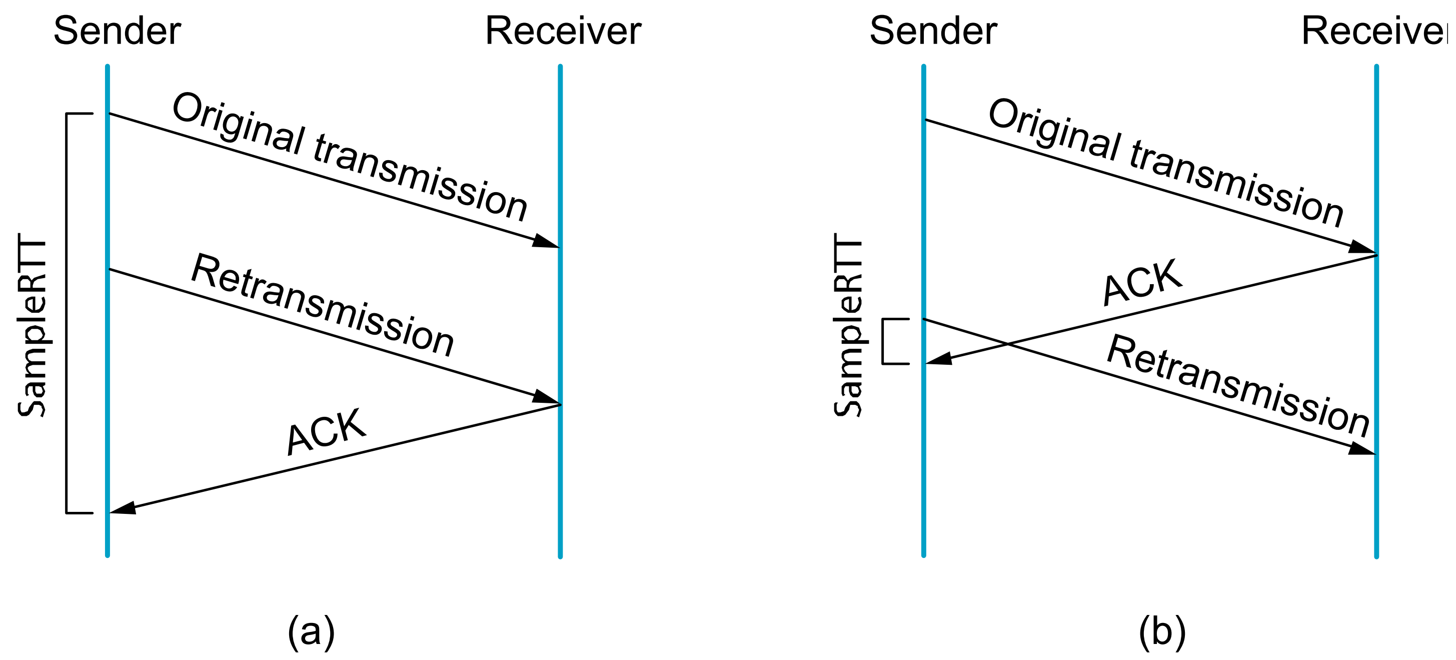
$$\text{TimeOut} = 2 \times \text{EstimatedRTT}$$

Karn/Partidge Algorithm

An obvious flaw in the original algorithm:

Whenever there is a retransmission it is impossible to know whether to associate the ACK with the original packet or the retransmitted packet.

Figure 5.10 Associating the ACK?



Karn/Partidge Algorithm

1. Do not measure SampleRTT when sending packet more than once.
2. For each retransmission, set TimeOut to double the last TimeOut.

   { Note – this is a form of exponential backoff based on the believe that the lost packet is due to congestion.}

Jacobson/Karels Algorithm

The problem with the original algorithm is that it did not take into account the variance of SampleRTT.

Difference = SampleRTT – EstimatedRTT

EstimatedRTT = EstimatedRTT + (δ x Difference)

Deviation = δ (|Difference| - Deviation)

where δ is a fraction between 0 and 1.

Jacobson/Karels Algorithm

TCP computes timeout using both the mean and variance of RTT

$$\text{TimeOut} = \mu \times \text{EstimatedRTT} + \Phi \times \text{Deviation}$$

where based on experience $\mu = 1$ and $\Phi = 4$.