Introduction to Computer Networks

Transport Layer Protocols

Outline

- Introduction to end-to-end protocols
- Simple Demultiplexer protocol (UDP)
- Reliable Byte Stream protocol (TCP)

End-to-end Protocols

- A transport protocol is usually expected to provide
  - Guaranteed message delivery
  - Delivers messages in the same order they were sent
  - Delivers at most one copy of each message
  - Supports arbitrarily large messages
  - Supports synchronization between the sender and the receiver
  - Allows the receiver to apply flow control to the sender
  - Supports multiple application processes on each host

End-to-end Protocols

- Typical limitations of the network service (like IP of Internet) on which transport protocol will operate
  - Drop messages
  - Reorder messages
  - Deliver duplicate copies of a given message
  - Limit messages to some finite size
  - Deliver messages after an arbitrarily long delay

  → Unreliable service

End-to-end Protocols

- Challenge for Transport Protocols
  - Develop algorithms that turn the unreliable service of the underlying network into the service required by application programs
  - *Unreliable service* → *Unreliable service* (UDP)
  - *Unreliable service* → *Reliable service* (TCP)

Outline

- Introduction to end-to-end protocols
- Simple Demultiplexer protocol (UDP)
- Reliable Byte Stream protocol (TCP)

Simple Demultiplexer (UDP)

- Extends host-to-host delivery service of the underlying network into a process-to-process communication service
- Adds a level of demultiplexing which allows multiple application processes on each host to share the network

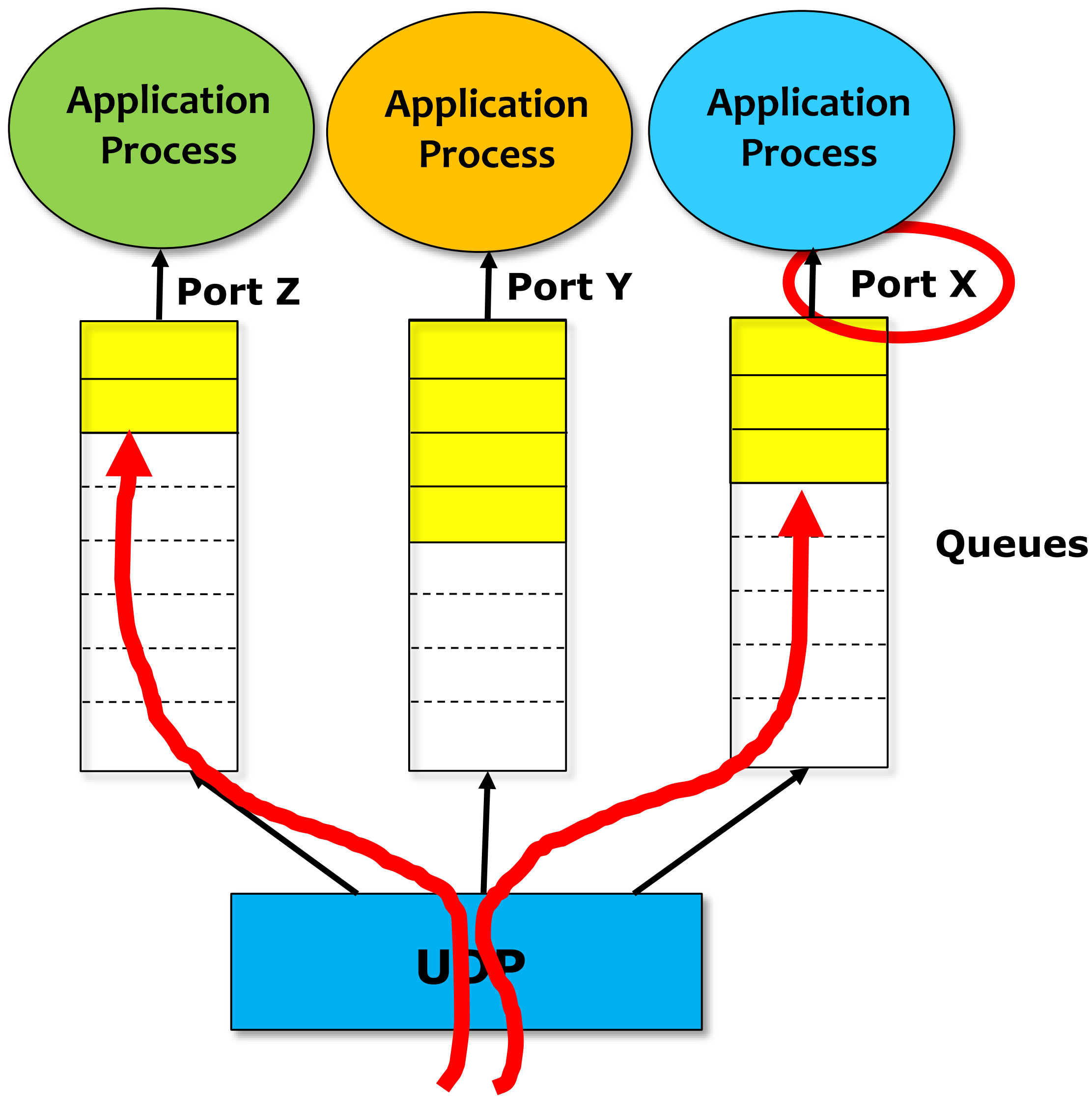
Simple Demultiplexer (UDP)

0 8 16 31

SrcPort	DstPort
Checksum	Length
Data	

Format for UDP header

Simple Demultiplexer (UDP)



UDP Packet Demultiplexer

Outline

- Introduction to end-to-end protocols
- Simple Demultiplexer protocol (UDP)
- Reliable Byte Stream protocol (TCP)

Reliable Byte Stream (TCP)

- In contrast to UDP, Transmission Control Protocol (TCP) offers the following services
  - Reliable
  - Connection oriented
  - Byte-stream service

Flow control VS Congestion control

- **Flow control** involves preventing senders from overrunning the capacity of the receivers
- **Congestion control** involves preventing too much data from being injected into the network, thereby causing routers/switches or links to become overloaded

End-to-end Issues

- TCP runs over the Internet rather than a point-to-point link
- The TCP sliding window algorithm need to consider:
  - TCP supports logical connections between processes that are running on two different computers in the Internet
  - TCP connections are likely to have widely different RTT times
  - Packets may get reordered in the Internet

End-to-end Issues

- TCP needs a mechanism using which each side of a connection will learn what resources the other side offers to the connection
- TCP needs a mechanism using which the sending side will learn the capacity of the network

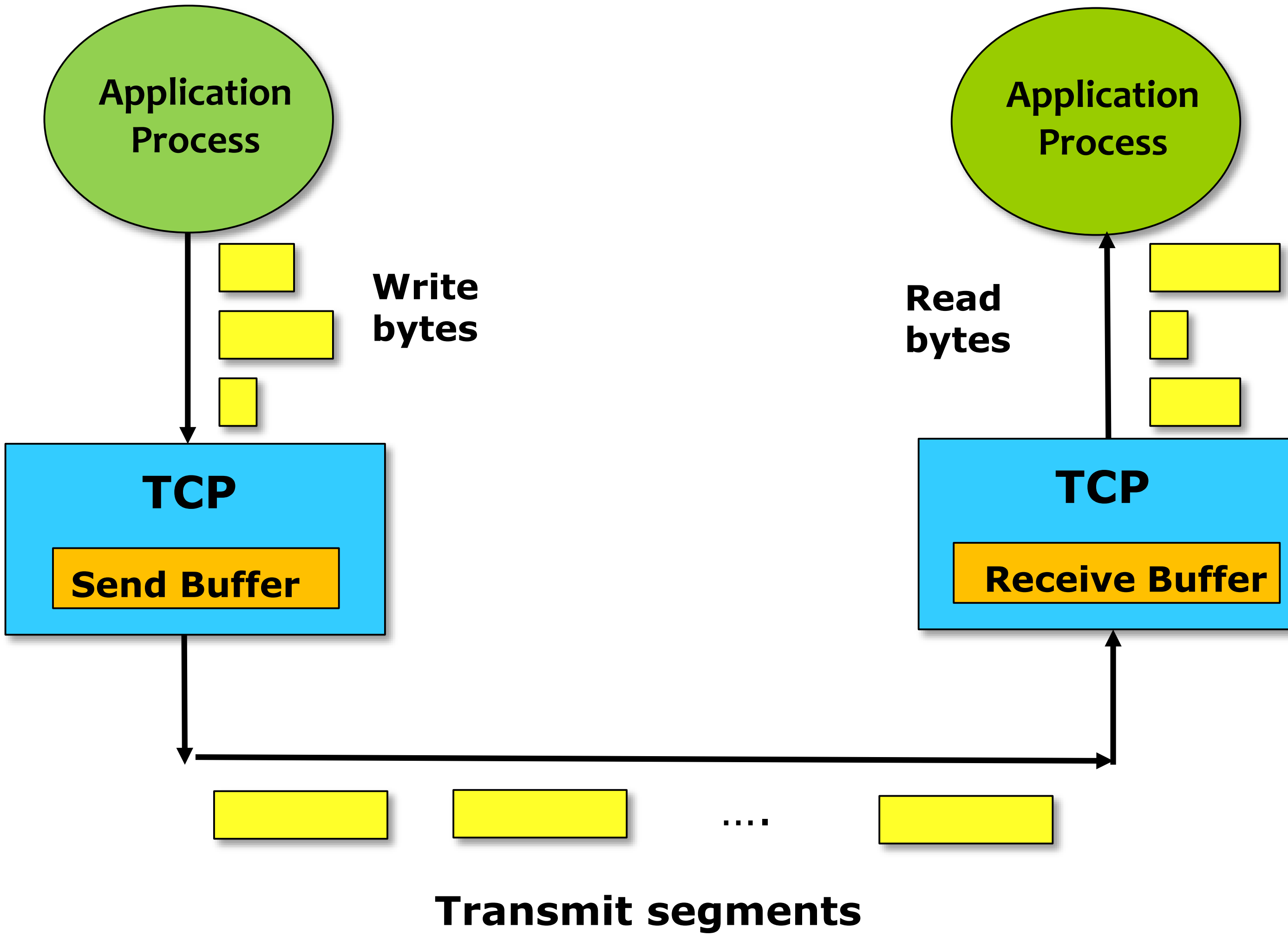
TCP Segment

- TCP is a byte-oriented protocol
- The sender writes bytes into a TCP connection and the receiver reads bytes out of the TCP connection.
- However, TCP does not transmit individual bytes over the Internet.

TCP Segment

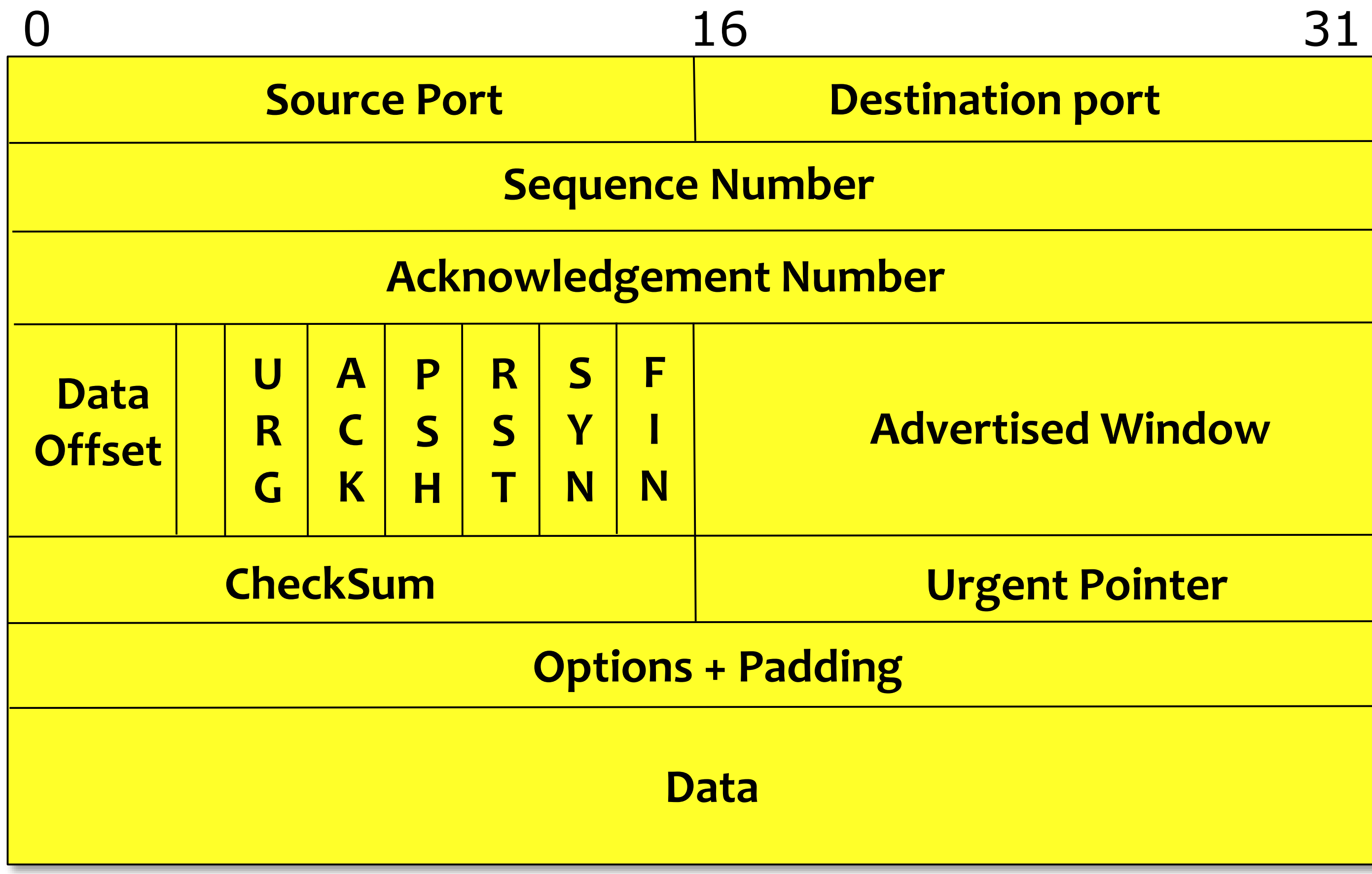
- The source TCP buffers enough bytes from the sending process to fill a reasonably sized packet and then sends this packet to its peer on the destination host.
- The destination TCP then puts the contents of the packet into a receive buffer, and the receiving process reads from this buffer.
- The packets exchanged between TCP peers are called *segments*.

TCP Segment



How TCP manages a byte stream.

TCP Header



TCP Header Format

TCP Header

- The SrcPort and DstPort fields identify the source and destination ports, respectively.
- The Acknowledgment, SequenceNum, and AdvertisedWindow fields are all involved in TCP's sliding window algorithm.
- Because TCP is a byte-oriented protocol, each byte of data has a sequence number; the SequenceNum field contains the sequence number for the first byte of data carried in that segment.
- The Acknowledgment and AdvertisedWindow fields carry information about the flow of data going in the other direction.

TCP Header

- The 6-bit Flags field is used to relay control information between TCP peers.
- The possible flags include SYN, FIN, RESET, PUSH, URG, and ACK.
- The SYN and FIN flags are used when establishing and terminating a TCP connection, respectively.
- The ACK flag is set any time the Acknowledgment field is valid, implying that the receiver should pay attention to it.

TCP Header

- The URG flag signifies that this segment contains urgent data. When this flag is set, the UrgPtr field indicates where the nonurgent data contained in this segment begins.
- The urgent data is contained at the front of the segment body, up to and including a value of UrgPtr bytes into the segment.
- The PUSH flag signifies that the sender invoked the push operation, which indicates to the receiving side of TCP that it should notify the receiving process of this fact.

TCP Header

- The RESET flag signifies that the receiver has become confused, it received a segment it did not expect to receive—and so wants to abort the connection.
- Finally, the Checksum field is used in exactly the same way as for UDP—it is computed over the TCP header, the TCP data, and the pseudoheader, which is made up of the source address, destination address, and length fields from the IP header.

TCP Connection Management

- TCP sender, receiver establish "connection" before exchanging data segments
- initialize TCP variables:
  - Sequence numbers
  - Buffers, flow control info (e.g. RcvWindow)
- *Client*: connection initiator

```
Socket clientSocket = new Socket("hostname","port number");
```

- *Server*: contacted by client

```
Socket connectionSocket = welcomeSocket.accept();
```

TCP Connection Management

Three-way handshake:

Step 1: Client sends TCP SYN segment to server

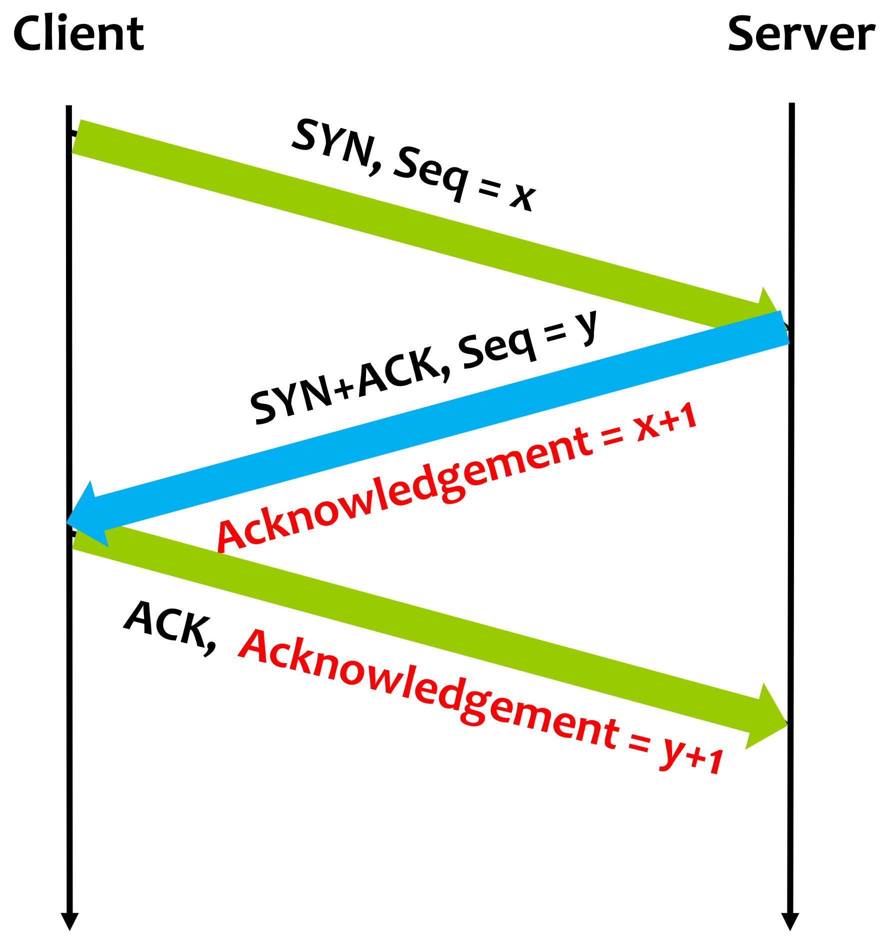
- specifies initial seq #
- no data

Step 2: Server receives SYN, replies with SYN/ACK segment

- server allocates buffers
- specifies server initial seq. #

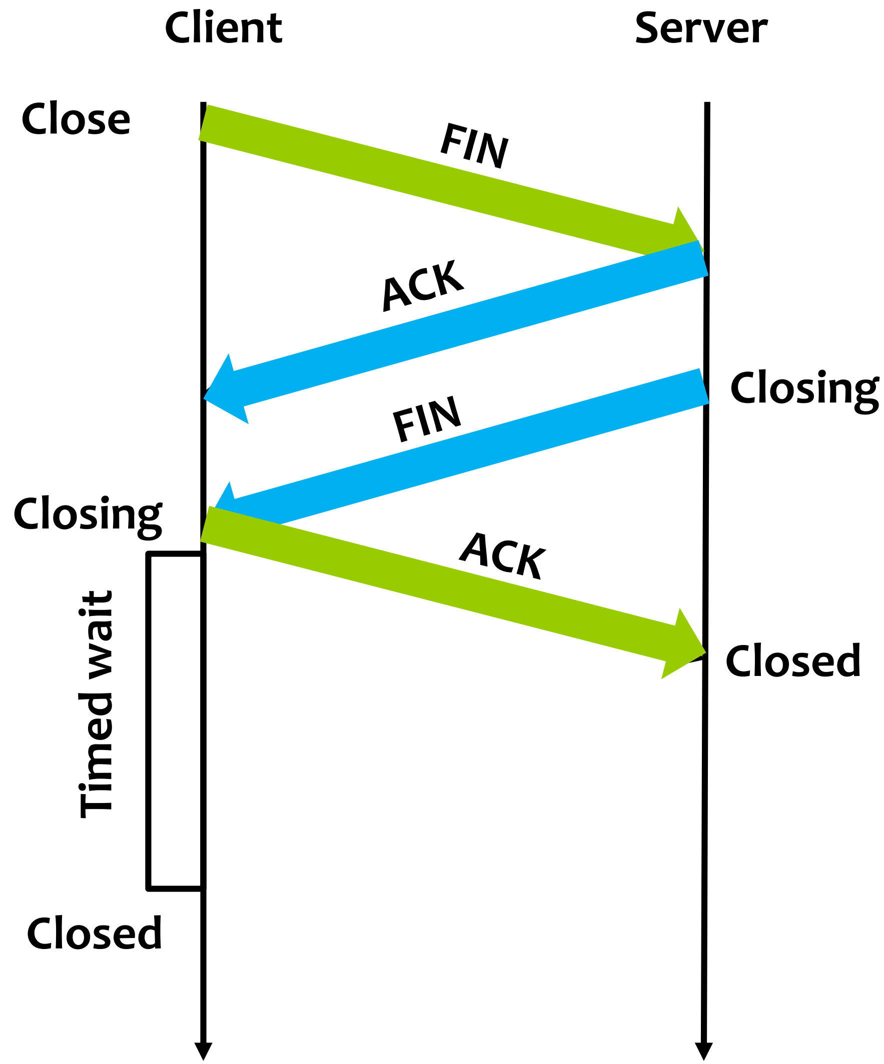
Step 3: client receives SYN/ACK, replies with ACK segment, which may contain data

Connection Establishment in TCP



Timeline for three-way handshake algorithm

TCP Connection Management (cont.)



Closing a connection:

client closes socket:

```
clientSocket.close();
```

Step 1: Client sends TCP FIN control segment to server

Step 2: Server receives FIN, replies with ACK. Closes connection, sends FIN.

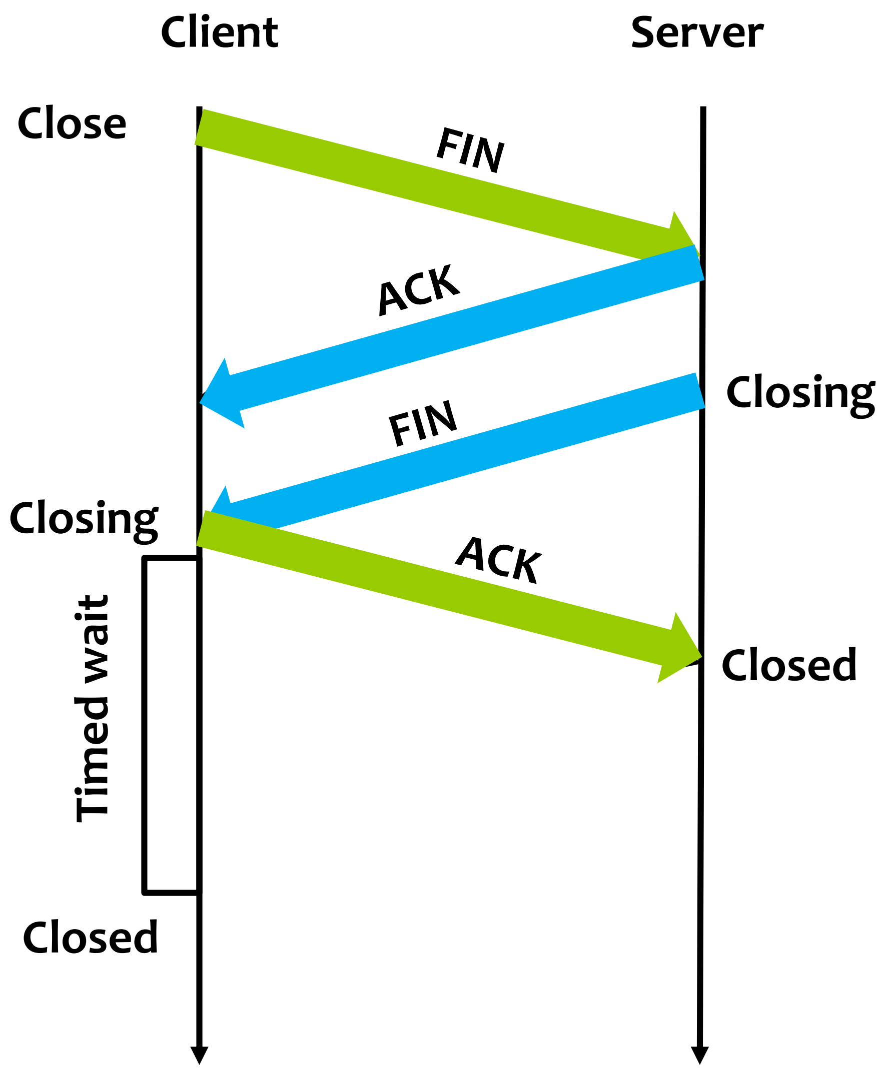
TCP Connection Management (cont.)

Step 3: Client receives FIN, replies with ACK.

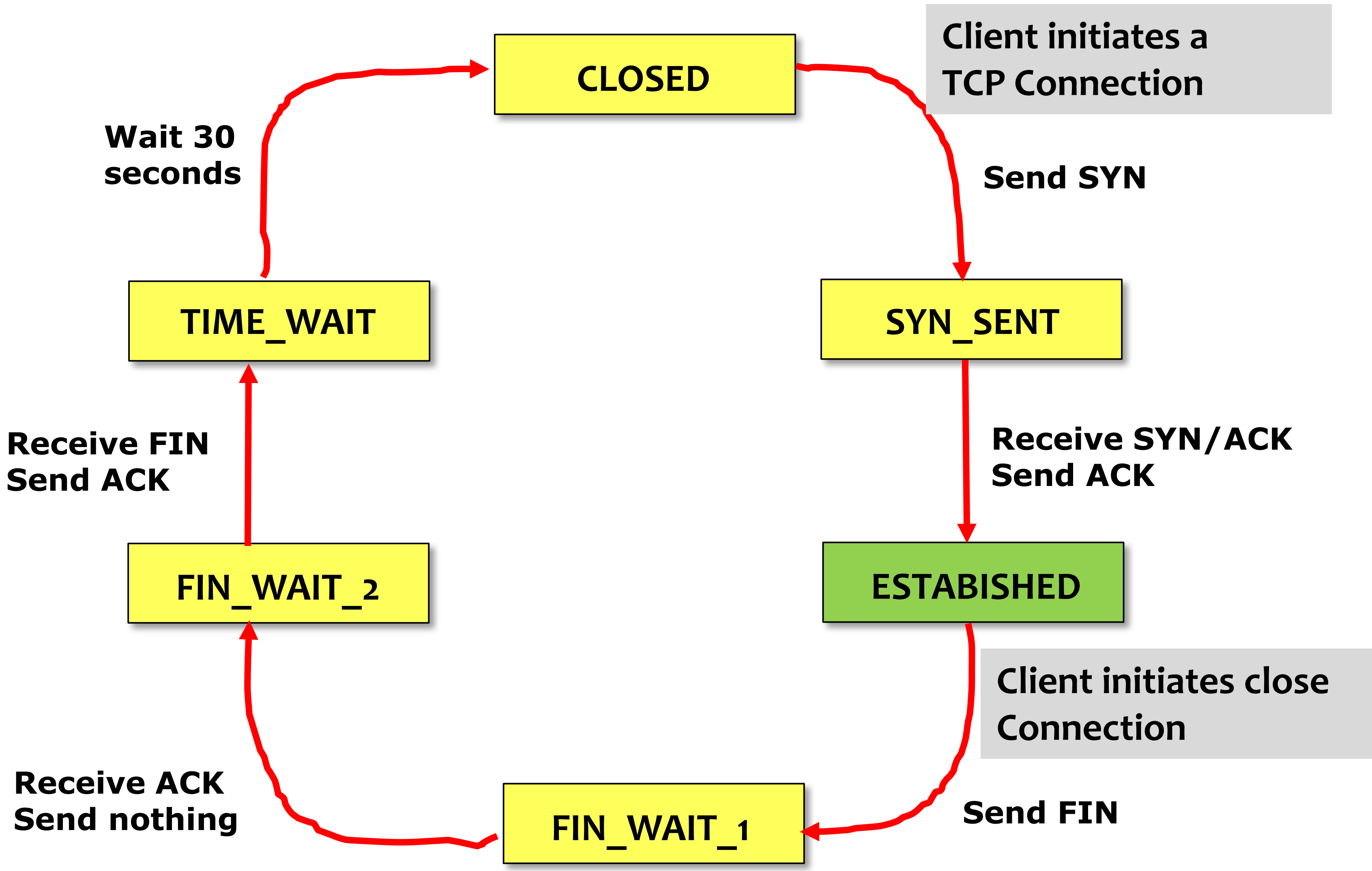
- Enters "timed wait" - will respond with ACK to received FINs

Step 4: Server receives ACK. Connection closed.

Note: with small modification, can handle simultaneous FINs.

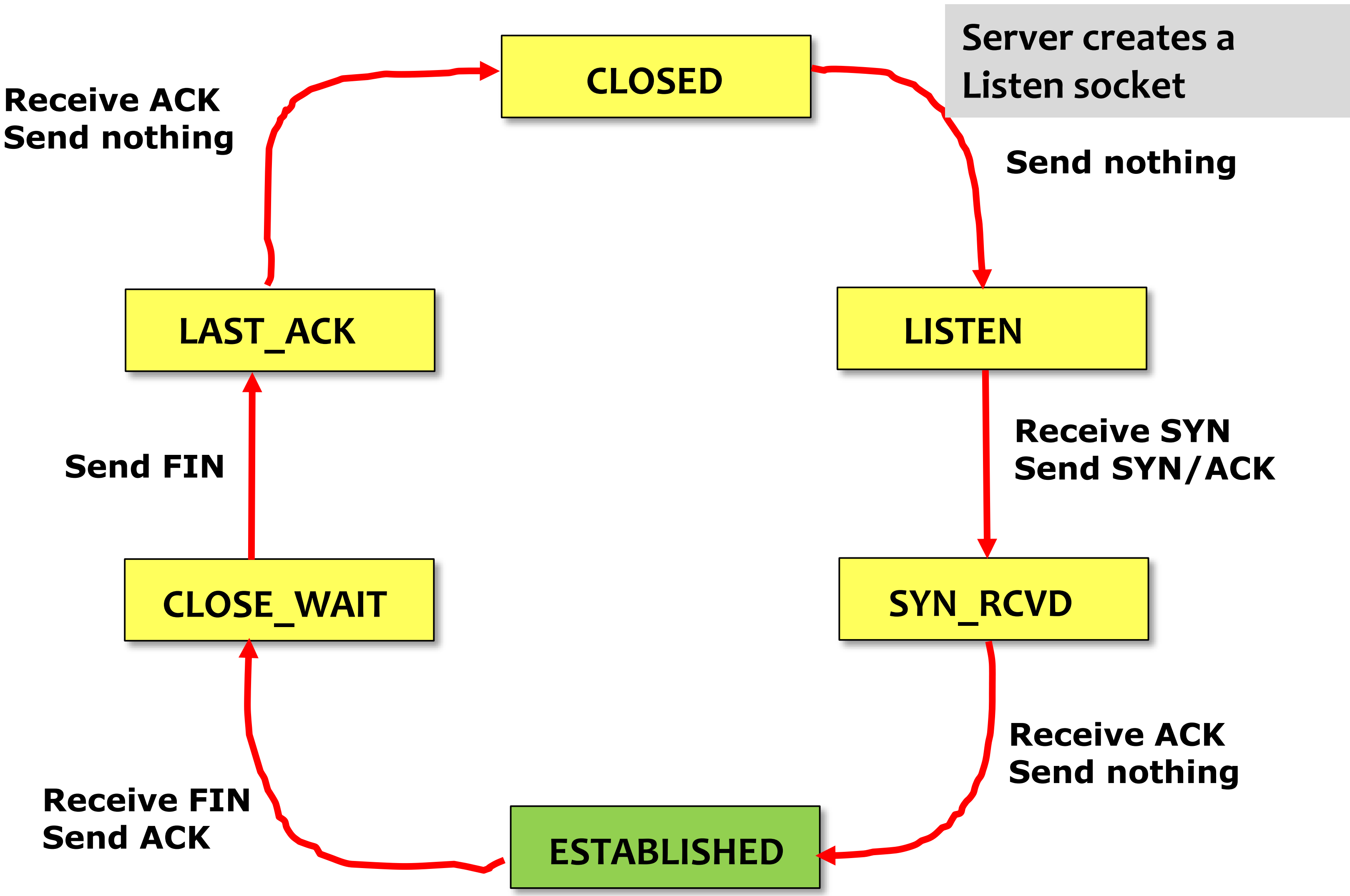


TCP Connection Management (cont)



TCP client state diagram

TCP Connection Management (cont)



TCP server state diagram

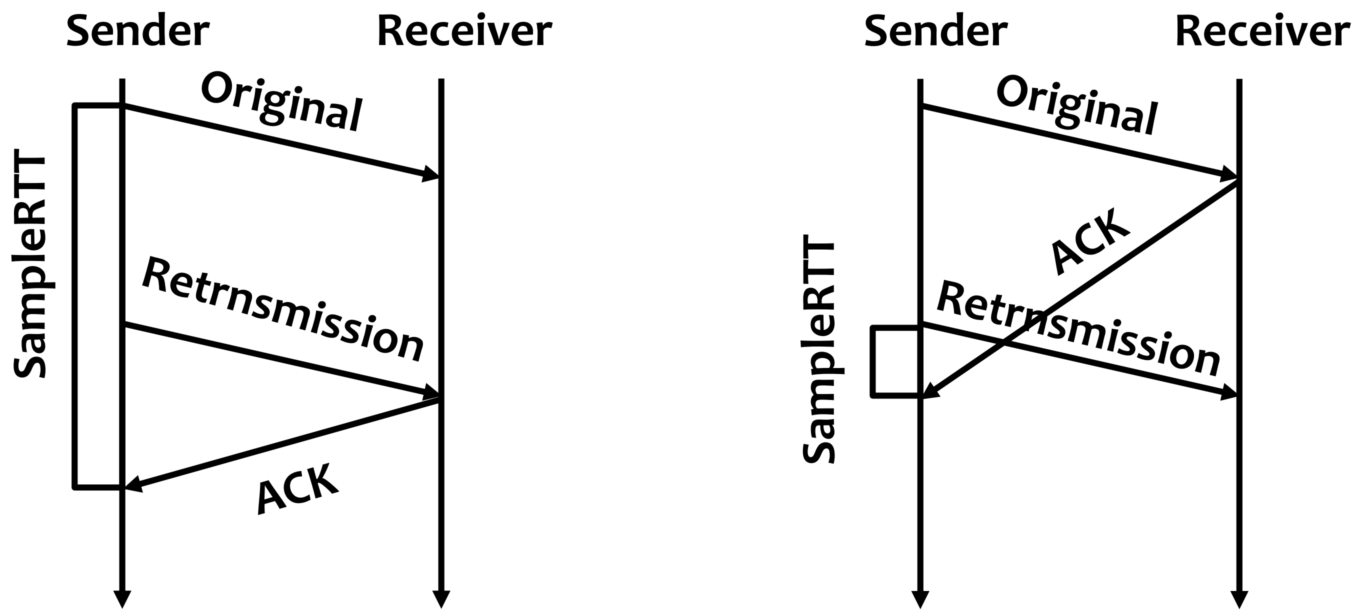
Timeout value for Retransmission

- **Original Algorithm**
  - Measure **SampleRTT** for each **segment/ ACK pair**
  - Compute **weighted average of RTT**
    - EstRTT = a x EstRTT + (1 - a)x SampleRTT
    - α between 0.8 and 0.9
  - Set timeout based on EstRTT
    - TimeOut = 2 x EstRTT

Timeout value for Retransmission

- Problem of calculating the SampleRTT
  - When a segment is retransmitted and then an ACK arrives at the sender
    - It is impossible to decide if this ACK should be associated with the first or the second transmission for calculating RTTs

Timeout value for Retransmission



Problems of associating the ACK with

(a) original transmission (should be retransmission)

(b) retransmission (should be original)

Karn/Partridge Algorithm

- Do not sample RTT when retransmitting
- Double timeout after each retransmission
- Karn-Partridge algorithm was an improvement over the original approach, but it does not eliminate congestion
- We need to understand how timeout is related to congestion
  - If you timeout too soon, you may unnecessarily retransmit a segment which adds load to the network

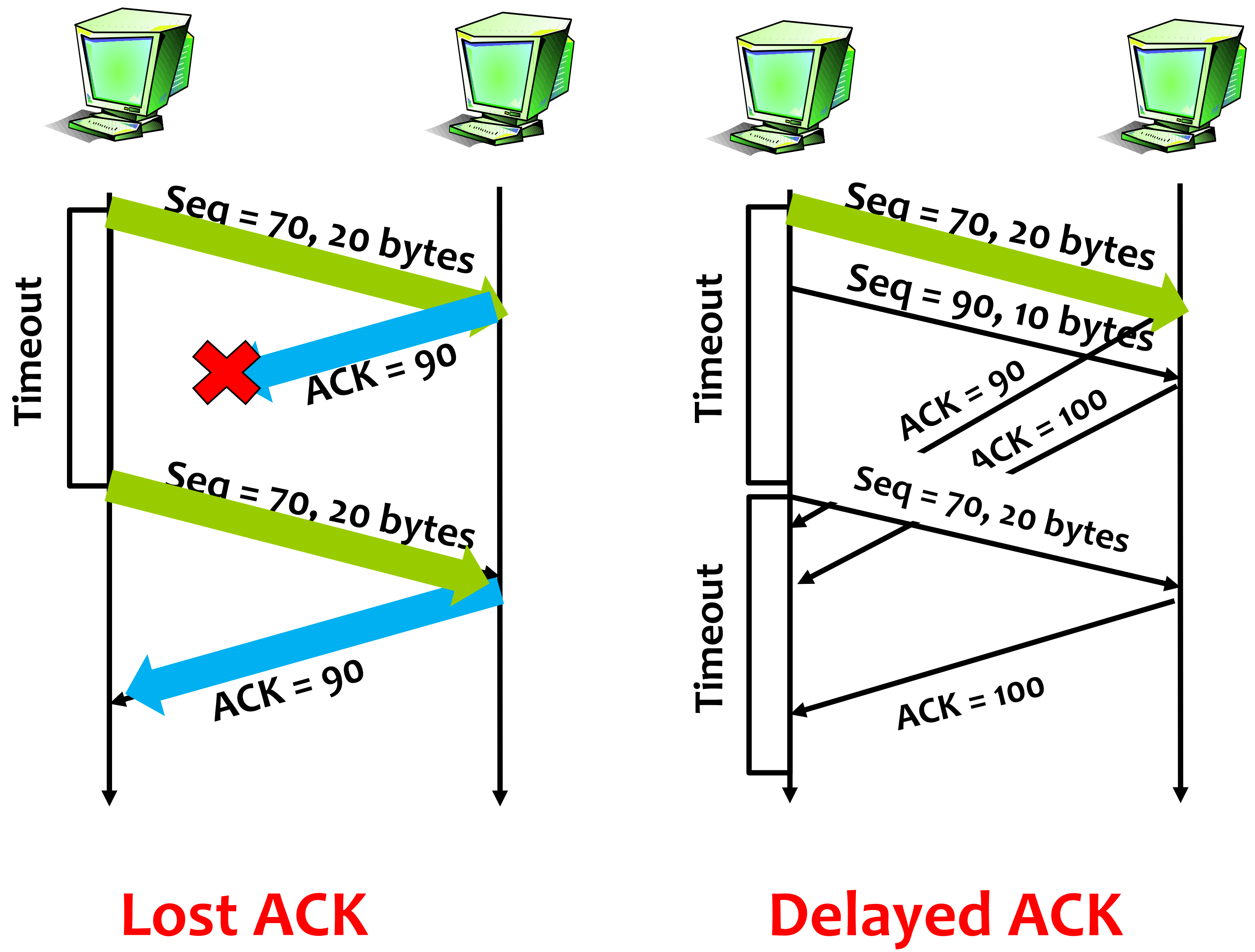
Karn/Partridge Algorithm

- Main problem with the original computation is that it does not take variance of SampleRTTs into consideration.
- For small variance among SampleRTTs
  - Then the EstimatedRTT can be better trusted
  - There is no need to multiply this by 2 to compute the timeout
- For large variance among SampleRTTs
  - The timeout value should not be tightly coupled to the Estimated RTT

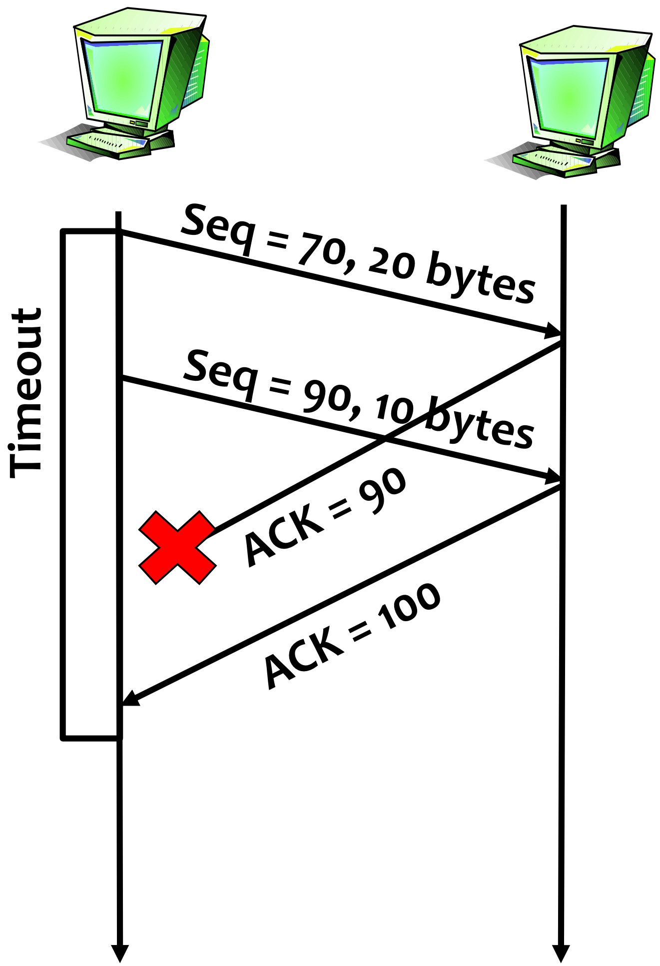
Jacobson/Karels Algorithm

- Jacobson/Karels proposed a new scheme for TCP retransmission
- Difference = SampleRTT – EstimatedRTT
- EstimatedRTT = EstimatedRTT + ($\delta \times$ Difference)
- Deviation = Deviation + δ (|Difference| – Deviation)
  - where δ is a factor between 0 and 1
- TimeOut = $\mu \times$ EstimatedRTT + $\phi \times$ Deviation
  - where based on experience, $\mu = 1$ and $\phi = 4$.
  - Thus, when the variance is small, TimeOut is close to EstimatedRTT; a large variance causes the deviation term to dominate the calculation.

TCP retransmission scenarios



TCP retransmission scenarios (more)



Cumulative ACKs

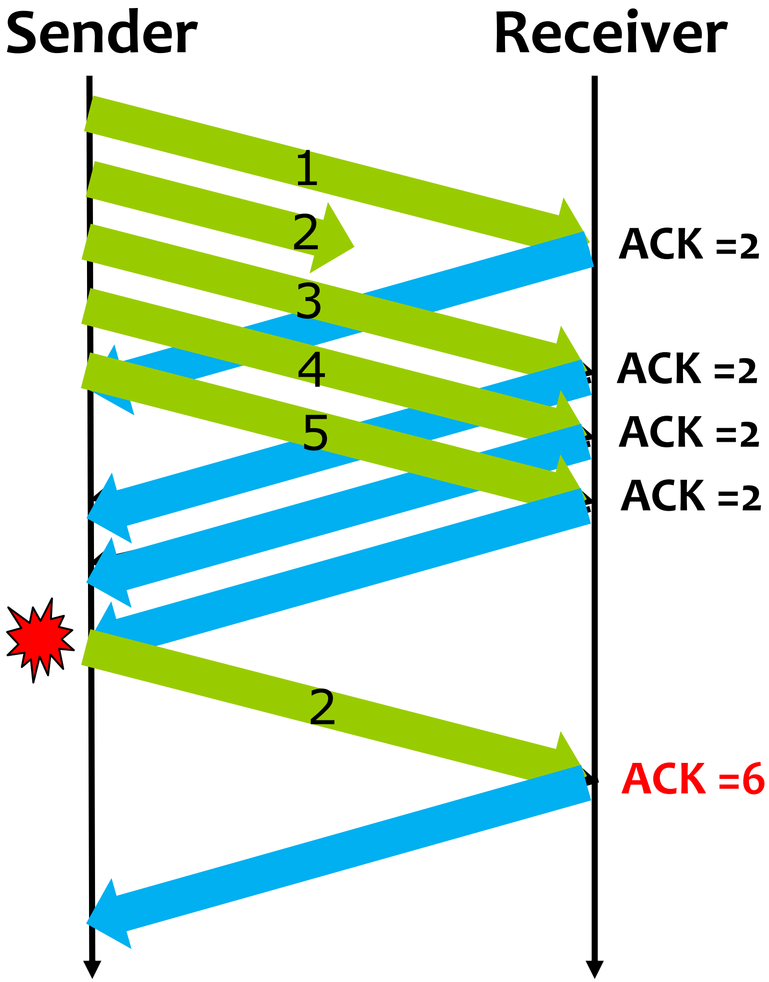
TCP Fast Retransmission

- **Fast Retransmit**
  - **Every time a data packet arrives at the receiving side, the receiver responds with an acknowledgment, even if this sequence number has already been acknowledged.**
  - **Thus, when a packet arrives out of order—TCP resends the same acknowledgment it sent last time.**
  - **This second transmission of the same acknowledgment is called a *duplicate ACK*.**

TCP Fast Retransmission

- **Fast Retransmit**
  - When the sending side sees a duplicate ACK, it knows that the other side must have received a packet out of order, which suggests that an earlier packet might have been lost.
  - Since it is also possible that the earlier packet has only been delayed rather than lost, the sender waits until it sees some number of duplicate ACKs and then retransmits the missing packet.
  - TCP waits until it has seen three duplicate ACKs before retransmitting the packet.

TCP Fast Retransmission



TCP Congestion Control

- The idea of TCP congestion control is for each source to determine **how much capacity is available in the network**, so that it knows how many packets it can safely have in transit.
- TCP is said to be ***self-clocking*** by **using ACKs** to pace the transmission of packets.

TCP Congestion Control

- **Additive Increase Multiplicative Decrease (AIMD)**
  - ***CongestionWindow*: used by the source to limit how much data it is allowed to have in transit simultaneously for a connection.**
  - **The congestion window is congestion control's counterpart to flow control's advertised window.**
  - **The maximum number of bytes of unacknowledged data allowed is now the minimum of the congestion window and the advertised window**
  - **Transmission rate**

$$\text{Rate} = \frac{\text{CongestionWindow}}{\text{RTT}} \text{ Bytes/sec}$$

TCP Congestion Control

- **Additive Increase Multiplicative Decrease (AIMD)**
  - TCP's effective window is revised as follows:
    - MaxWindow = MIN (CongestionWindow, AdvertisedWindow)
    - EffectiveWindow = MaxWindow – (LastByteSent – LastByteAcked).
  - A TCP source is allowed to send no faster than the slowest component can accommodate
    - the network or
    - the destination host

TCP Congestion Control

- **Additive Increase Multiplicative Decrease (AIMD)**
  - How to determine the value for CongestionWindow ?
  - The AdvertisedWindow is sent by the receiver.
  - But no one to send a suitable CongestionWindow to the sending side of TCP.
  - TCP source sets the CongestionWindow based on the congestion level it observed.
    - Decreasing the congestion window when the level of congestion goes up and
    - Increasing the congestion window when the level of congestion goes down.
  - Called *additive increase/multiplicative decrease (AIMD)*

TCP Congestion Control

- **Additive Increase Multiplicative Decrease (AIMD)**
  - **How does the source determine that the network is congested and that it should decrease the congestion window?**
    - **TCP interprets packet lose (3-duplicate ACK) as a sign of congestion and reduces the rate.**
    - **Each time a packet lose occurs, the source sets CongestionWindow to half of its previous value.**
    - **This corresponds to the "multiplicative decrease" part of AIMD.**

TCP Congestion Control

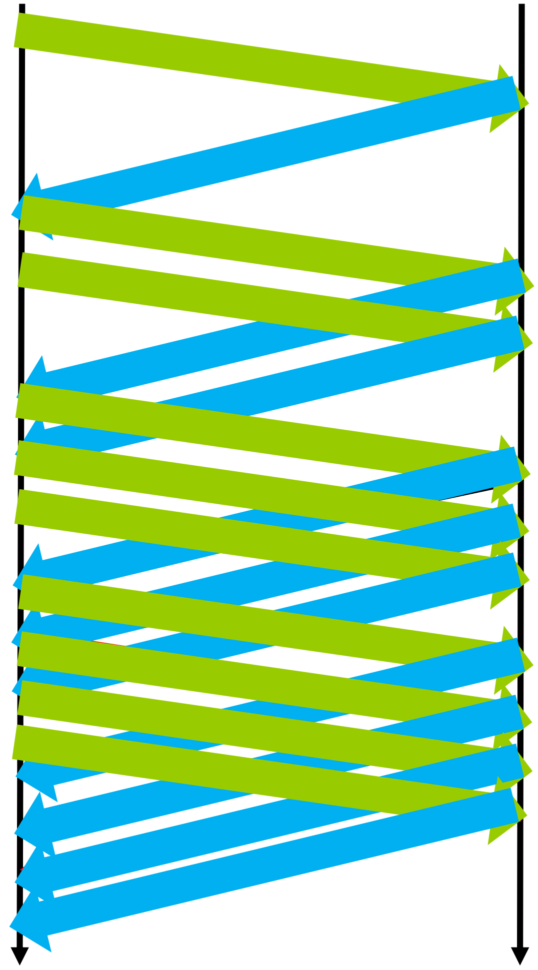
- **Additive Increase Multiplicative Decrease (AIMD)**
  - **Although CongestionWindow is defined in terms of bytes, it is easiest to understand multiplicative decrease if we think in terms of whole packets.**
    - **For example, suppose the CongestionWindow is currently set to 16 packets. If a loss is detected, CongestionWindow is set to 8.**
    - **Additional losses cause CongestionWindow to be reduced to 4, then 2, and finally to 1 packet.**
    - **CongestionWindow is not allowed to fall below the size of a single packet, or in TCP terminology, the *maximum segment size (MSS)*.**

TCP Congestion Control

- **Additive Increase Multiplicative Decrease (AIMD)**
  - **Increase the congestion window when the newly capacity of network is available.**
  - **Every time the source successfully sends a CongestionWindow's worth of packets (all packets sent out during the last RTT have been ACKed), it adds the equivalent of 1 packet to CongestionWindow.**

TCP Congestion Control

- Additive Increase Multiplicative Decrease (AIMD)



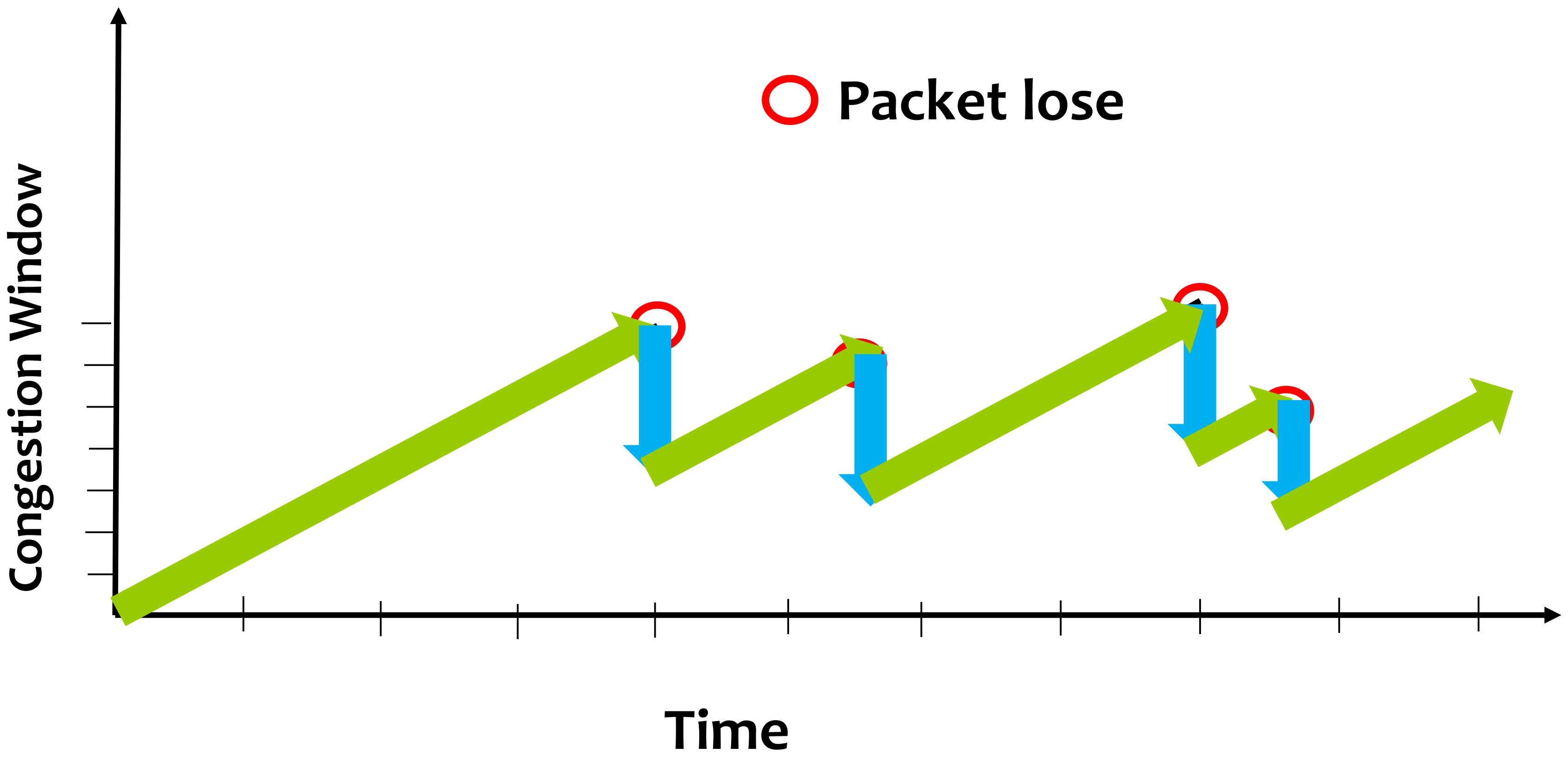
One more packet is added for each RTT

TCP Congestion Control

- Additive Increase Multiplicative Decrease (AIMD)
  - TCP does not wait for an entire window's worth of ACKs to add 1 packet's worth to the congestion window, but instead increments CongestionWindow by a little for each ACK.
  - 不等所有 ACK 都收到才加 1 (MSS bytes), 每收到一個 ACK 就先加一部分 (CW 可以較早滑動)
  - 例如
    - CW = 5 x MSS,每收到一個 ACK 就先加 1/5 MSS
    - CW = 8 x MSS,每收到一個 ACK 就先加 1/8 MSS
  - Increment = MSS × (MSS/CongestionWindow)
  - CongestionWindow += Increment

TCP Congestion Control

- Additive Increase Multiplicative Decrease (AIMD)
- Trace: Sawtooth behavior



TCP Congestion Control

- Slow Start
  - The additive increase mechanism is good when the source is operating close to the available capacity of the network, but it takes too long to ramp up a connection when it is starting from scratch.
  - Slow start: to increase the congestion window rapidly from a cold start.
  - Slow start effectively increases the congestion window exponentially, rather than linearly.

TCP Congestion Control

- Slow Start
  - Initially, the CongestionWindow = 1 packet.
  - Example: MSS = 500 bytes, RTT = 200 msec
    - initial rate = 20 kbps
  - When the ACK for this packet arrives, TCP adds 1 to CongestionWindow and then sends two packets.
  - 每收到一個 ACK 就加 1 packet (MSS)
  - Upon receiving the corresponding two ACKs, TCP increments CongestionWindow by 2— one for each ACK—and next sends four packets.
  - TCP effectively doubles the number of packets it has in transit every RTT.

TCP Congestion Control (Slow Start)

- **Slow Start**
- **When connection begins, increase rate exponentially until first loss event:**
  - double `CongWin` every RTT
  - done by incrementing `CongWin` for every ACK received
- **initial rate is slow but ramps up exponentially fast**

Packets in transit during slow start

TCP Congestion Control

- After 3 dup ACKs:
  - CongWin is cut in half
  - window then grows linearly
- But after timeout event:
  - CongWin instead set to 1 MSS;
  - window then grows exponentially
  - to a threshold, then grows linearly

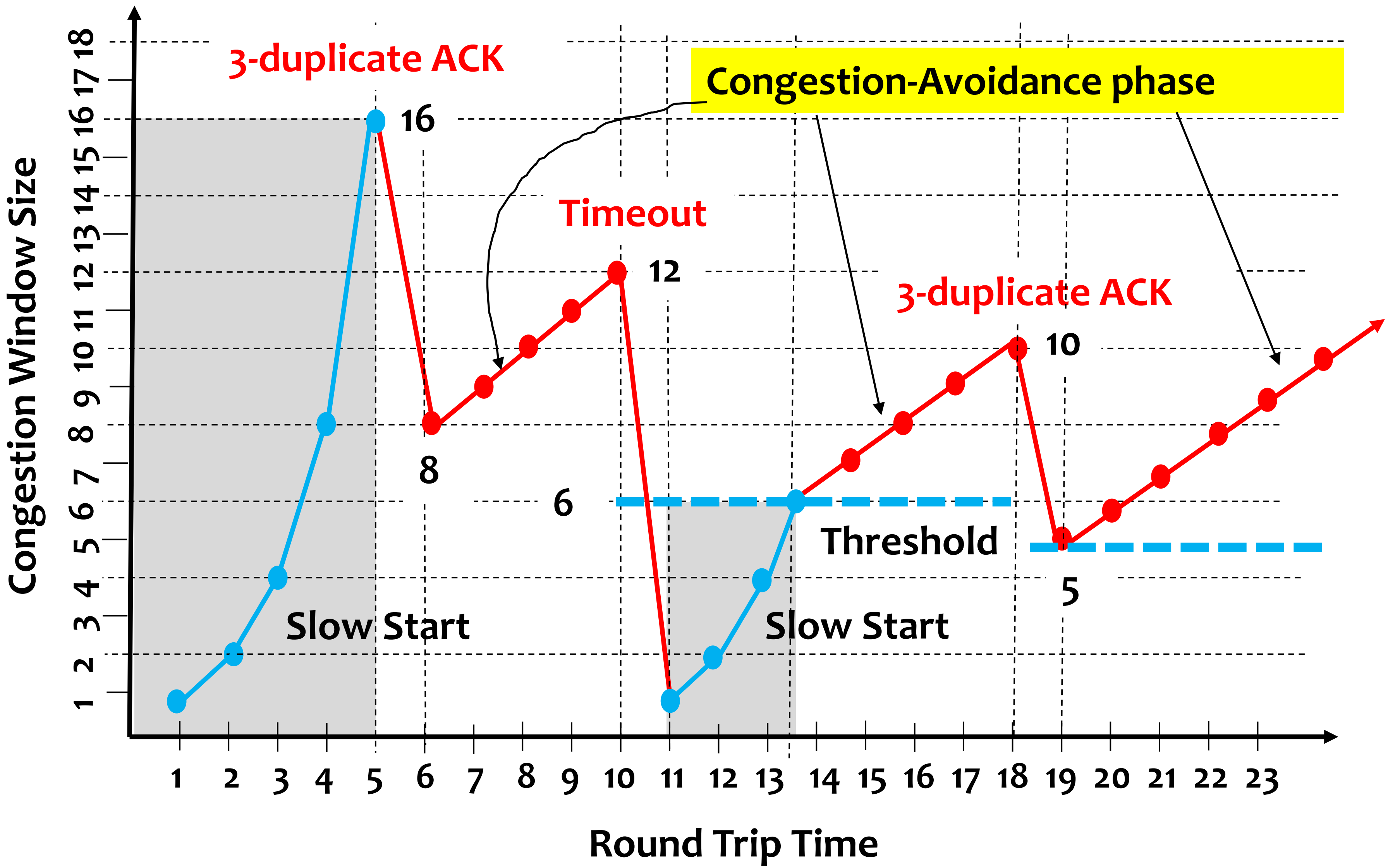
note

- 3 dup ACKs indicates network capable of delivering some segments
- Timeout indicates a "more alarming" congestion scenario

TCP Congestion Control

- Summary :
- When CongWin is below Threshold, sender in slow-start phase, window grows exponentially.
- When CongWin is above Threshold, sender is in congestion-avoidance phase, window grows linearly.
- When a triple duplicate ACK occurs, Threshold set to CongWin/2 and CongWin set to Threshold.
- When timeout occurs, Threshold set to CongWin/2 and CongWin is set to 1 MSS.

TCP Congestion Control



TCP Sender Congestion Control

State	Event	TCP Sender Action	Commentary
Slow Start (SS)	ACK receipt for previously unacked data	CongWin = CongWin + MSS, If (CongWin > Threshold) set state to "Congestion Avoidance"	Resulting in a doubling of CongWin every RTT
Congestion Avoidance (CA)	ACK receipt for previously unacked data	CongWin = CongWin+MSS * (MSS/CongWin)	Additive increase, resulting in increase of CongWin by 1 MSS every RTT
SS or CA	Loss event detected by triple duplicate ACK	Threshold = CongWin/2, CongWin = Threshold, Set state to "Congestion Avoidance"	Fast recovery, implementing multiplicative decrease. CongWin will not drop below 1 MSS.
SS or CA	Timeout	Threshold = CongWin/2, CongWin = 1 MSS, Set state to "Slow Start"	Enter slow start
SS or CA	Duplicate ACK	Increment duplicate ACK count for segment being acked	CongWin and Threshold not changed

TCP throughput

- What's the average throughout of TCP as a function of window size and RTT ?
  - Ignore slow start
- Let W be the window size when loss occurs.
- When window is W, throughput is W/RTT
- Just after loss, window drops to W/2, throughput to W/2RTT.
- Average throughout: 0.75 W/RTT

Summary

- We have introduced how to convert host-to-host packet delivery service to process-to-process communication channel.
- UDP for unreliable transmission service
- TCP for reliable transmission service
  - 3-way handshaking connection establishment
  - TCP connection state diagram
  - TCP timeout value calculation
  - TCP retransmission scenarios
  - TCP fast retransmission

Summary

- TCP Congestion Control
  - AIMD (additive Increase Multiplicative Decrease)
  - Slow start
  - 3-duplicate ACKs (packet lose)
  - Timeout