Transport Layer Protocols:

TCP (Transmission Control Protocol)

Transport Layer Functions

- Multiplex/Demultiplex traffic for different applications
- Other possible function a protocol may or may not implement
  - Establish and maintain end-to-end connections
  - Guarantee reliable, inorder, end-to-end transfer
  - Provide end-to-end flow control
  - Congestion control
  - TCP implements all of the above, UDP implements none
- Lowest level end-to-end protocol
  - Header generated by sender is interpreted only by the destination, not by routers
  - Routers view transport header as part of the payload/data

Transmission Control Protocol (TCP)

- End-to-end, reliable, in-order transfer
- Connection-oriented protocol
  - Explicit connection establishment and termination
- Stream oriented
  - Data to be sent is interpreted as a stream of bytes, with no boundaries
  - Actual transfer breaks this stream into packets of arbitrary size
- Full duplex
- Flow control, Error detection and control
- Congestion Control
- Specified originally in RFC 793, many other related RFCs are there

Port, Endpoint, and Connection

- Port
  - A 16 bit integer used to identify an application using
- Endpoint
  - A 2-tuple <host, port>
  - host is an IP address
  - Commonly called a *socket*
- Connection
  - Defined by two endpoints
  - Two connections will have at least one endpoint different (but can have one endpoint same)
  - Messages are demultiplexed based on connections, not ports

Classification of Ports

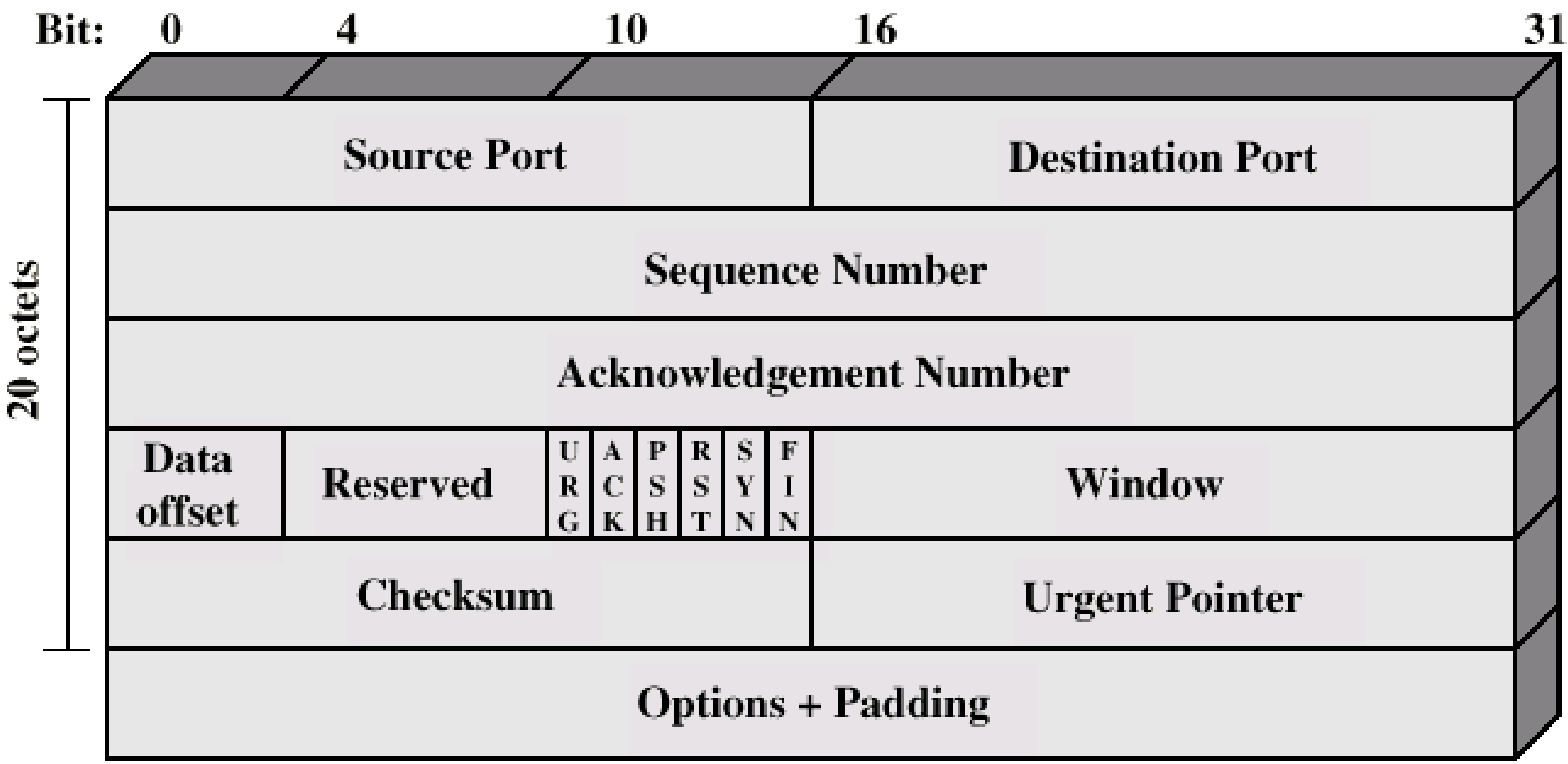
- Well-known/reserved ports
  - Ports upto 1023
  - Normally used for protocols with wide applicability
  - Assigned by IANA
  - Examples
    - ftp – 21,20, telnet – 23 etc.
- Registered ports
  - Ports 1024 – 49151
  - Used to avoid port collisions between user-level applications developed independently when installed on the same machine
  - Registration in ICANN registries is voluntary, though recommended
- Dynamic or private ports (also called ephemeral ports)
  - Ports 49152 – 65535
  - Can be used by anyone for anything

Stream and Segment

- Data viewed as a byte stream, i.e., a sequence of bytes
- Segment – the unit of transfer between TCP s/w on two machines
  - The stream of bytes is divided into segments, each segment is given a TCP header, and transmitted
  - Usually 1 segment is encapsulated in 1 IP datagram
  - Segments may not contain any data
    - Ex – segments used to establish/terminate connections, send acks etc.
- Sequence Number – used to specify position within the stream
  - Each TCP segment will contain a 32-bit sequence no. to identify its position in the stream

- Maximum Segment Size (MSS)
  - Maximum size of a segment (excluding header)
  - Ideally, should be (Minimum MTU of any link from the source to the destination – TCP header size – IP header size)
    - Avoids fragmentation and reassembly at IP layer, which is costly
  - Hard to know minimum MTU of links end-to-end, though can be known in some cases, for ex, if all links are Ethernet
  - Default MSS = 536
    - All IP based networks must support a MTU of 576
  - Can be changed during connection establishment time using TCP options (will see shortly)
    - Cannot be changed once connection is established

TCP Header


Bit: 0 4 10 16 31
20 octets
Source Port
Destination Port
Sequence Number
Acknowledgement Number
Data offset
Reserved
URG
ACK
PSH
RST
SYN
FIN
Window
Checksum
Urgent Pointer
Options + Padding

- Before looking at the fields, note that TCP is full-duplex
  - If A establishes a connection to B, data can flow from either A to B or B to A
- For each-way transfer, say from A to B, some header fields are used for the forward direction (for example, data from A to B) and some for the reverse direction (for example, ack from B to A for the data sent from A to B)
- Piggybacking is used to reduce number of messages
  - Example: if data is transferred from A to B, the ack for it from B to A can be piggybacked (as value in a header field) when data is transferred from B to A

Header Fields

- Source Port and Destination Port: identifies sending and receiving applications
- Sequence Number: byte number of the first byte in this segment in the data stream sent by the application
  - Note that this is an actual byte number and not offset from the start, as first byte's sequence number can be random and need not be 0 or 1
- Acknowledgement Number: byte number that the sender of this message expects to receive next (for the data stream being sent in the opposite direction)

- Data offset (also called HLEN): header length in multiples of 4 bytes
  - Specified where does the data start in the segment
  - Needed because Options filed in the header can have variable length
- Window: How much data (in bytes) is the sender of the message (i.e., the receiver of the stream being sent in the opposite direction) willing to accept
  - Basically, size of free space in the buffer for the sender of the message
  - This field is advertised on all segments, carrying data or ack

- TCP Options
  - Window Scaling: provides multiple by which the window size advertised is to be multiplied
  - Maximum segment size: allows a receiver to specify the maximum size of a segment it is willing to accept
  - Selective Acknowledgments (SACK)
  - Some others, we will not do
- Checksum
  - For error detection
  - 16-nit word size, one's complement of the one's complement sum of words
  - However, uses a pseudo-header (will see shortly)

- TCP Flags

U R G	**A C K**	**P S H**	**R S T**	**S Y N**	**F I N**

  - URG - urgent pointer is valid
  - ACK - the acknowledgment number is valid
  - PSH - The receiver should pass this data to the application as soon as possible ("push")
  - RST - reset the connection
  - SYN - synchronize the sequence numbers to initiate a connection
  - FIN - sender is finished sending data

- Note that IP address is not part of TCP header
  - This is as it should be, as IP address is a network layer information
- Then how does TCP software use the IP addresses to identify connections?
  - IP header is stripped off before the data is passed up to TCP layer
- IP software passes the source and destination addresses separately with each segment sent up

Computing TCP Checksum

- Prepends a psuedo-header before the TCP segment
- Pads octets of zeros to make (psuedo-header + TCP segment + pad length) multiple of 16
- Checksum computed on this entire thing
- Psuedo-header and pad octets are not transmitted, just used for calculating the checksum (*then why have it?*)
- Receiver will do the same and compare checksum

TCP psuedo-header

0		16 … 31
Source IP address		
Destination IP address		
Zero	Protocol	TCP Length

- Protocol = 6 (value for TCP in protocol field of IP datagram)
- TCP Length = length of TCP segment incl. TCP header (but not incl. psuedo-header and pads, it is computed (*how?*))

Basic Data Transfer

- Connection established
- All transmission involves TCP segments
- Application generates data
- TCP software puts application data in send buffer
- TCP segments are formed from the data in the send buffer
  - Can be of different sizes, formed at different times (we will see more details)
  - TCP header added
- Sender sends the TCP segments one by one as they are formed subject to send window size
- Timer set for each segment set, sender waits for acknowledgments
- If timeout, retransmit. If ack received, change window and transmit more segments if possible

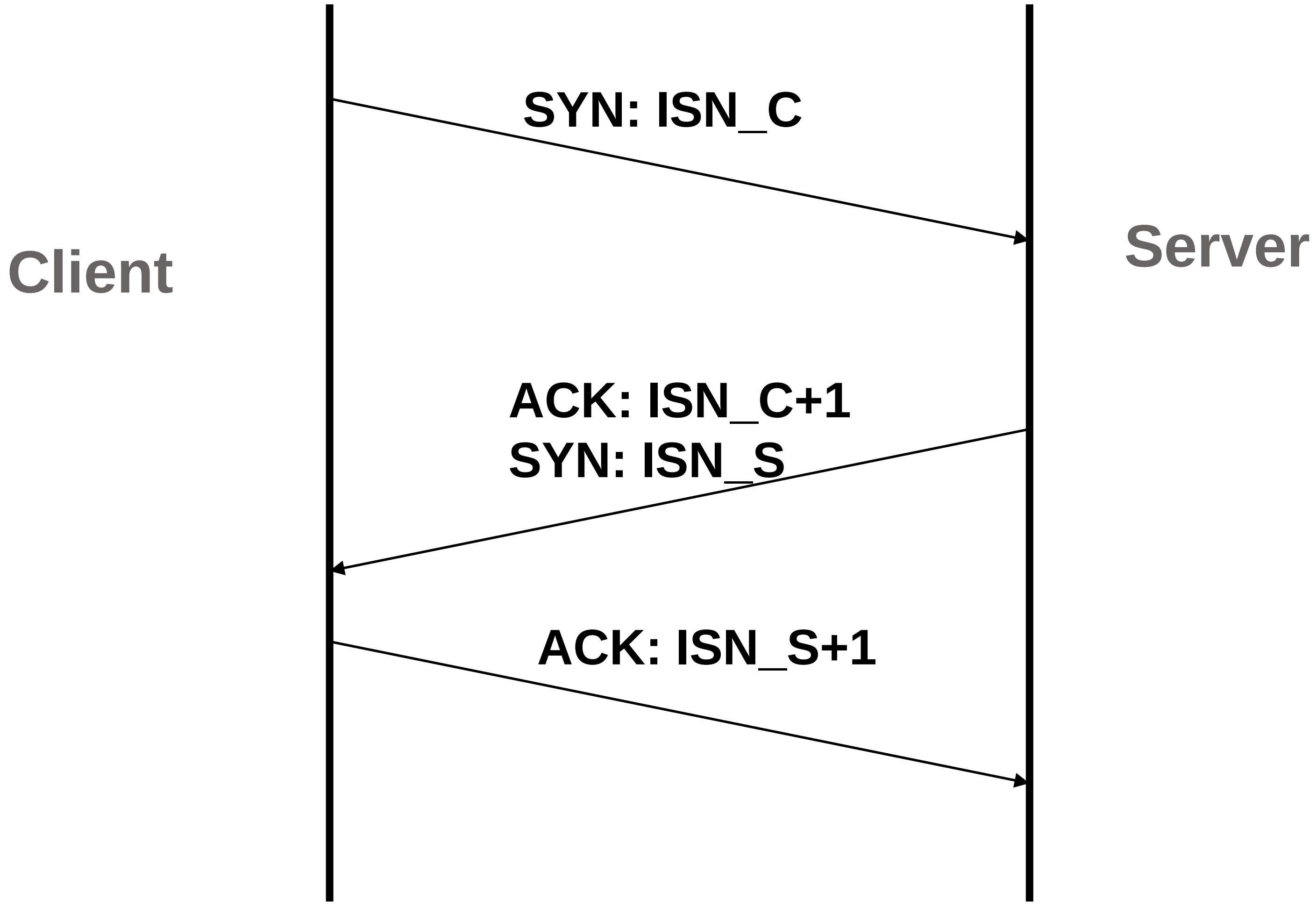
- Receiver sends acknowledgments by sending another TCP segment
  - Can be an acknowledgements segment (ACK flag set and a valid acknowledgement no. field) with no data (how to know this?)
  - Acknowledgements can be piggybacked on TCP segments carrying data in the other direction
  - Acknowledgement specifies next sequence no. the receiver expects
  - Cumulative positive acknowledgement, no NAK
- The above repeats until connection is terminated

Connection Establishment

- Purpose
  - Both sides should know that both sides are ready for data transfer
  - Each side should know the other side's Initial Sequence Number (ISN) – the starting sequence number of the first byte in the stream that will be sent
    - First byte address cannot always start at 0 or 1, there are problems of confusion between old and new connections
    - ISN's are usually chosen randomly
    - For full-duplex communication between A and B, A should know B's ISN and B should know A's ISN
  - Can negotiate certain TCP options also
- Done through Three-Way Handshake

Three-Way Handshake

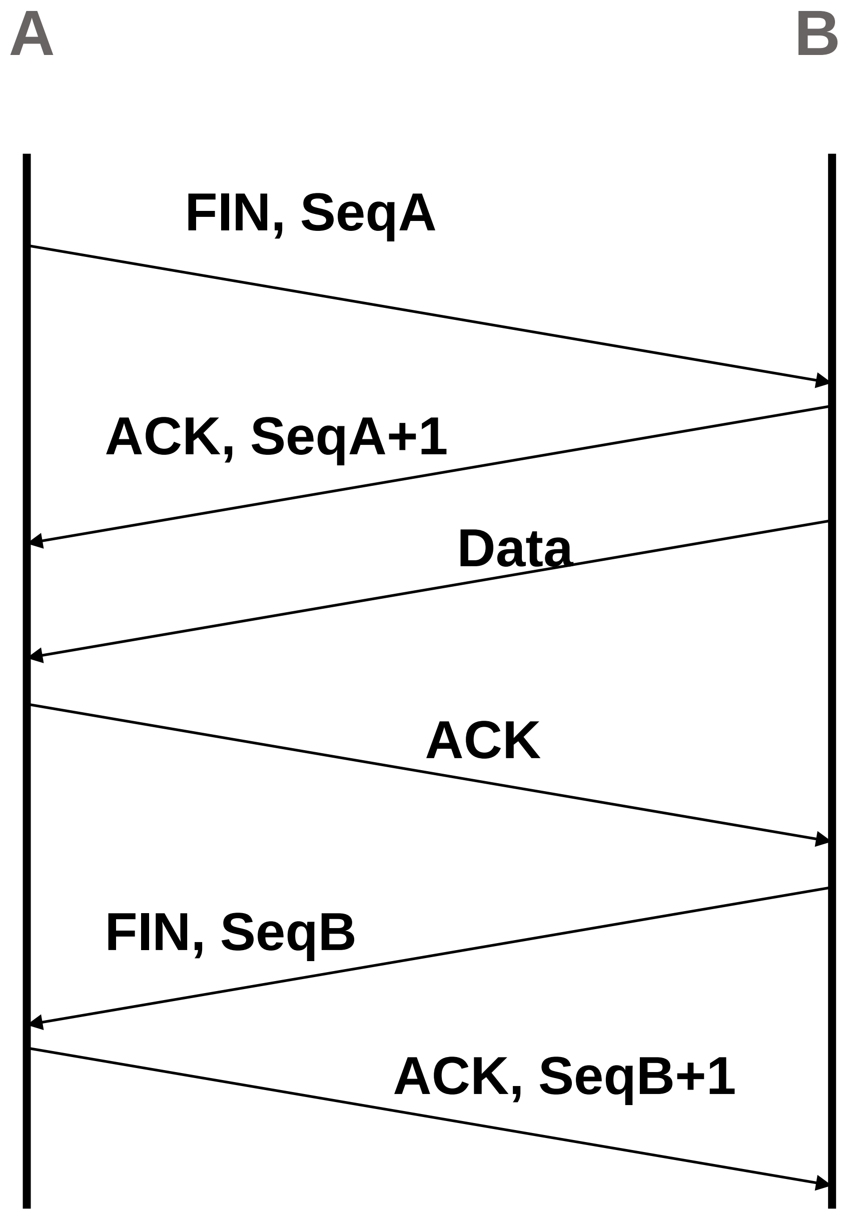
- The client sends a SYN segment (SYN flag set) specifying the port number of the server and the client's ISN
- The server responds with a SYN + ACK segment (SYN and ACK flags set)
  - Sequence No. field contains server's ISN
  - Server acknowledges the client SYN using client ISN+1 in the acknowledgement no. field.
- The client acknowledges the SYN from the server using an ACK segment (ACK flag set) with the server's ISN+1 in the acknowledgement no. field
- The side sending the first SYN is said to perform an active open. The other side performs a passive open.
- However, after a connection is established, it is full-duplex communication, with no master/slave



- Full duplex communication can now happen

Normal Connection Termination

- First note that a TCP connection opened is full-duplex. For termination, it is viewed as 2 separate connections, one in each direction
  - Each can be terminated separately
- Either side can initiate termination
  - Send FIN segment (FIN flag set)
  - Indicates that FIN sender is not going to send any more data
- Acknowledging FIN
  - FIN receiver must acknowledge the FIN segment by sending an ACK segment (ACK flag set) with acknowledgement no. = FIN segment sequence number + 1
- FIN receiver can continue sending data
  - Half open connection
  - FIN sender must continue to acknowledge, just cannot send any more data
- The above repeats when the other side wants to close connection


A
B
FIN, SeqA
ACK, SeqA+1
Data
ACK
FIN, SeqB
ACK, SeqB+1

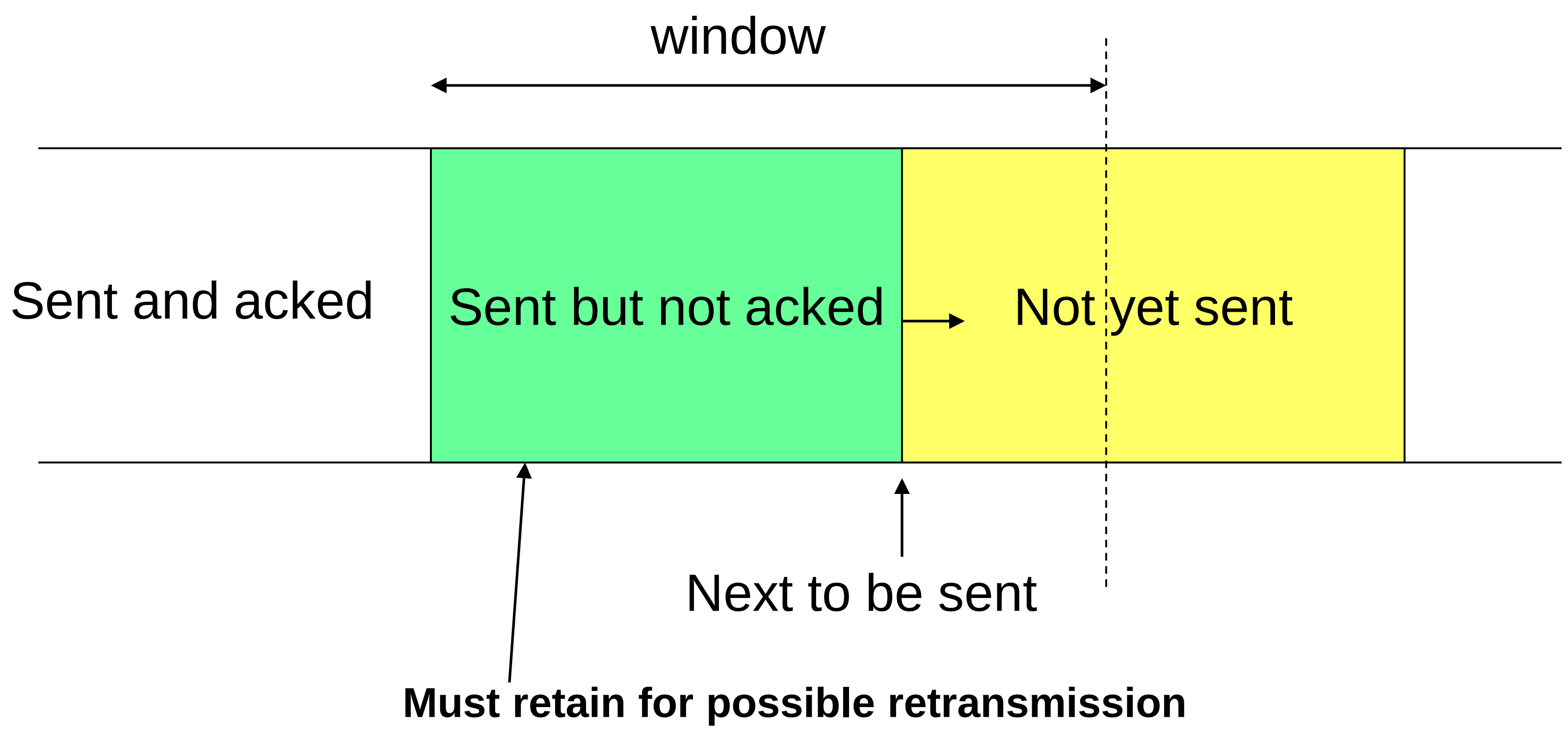
TCP Flow Control

- TCP uses a sliding window protocol without selective or negative acknowledgments.
  - Selective acknowledgments would let the protocol say it's missing a range of bytes. TCP can only say that it has received "up to byte N".
  - The protocol has no way to specify a negative acknowledgment. It can only say what has been received
- Concepts same as discussed earlier, with some differences

Sliding Window Based Flow Control

- TCP sliding window works at byte level
  - So we talk about how many bytes are sent and ack'ed, utpto what byte can be sent etc.
  - NOT how many segments are sent etc.
- Sender maintains a window of size n and start of window X
- Sender can send up to n bytes starting from byte X without receiving an acknowledgement
- When the first p bytes of data are acknowledged then the window slides forward by p bytes to $X+p$. Sender can now send n bytes starting from $X+p$
- Window size determines how much unacknowledged data can the sender send as usual

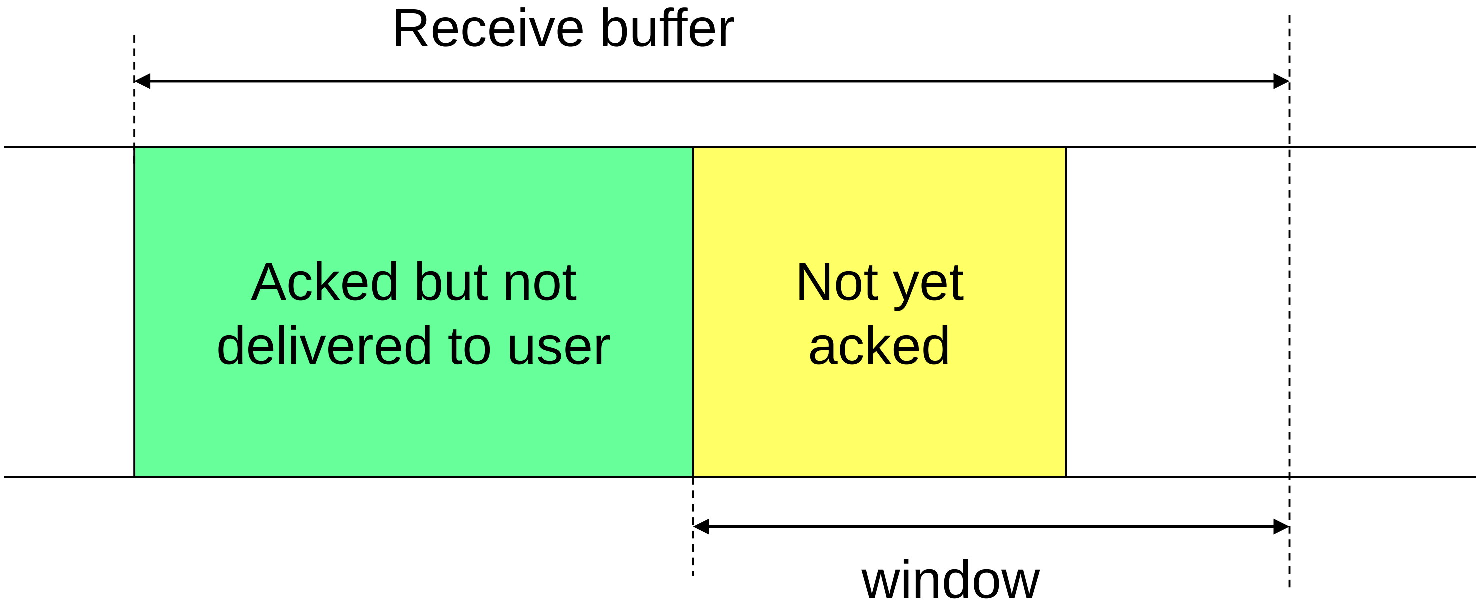
Sender Side Window



Problem

- Acknowledgment may be sent immediately by receiver, but receiver can delete acknowledged data from its buffer only after the data has been delivered to the application
- Application may read the data at different speeds and at different times
- So, depending on when and how fast the application reads the data, the receiver's window size may change
  - So even if the receiver current window size is W bytes and the receiver receives and acknowledges p bytes, the window size may not be reset to W, may still be only W – p until the application picks up the data

Receiver Side Window



Solution

- Receiver tells sender what is the current window size in every segment it transmits to the sender (in Window field of header)
  - This can be data sent from receiver to sender in the other direction, or ack's for the data received from the sender
- Sender uses this advertised window size instead of a fixed value
- Window size (also called advertised window) is continuously changing, = current free buffer space at receiver
- Can go to zero; sender not allowed to send anything!
- Naïve implementations can cause silly window syndrome

Silly Window Syndrome

- The problem of TCP sending very small segments
  - Small segments are bad
    - Too much overhead given headers have to go with each segment
- Can be caused by both sender side and receiver side applications
  - Sender side application sending data very slowly
  - Receiver side application reading data from the receive buffer very slowly

Receiver Side Silly Window Syndrome

- Suppose that sender has lot of data to send, and sends one window full of data
- The application on the receiver side does not read the data yet, so receiver's buffer is full, so receiver advertises a window size of 0, sender blocks (but has more data to send)
- The application now reads very slowly, say 1 byte at a time
- For each 1 byte read, the receiver advertises a window of size 1
- Sender sends 1 more byte of data
- This repeats, causing a lot of 1-byte segments to be sent

- Solutions
  - Do not advertise small sized windows
    - Acknowledge data frames that arrive, but keep advertising a 0-sized window until either (i) half of the receive buffer is free, or (ii) receiver buffer has at least MSS amount of free space
  - Delay the acknowledgements for data frames that arrive
    - Sender side window cannot move, so cannot transmit more data than current window

Sender Side Silly Window Syndrome

- Caused when the application generates data slowly
  - Say 1 byte every time
- If data is sent immediately, lots of small segments are sent
- If it is to be delayed, how long to delay? No clue when the Sender side application will generate data again
- Nagle's Algorithm:
  - If there is previous data that is sent but not acknowledged, place any further data to be sent in the send buffer but do not send until:
    - Either an acknowledgement is received, or
    - MSS sized data is available for sending
- Nagle's algorithm decides when are segments sent by a TCP sender, subject to window restrictions

When are ACKs sent?

- Suppose A is sending data to B
- TCP acks are cumulative, acknowledges the longest contiguous sequence from the start that is received
  - If ISN of A = 1000, A has sent 4 segments with sequence numbers 1001 (why not 1000?), 1600, 2800, and 3100, and B has received the 1st, 2nd and 4th segments only, then
    - Longest contiguous sequence from start received is byte numbers1001 to 2799
    - TCP will ack with ack no. = 2800 (next byte expected)
    - Segment 1 and 2 are received in-order, Segment 3 is a missing segment, Segment 4 is a segment received out-of order
      - Note that receiver does not know there is one missing segment, it just knows there is at least one

- B sends an acknowledgement if/when
  - If B has data to send to A, always piggyback the ack for the data received from A (ACK flag set, byte no. of next byte to expect put in Acknowledgement Number field)
  - If B has no data to send, receives a segment from A in-order (sequence number = next sequence number expected)
    - If all previous in-order segments are acknowledged, delay sending the acknowledgement until one more segment arrives or a time elapses (typically 500 milliseconds), then send
  - If B gets an out-of-order segment having a higher than expected sequence number, send an ack with next sequence number expected
  - If the receiver gets a missing segment which extends the longest contiguous byte stream from the start that it has received, send an ack with the next sequence number to expect
  - If a duplicate segment arrives, discard the segment but send an ack with the next sequence number expected

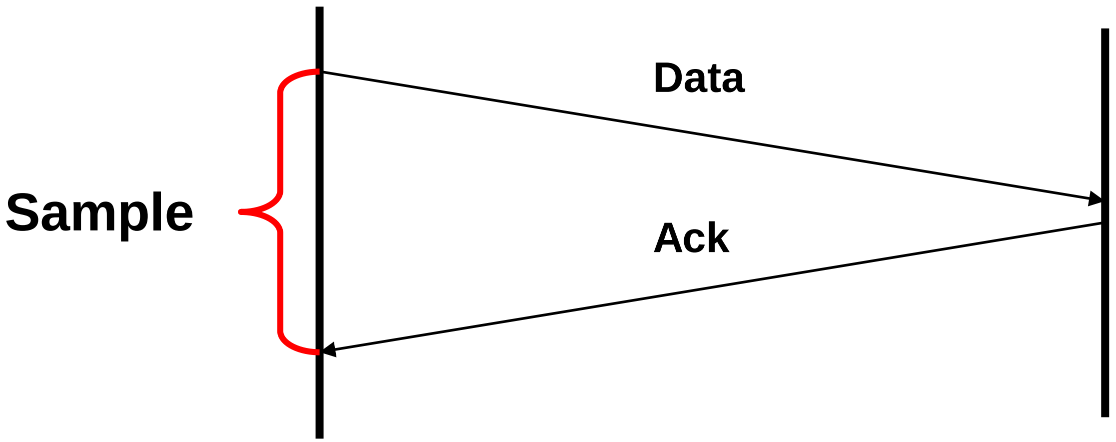
Ensuring Reliable, Inorder Transfer

- Checksum (mostly) guarantees end-to-end data integrity
- Sequence numbers detect packet sequencing problems:
  - Duplicate: ignore
  - Reordered: reorder or drop
  - Lost: retransmit
- Lost segments detected by sender
  - Use timeout to detect lack of acknowledgment
- Retransmission requires that sender keep copy of the data.
  - Copy is discarded when ack is received

Ensuring Reliability

- Keeps separate timer for each unacknowledged segment
- Uses retransmissions of segments whose timers expire
- Retransmission time-out depends on round-trip delay
  - Round-trip delay varies based on path followed, network condition etc. So how to set?
  - Solution - estimate RTT dynamically

Estimating Round-trip Delay



- Every Data/Ack pair gives new RTT estimate
- Can get lots of short-term fluctuations

TCP Round-trip Estimator

- Original
  - Round trip times estimated as a moving average:
    - New RTT = α (old RTT) + (1 – α) (new sample)
  - Set timeout to $\beta \times$RTT
  - Typically, $\alpha = 0.8$-0.9, $\beta = 2$ originally
- However, a fixed β does not adapt well to high variance in RTT
  - Ideally, timeout > real RTT
    - = estimated RTT + X
  - If there is a high variance in the RTT, need a higher X to ensure timeout is always greater than RTT
  - RTT variance is high at high loads

TCP Round-trip Estimator

- Modified:
  - Set timeout = Estimated RTT + δ × deviation
  - $\delta = 4$ typically
- How to compute the deviation in RTT?
  - Deviation = $(1-\rho)\times$deviation + ρ × (sample RTT – estimated RTT)
  - Typically, $\rho = 0.25$

Acknowledgement Ambiguity

- If a segment is retransmitted, and an ack for it is received, is it an ack for the retransmitted frame or the original frame? What RTT sample value to take?
- Solution: Karn's algorithm
  - Do not accept samples for segments that are retransmitted
  - Use a timer backoff scheme to increase timeout to account for scenarios when round trip delay increases suddenly

Karn's Algorithm

- Compute timeout using the estimated round trip time as before
  - Use only samples for segments that are not retransmitted
- If timer expires and retransmission occurs
  - For every retransmission, set timeout $= \gamma \times$ timeout, subject to an upper limit
  - Typically, $\gamma = 2$

Ensuring In-order Delivery

- TCP uses sequence number field in segment headers to reconstruct the data stream at the receiver side
- What happens to segments received out-of-order?
  - TCP specification does not restrict, up to implementations

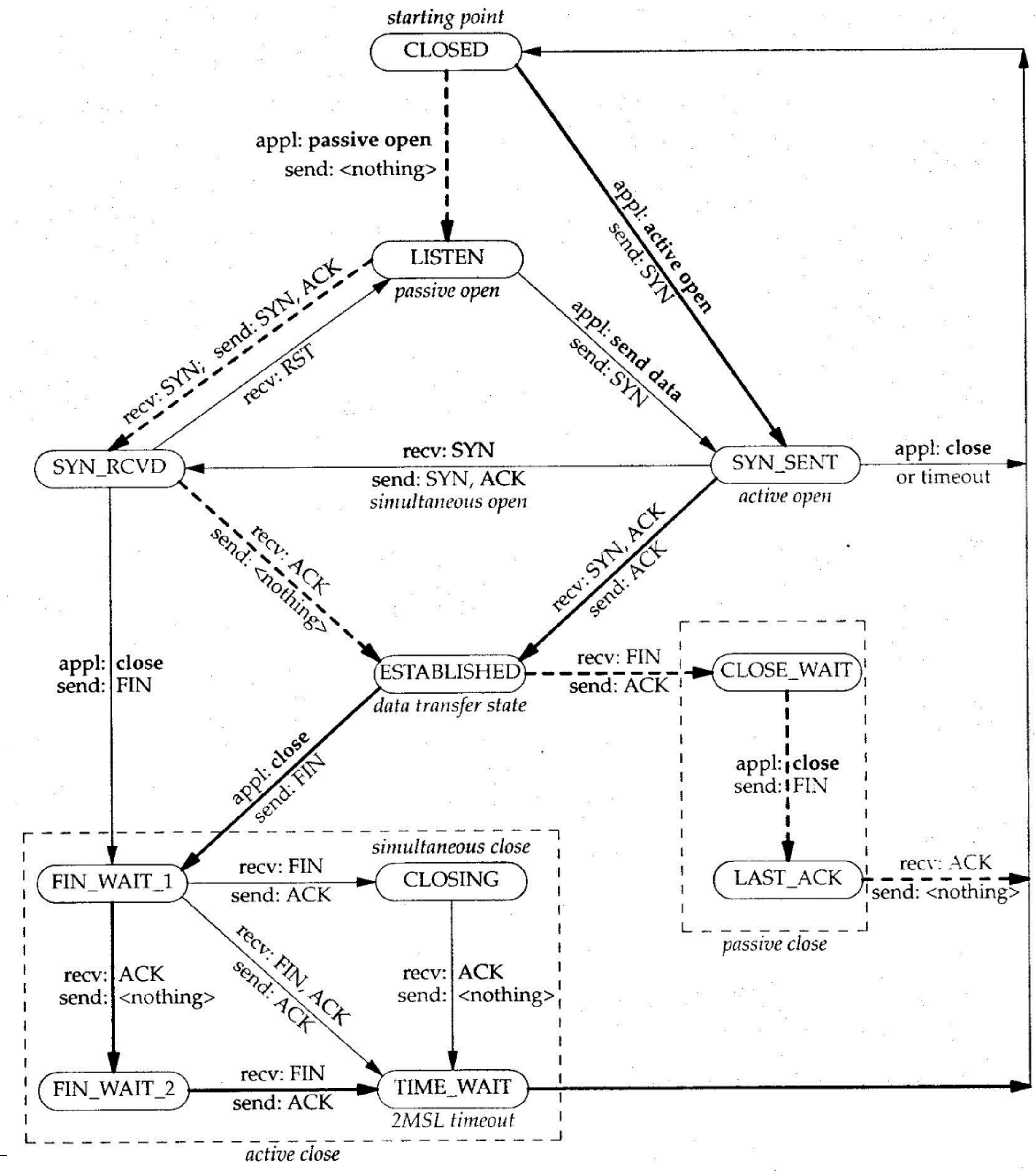
Out of Band Data

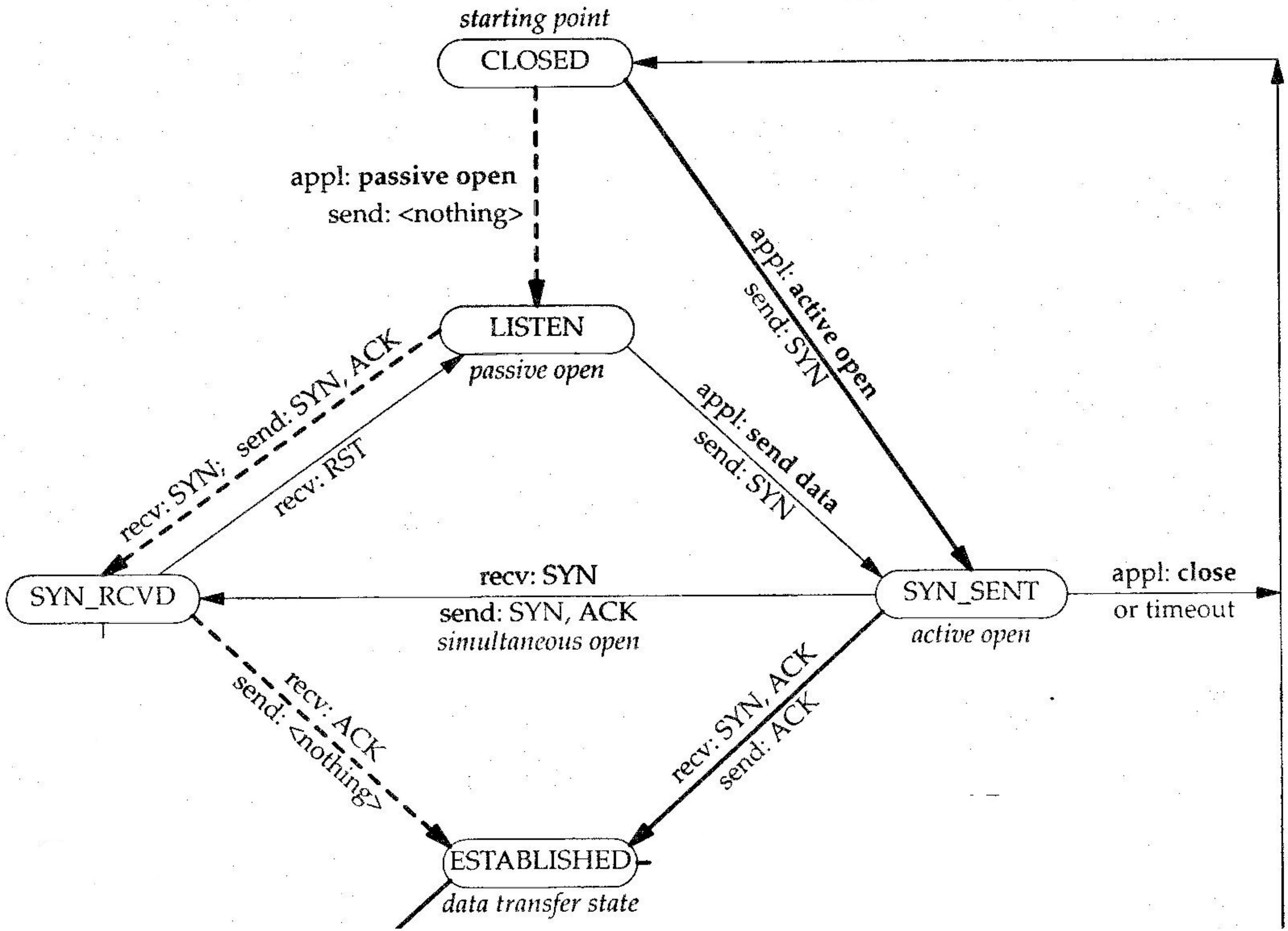
- Sometimes there is need to send urgent data that needs to be delivered to the application out of turn
- Set URG flag bit to indicate presence of out-of-band data in segment
- Set URGENT pointer to position in segment where urgent data ends

Resetting a TCP Connection

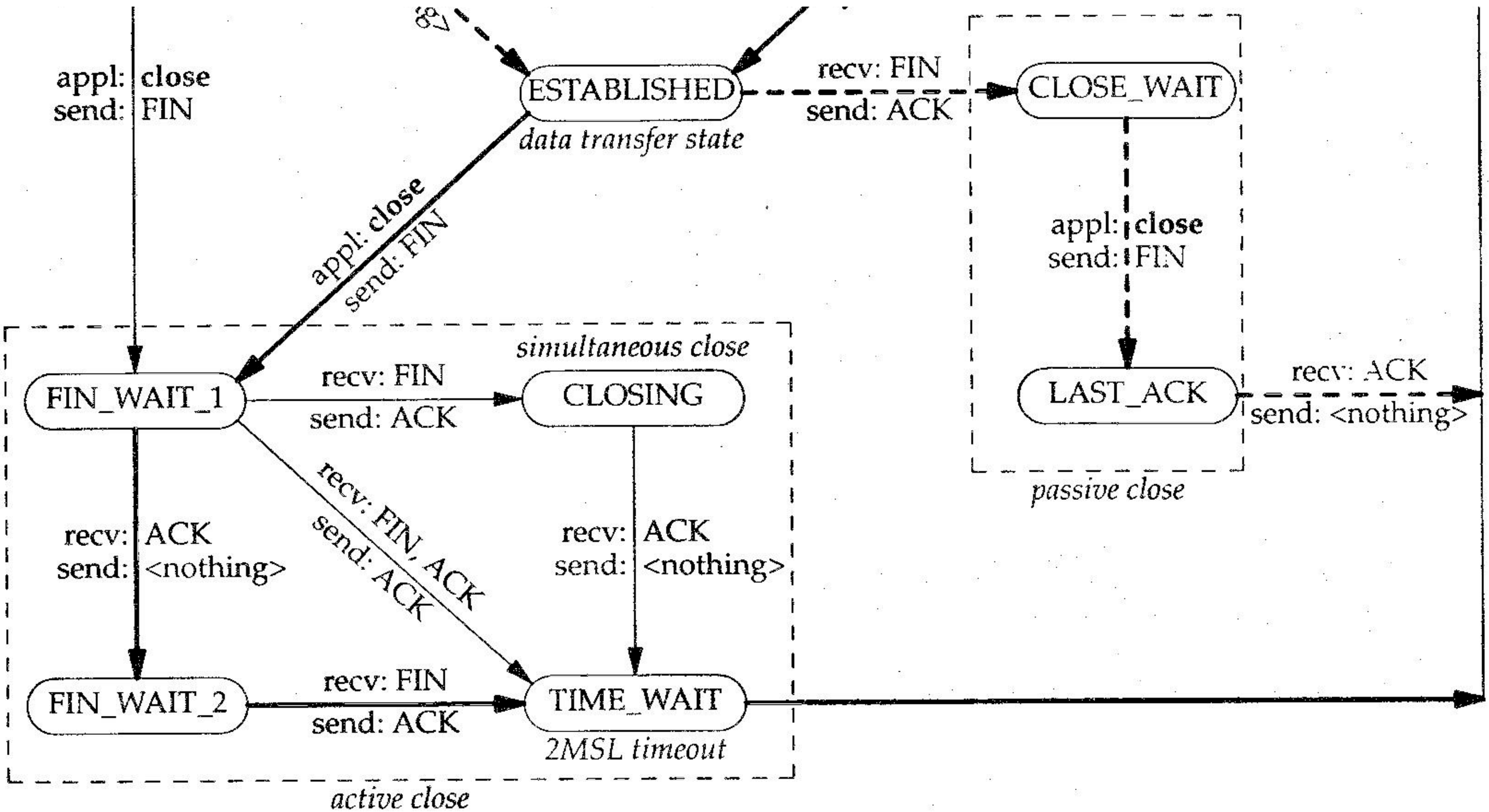
- Can be used to abort a TCP connection in case of any abnormal condition
- Initiated by one side, sends a TCP segment with the RST flag set in header
- The other side responds with a TCP segment with the RST flag set in header
- Immediately releases all resources and closes the connection in both directions

TCP State Diagram





starting point
CLOSED
appl: passive open
send: <nothing>
LISTEN
passive open
appl: active open
send: SYN
recv: SYN; send: SYN, ACK
recv: RST
appl: send data
send: SYN
SYN_RCVD
recv: SYN
send: SYN, ACK
simultaneous open
SYN_SENT
active open
appl: close
or timeout
recv: ACK
send: <nothing>
recv: SYN, ACK
send: ACK
ESTABLISHED
data transfer state



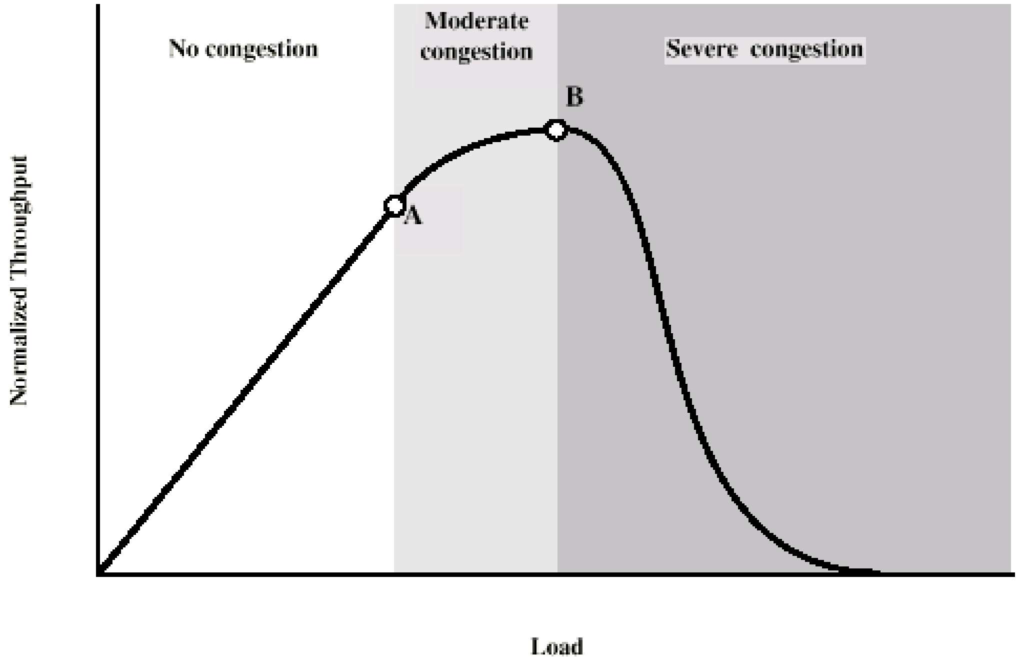
appl: close
send: FIN
ESTABLISHED
data transfer state
recv: FIN
send: ACK
CLOSE_WAIT
appl: close
send: FIN
LAST_ACK
recv: ACK
send: <nothing>
passive close
appl: close
send: FIN
simultaneous close
FIN_WAIT_1
recv: FIN
send: ACK
CLOSING
recv: FIN, ACK
send: ACK
recv: ACK
send: <nothing>
recv: ACK
send: <nothing>
FIN_WAIT_2
recv: FIN
send: ACK
TIME_WAIT
2MSL timeout
active close

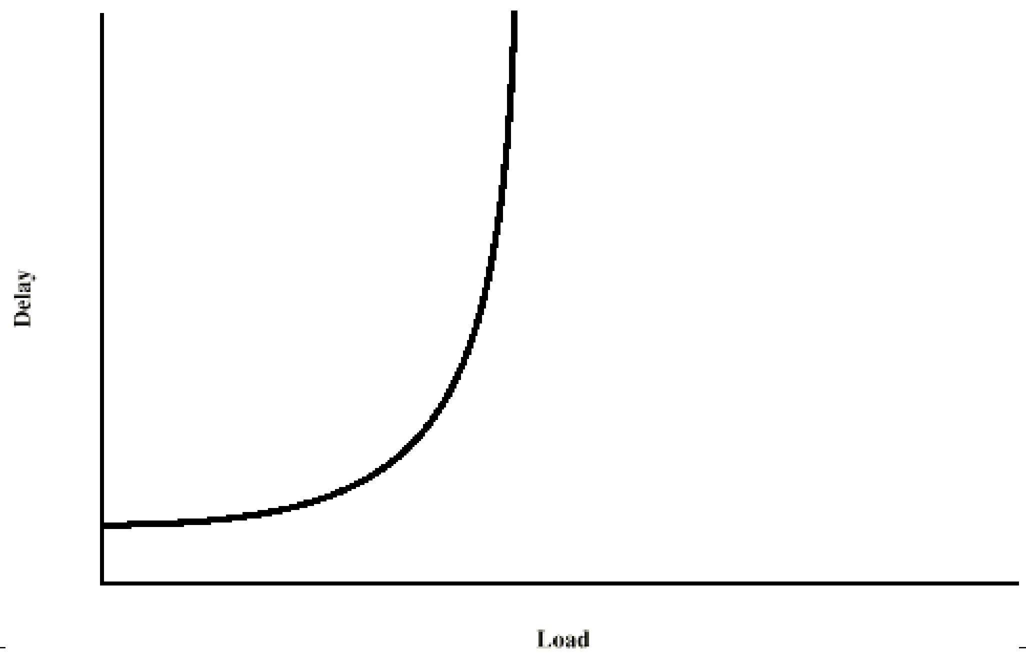
TCP Congestion Control

What is Congestion

- The number of packets transmitted on the network is greater than the capacity of the network
- More specifically
  - Packets come to some router at a rate higher than the rate at which the router can process packets and send them out
  - Causes packets to get queued at router buffers
  - Eventually the buffer fills up, causes packets to start getting dropped
- Why is it bad?
  - Retransmissions cause bandwidth wastage
  - Delay is increased
  - Congestion Collapse - retransmissions due to drop due to congestion can further increase congestion!!

Effect of Congestion





Congestion Control

- Congestion control aims to keep number of packets below level at which performance falls off dramatically
- End-to-end flow control is not enough!
  - Independent senders can each have flow control with their receivers, but together can still inject large number of packets in the network

How to detect congestion?

- Implicit congestion Signaling
  - Infers congestion indirectly from other events such as delay, drop
  - Transmission delay may increase with congestion
  - Packet may be discarded
  - Source can detect these as implicit indications of congestion and dynamically adjust packet sending rate
  - Basis for TCP congestion control

- Explicit congestion signaling
  - Congested router may send special packets back to the source
    - Ex. ICMP source quench packets
      - Source can cut back on sending rate until no more ICMP source quench packets received
  - ECN (Explicit Congestion Notification)
    - Router sets flag in header of IP packet from sender to receiver to indicate congestion
      - Part of TOS bits in IP header used
    - Receiver echoes back the flag in TCP header back to sender so that sender can adjust rate
      - Additional flags defined

TCP Congestion Control

- Sender estimates level of congestion from packet delay/drop
- Send more when no congestion detected
- Slow down when congestion detected
- Two issues to address:
  - Detecting congestion
  - Adjusting sending rate

Detecting Congestion

- Detected by detecting
  - Increasing round trip delay
    - TCP Vegas
    - We will not do
  - Packet drops
    - Most commonly used
    - TCP Tahoe, Reno, New Reno,…
- How to detect potential packet drops?
  - Timeout
  - Too many duplicate acknowledgements
    - Indicates packets are reaching (since ack's are sent) but some earlier packet has not reached yet
    - Probably lost because if out of order, only a few duplicate acks will be there

Adjusting Sending Rate

- Based on TCP Congestion Window (cwnd)
- Limits how much data can be in transit (similar to receiver window advertisement, but purpose is different)
- Max. window size at sender at any time =

  min (cwnd, recvAdvertisedWindow)
- cwnd is varied to control sending rate to address congestion
  - Varied by sender, congestion control is sender-side task
- Receiver advertised window varied to address end-to-end flow control
  - Varied by receiver, as discussed earlier

- Basic principle
  - On detecting packet drop, decrease cwnd
  - On receiving ack, increase cwnd
- Two phases of TCP congestion control
  - slow start
  - congestion avoidance
- Which phase to do?
  - Based on a variable ss_thresh (slow start threshold)
  - cwnd initially set to maximum segment size (MSS)
  - slow start : cwnd $\leq$ ss_thresh
  - congestion avoidance : cwnd $>$ ss_thresh

Before we go any further

Lets clear up some confusion that may arise
in interpreting cwnd

- cwnd is implemented as number of bytes (as it should be as TCP is byte-oriented)
- However, most descriptions talks about cwnd in terms of number of segments
- A segment in the context of cwnd means a full-size segment (size = maximum segment size, MSS), so easy to convert
- RFC 2581 talks in terms of both
- We will also talk about cwnd in term of no. of segments (easier to follow from text for students)

- Various implementations of TCP congestion control exist, differing mostly in how and when cwnd is computed
- We will do TCP Tahoe (the original TCP) and TCP Reno (the next extension, which has the basic things still used)

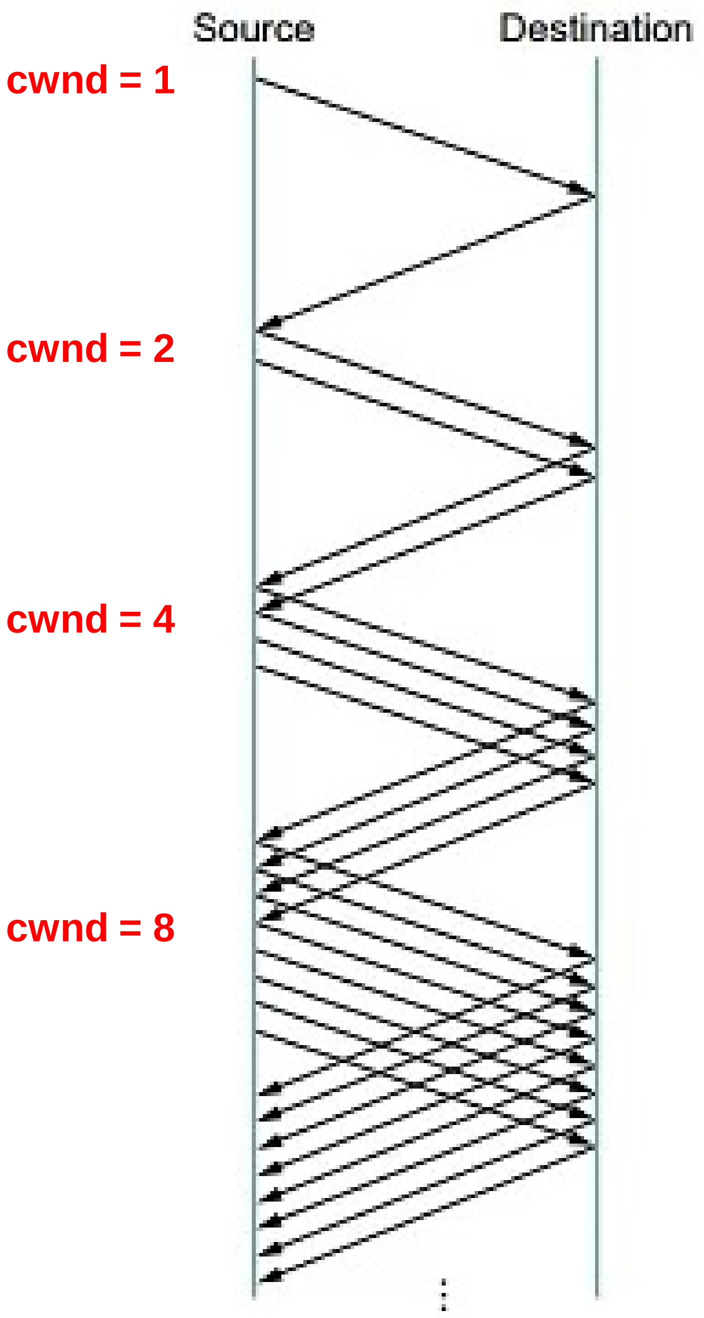
TCP Tahoe

- ARPANET switches to TCP in 1963
- Internet suffers congestion collapse in 1986
- Van Jacobson fixes TCP, TCP Tahoe introduced in 1987-88
  - Slow start, congestion avoidance, fast retransmit
- Extended with Fast Recovery in TCP Reno in 1990

Slow Start

- Used to find a good sending rate initially at startup, or while recovering after congestion
- Whenever starting traffic on a new connection, or whenever increasing traffic after congestion was detected:
  - Intialize *cwnd* = 1
  - Each time a segment is acknowledged, increment *cwnd* by 1
- Continue until
  - *ss_thresh* is reached, or
  - Timout, or
  - 3 duplicate ACKs received

- Slow start is not so slow!!



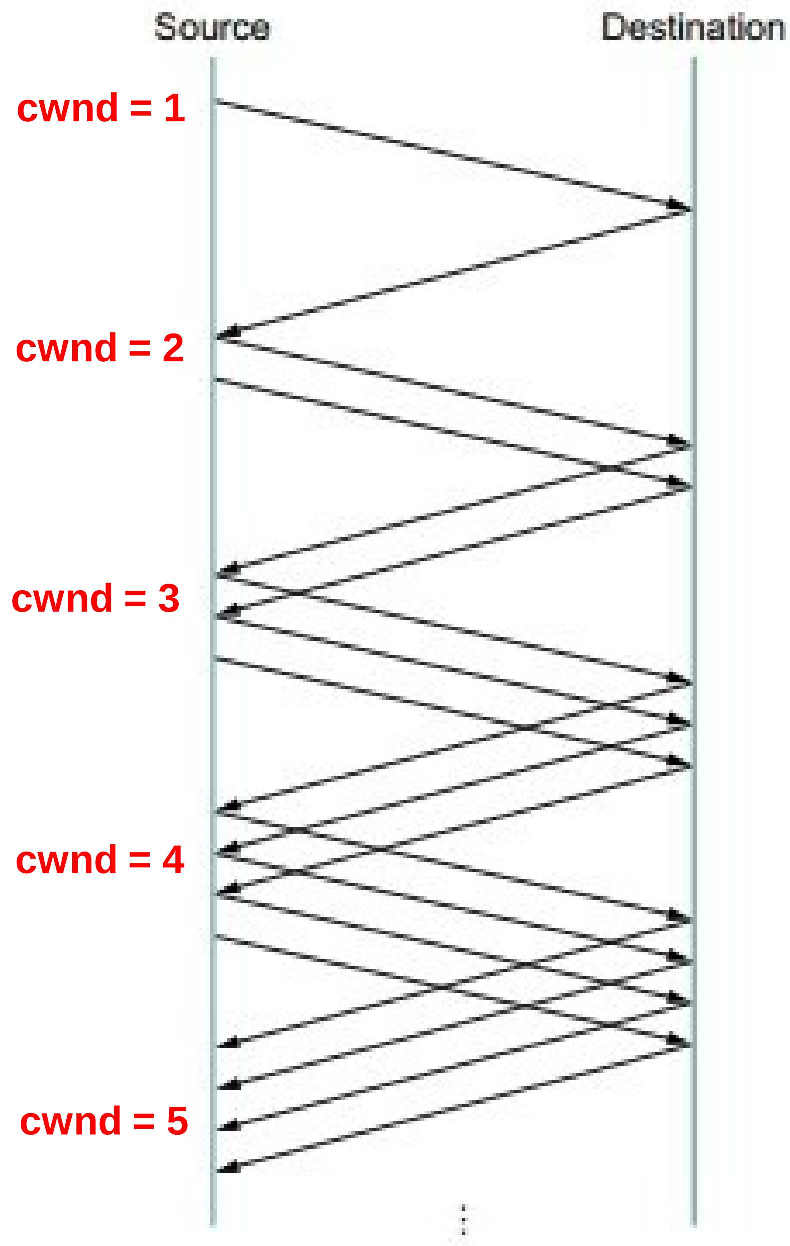
- The change in congestion window is per ack
  - So a cumulative ack will still increase the window by 1 only
- Given a *ss_thres* = 8, work out how many RTTs it will take for slow start to reach the threshold if there is one ack per 2 segments

Congestion Avoidance

- Slow start sets a “good” congestion window fast
- Congestion avoidance slows down the increase in cwnd
  - Why do we need to slow down even if there is no timeout?
- If *cwnd* > *ss_thresh* then

  each time a segment is acknowledged

  increment *cwnd* by 1/*cwnd* (*cwnd* += 1/*cwnd*)
- *cwnd* is increased by one only if all segments have been acknowledged
  - Increases by 1 per RTT, vs. doubling per RTT in slow start
  - Additive Increase


Source
Destination
cwnd = 1
cwnd = 2
cwnd = 3
cwnd = 4
cwnd = 5

On Detecting Congestion

- On a timeout
  - *ss_thresh* is set to half the current size of the congestion window:

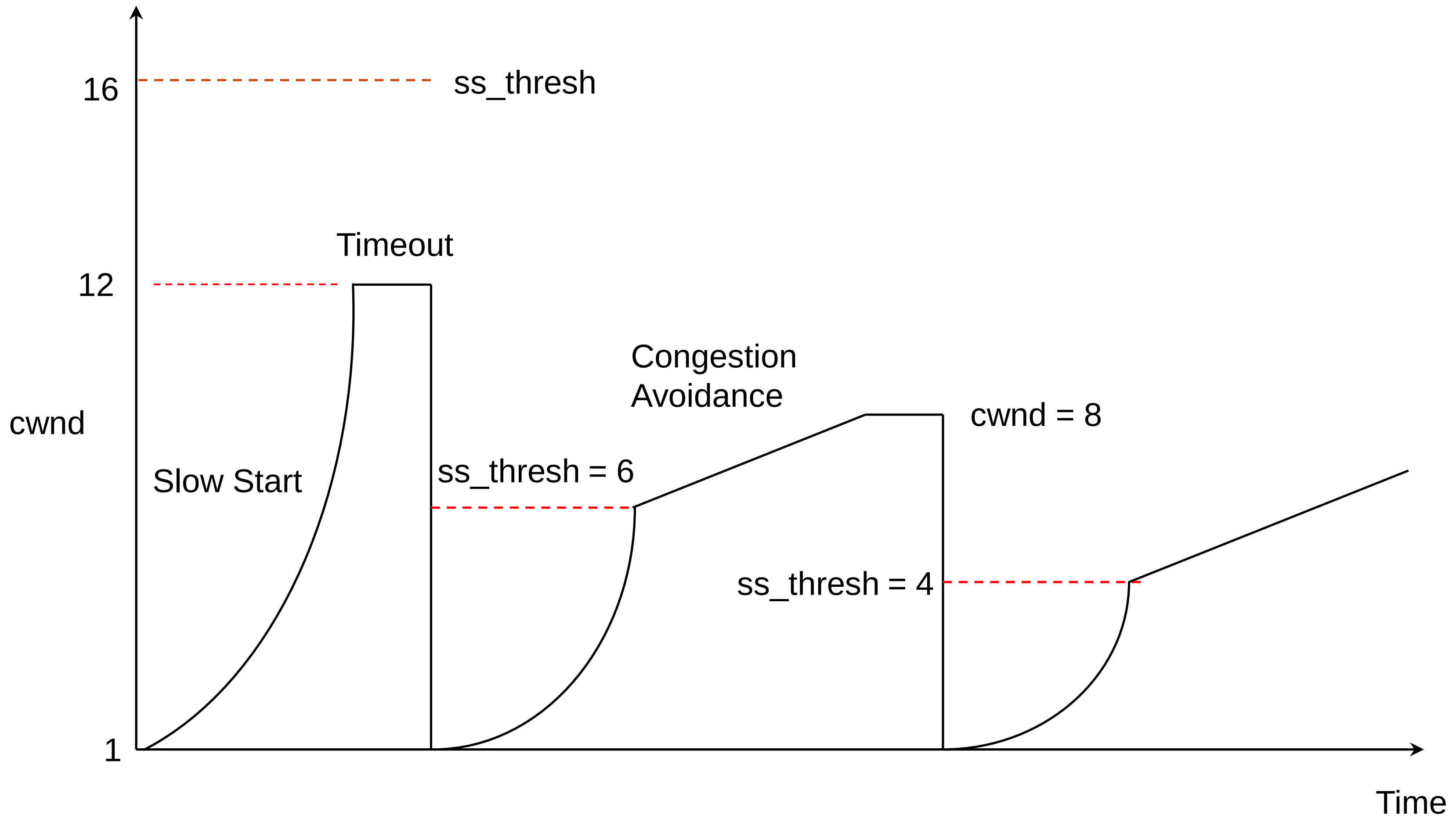
    $$ss\_thresh = cwnd \;/\; 2$$

  - *cwnd* is reset to one

    $$cwnd = 1$$

  - Slow-start phase is entered
  - This is called multiplicative decrease

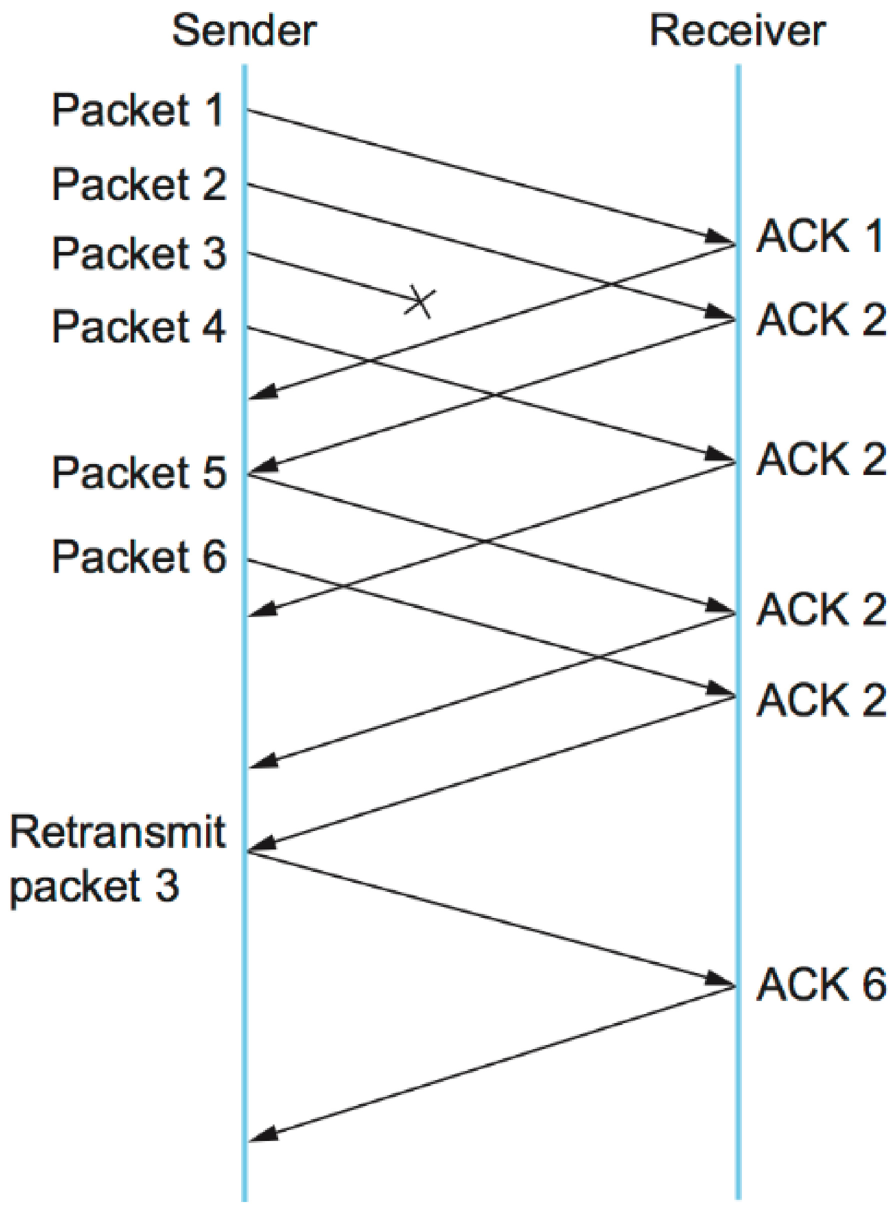
Example



16
ss_thresh
Timeout
12
Congestion
Avoidance
cwnd
cwnd = 8
Slow Start
ss_thresh = 6
ss_thresh = 4
1
Time

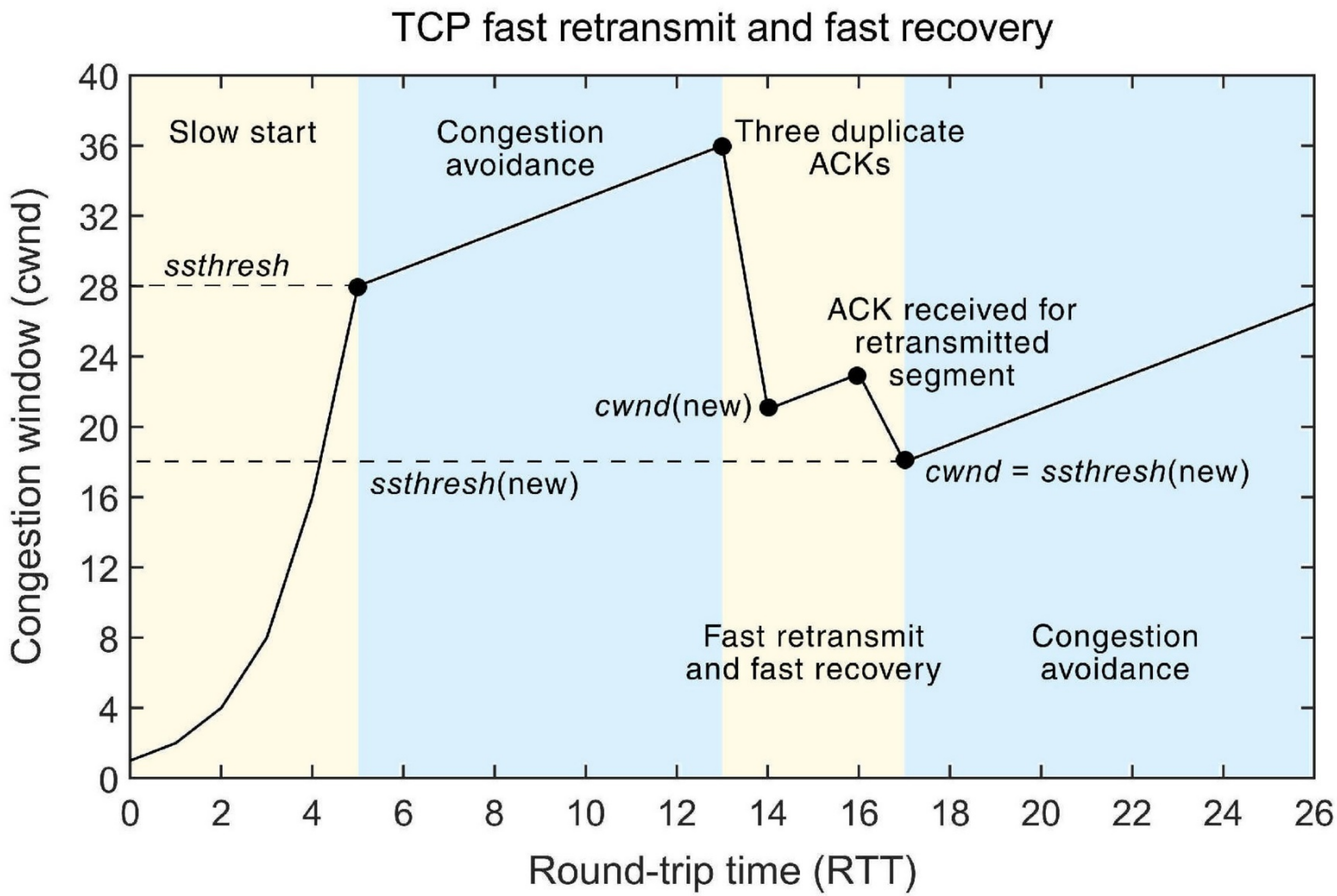
Fast Retransmit

- On receiving 3 duplicate ACKs (total 4) for the same segment,
  - Retransmit the segment without waiting for timeout
  - Set $ss\_thresh = cwnd/2$
  - Set $cwnd = cwnd/2 + 3$
    - Why the +3?


Sender
Receiver
Packet 1
Packet 2
Packet 3
Packet 4
Packet 5
Packet 6
Retransmit
packet 3
ACK 1
ACK 2
ACK 2
ACK 2
ACK 2
ACK 6

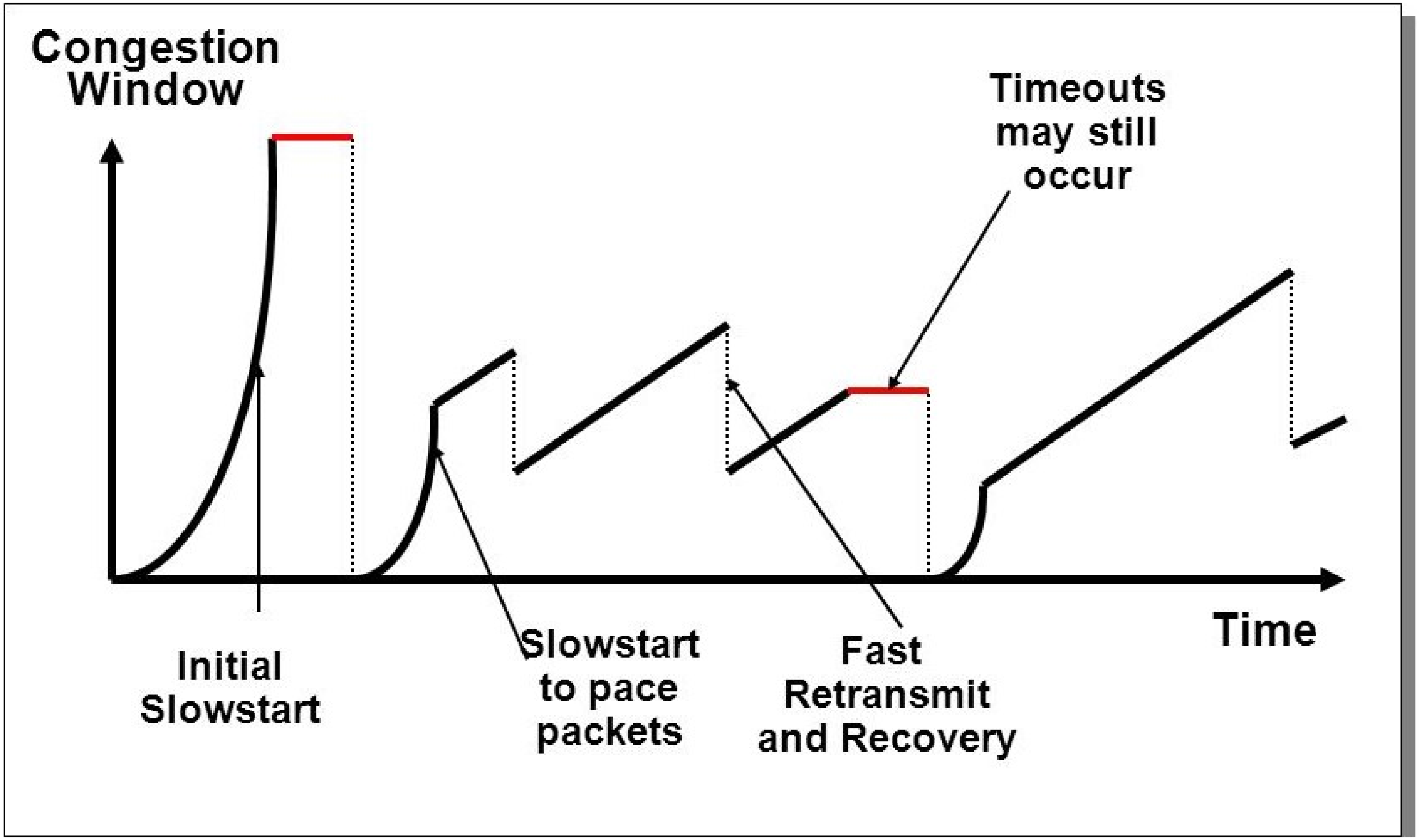
TCP Reno

- Adds Fast Recovery
  - On 3 duplicate ACKs, go back to congestion avoidance
- Fast Retransmit and Fast Recovery are implemented together
- On receiving 3 duplicate ACKs for the same segment,
  - Retransmit the segment without waiting for timeout
  - Set $ss\_thresh = cwnd\ /\ 2$
  - Set $cwnd = cwnd/2 + 3$
- On subsequent duplicate ACK, $cwnd = cwnd + 1$
- On new ACK, $cwnd = ss\_thres$



TCP fast retransmit and fast recovery
Slow start
Congestion avoidance
Three duplicate ACKs
ssthresh
ACK received for retransmitted segment
cwnd(new)
ssthresh(new)
cwnd = ssthresh(new)
Fast retransmit and fast recovery
Congestion avoidance
Congestion window (cwnd)
Round-trip time (RTT)
0
4
8
12
16
20
24
28
32
36
40
0
2
4
6
8
10
12
14
16
18
20
22
24
26

TCP Saw Tooth Behavior



TCP Variants

- Many implementations of TCP has been done since Reno, some to address some problems, some suited for specific scenarios (ex. links with high bandwidth delay product (Long, fat links) etc.)
  - TCP New Reno (1999)
  - TCP Vegas (1995)
  - TCP SACK (1996)
  - TCP Westwood (2001)
  - Fast TCP (2006)
  - HSTCP (High Speed TCP) (2003)
  - TCP Bic (2004)
  - Cubic TCP
    - Default implementation for Linux

Transport Layer Protocols:
UDP (User Datagram Protocol)

User Datagram Protocol (UDP)

- Transport layer protocol like TCP, notion of port to identify application layer service
- Provides multiplexing/demultiplexing of applications
- Connectionless (no connection setup/termination)
  - So demultiplexing done on the basis of one endpoint (<IP, port> pair)
- Each block of message given by user is sent independently and separately
  - An UDP datagram
- Datagrams can be lost, or arrive in different order from the order sent

- No flow control
  - So no acknowledgement, window maintenance etc.
- No error control
  - So no acknowledgement, timeout, retransmission etc.
  - Unreliable service
- No congestion detection or control
- So provides only multiplexing/demultiplexing of applications
- But simple and fast, as there is no complex operations like flow control, error control etc.
  - So no need of any connection state maintenance also

UDP Message Format

0 16 32

Source Port	Destination Port
Length	Checksum
Data	

- Header Fields
  - Source port, destination port: Identifies the UDP applications at the two ends
  - Length: Size of the datagram in bytes, including header
  - Checksum: Checksum of Psuedoheader + UDP datagram
    - Psuedoheader computed the same way as for TCP

- Simple, fast protocol used by many applications when reliability is not a big issue
  - DNS (53) – Domain Name System
  - tftp (69) – Trivial File Transfer Protocol
  - ntp (123) – Network time Protocol
  - snmp (161) – Simple Network Management Protocol
  - RIP (520) – Routing Information Protocol
  - DHCP (546,547) – Dynamic Host Configuration Protocol
  - Many other well known applications