Transports and TCP

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CPS 214

Host-to-Host vs. Process-to-Process Communication

- Until now, we have focused on delivering packets between arbitrary *hosts* connected to Internet
  - Routing protocols
  - IP best effort delivery model
  - Scalability and robustness through hierarchy and soft state
- Transition to arbitrary *processes* communicating together
  - One goal: provide illusion that all processes located on one large computer
  - Can address (*name)* and reliably communicate with any process

    Ports

UDP

- User Datagram Protocol (UDP)
  - Simple demultiplexing

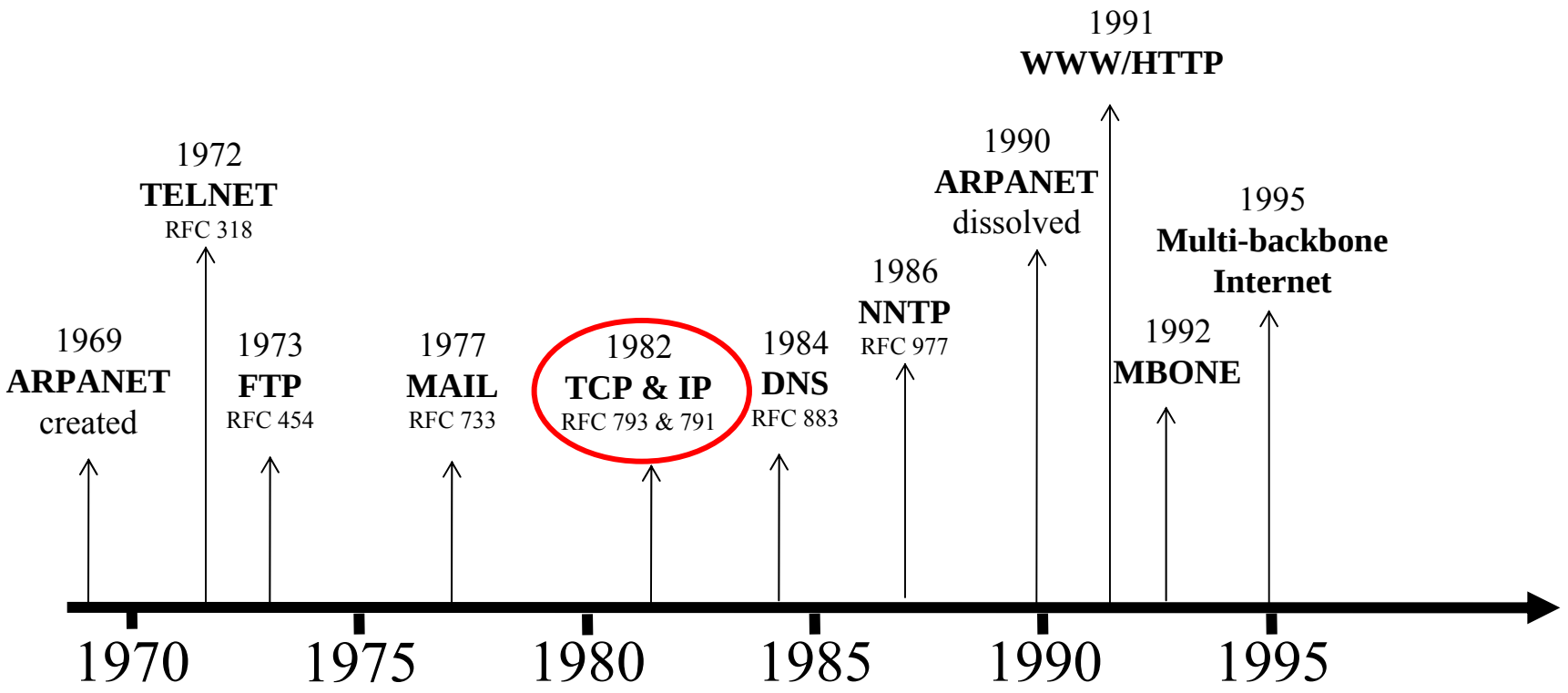
    No guarantees about reliability, in-order delivery
  - Thin veneer on top of IP adds src/dest port numbers

    16 bit port number allows for identification of 65536 unique communication endpoints per host

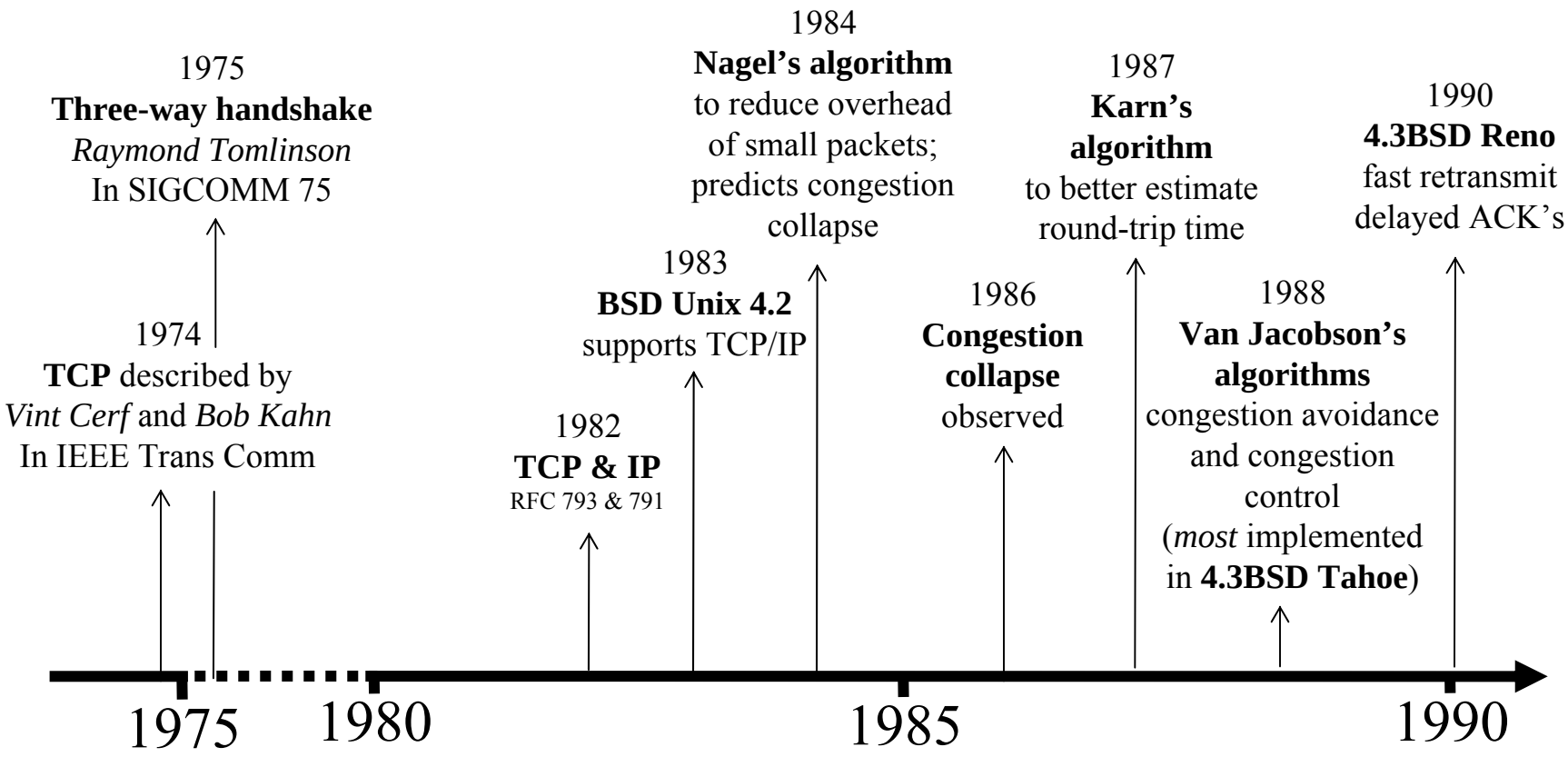
    Note that a single process can utilize multiple ports

    IP addr + port number uniquely identifies all Internet endpoints
  - UDP Packet

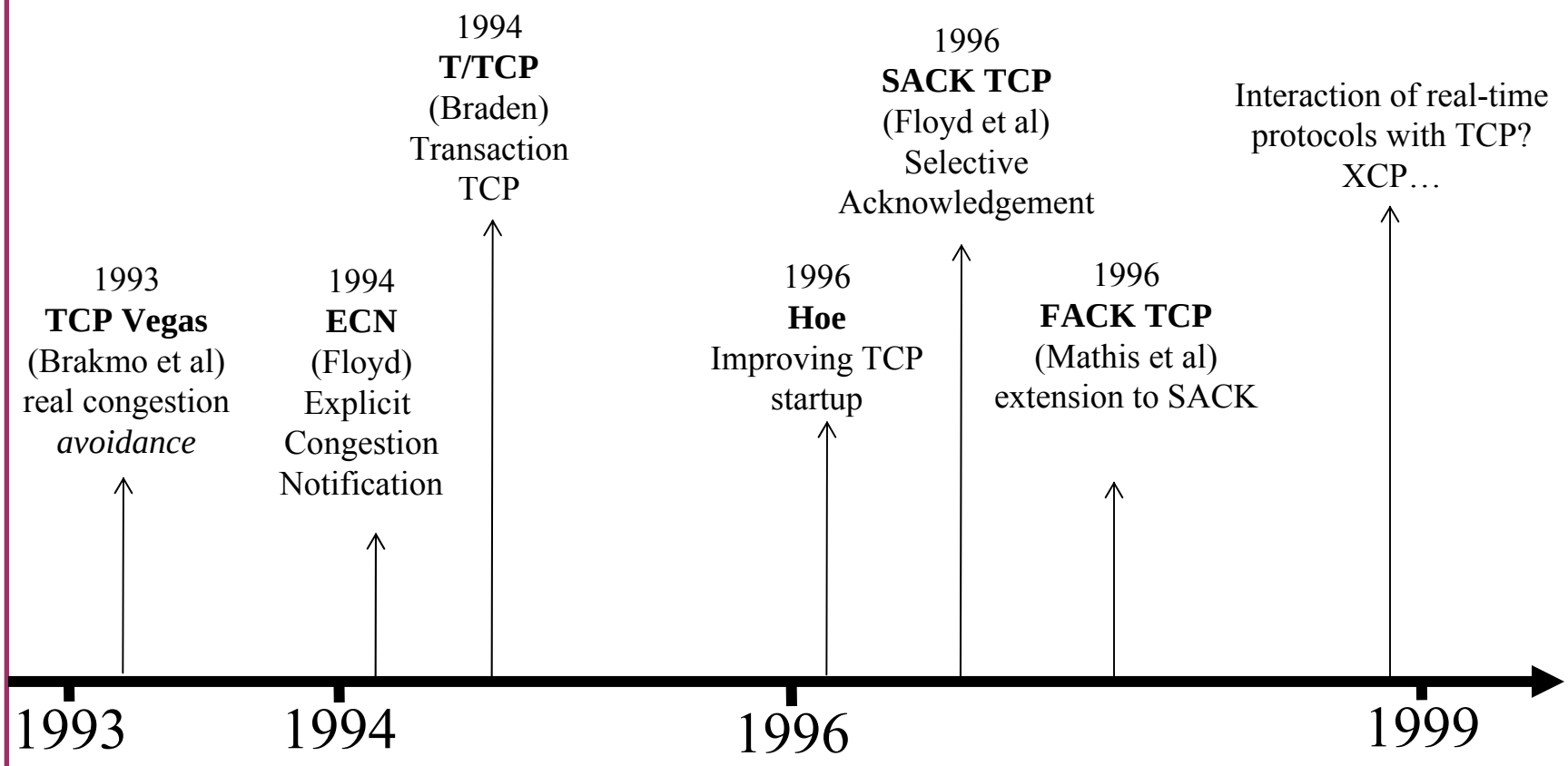
A Brief Internet History



TCP Timeline



TCP: After 1990



TCP

- Transmission Control Protocol (TCP)
  - Reliable in-order delivery of byte stream
  - Full duplex (endpoints simultaneously send/receive)

    e.g., single socket for web browser talking to web server
  - Flow-control

    To ensure that sender does not overrun *receiver*

    Fast server talking to slow client
  - Congestion control

    Keep the sender from overrunning the *network*

    Many simultaneous connections across routers (cross traffic)

TCP

- Utilize sliding window protocol, plus:
  - Need for connection establishment (no dedicated cable)
  - Varying round trip times over life of connection

    Different paths, different levels of congestion
  - Ready for *very* old packets
  - Delay-bandwidth product highly variable

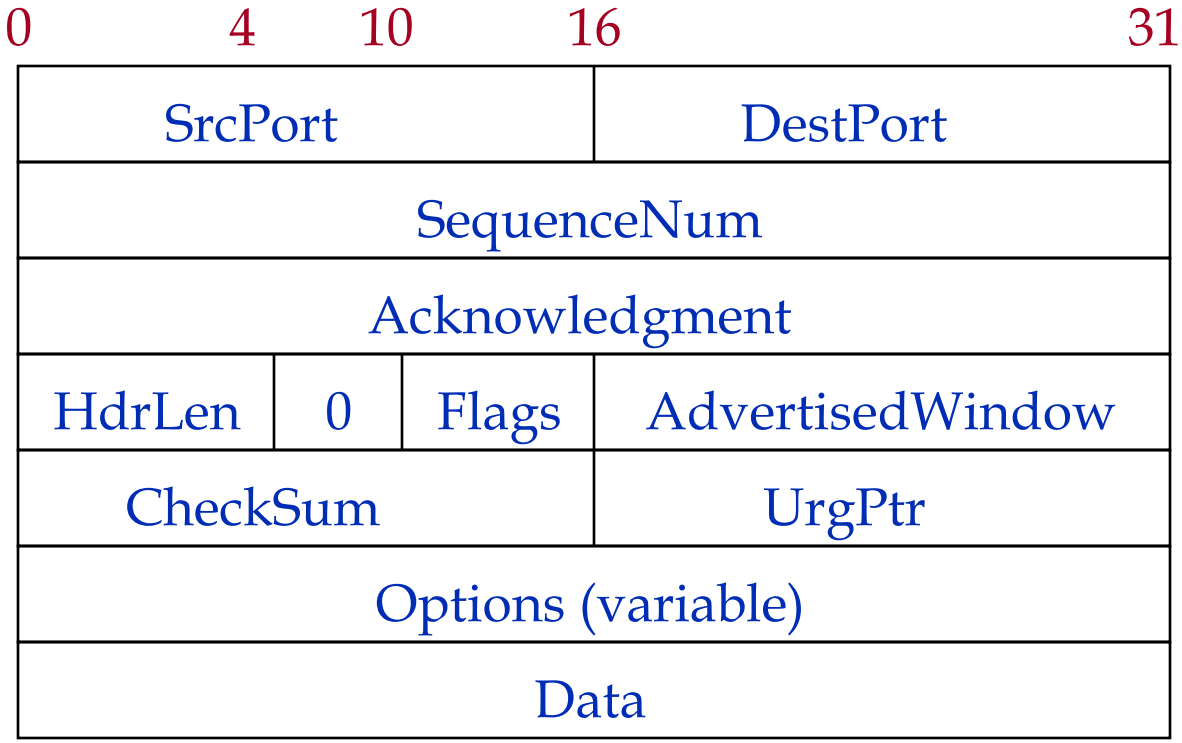
    Amount of available buffer space at receivers also variable
  - Sender has no idea what links will be traversed to receiver

    Must dynamically estimate changing end-to-end characteristics

TCP Flavors

- TCP Tahoe
  - Jacobson's implementation of congestion control (AIMD)
- TCP Reno
  - Fast recovery
  - Fast retransmit
  - Delayed ACK's
- TCP Vegas
  - Source-based congestion *avoidance* rather than control
  - TCP Reno needs to cause congestion to determine available bandwidth

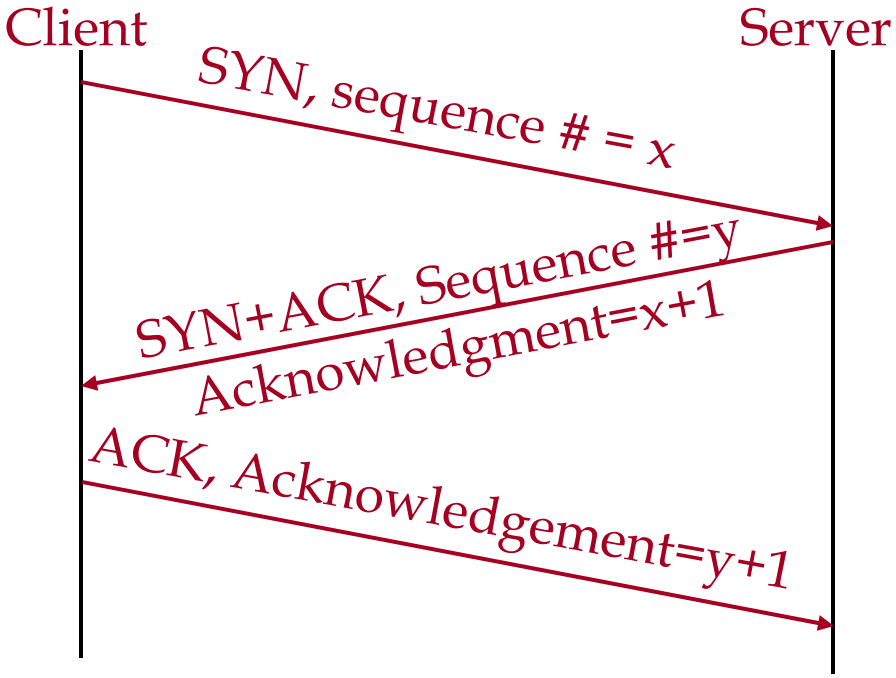
TCP Header Format



- Without options, TCP header 20 bytes
  - Thus, typical Internet packet minimum of 40 bytes

TCP Connection Establishment

- Exchange necessary information to begin communication
- Three-way handshake
  - E.g., server listening on socket



TCP Connection Teardown

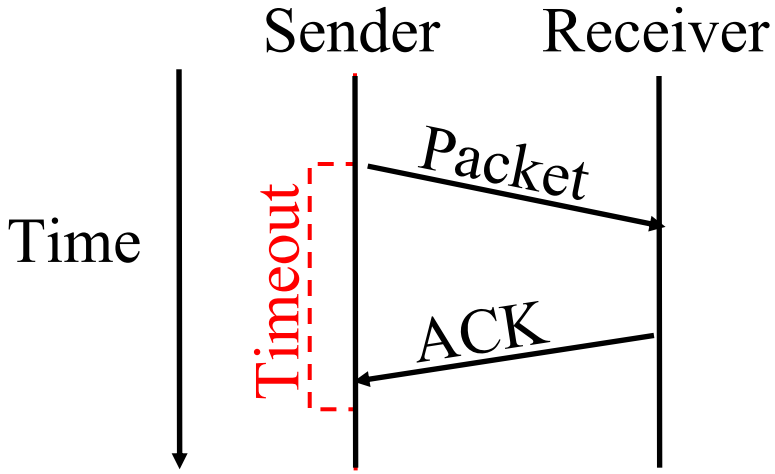
- Closing process sends a FIN message
  - Waits for ACK of FIN to come back
  - This side of the connection is now closed
- Each side of a TCP connection can independently close the connection
  - Thus, possible to have a half duplex connection

Reliable Transmission

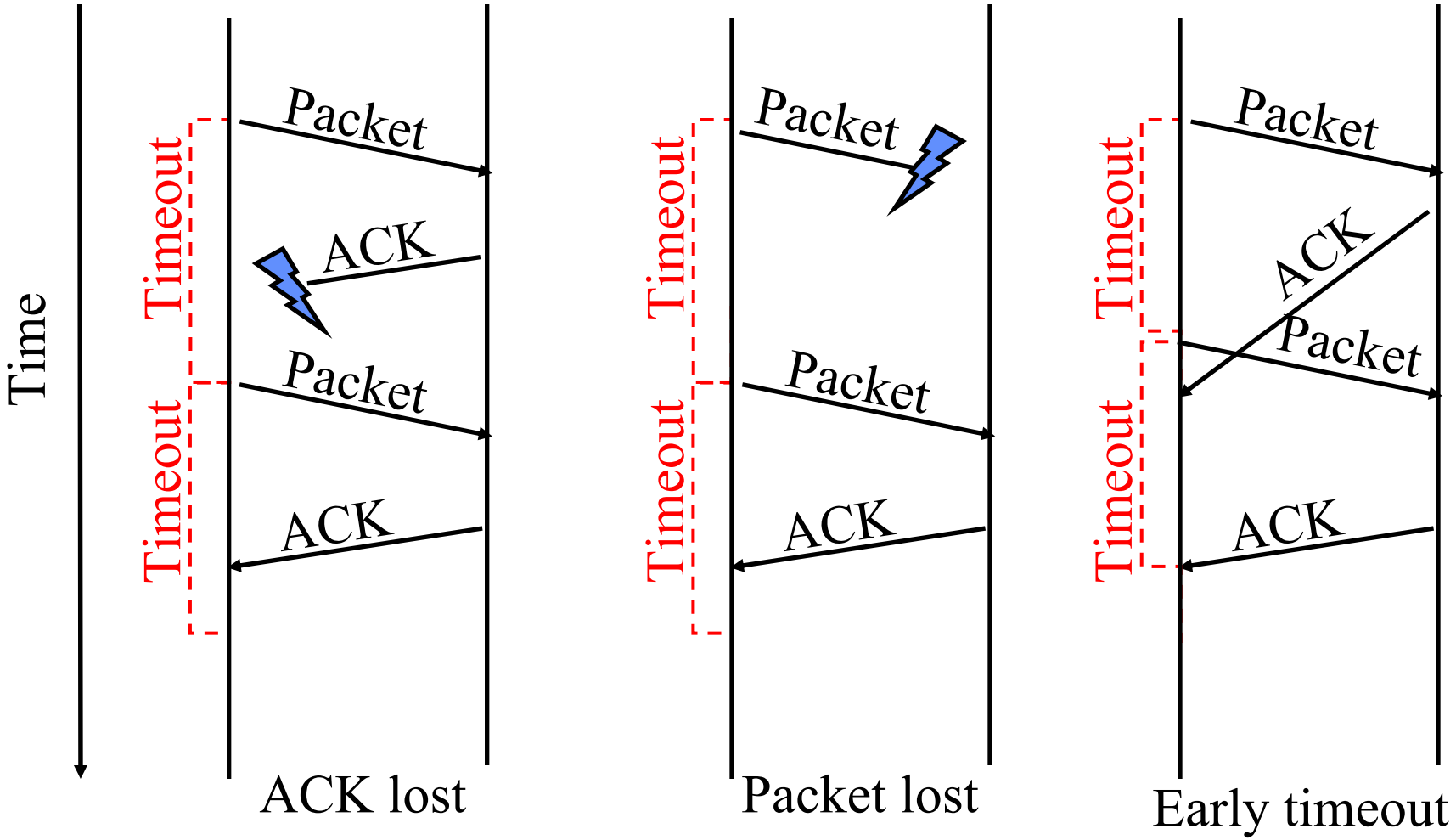
- How do we send a packet reliably when it can be lost?
- Two mechanisms
  - Acknowledgements
  - Timeouts
- Simplest reliable protocol: Stop and Wait

Stop and Wait

Send a packet, stop and wait until acknowledgement arrives



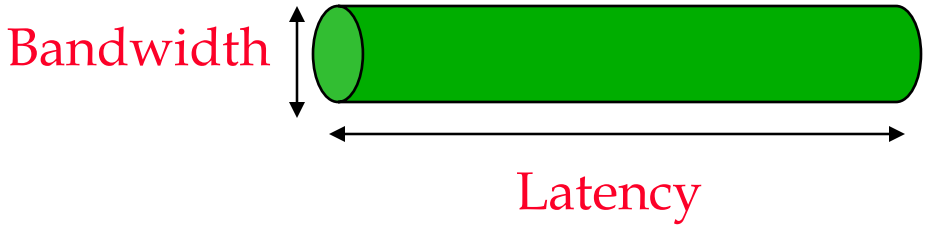
Recovering From Error



Problems with Stop and Wait

- How to recognize a duplicate transmission?
  - Solution: put sequence number in packet
- Performance
  - Unless Latency-Bandwidth product is very small, sender cannot fill the pipe
  - Solution: sliding window protocols

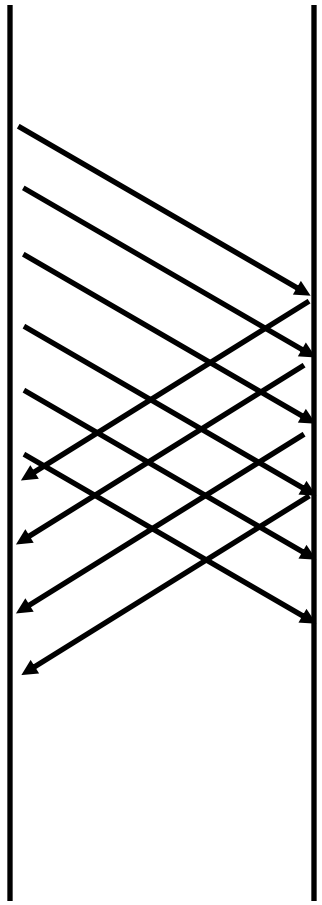
Keeping the Pipe Full



- Bandwidth-Delay product measures network capacity
- How much data can you put into the network before the first byte reaches receiver
- Stop and Wait: 1 data packet per RTT
  - Ex. 1.5-Mbps link with 45-ms RTT
  - Stop-and-wait: 182 Kbps
- Ideally, send enough packets to fill the pipe before requiring first ACK

How Do We Keep the Pipe Full?

- Send multiple packets without waiting for first to be ACK'd
- Reliable, unordered delivery:
  - Send new packet after each ACK
  - Sender keeps list of unack'd packets; resends after timeout
- Ideally, first ACK arrives immediately after pipe is filled
  - Opens up another "slot"



TCP Flow Control

- TCP is a sliding window protocol
  - For window size *n*, can send up to *n* bytes without receiving an acknowledgement
  - When the data is acknowledged then the window slides forward
- Each packet advertises a window size in TCP header
  - Indicates number of bytes the receiver is willing to get
- Original TCP always sent entire window immediately
  - Too bursty?

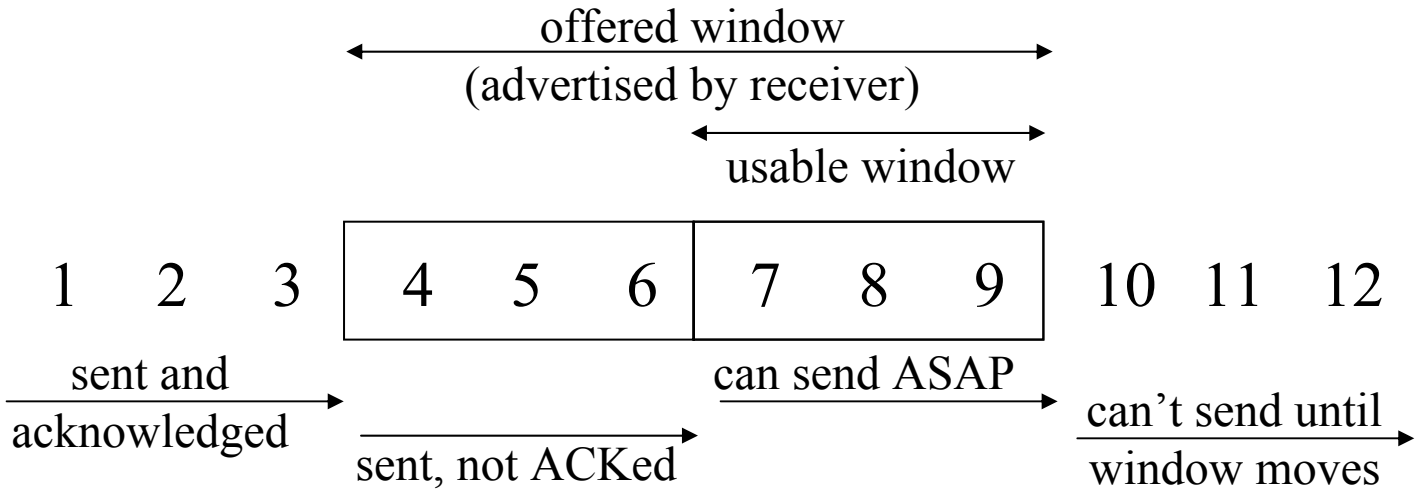
Sliding Window

- Receivers buffer later packets until prior packets arrive
  - For out of order delivery
- Sender must prevent buffer overflow at receiver
  - Flow control
- Solution: sliding window
  - Circular buffer at sender and receiver

    Packets in transit <= buffer size

    Advance when sender and receiver agree packets at beginning have been received

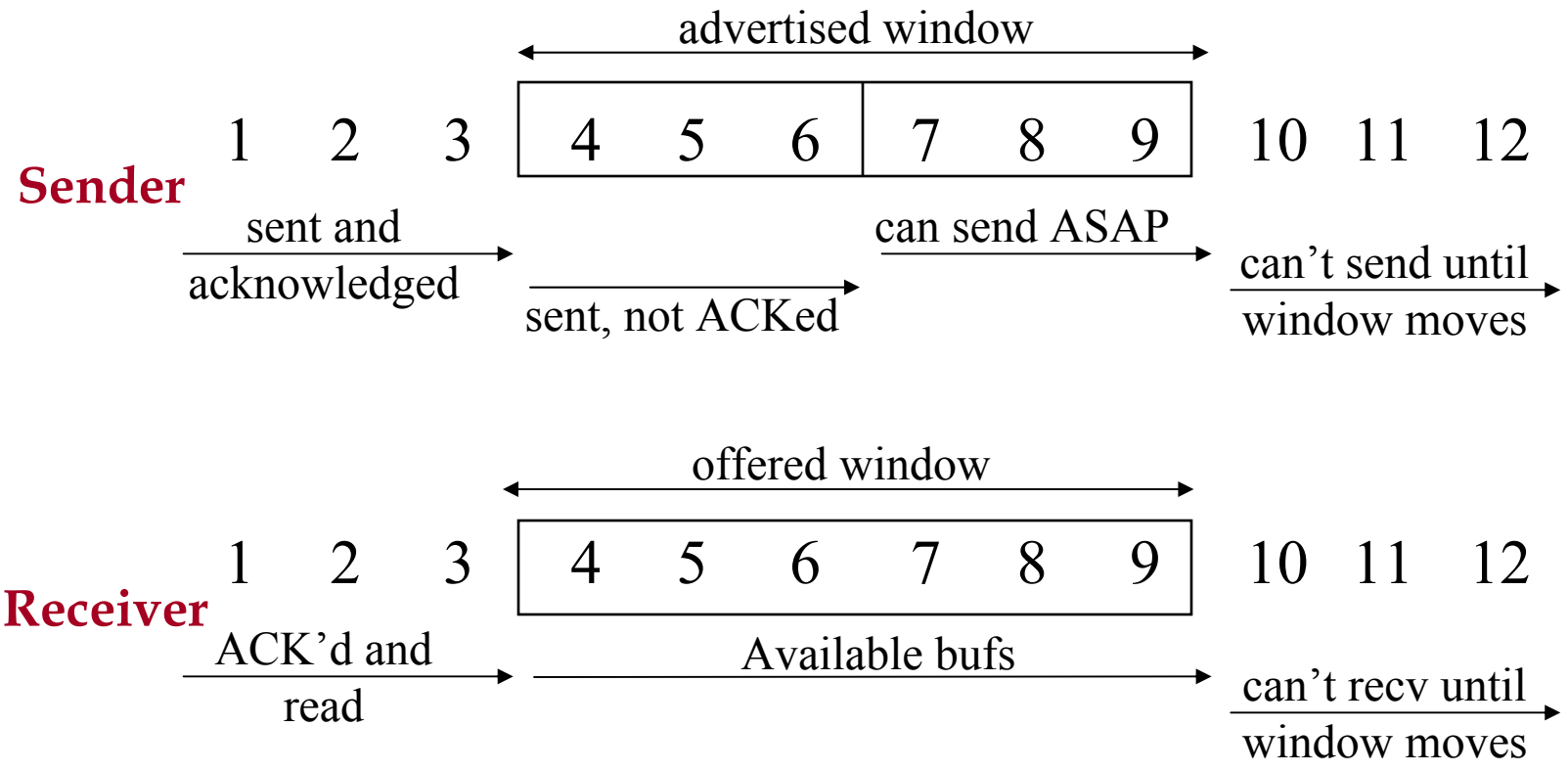
Visualizing the Window



Left side of window advances when data is acknowledged.
Right side controlled by size of window advertisement.

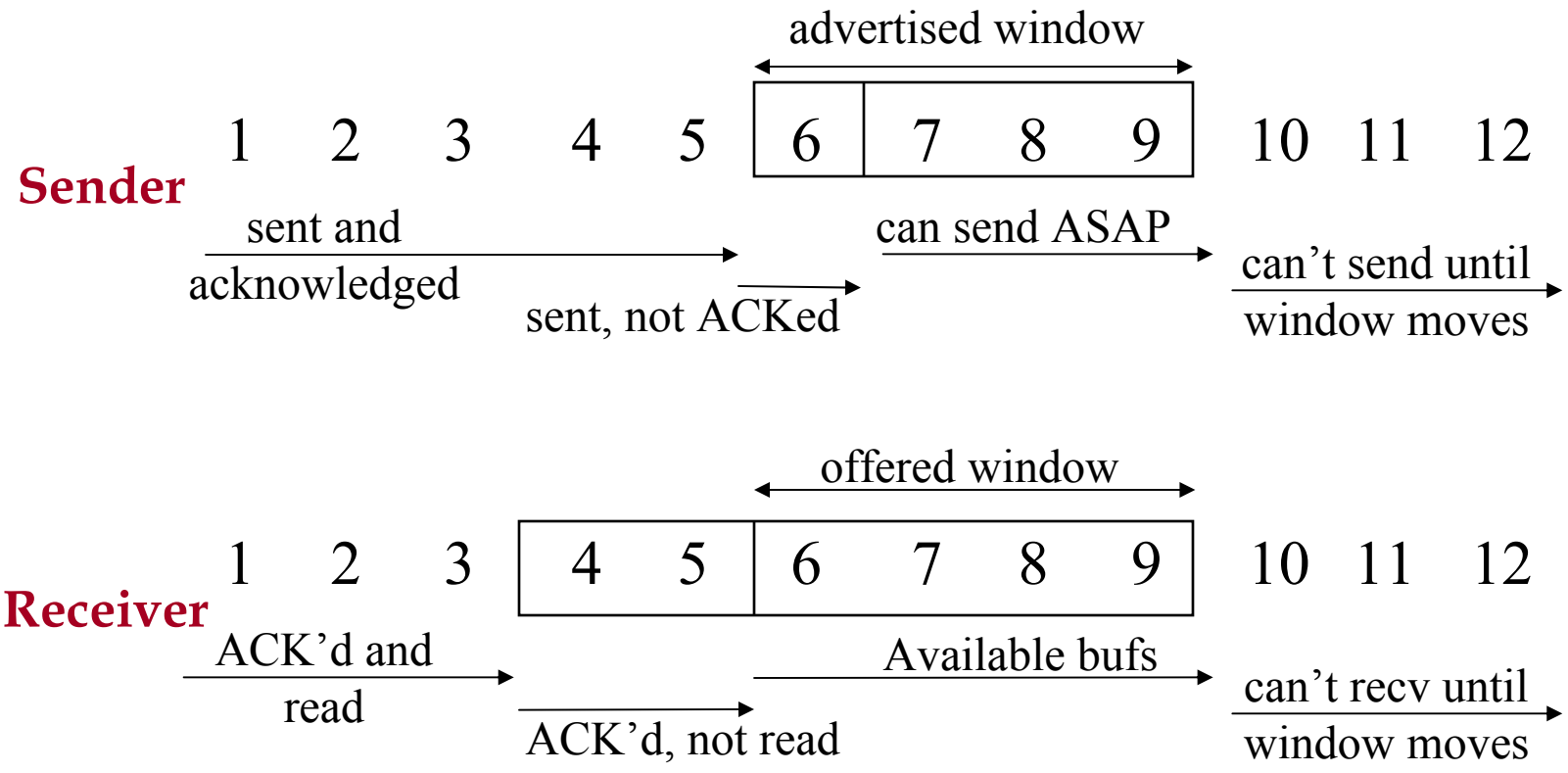
Visualizing the Window: Example

Initial State, Receiver has 6 slots to buffer packets
Packets 4, 5, 6 sent, but not yet received



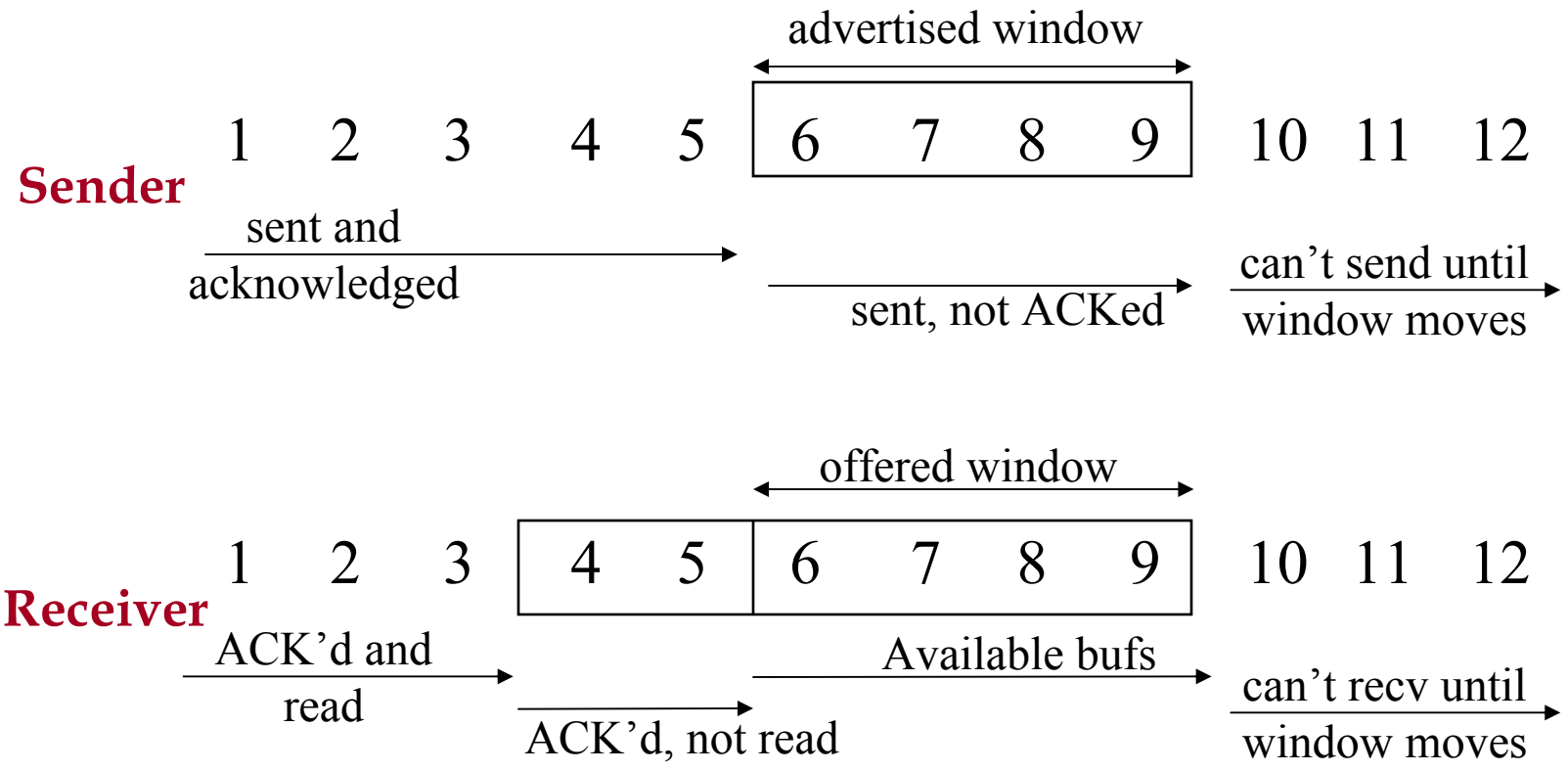
Visualizing the Window: Example

Receiver to Sender➔ ACK 5, Window 4



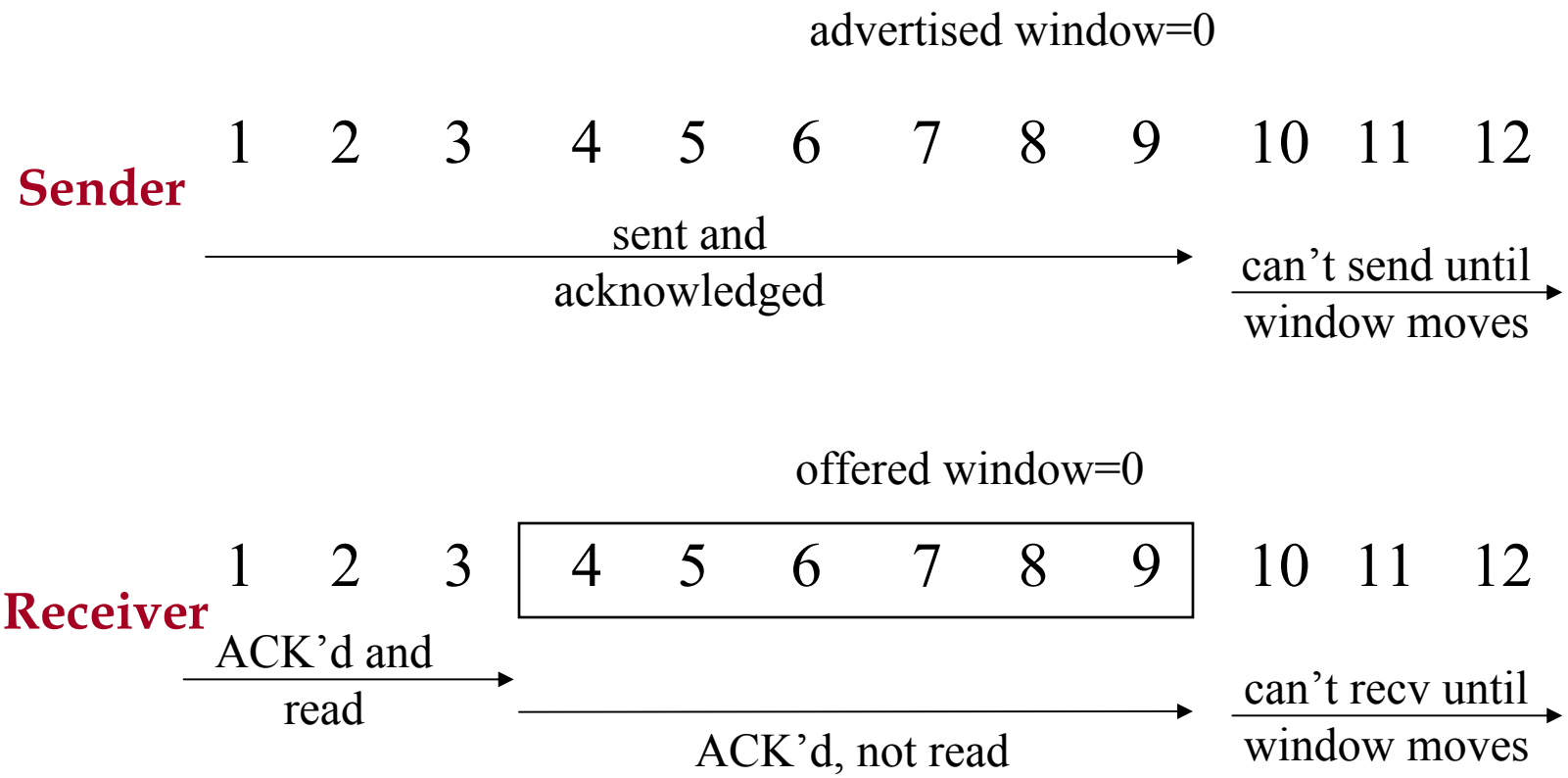
Visualizing the Window: Example

Sender to Receiver➔ Send 7, 8, 9



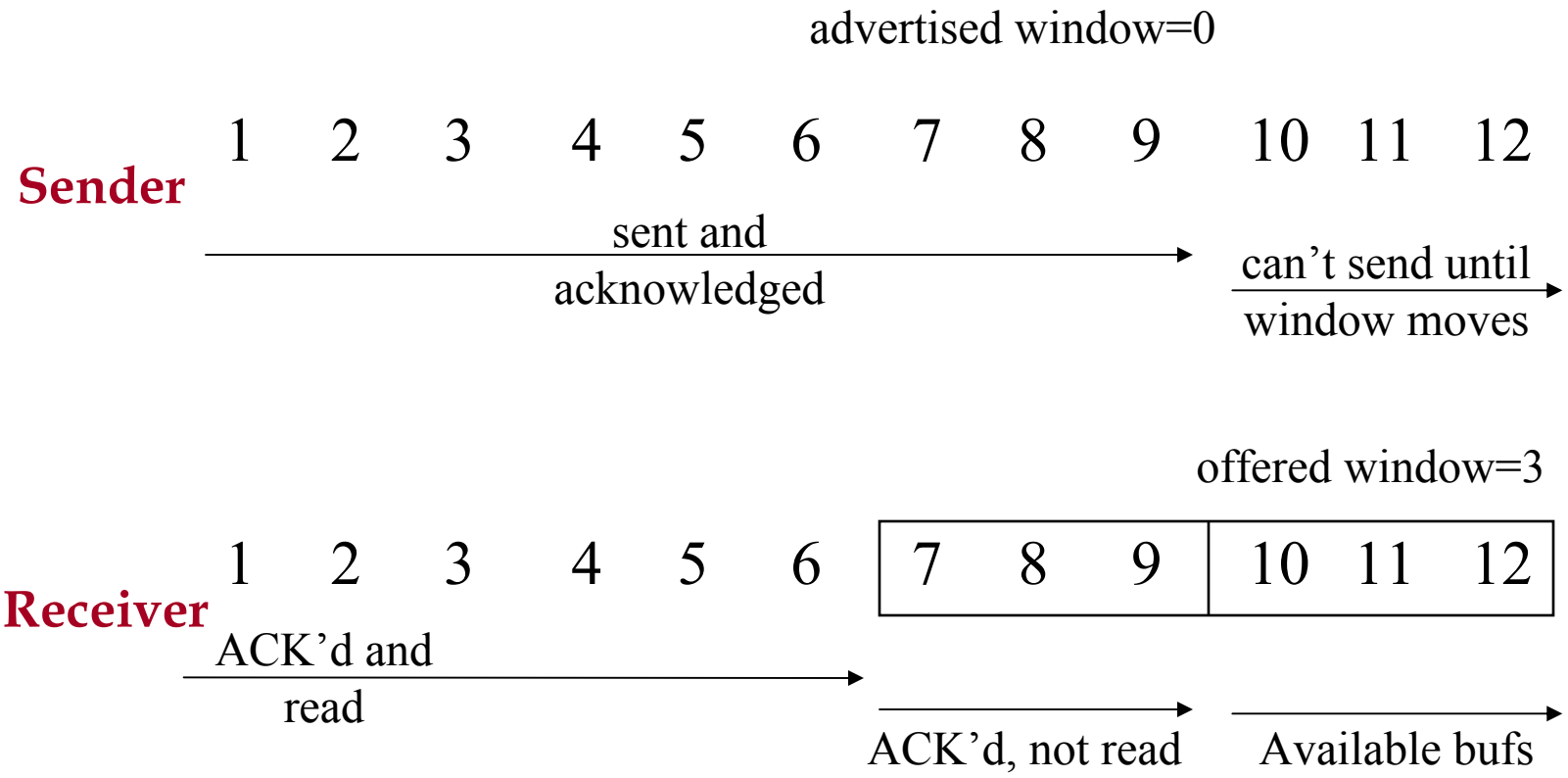
Visualizing the Window: Example

Receiver to Sender➔ ACK 9, Window 0



Visualizing the Window: Example

Receiver App reads packets 4, 5, 6
But sender has no way of knowing that more room is available!



Options for Sender Discovery of Increased Advertised Window

- Receiver sends duplicate ACK with a larger advertised window
  - Complicates receiver design
  - TCP design philosophy: keep receiver simple

    Also explains slow deployment of SACK, NACK, etc.
- Sender periodically transmits a 1-byte packet
  - If no space available at receiver➔packet dropped, no ACK
  - If additional space became available➔ACK contains new advertised window
- NOTE: advertised window in bytes, not packets

Sequence Numbers

- TCP uses 32-bit sequence number
  - TCP assumes that packet will not live in Internet for > 1 min
  - On 622 Mbps link, can wrap 32-bit sequence number in 55 seconds

    Gbps links becoming common
  - Why is this a problem?

Sequence Numbers

- TCP uses 32-bit sequence number
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    Gbps links becoming common
  - Proposal: extend sequence number with timestamp to distinguish between old and new incarnations of packets

Advertised Window

- TCP uses a 16-bit advertised window field (flow control)
  - Specifies number of bytes that can be sent from sender to receiver
  - Recall "keeping pipe full" to obtain available bandwidth
  - 16-bit field translates to max 64KB advertised window
  - For 100 ms RTT T3 link (45 Mbps), delay-bandwidth product is 549 KB

    Advertised window not large enough to keep pipe full

    Poor bandwidth utilization

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  - Proposal: advertised window specifies chunks larger than byte granularity

Adaptive Retransmission for Reliable Delivery

- TCP retransmits packet if ACK not received within timeout period
  - Necessary for reliability on top of "best-effort" IP
- Round trip time varies with congestion, route changes, …
  - If timeout too small, useless retransmits
  - If timeout too big, low utilization
- TCP: estimate RTT by timing ACKs
  - Exponential weighted moving average
  - Factor in RTT variability

Retransmission

- How long a timeout to set?
- Original TCP: Estimate round-trip time R
  - $R = \alpha R + (1- \alpha)M$
  - α is a smoothing factor of 0.9

    Places much more weight on historical result

    Smooth out outlying measurements
  - M is measured round-trip time (ACK's to data)
- Timeout at $R\beta$, where β is a delay variance factor of 2.0
  - Conservative: do not retransmit until two RTT's have passed

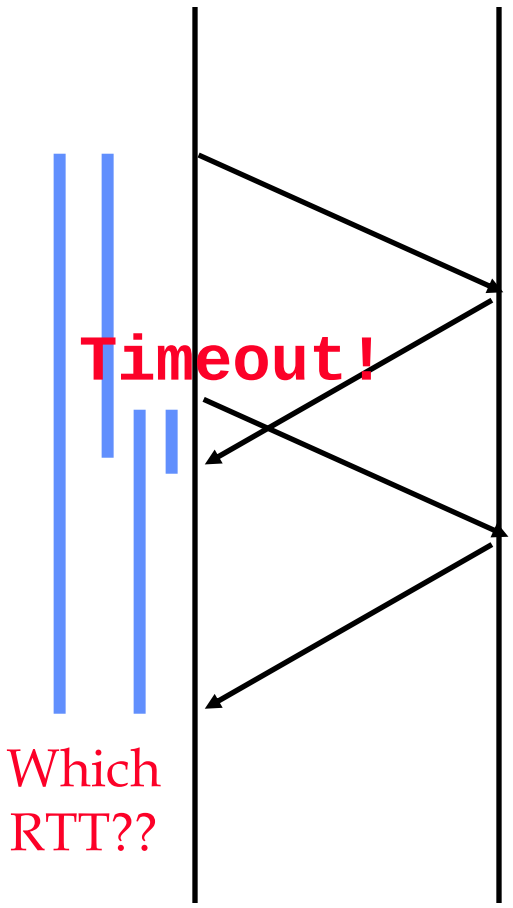
    Jacobson's TCP modifications allows for varying β

Retransmission Ambiguity

- How do we distinguish first ACK from retransmitted ACK?
  - First send to first ACK

    What if ACK dropped?
  - Last send to last ACK

    What if last ACK dropped?



Retransmission Ambiguity: Solutions

- TCP: Karn-Partridge
  - Ignore RTT estimates for retransmitted packets
  - Double timeout on every retransmission

    Exponential backoff similar to Ethernet for congestion avoidance
- Add sequence #'s to retransmissions (retry #1, retry #2)
- TCP proposal: Add timestamp into packet header; ACK returns timestamp

Jacobson's RTT estimator

- Problem:
  - Originial TCP does not adapt to wide variance in RTT
  - Uses fixed β of 2.0
- Need to account for both estimate RTT *and* variance
- Jacobson:
  - Low variance➔estimate RTT sufficient
  - High variance➔ estimate RTT could be far off
- Solution:
  - Timeout = μ*EstimateRTT + φ*Deviation
  - μ=1, φ=4

Can We Shortcut Timeout?

- If packets usually arrive in order, out of order signals a drop
  - Negative ACK (NACK)

    Receiver requests missing packet
  - Selective ACK (SACK)

    Receiver describes state of receive window
  - Fast retransmit

    Sender detects missing ACK from multiple duplicate ACKs

    Recall: receiver ACKs highest sequence # received in order

    Triple duplicate ACKs for fast retransmission (shortcut timeout)
- How is retransmission timer related to congestion control?

Transport Protocol Summary

- TCP designed to connect arbitrary hosts on the Internet
  - Difficult to determine link characteristics
  - Difficult to determine receiving host characteristics
  - Both host/network characteristics change over time

    Network becomes more congested➔larger RTT

    Receiving becomes overloaded➔smaller advertised window
- TCP provides
  - Reliable, in-order delivery of byte stream
  - Flow control
  - Congestion control

Silly Window Syndrome

- Problem: (Clark, 1982)
  - If receiver advertises small increases in the receive window then the sender may waste time sending lots of small packets

    e.g., application reads small number of bytes, freeing up small amount of kernel buffer space
- Solution:
  - Receiver must not advertise small window increases

Nagel's algorithm (self-clocking)

- Small packet problem:
  - Don't want to send a 41 byte packet for each keystroke

    IP 20 bytes, TCP 20 bytes, keystroke 1 byte
  - How long should OS/app buffer keystrokes?
- Solution:
  - Only one outstanding small segment not yet ACK'd

    e.g., telnet cannot echo last character until ACK'd anyway
- Can turn of with TCP_NODELAY option
  - What's the story with these options anyway?