

CS120: Computer Networks

Lecture 16. TCP 2

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Transmission Control Protocol (TCP)

- RFC: 793,1122,1323, 2018, 2581
- Goal: Reliable, In-order Delivery
 - Connection oriented
 - Flow control
 - Congestion control
- ➤ Core Algorithm: Sliding Window

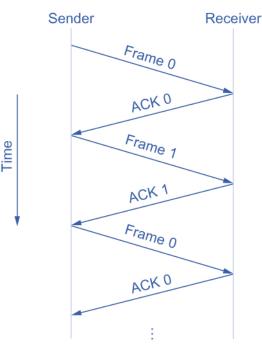
Delay × Bandwidth

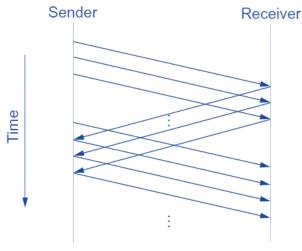
Quantify the utilization of the link

Table 1.1 Sample Delay $ imes$ Bandwidth Products				
Link type	Bandwidth (typical)	One-way distance (typical)	Round-trip delay	RTT imes Bandwidth
Dial-up	56 kbps	10 km	87 μs	5 bits
Wireless LAN	54 Mbps	50 m	0.33 μs	18 bits
Satellite	45 Mbps	35,000 km	230 ms	10 Mb
Cross-country fiber	10 Gbps	4,000 km	40 ms	400 Mb

Delay × Bandwidth

- Efficiency Problem
 - 1.5Mbps bandwidth
 - 45ms RTT
 - 1KB per packet
 - Effective Rate = 1024*8/(1024*8/1.5M+45ms)=162kbps
- Solution
 - Pipeline
 - Full pipe
 - 1.5Mbps*45ms/1kB = 8 packets





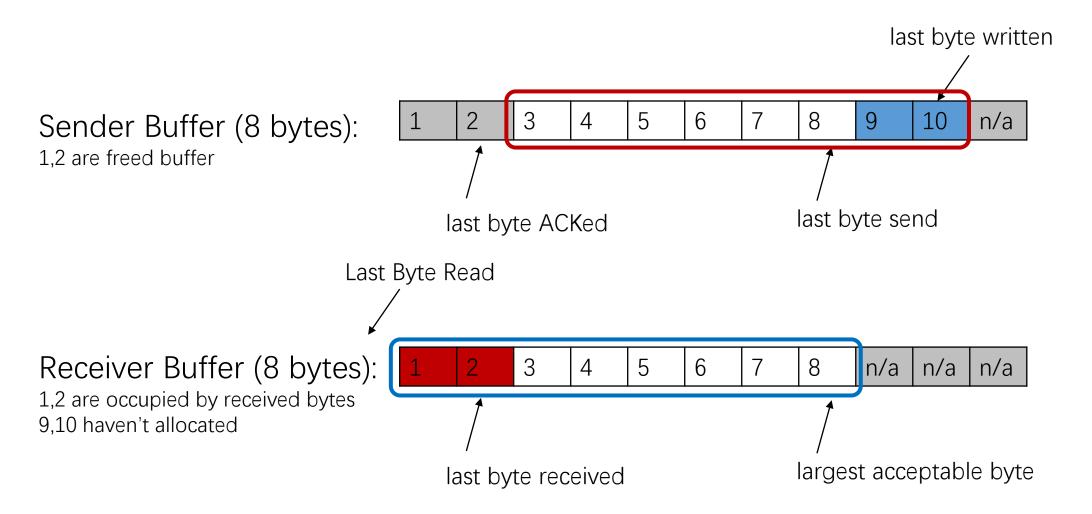
Sliding Window

- Sender Buffer
 - Retransmit
- Receiver Buffer
 - Frame order

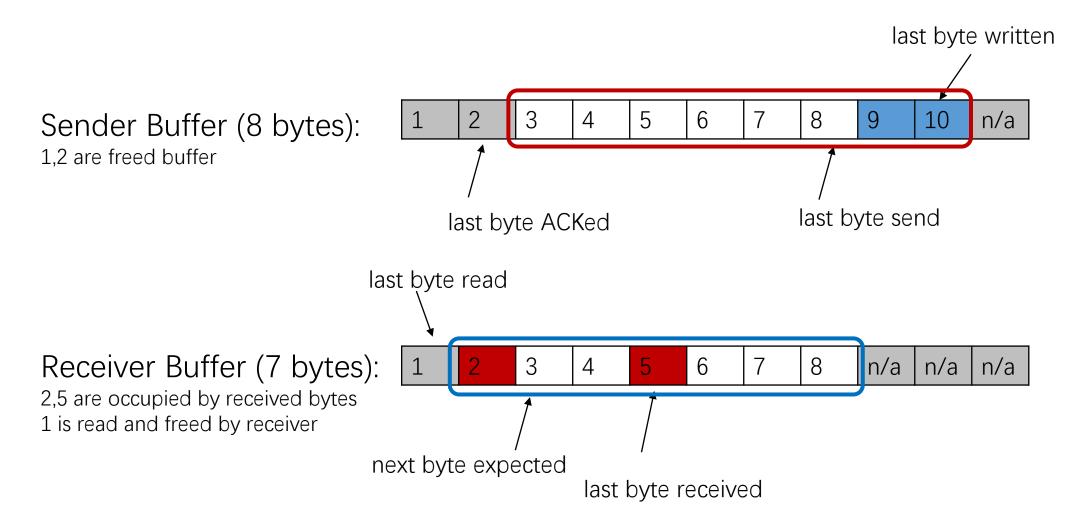
Sliding Window

SWS send window size 3 6 8 10 Sender Buffer: LFS LAR last frame send last ACK received RWS receive window size Receiver Buffer: 3 6 10 5 8 LAF LFR largest acceptable frame last frame received

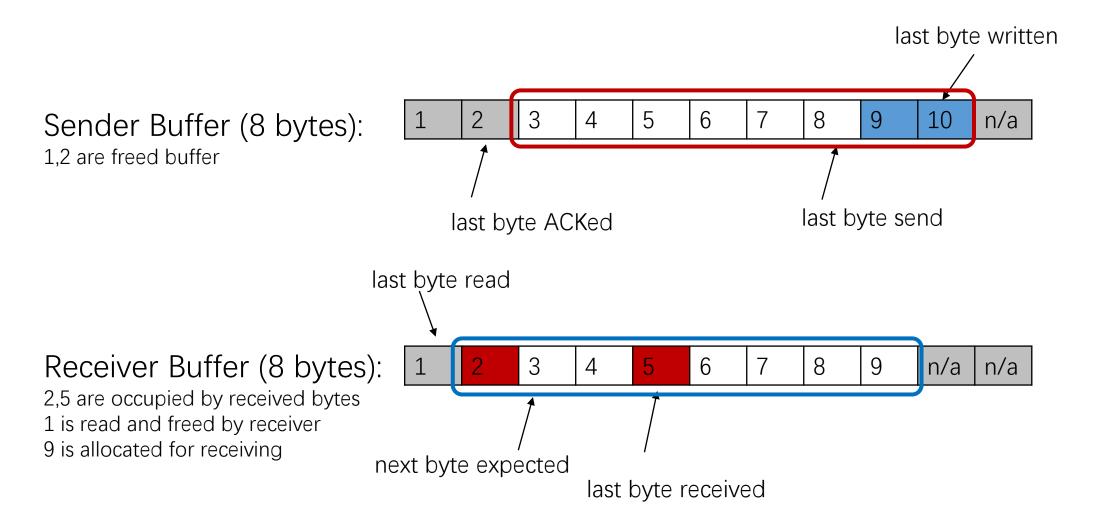
Sliding Window in TCP



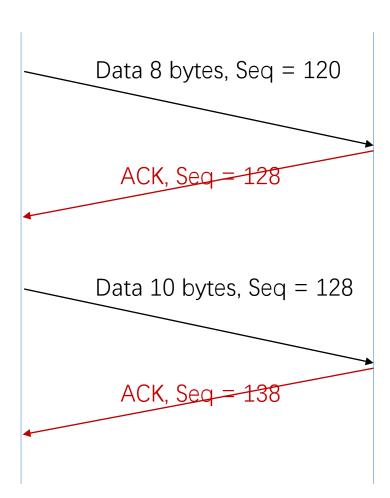
Sliding Window in TCP

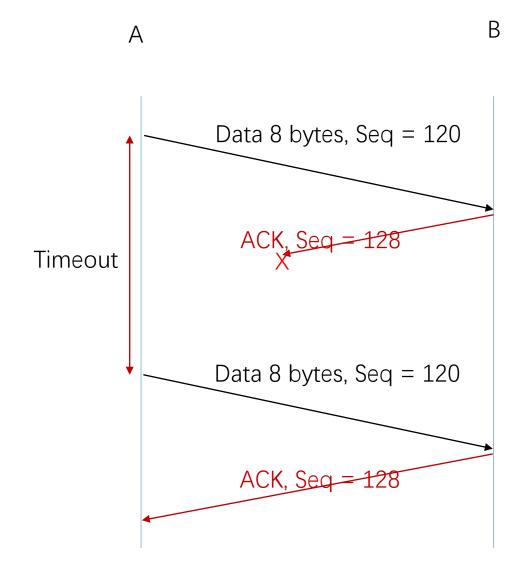


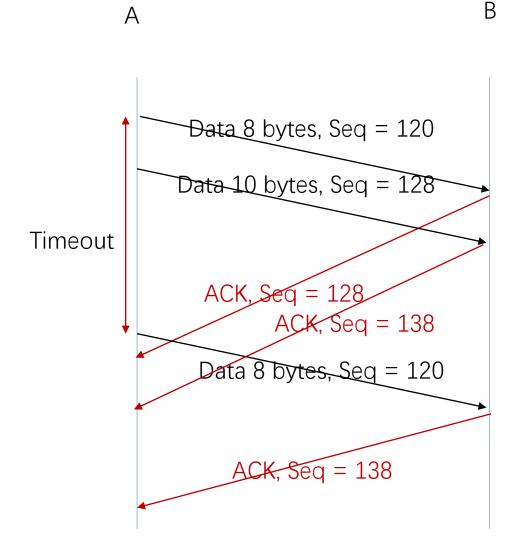
Sliding Window in TCP



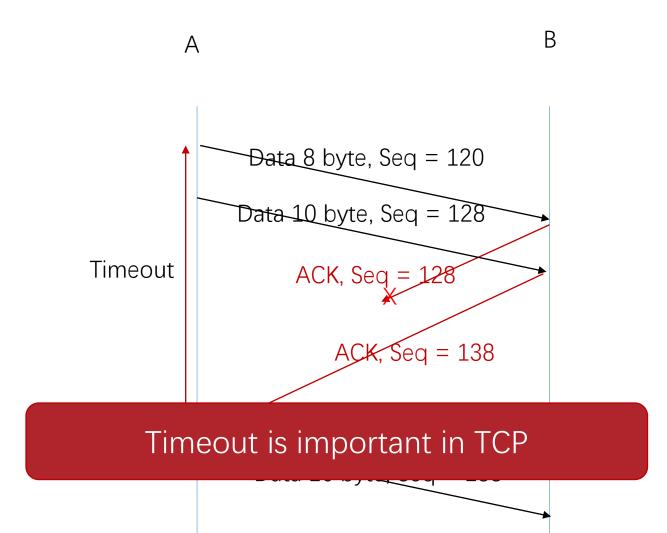
A







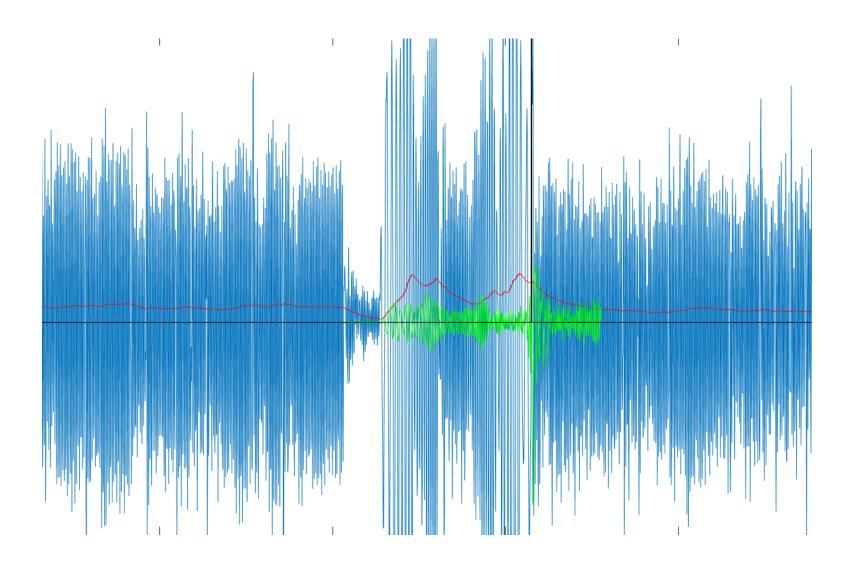
Accumulative ACK



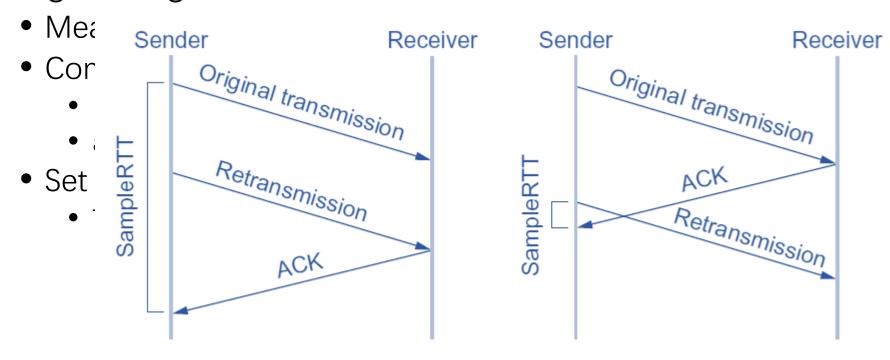
Accumulative ACK

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





Original Algorithm



- Karn/Partridge Algorithm
 - Do not sample RTT when retransmitting
 - Double timeout after each retransmission
 - (But do not increase sliding window)
 - Retransmission => network is congested, should slow down
 - Congestion Control (will introduce in next few lectures)
- The estimation for RTT is not accurate
 - RTT is a critical signal for identifying network congestion

- Jacobson/Karels Algorithm
 - Take RTT variance into account

Difference = SampleRTT - EstimatedRTT

EstimatedRTT = EstimatedRTT + $(\delta * Difference)$

Deviation = Deviation + $\delta * (|Difference| - Deviation)$

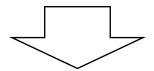
• δ is typically set to 1/8

TimeOut = μ * EstimatedRTT + ϕ * Deviation

• μ is typically set to 1 and ϕ is set to 4

Jacobson/Karels Algorithm Implementation

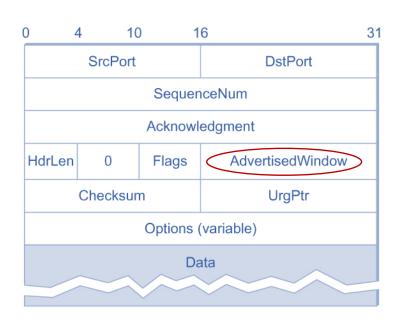
```
Difference = SampleRTT - EstimatedRTT
EstimatedRTT = EstimatedRTT + (\delta*Difference)
Deviation = Deviation + \delta*(|Difference| - Deviation)
TimeOut = \mu* EstimatedRTT + \phi* Deviation
```

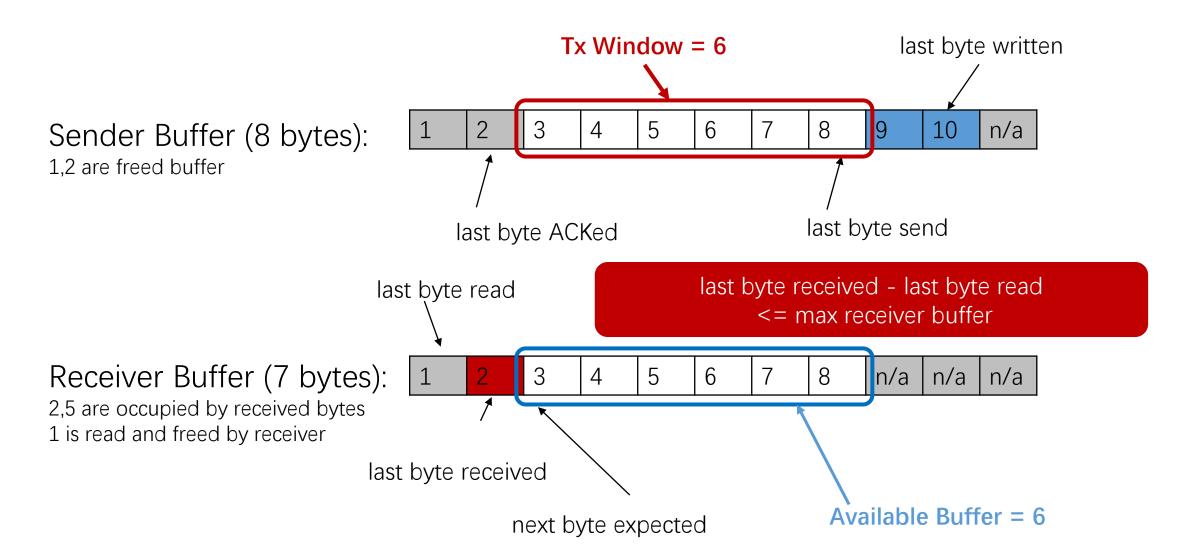


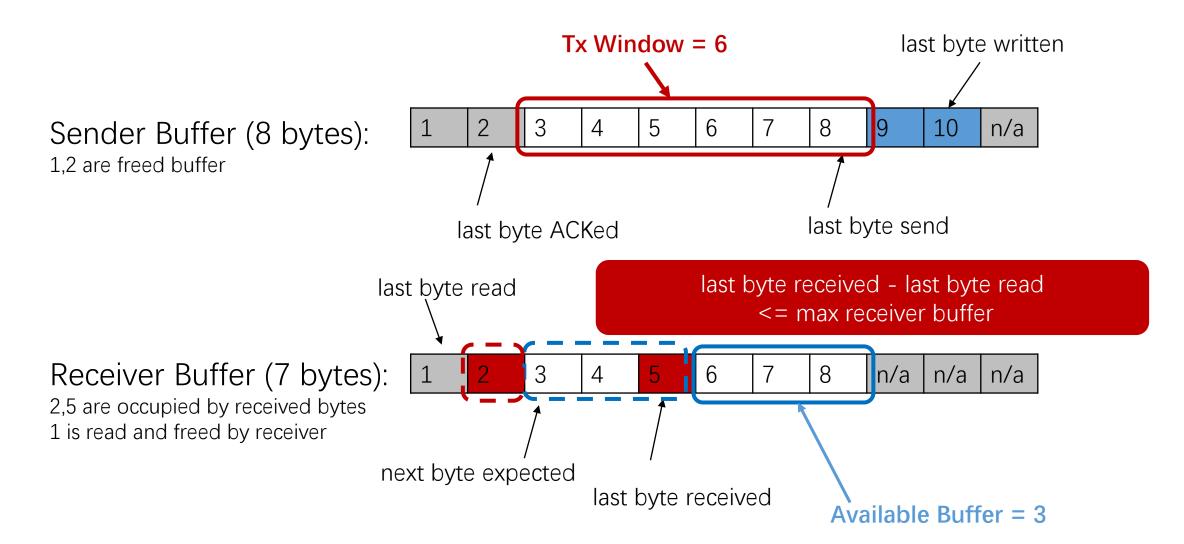
Transmission Control Protocol (TCP)

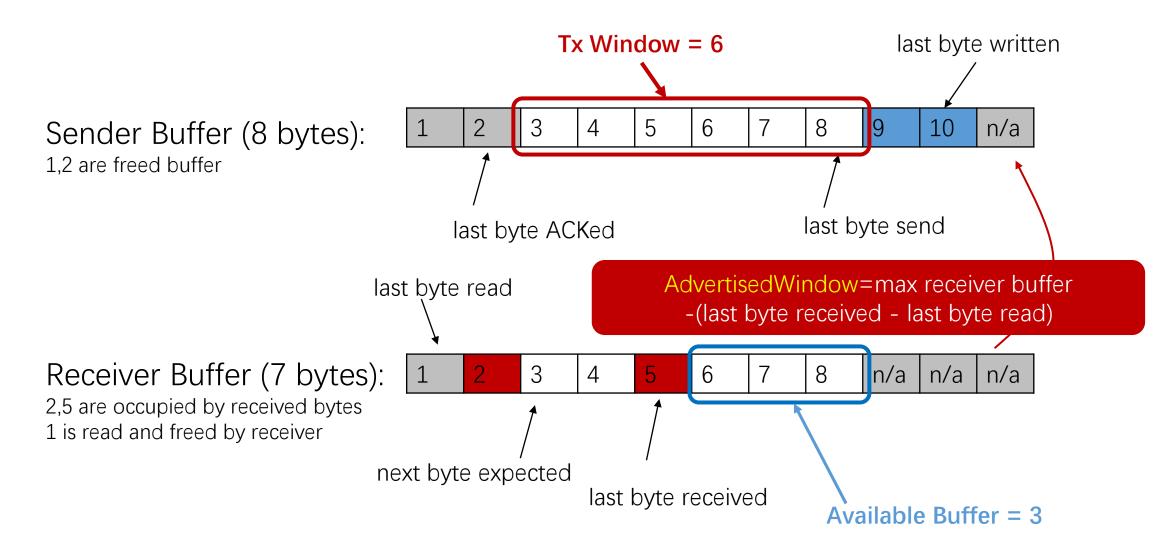
- RFC: 793,1122,1323, 2018, 2581
- Goal: Reliable, In-order Delivery
 - Connection oriented
 - > Flow control
 - Congestion control
- ➤ Core Algorithm: Sliding Window

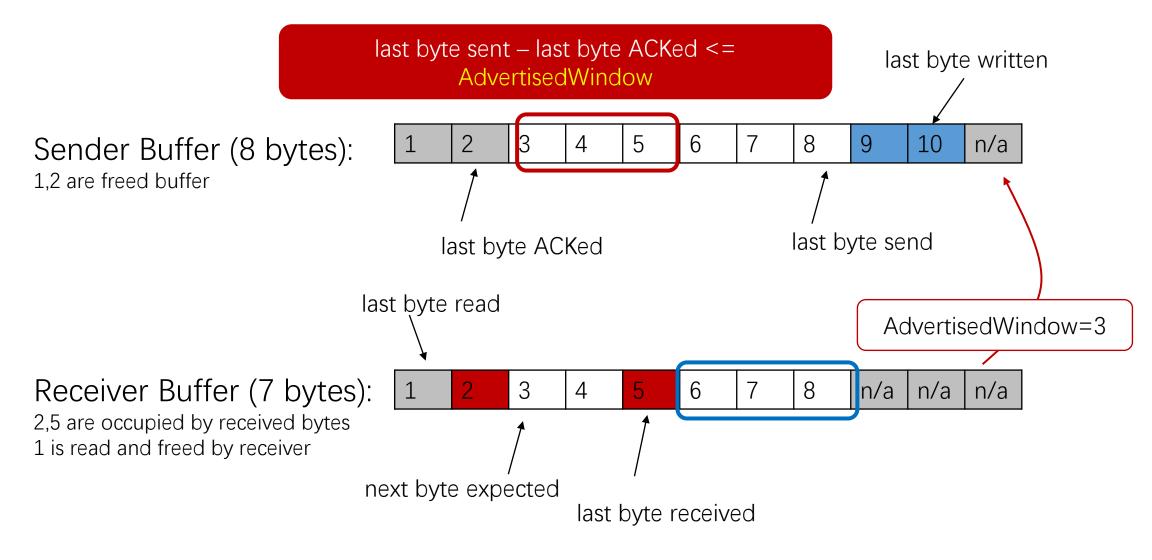
- Goal:
 - Sender does not overrun receiver's buffer
- Approach:
 - Rx window is not fixed, adjusted according to receiving buffer
 - The receiver advertises an adjustable window size (AdvertisedWindow field in TCP header)
 - Sender is limited to having no more than AdvertisedWindow bytes of unACKed data at any time

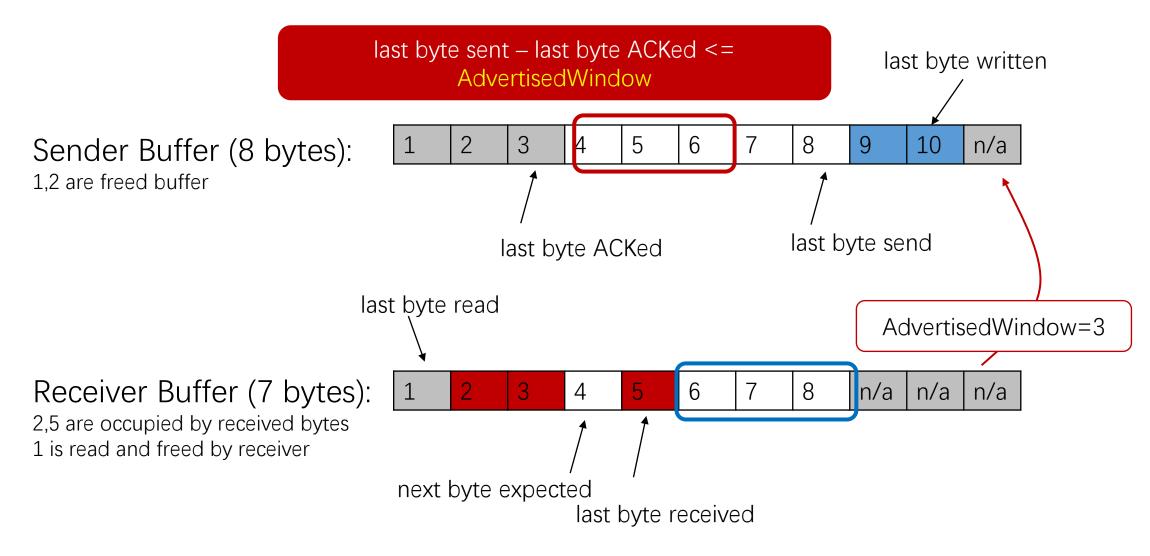






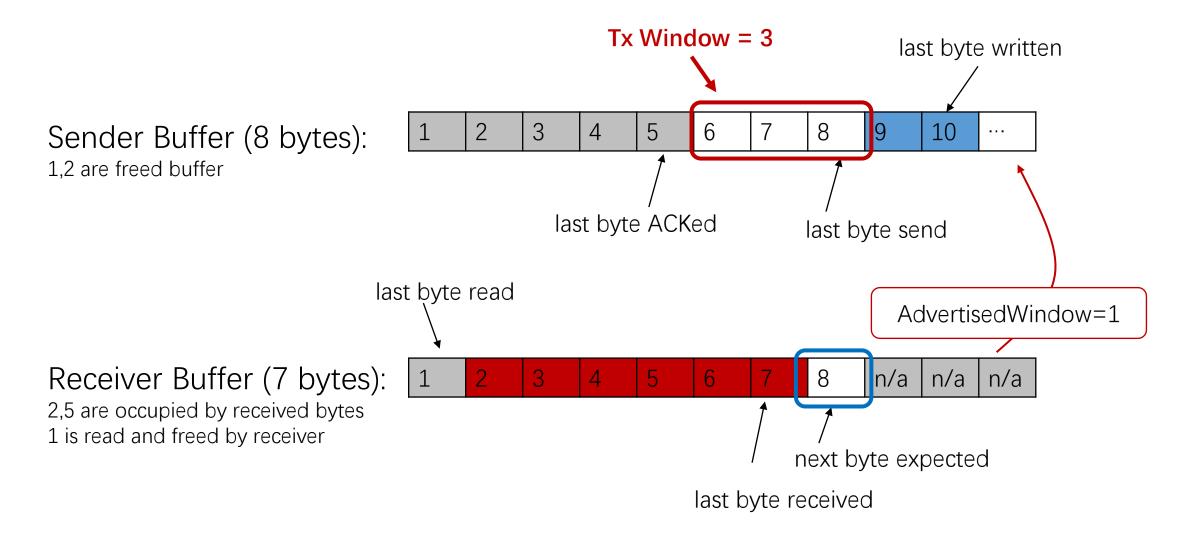


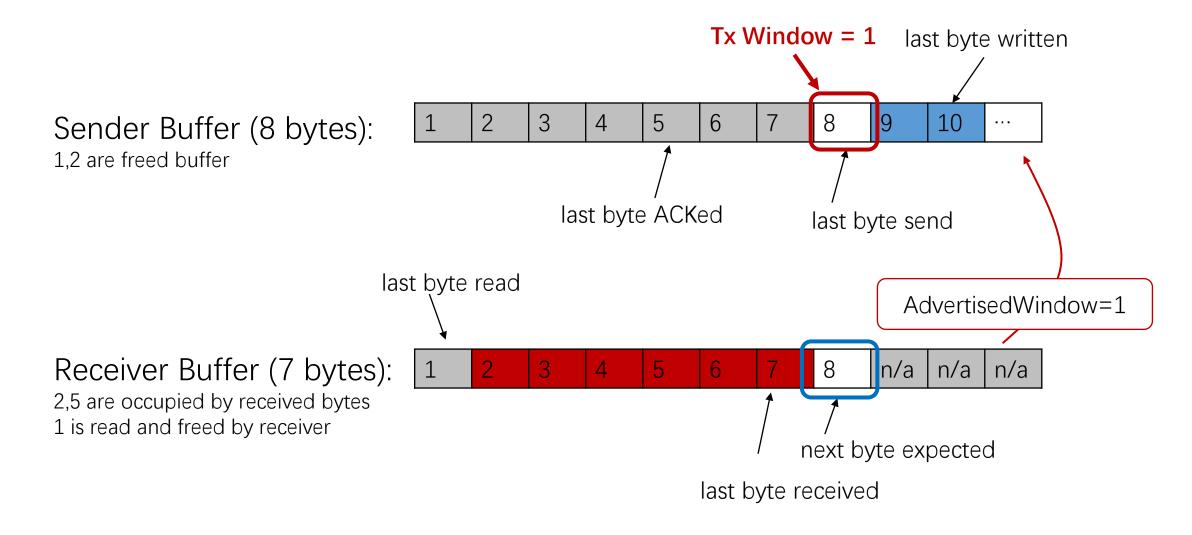


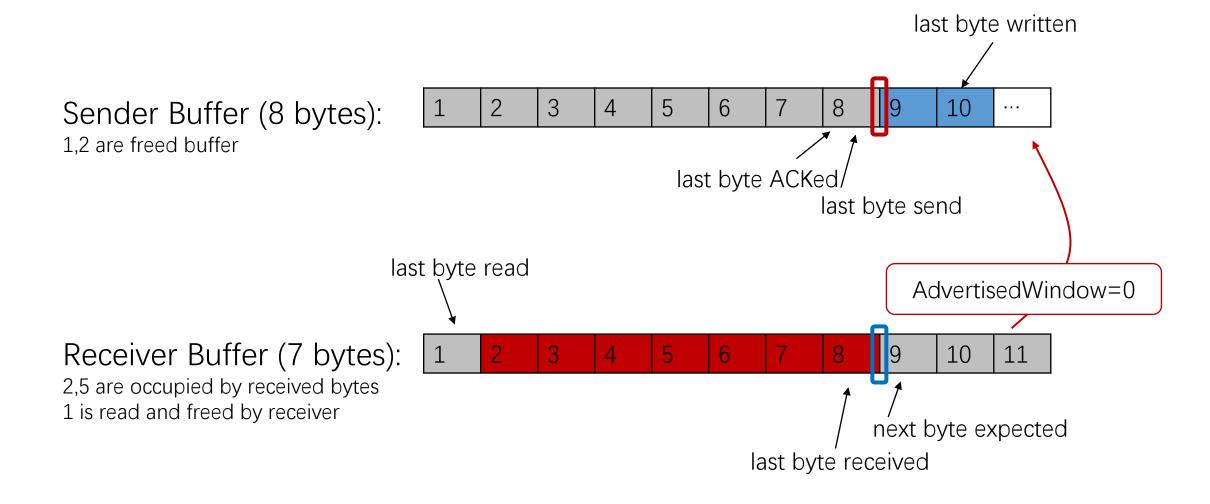


Tx Window = 3last byte written 3 10 2 5 8 Sender Buffer (8 bytes): 4 6 1,2 are freed buffer last byte ACKed last byte send last byte read Receiver Buffer (7 bytes): n/a n/a n/a 2,5 are occupied by received bytes 1 is read and freed by receiver next byte expected last byte received

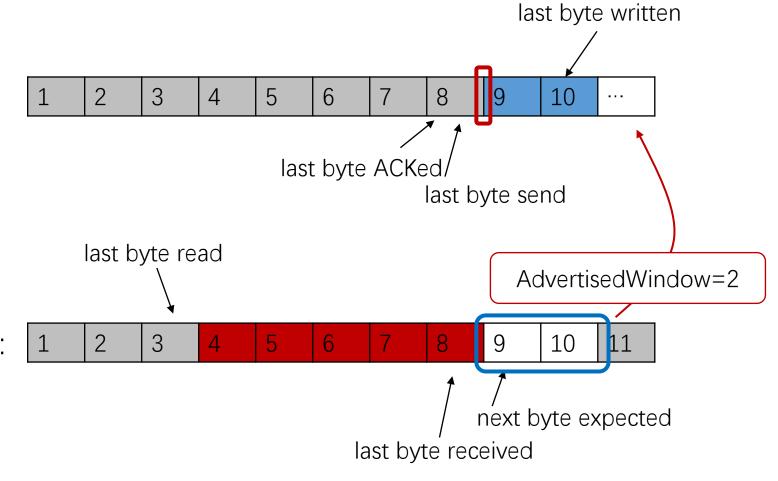
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Sender Buffer (8 bytes): 1,2 are freed buffer

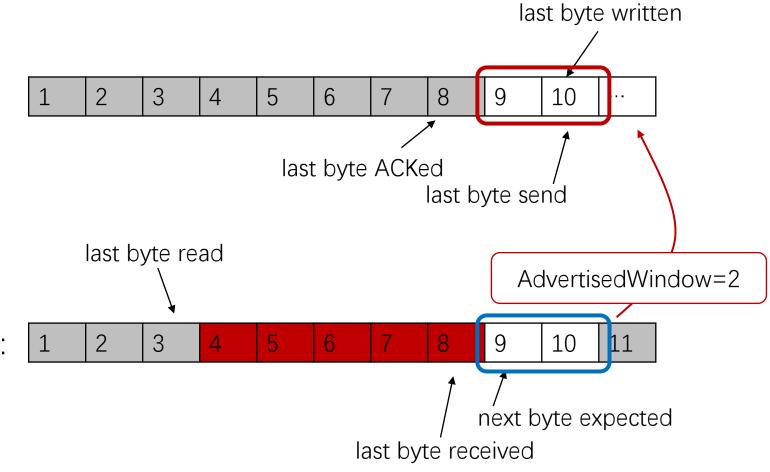


Receiver Buffer (7 bytes):

2,5 are occupied by received bytes 1 is read and freed by receiver

Sender Buffer (8 bytes):

1,2 are freed buffer

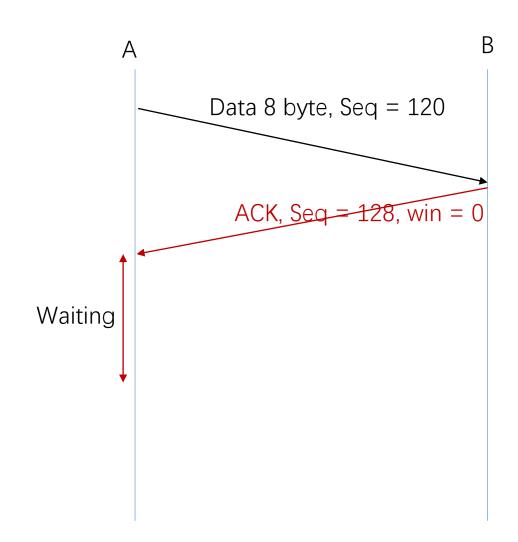


Receiver Buffer (7 bytes):

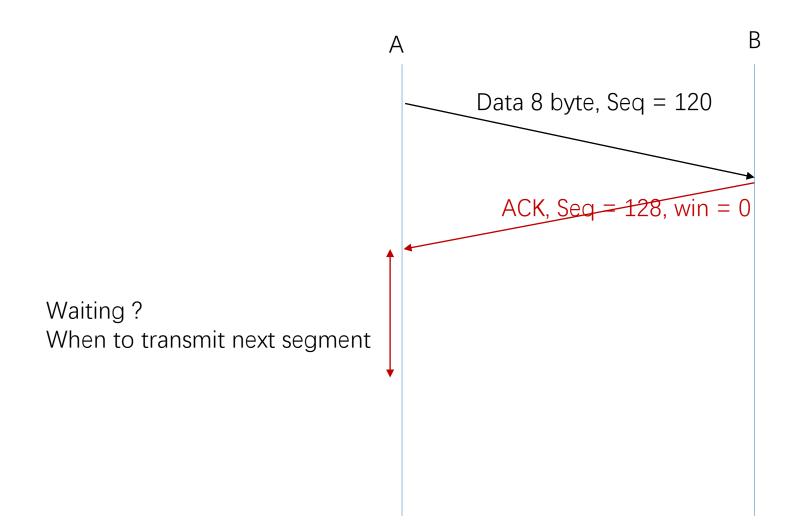
2,5 are occupied by received bytes 1 is read and freed by receiver

- Receiver throttles sender by advertising a window size no larger than the amount it can buffer:
 - AdvertisedWindow=max receiver buffer-(last byte received last byte read)
 - Indicate the amount of free space available in the receive buffer
- Sender must adhere to AdvertisedWindow from the receiver such that:
 - last byte sent last byte ACKed <= AdvertisedWindow

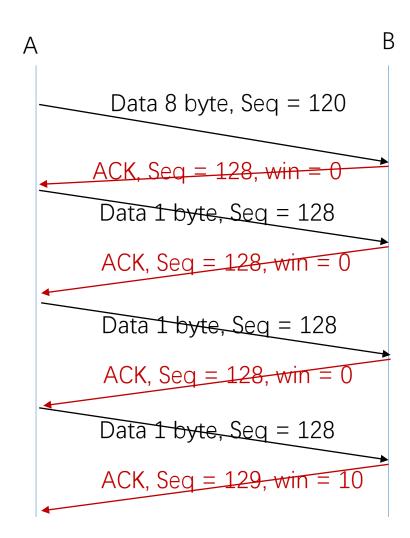
TCP Flow Control: 0 window case



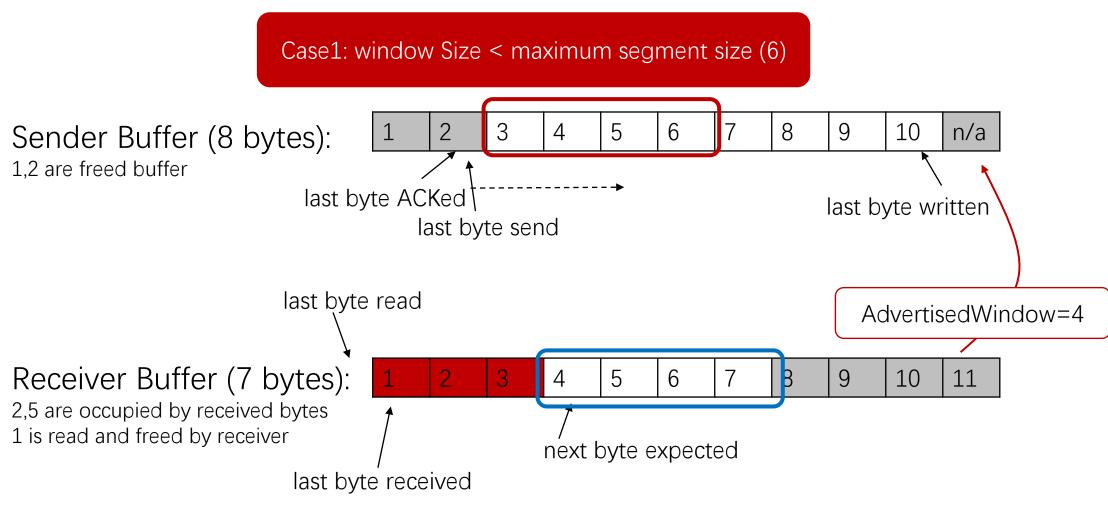
TCP Flow Control: 0 window case



TCP Flow Control: 0 window case



TCP Flow Control: Silly Window Syndrome



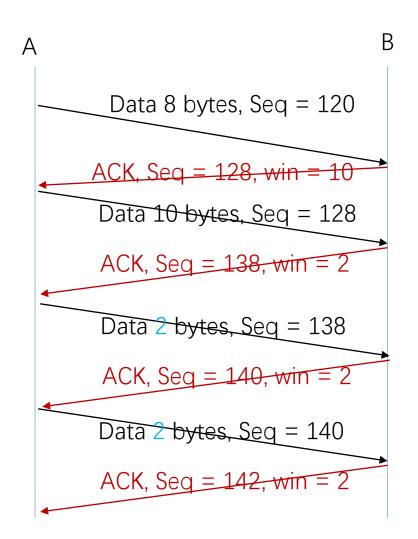
TCP Flow Control: Silly Window Syndrome

Case2: available data < maximum segment size (6) 3 9 10 n/a Sender Buffer (8 bytes): 5 6 8 1,2 are freed buffer last byte ACKed last byte send last byte written last byte read AdvertisedWindow=7 Receiver Buffer (7 bytes): 9 2 3 6 8 10 2,5 are occupied by received bytes 1 is read and freed by receiver

last byte received

next byte expected

TCP Flow Control: Silly Window Syndrome



TCP: How to Trigger Transmission

- Best of Effort
 - Silly Window Syndrome
- Wait Until Full
 - Less Timely
- **≻**Hybrid
 - Wait Until Full + Timer

TCP: Nagle's Algorithm

When the application produces data to send if both the available data and the window \geq MSS send a full segment else if there is unACKed data in flight buffer the new data until an ACK arrives else send all the new data now

Transmission Control Protocol (TCP)

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- Core Algorithm: Sliding Window

Some Patches

TCP Record Boundaries

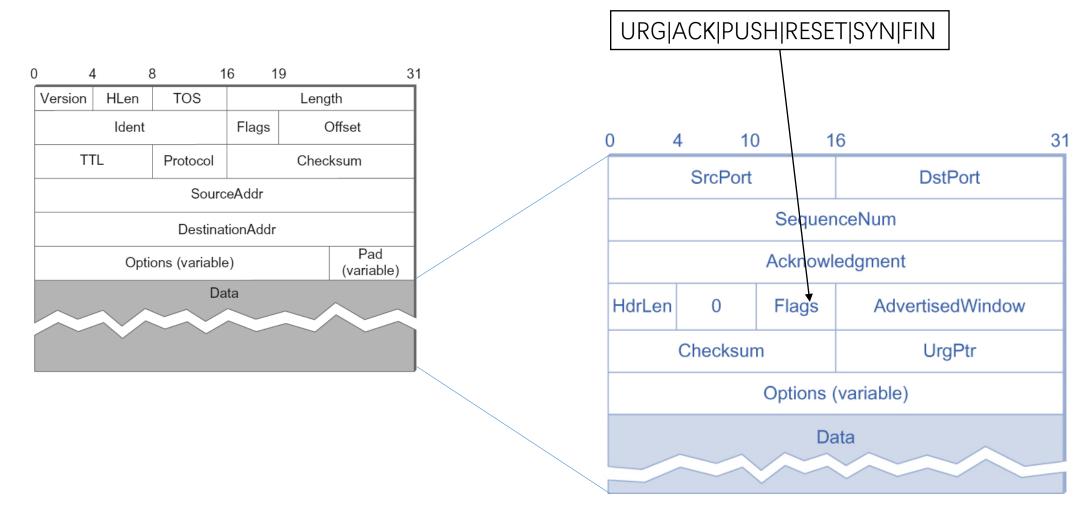
- Problem: TCP is a byte stream, how to specify a message in the stream
 - UDP: one message per frame

Tx: 10 2 2 Rx: 7 7

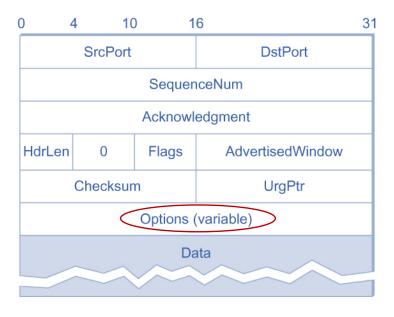
- Solution
 - Sentinel-based Checking
 - URG flag
 - PUSH flag

TCP will inform the application that a URG/PUSH segment is received

TCP: Header

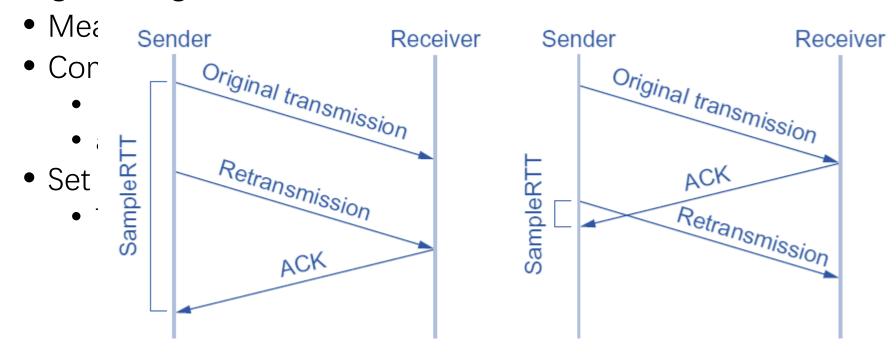


• TCP keeps evolving to incorporate high performance network

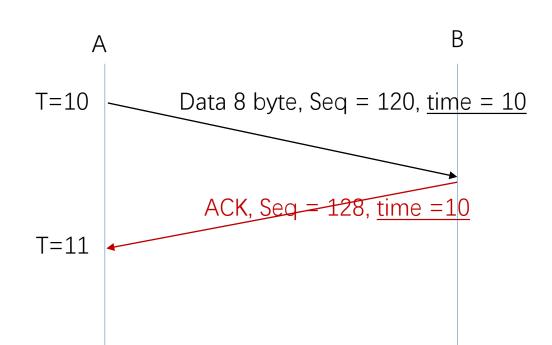


Sliding Window in TCP: Adaptive Timeout

Original Algorithm



- Better Estimation of RTT
 - Introduce 32-bit timestamp



- Sequence Number Wraparound: sequence number can be easily used up in high speed network (too many bytes are out per unit time)
 - Use Seq + Timestamp to identify wraparound

Table 5.1 Time Until 32-Bit Sequence Number Space Wraps Around	
Bandwidth	Time until Wraparound
T1 (1.5 Mbps)	6.4 hours
Ethernet (10 Mbps)	57 minutes
T3 (45 Mbps)	13 minutes
Fast Ethernet (100 Mbps)	6 minutes
OC-3 (155 Mbps)	4 minutes
OC-12 (622 Mbps)	55 seconds
OC-48 (2.5 Gbps)	14 seconds

- Keeping the Pip Full: Advertised Window is not large enough for the Rx buffer
 - Advertised Window (16bits) + Scaling Factor (0 to 14)
 - Negotiated in SYN packet

Table 5.2 Required Window Size for 100-ms RTT	
Bandwidth	Delay $ imes$ Bandwidth Product
T1 (1.5 Mbps)	18 KB
Ethernet (10 Mbps)	122 KB
T3 (45 Mbps)	549 KB
Fast Ethernet (100 Mbps)	1.2 MB
OC-3 (155 Mbps)	1.8 MB
OC-12 (622 Mbps)	7.4 MB
OC-48 (2.5 Gbps)	29.6 MB

Selective ACK

Demo

- TCP Handshake
 - filter: tcp.stream eq X
- TCP Sliding Window
- TCP Extensions

TCP: Design Summary

Stream-oriented v.s. Request/reply





- Byte-stream v.s. Message
 - Arbitrary long message v.s. Limited message
- In Order v.s. Out Order





- Explicit Setup v.s. Implicit Setup
- Window-based v.s. Rate-based

Reference

- Textbook 5.2
- TCP Extenstions https://www.ietf.org/rfc/rfc1323.txt