# TCP congestion control

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## TCP protocol

- TCP (*Transmission Control Protocol* )
- Already reviewed
  - Fundamentals
  - Port mechanism
    - Socket
  - Header format

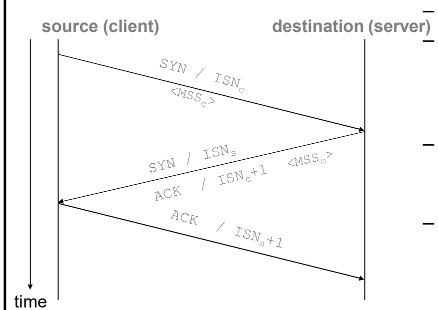


## References


- Richard Stevens: TCP Illustrated
- RFC 793 (1981)
  - Transmission Control Protocol
  - Updated by RFC 3168 (ECN) RFC 6093, RFC 6528
- RFC 7323 (updates RFC 1323 in1992)
  - TCP Extensions for High Performance
- RFC 5681 (obsoletes RFC 2581):
  - TCP Congestion Control
- RFC 6582 (obsoletes RFC 3782 and RFC 2582):
  - The NewReno Modification to TCP's Fast Recovery Algorithm
- RFC 2883 (obsoletes RFC 2018 defined in 1996):
  - An Extension to the Selective ACKnowledgement (SACK ) Option for TCP
- RFC 6298 (obsoletes RFC 2988):
  - Computing TCP's Retransmission Timer



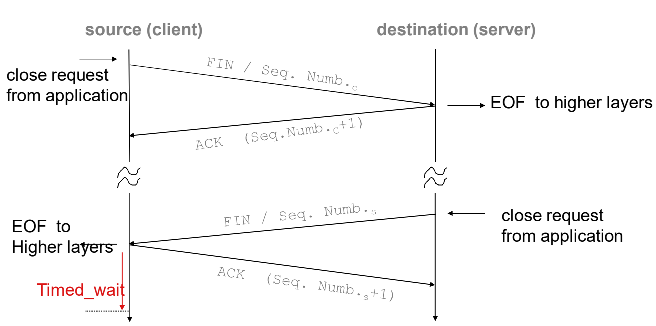

## TCP connection opening (three-way handshake)




- Three-way handshake
- Client executes an active open, server executes a passive open
- ISN (initial sequence number) is randomly generated
- A SYN uses one sequence number

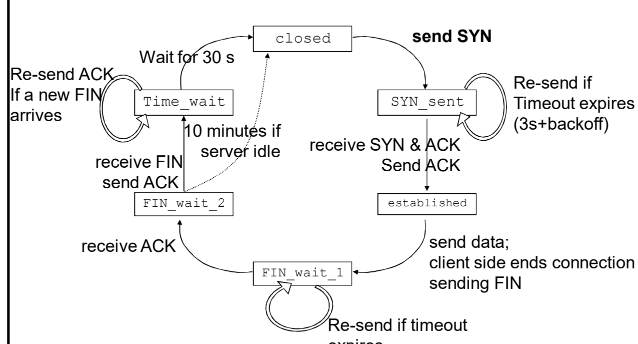


## TCP connection closing (half-close)






## Connection management: client

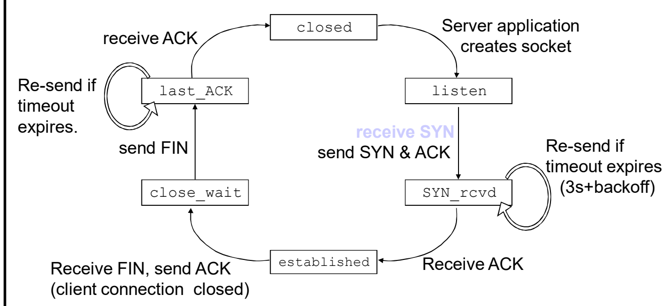


## Notes

- The `Timed_wait` state avoids that old segments belonging to closed connections may interfere with new connections
- `Timed_wait` should be "aligned" to TTL, today a timer set to 30s is used
- During the `Timed_wait` state, socket (ports) cannot be used
- BSD implementation passes from `FIN_wait_2` to `closed` in 10 minutes, of the server does not send any data in the meantime

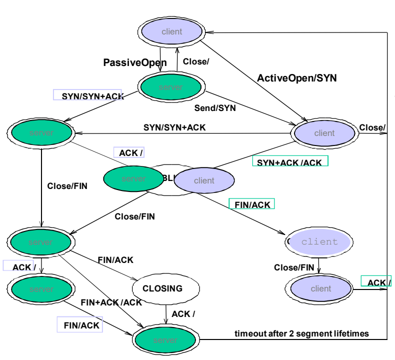


## Connection management: server





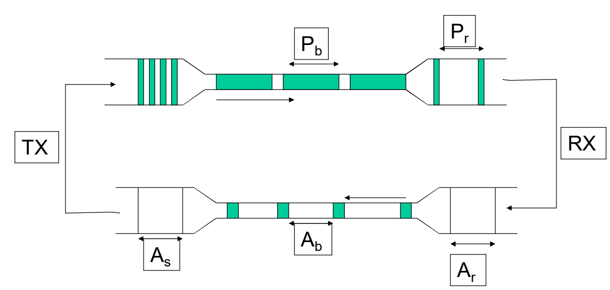

## TCP State Transition Diagram




- Example of TCP connections opened by the client and closed by the server



## Self-clocking behavior




- Segments are spaced within a RTT according to the bottleneck link rate



## TCP transmitter

- Fragments data application in segments
- Computes and transmits checksum over header and data
- Window with Go BACK N retransmission (but!)
- Activates timer when sending segments:
  - Unacknowledged segments induce retransmissions after a timeout expiration
- Like any window protocol, transmission speed ruled by window size
  - Flow and congestion control



## TCP receiver

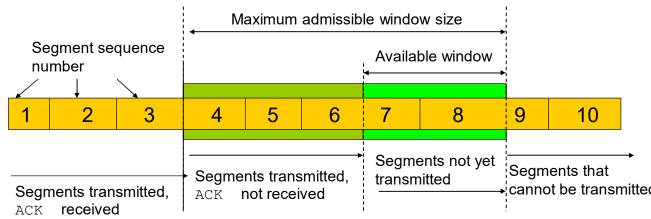
- Discards segments with CRC errors
- Stores out of sequence segments
  - Selective repeat like behaviour
- Re-orders out of sequence segments
  - Delivers an ordered and correct data stream to application process
- Cumulative ACKs
- Declares in the window field of the TCP header the amount of available buffer space to control transmitter sending rate (flow control)

## TCP receiver

- In sequence and correct segment
  - Store the segment (eventually passing it to higher layer protocols) and send a cumulative ACK
- Duplicate segment
  - Discard the segment and send a cumulative ACK with the number of the last segment received in sequence
- Segment with checksum error
  - Discard the segment; no ACK sent
- Out of sequence segment
  - Store the segment (non mandatory, but de facto standard) and send a cumulative ACK with the number of the last segment received in sequence (duplicate ACK )



## Transmitter window






## Transmitter window dynamics

- When an ACK referring to a new segment is received, the transmitter window:
  - Move to the right by the segment size
  - It is possible to transmit a new segment
- When a new segment is transmitted, the available window is reduced by a segment
- If the available window goes to zero, segment transmission is stopped



## Flow and congestion control

- For any window protocol, the transmission bit rate in absence of errors is:

$$\frac{\text{Transmission window}}{\text{Round trip time}}$$

- “Short” connections (small RTT) obtain higher bit rate
- To regulate transmission bit rate (objective of both flow and congestion control), control
  - Round trip time (delay ACK transmission)
    - But generates retransmissions due to timer at the sender
  - Transmission window size



## Flow and congestion control

- TCP: transmitter bit rate regulated by both:
  - Flow control
  - Congestion control
- Flow control: avoid to saturate a slow receiver
  - The receiver controls the speed of the sender
- Congestion control: avoid to saturate the network (more precisely, the link which becomes the bottleneck link)
  - The network controls the speed of the sender
  - Data are stored in node buffer
    - Under congestion
      - Buffer occupancy increases
      - Round trip increases, and bit rate decreases
      - This is not enough to control congestion: packet get dropped because of finite queue size



## Flow and congestion control

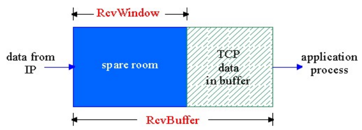
$$\frac{\text{Transmission window}}{\text{Round trip time}}$$

- TCP transmitter window size is regulated:
  - Flow control: receiver declares the available window size (rwnd) (available receiver buffer)
  - Congestion control: the transmitter computes a congestion window (cwnd) value as a function of segment losses detected by missing ACKs
    - Timeout expiration
    - Duplicate ACKs
- The actual transmitter window size is the minimum between the two above values

  transmission window= min(cwnd, rwnd)

## TCP flow control

- TCP receiver explicitly declares the available buffer space (which varies over time)
  - *RcvWindow* or rwnd field in the TCP header
- TCP transmitter window (amount of data sent without receiving ACKs) never exceeds the declared receiver window size (in bytes)




Reception buffer



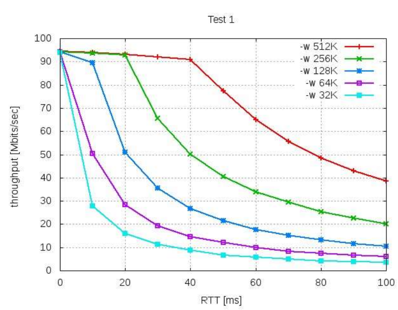
## Impact of flow control

- One TCP sender
- Limited RWND at the receiver
- Line speed C=100Mb/s
- Increasing RTT

- What is the maximum throughput that can be obtained?



## Impact of RWND






## TCP window scaling option

- The RWND field is 16 bit => MaxRWND = $2^{16}$=64kB
- This limits the throughput to

  Throughput <= 64kB/RTT
- To allow faster throughput on high speed/large RTT paths, scale the window (RFC 1323)
  - During the three way handshake, the client and server agree on a scaling factor
  - Uses option field in TCP header
- Default for modern OSes
  - Only Windows XP did not enable this by default
  - Can limit the download speed even on a 20Mb/s ADSL line



## TCP congestion control

- Originally (<1988) TCP was relying only on the window control operated by the receiver to enforce flow control
  - Relatively lightly loaded networks
  - TCP connection limited by the receiver speed
- Congestion effect is segment drops, which implies throughput reduction due to frequent retransmissions
- Goals of congestion control
  - Adjusting to the bottleneck bandwidth
  - Adjusting to bandwidth variations
  - Fairly sharing bandwidth between flows
  - Maximizing throughput



## TCP congestion control

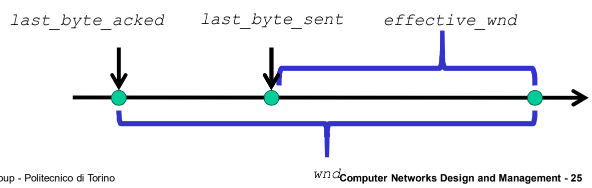
- Besides the limitation imposed by the receiver through the receiver window (rwnd), the TCP transmitter controls the network congestion through the congestion window (cwnd)
- TCP transmitter can send up to B bytes without receiving an ACK, where

  B = min (rwnd, cwnd)
- Several versions of TCP congestion control defined to compute cwnd
  - Reno (NewReno)
  - SACK
  - BIC and CUBIC
  - Many others (Tahoe, Vegas, Westwood, fastTCP, highspeedTCP,…)

## Congestion Window (cwnd)

- Limits amount of in transit data
- Measured in bytes

```
wnd = min(cwnd, rwnd)
effective_wnd = wnd –
(last_byte_sent – last_byte_acked);
```



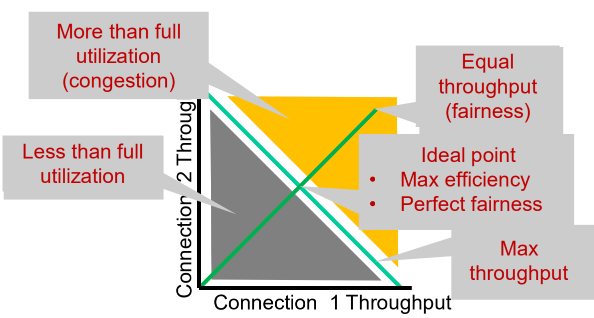



## TCP congestion control

- Obvious idea
  - Try to adapt rate to available resources
  - Increase rate (cwnd) when not congested
  - Decrease rate (cwnd) when congestion detected
- Issues
  - How much to decrease/increase?
  - How to detect congestion?
    - Packet loss => congestion
      - Timeout expiration
      - Duplicate ACKs
  - Need to probe for available bandwidth
    - How to start?
    - How to proceed when congestion is detected?
  - Must work for greedy source but also for …. (e.g. telnet)



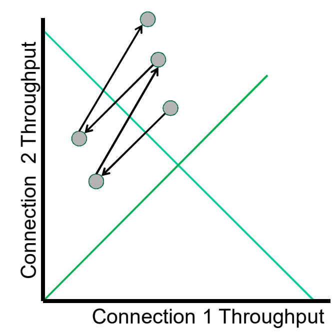
## Utilization and Fairness






## Multiplicative Increase Additive Decrease

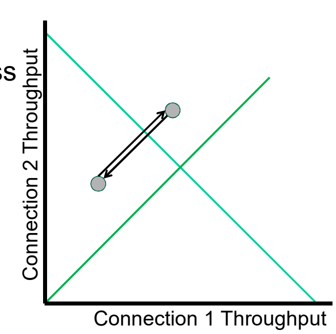
- Not stable!
- Moves away from fairness






## Additive Increase Additive Decrease

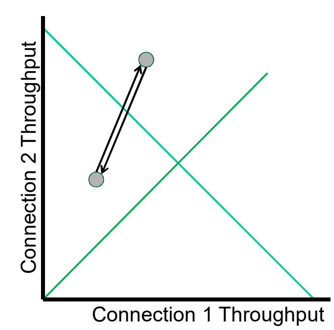
- Stable
- But does not converge to fairness





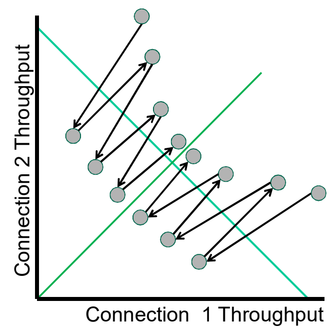

## Multiplicative Increase Multiplicative Decrease

- Stable
- Does not converge to fairness



## Additive Increase Multiplicative Decrease

- Stable
- Converges to ideal working point
- AIMD algorithm






## TCP congestion control algorithm

- Tahoe version (1988)
- Maintains an additional variable (besides cwnd and rwnd)
  - ssthresh: threshold
  - Heuristically set to represent an “optimal” window value
- Two phases of congestion control
  - Slow start (cwnd < ssthresh)
    - Probe for bottleneck bandwidth
  - Congestion avoidance (cwnd >= ssthresh)
    - Probe for bottleneck bandwidth
    - AIMD
- Note: algorithm description assumes for simplicity that each TCP segment has a size equal to 1 MSS

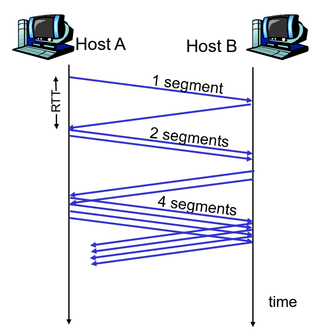


## Slow Start agorithm

- Main ideas
  - Run when cwnd<ssthresh
  - Starts at slow pace but increase fast
- At connection startup
  - cwnd = 1 segment (more precisely, cwnd=1MSS)
  - sstresh = rwnd
- For each in sequence ACK received, cwnd = cwnd + 1MSS
- Exponential window growth
  - For each RTT, cwnd size doubles
  - Not slow!
- Continues until
  - ssthresh is reached
  - A segment is lost

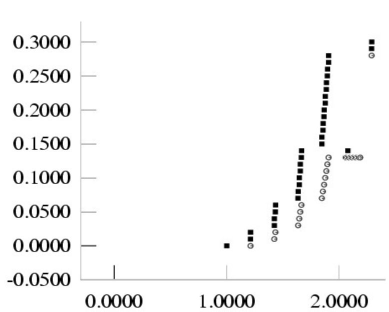


## *Slow Start* algorithm






## Slow Start: example

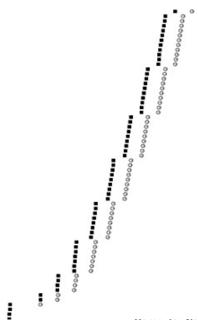





## Congestion Avoidance algorithm

- Main ideas
  - Run when cwnd>=ssthresh
  - Slow down window growth but keep increasing to probe for additional available bandwidth
- For each in sequence ACK received
  - cwnd = cwnd + 1/ cwnd or
  - cwnd = cwnd + MSS/ cwnd (in byte)
- Linear window growth
  - Every RTT, the window increases by 1 MSS in absence of losses
  - ADDITIVE increase
- Continues until a segment is lost

## Congestion Avoidance: example





## If one segment is lost…

- …congestion indication
  - Transmitter bit rate overcame available bit rate
- Main ideas:
  - TCP transmitter re-send the missing segment if the proper ACK is not received within the timeout expiration (“all segments lost” is a severe congestion scenario)
  - Reset the window value (cwnd=1)
  - Set the threshold to half the current window to ensure a fast cwnd increase
    - ssthresh = max(min(cwdn,rwnd)/2,2),



## Summary

1) `cwnd` = 1 MSS
   `ssthresh` = `rwnd`

SLOW START

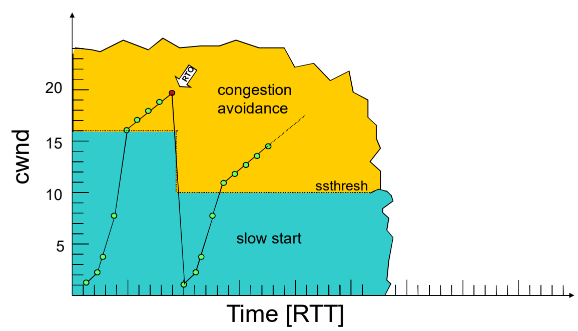
2) `cwnd` = `cwnd` + 1 for each `ACK` `until`
   `cwnd` > `ssthresh (goto 3)`
   if timeout expires:
   `ssthresh` = min(`cwnd`, `rwnd`)/2
   `cwnd = 1`
   `goto 2)`

CONGESTION AVOIDANCE

3) `cwnd` = `cwnd` + 1/ `cwnd` for each `ACK`
   if timeout expires:
   `ssthresh` = min(`cwnd`, `rwnd`)/2
   `cwnd = 1`
   `goto 2)`



## Summary






## Fast Retransmit and Fast Recovery

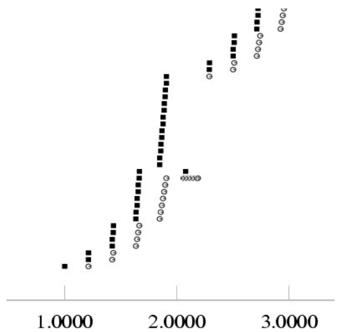
- Further modification to the congestion control algorithm proposed in 1990 (RFC 2001, Stevens)
- It allows the “immediate” retransmission of a single segment lost (Fast Retransmit)
  - Single segment loss is an indication of mild congestion
- …and avoids to re-start the algorithm in the Slow Start phase when a single segment was lost (Fast Recovery)



## Fast Retransmit

- Observe duplicate ACKs
  - If few duplicate ACKs, it may be an out of order segments delivery
  - If more duplicate ACKs are lost, strong indication of segment loss
    - However, since duplicate ACKs are received at the transmitter, other segments were received, which implies mild congestion
- If three duplicate ACKs are received, re-transmit the missing segment without waiting for timeout expiration (Fast Retransmit)

## Fast Retransmit: example



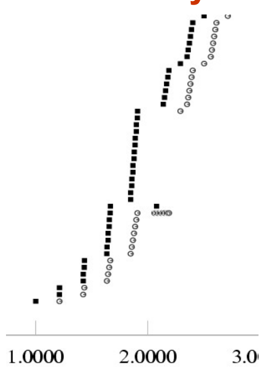



## Fast Recovery

- When congestion detected, go into congestion avoidance phase, and avoids slow start
- When the 3rd duplicate ACK is received:
  - ssthresh = min(cwnd,rwnd)/2
  - re-transmit the missing segment
  - cwnd=ssthresh+3
    - To keep constant the number of segments in the pipe
- For each successive duplicate ACK
  - cwnd=cwnd+1
  - enable segment transmission also during Fast Recovery
- When an ACK confirms the missing segment:
  - cwnd=ssthresh
  - cwnd=cwnd+1/cwnd for each correct and in sequence ACK

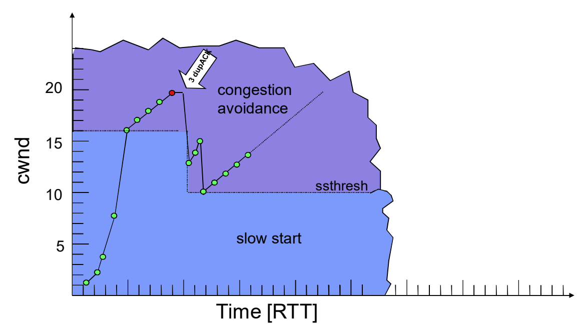


## Fast Recovery: example



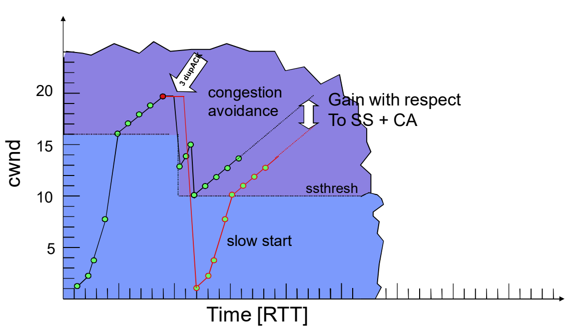



## Summary



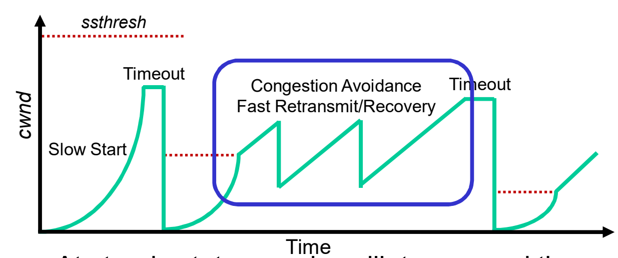



## Summary






## Fast Retransmit and Fast Recovery




- At steady state, cwnd oscillates around the optimal window size
- TCP always forces packet drops

## TCP versions

- TCP Tahoe (Included in 4.3BSD Unix)
  - Originally proposed by Van Jacobson
    - Slow start
    - Congestion avoidance
    - Fast retransmit
- TCP Reno (Proposed in 1990)
  - All TCP Tahoe algorithms
  - Adds
    - Fast-recovery
    - Delayed ACKs
    - Header prediction to improve performance in HW



## TCP Reno: Delayed ACK

- Motivations to delay ACK transmission
  - To reduce the number of ACKs sent (reduce control traffic)
  - To exploit piggybacking to send ACKs
    - The application may create data as a response to received segment
  - To declare a larger rwnd
    - The receiver may empty the reception buffer, declaring larger available window rwnd
- Disadvantages
  - Increases connection RTT (Round Trip Time)
  - Window growth is slowed down



## Delayed ACK: RFC

- The delayed ACK algorithm (RFC 1122, 1989) SHOULD be used by a TCP receiver. When used, a TCP receiver MUST NOT excessively delay acknowledgments. Specifically, an ACK SHOULD be generated for at least every second full-sized segment, and MUST be generated within 500ms of the arrival of the first unacknowledged segment.
- Out-of-order data segments SHOULD be acknowledged immediately, to accelerate loss recovery.



## Delayed ACK : algorithm

- ACKs are sent
  - either every 2 received-in-sequence segments
    - Window growth halved
  - or 200ms after segment reception
- Immediate ACK transmission only for out-of-sequence segments
  - Send ACK for the last in sequence and correctly received segment
    - Generates duplicate ACKs



## TCP ACK generation
### [RFC 1122, RFC 2581]

| Event | TCP Receiver action |
|---|---|
| in-order segment arrival, no gaps, everything else already acked | delayed ACK . Wait up to 500ms for next segment. If no next segment, send ACK |
| in-order segment arrival, no gaps, one delayed ACK pending | immediately send single cumulative ACK |
| out-of-order segment arrival higher-than-expect seq. # gap detected | send duplicate ACK, indicating seq. # of next expected byte |
| arrival of segment that partially or completely fills gap | immediate ACK if segment starts at lower end of gap |





## TCP NewReno

- RFC2582, proposed in 1999
- Solves the TCP-Reno problem
  - Multiple segment drops make useless the fast recovery-fast retransmit mechanism
- Considers partial ACKs reception during a Fast Recovery as a signal of loss of another segment
  - Retransmits immediately
- A new status variable, named recovery, is needed
- When ACK received
  - The Fast Recovery phase is declared ended

## TCP NewReno

- When the 3rd consecutive duplicate ACK is received :
  - ssthresh = min(cwnd,rwnd)/2
  - Recovery=highest sequence number transmitted
  - Retransmit the missing segment
  - cwnd=ssthresh+3
- For each successive duplicate ACK
  - cwnd=cwnd+1
  - Send new segments if possible



## TCP NewReno

- When an ACK which confirms the missing segment is received:
  - If ACK > recovery, then
    - cwnd=ssthresh
    - Fast Recovery procedure ends
  - Else [partial ACK]
    - Shrink transmission window by an amount equal to the confirmed segment size
    - cwnd=cwnd+1
    - Send new segments if cwnd permits



## TCP SACK

- RFC 2018 - 1996
- Introduces selective acknowledge in ACK
  - It changes the semantic and format of ACKs
- Must be negotiated by TCP transmitter and receiver
  - Must understand the new format
- Exploits Option field in TCP header to transport SACK information
  - The receiver tells the sender what it has and what it is missing
- More than one segment per RTT can be retransmitted
  - The sender can then retransmit the missing segments in a single RTT



## TCP SACK

```
                  +--------+--------+
                  | Kind=5 | Length |
+--------+--------+--------+--------+
|      Left Edge of 1st Block       |
+--------+--------+--------+--------+
|      Right Edge of 1st Block      |
+--------+--------+--------+--------+
|                                   |
/            . . .                  /
|                                   |
+--------+--------+--------+--------+
|      Left Edge of nth Block       |
+--------+--------+--------+--------+
|      Right Edge of nth Block      |
+--------+--------+--------+--------+
```

- A block represents a contiguous sequence of bytes correctly received and buffered at the receiver
- The receiver sends SACK info only if some out of sequence segments were received
- May be used to indicate duplicated segments



## TCP variants today

- The most popular version try to address three key problems
  - TCP poor performance on high bandwidth-delay product network
    - How much time is needed to increase cwnd on a 1Gbps link from half utilization to full utilization?
      - Using 1500-byte PDU and 100 ms RTT
      - Full utilization cwnd = 1Gbps/1500byte ~= 8333 segments
      - Half utilization cwnd = 8333/2 = 4166 segments
      - cwnd is increased by 1 for each RTT
        - » 4167 RTTs are needed to fully utilized the link
        - » 4167 RTT * 100ms(RTT time) = 6.95minutes
  - TCP throughput depends on RTT
    - Keep a separate delay based window (Microsoft Windows solution)
  - Vast majority of Internet traffic is made by short flows (e.g., HTTP)
    - Most TCP flows never leave slow start!
    - Increase initial cwnd to 10 (Google, RFC 6928 – 2013)





## TCP today

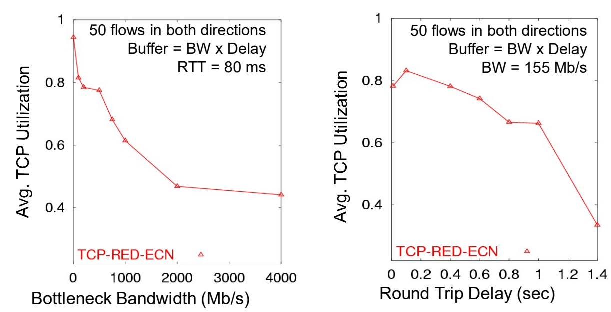
- Compound TCP (Windows)
  - Based on Reno
  - Uses two congestion windows: delay based and loss based
  - Thus, it uses a *compound* congestion controller
- TCP CUBIC (Linux)
  - Enhancement of BIC (Binary Increase Congestion Control)
  - Window size controlled by cubic function
  - Parameterized by the time *T* since the last dropped packet

## High Bandwidth-Delay Product

- Key Problem: TCP performs poorly when
  - The capacity of the network (bandwidth) is large
  - The delay (RTT) of the network is large
  - Or, when bandwidth * delay is large
    - b * d = maximum amount of in-flight data in the network
    - a.k.a. the bandwidth-delay product
- Why does TCP perform poorly?
  - Slow start and additive increase are slow to converge
  - TCP is ACK clocked
    - i.e. TCP can only react as quickly as ACKs are received
    - Large RTT → ACKs are delayed → TCP is slow to react



## Poor Performance of TCP Reno






## Goals

- Speed up cwnd growth
  - Slow start and additive increase are too slow when bandwidth-delay is large
  - Want to converge more quickly
- Maintain fairness with other TCP variants
  - Window growth cannot be too aggressive
- Improve RTT fairness
  - TCP Tahoe/Reno flows are not fair when RTTs vary widely
- Simple implementation

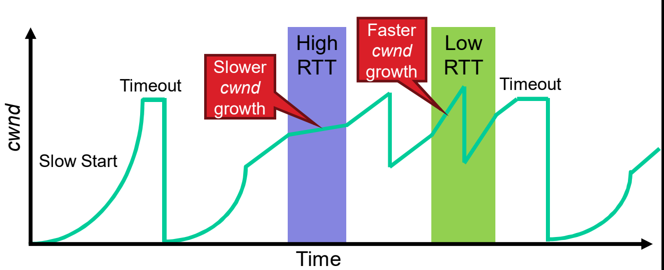


## Compound TCP Implementation

- Default TCP implementation in Windows
- Key idea: split cwnd into two separate windows
  - Traditional, loss-based window
  - New, delay-based window
- wnd = min(cwnd + dwnd, rwnd)
  - cwnd is controlled by AIMD
  - dwnd is the delay window
- Rules for adjusting dwnd:
  - If RTT is increasing, decrease dwnd (dwnd >= 0)
  - If RTT is decreasing, increase dwnd
  - Increase/decrease are proportional to the rate of change



## Compound TCP Example




- Advantages: fast ramp up, more fair to flows with different RTTs
- Disadvantage: must estimate RTT

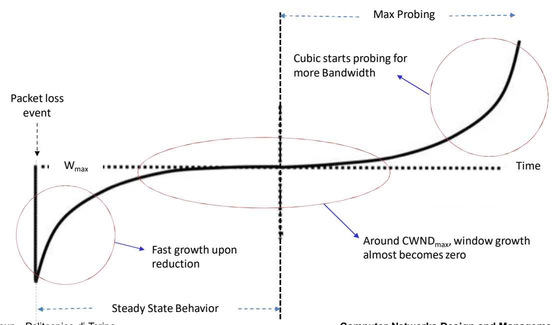




## TCP CUBIC Implementation

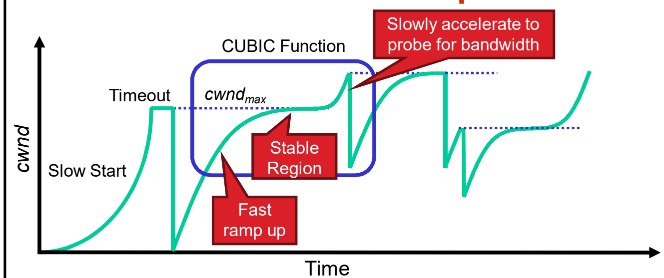
- Default TCP implementation in Linux
- Make window size growth independent of RTT
  - Use elapsed real time since the last loss event
- Replace AIMD with cubic function
- $$cwnd = C * \left(T - \sqrt[3]{\frac{cwnd_{max} * \beta}{C}}\right)^3 + cwnd_{max}$$
  - C → a constant scaling factor
  - β → a constant fraction for multiplicative decrease
  - T → time since last packet drop
  - $cwnd_{max}$ → cwnd when last packet dropped

## TCP CUBIC






## TCP CUBIC Example




- Less wasted bandwidth due to fast ramp up
- Stable region and slow acceleration help maintain fairness
  – Fast ramp up is more aggressive than additive increase



## Timeout setting and RTT estimation

- The timeout value is essential to obtain an efficient ARQ mechanism
- Timeout cannot be smaller than 200ms (delayed ACK and transmitter clock granularity)
- The timeout should be a function of connection RTT, which varies over time, depending on network load (and queueing delay)
- A round trip time estimate is needed to set a proper timeout value



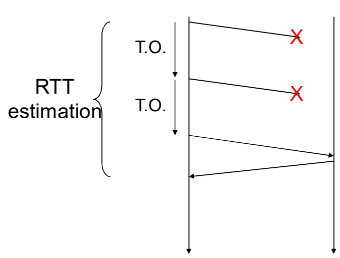
## Timeout setting

- For each transmitted segment, compute the time difference M between segment transmission and corresponding ACK reception
  – Instantaneous RTT sample
  M=t_ack-t_segment
- RTT estimate by weighting through an exponential filter with coefficient $\alpha$:
  – RTT=$\alpha$*RTT+(1-$\alpha$)*M ($\alpha$=0.875)
- Timeout (RTO) set to:
  – RTO= $\beta$ *RTT ($\beta$ >1, typically 2)



## Problems in RTT estimate

- Re-transmitted segment: RTT estimate?



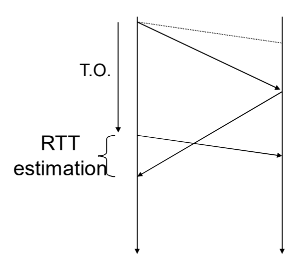

RTT estimate may increase without bound if associating the ACK to the first segment transmission!!





## Problems in RTT estimate

- Re-transmitted segment: RTT estimate?




RTT estimation too small if associating the ACK to segment re-transmission!

## Exponential backoff on the timeout value

- RTT samples of re-transmitted segment may provide a wrong estimante
- Karn algorithm:
  - RTT estimate is not modified unless an ACK for a non retransmitted segment is received
    - Not enough! Indeed, if then RTT increase, a new RTT estimate is never obtained since all segment are re-transmitted
  - Increase timeout value according to an exponential backoff algorithm for each lost segment, since the RTT estimate is not reliable
    - Sooner or later the timeout will assume a value larger than the current RTT; and a new RTT estimate is obtained



## Problems in RTT estimate

- Delay variations may create fluctuations on RTT estimate
  - Use more complex formulas to estimate RTT
  - Take into account the average estimation error (RFC6298 – 2011, RFC2988 – 2000)
    - timeout=average+4*standard_deviation



## Jacobson/Karels Algorithm

- New proposal for RTT estimation
  - Diff = SampleRTT - EstimatedRTT
  - EstimatedRTT = EstimatedRTT + ($\delta$ Diff)
  - Deviation = Deviation + $\delta$(|Diff|- Deviation)
  - Where $\delta$ ranges from 0 to 1
- Standard deviation is considered when computing RTO
  - RTO = $\mu$ EstimatedRTT + $\phi$ Deviation
    where $\mu$ = 1 and $\phi$ = 4



## Notes on RTT estimate

- Estimate is always constrained by timer granularity (10ms on recent systems, 200ms on older systems)
  - The RTT may be comparable with timer granularity (RTT=100-200ms for long distance connections)
- Accuracy in RTT estimation is fundamental to obtain an efficient congestion control (avoids useless re-transmissions or excessively long waits)



## Timeout setting: problems

- Initial value?
- Since an RTT estimation is missing, the initial timeout value is chosen according to a conservative approach
  - Initial timeout set to 1s (RFC6298)
- TCP connections are very sensible to the first segment loss since the timeout value is large



## Silly Window Syndrome

- Excessive overhead problem due to
  - Slow receivers or
  - Transmitter sending only small segments
- If the receiver buffer fills up, the receiver declares increasingly smaller rwnd
- The transmitter sends tinygrams if the applications generates few data (e.g., telnet application)

## TCP connections for telnet traffic

- Telnet application
  - When pressing a key on the terminal keyboard
  - A TCP segment TCP of 1B is sent in a dedicated IP datagram: (20B+20B)header +1B data
- Even worse, if local echo disabled, 4 1B segments are sent: key + ACK + echo + ACK
- Exploiting piggybacking of the first ACK on the echo segment, one segment is saved
  - Delayed ACK helps



## Silly Window Syndrome avoidance

- At the receiver side:
  - Declare the new available receiver window only if equal to
    - 1 MSS or
    - Half of the receiver buffer
  - Delayed acknowledgment
- At transmitter side:
  - Nagle algorithm



## Nagle algorithm (RFC 896)

- When opening the connection, all data in the transmission buffer are sent
- Then, wait for
  - at least 1 MSS data in the transmission buffer or
  - ACK reception
- A host never has more than one tinygram without an ACK



## Nagle algorithm

- When running a telnet application, successive characters following the first one are collected in a single segment, sent after receiving the first ACK
- Ftp, smtp, http connections are not penalized
- The number of tinygrams is drastically reduced
- Is congestion friendly
  - Being ACK clocked, when the network is lightly loaded ACKs are frequently and fastly received and segment transmission is speeded-up
  - When network becomes congested, ACKs are delayed and less segments are sent