The transport layer

INF 3190, Vår 2010

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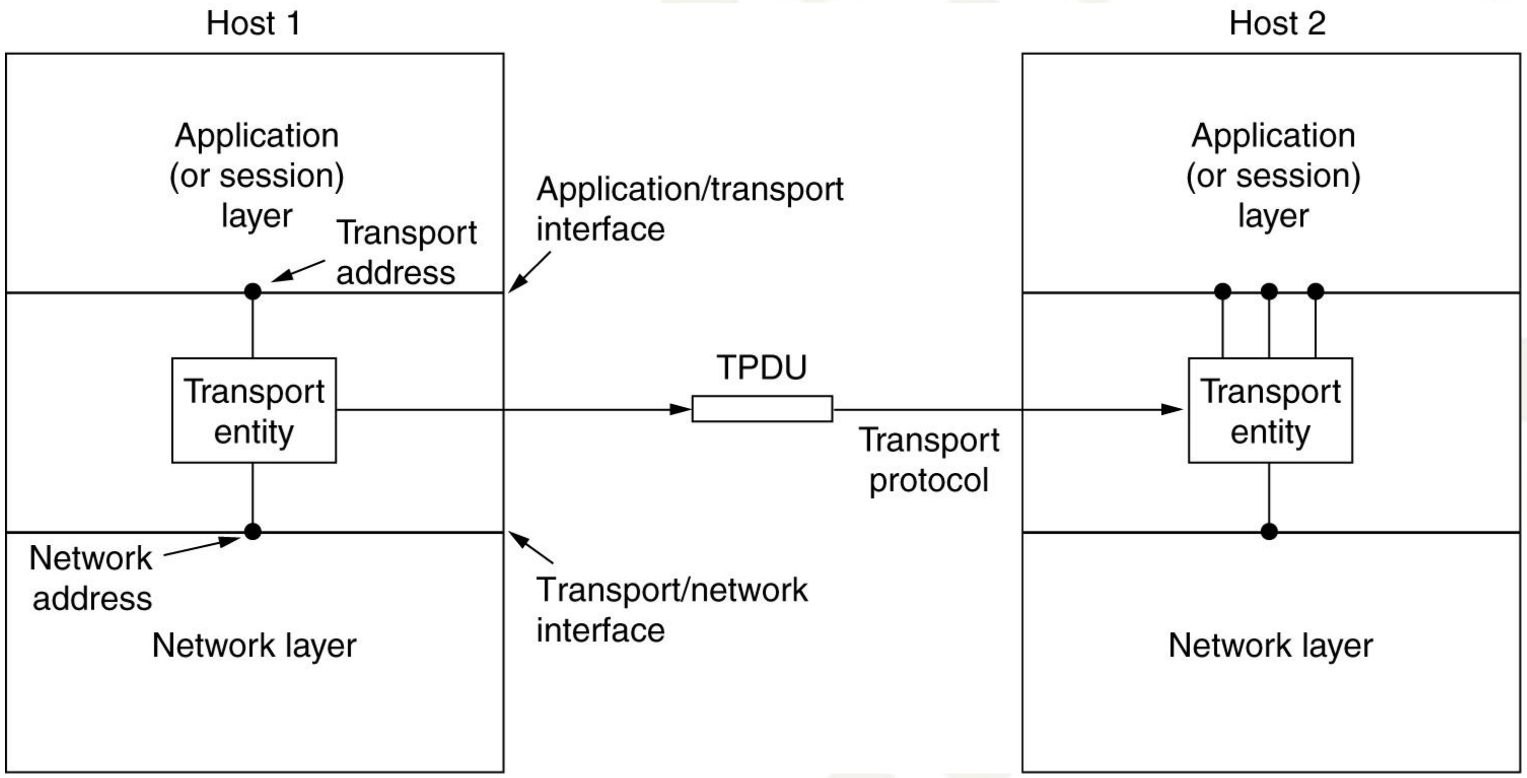
Where we are in the stack...



Ambiguities: bandwidth

Mirriam-Webster online (http://www.m-w.com):

- Main Entry: band*width, Pronunciation: 'band-"width
  Function: noun, Date: circa 1937
  - 1 : a range within a band of wavelengths, frequencies, or energies; especially : a range of radio frequencies which is occupied by a modulated carrier wave, which is assigned to a service, or over which a device can operate
  - 2 : the capacity for data transfer of an electronic communications system <graphics consume more bandwidth than text does>; especially : the maximum data transfer rate of such a system

Traditional, “real“ definition!

“Information rate“

- Unit: definition 1 - “Hz“, definition 2 - “bit/s“ (bps)

- Common interpretation in CN context:
  How many bits/sec can be transferred (“how thick is the pipe”)

Ambiguities: bandwidth /2

- Various wooly “bandwidth“ terms
  - Nominal bandwidth: Bandwidth of a link when there is no traffic
  - Available bandwidth: (Nominal bandwidth - traffic) ... during a specific interval
  - Bottleneck bandwidth: smallest nominal bandwidth along a path, but sometimes also smallest available bandwidth along a path

- Throughput: bandwidth seen by the receiver

- Goodput: bandwidth seen by the receiving application (e.g. TCP: goodput != throughput)

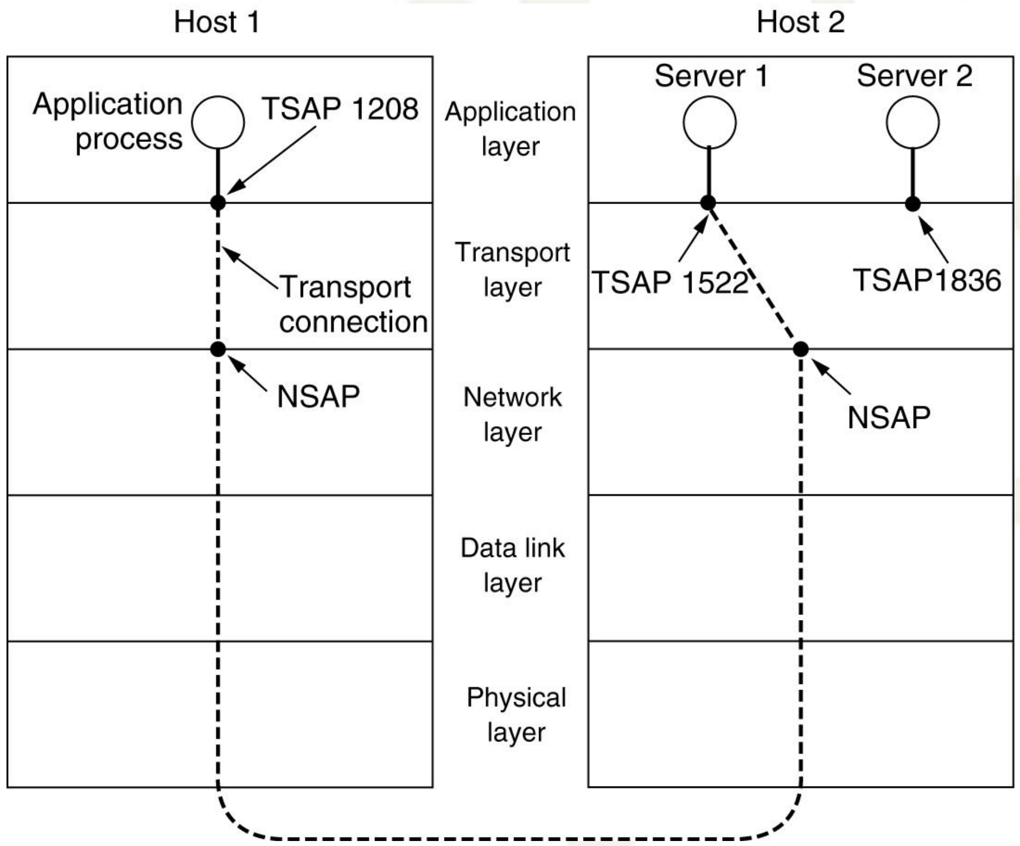
Ambiguities: delay

- Latency - time to transfer an "empty" message
  - also: “propagation delay”
  - limit: speed of light!
- End2end delay = *latency* + *msg_length / bottleneck bandwidth*
  + queuing delay
  - just a rough measure; e.g., processing delay can also play a role, esp. in core routers (CPU = scarce resource!)
- Jitter - delay fluctuations, very critical for most real-time applications
- Round-trip time (RTT) - time a messages needs to go from sender to receiver and back

More ambiguities

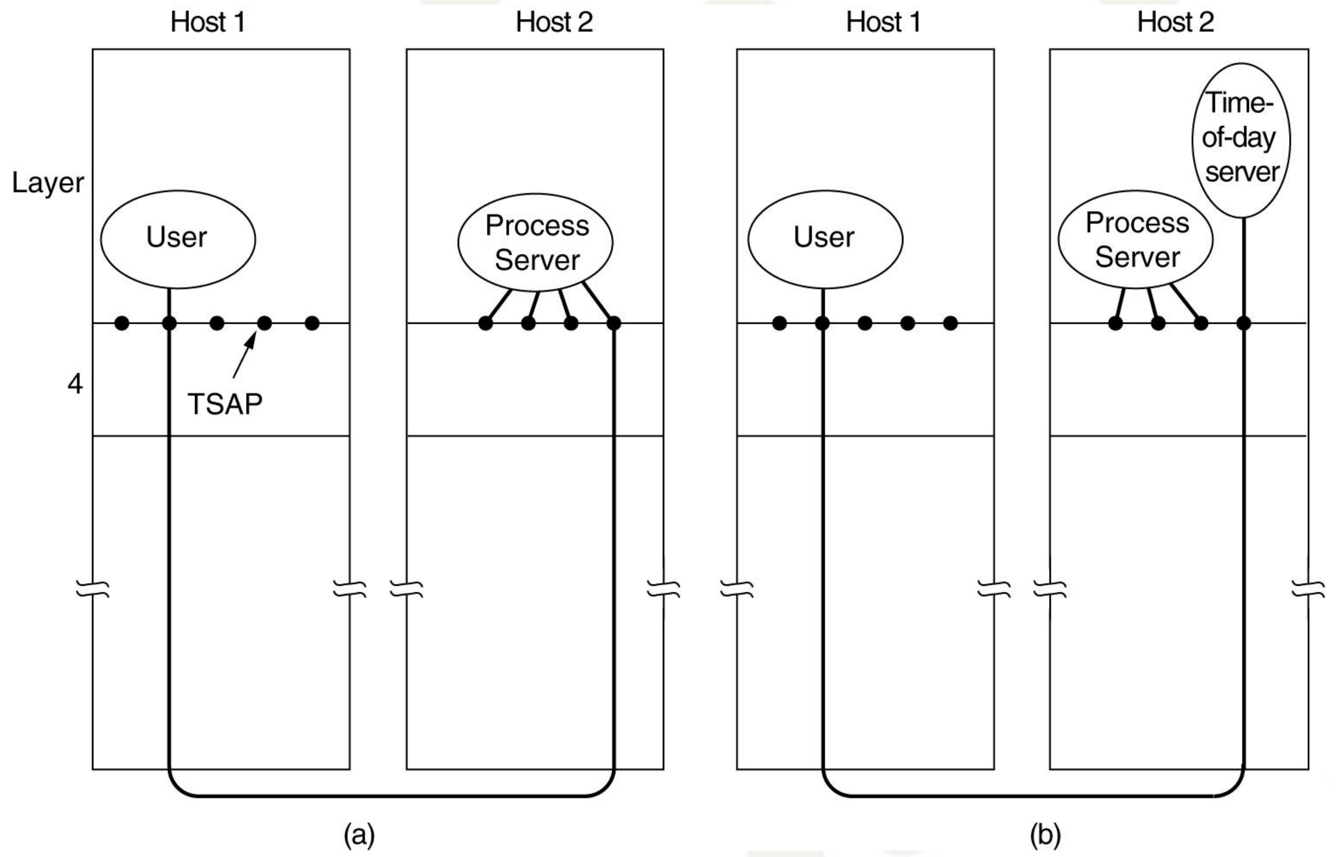
- mbit / mb / Mb / Mbit ... ?
- Latency: sometimes end2end-delay
- link: physical connection between one or more hosts or routers, or link between IP routers (may consist of multiple physical links!)
- capacity: often physical capacity, but different if you talk about TCP
- In general:
  make sure you know which layer you are talking about!

Addressing



TSAPs, NSAPs and transport connections

Connection establishment



How a user process in host 1 could establish a connection with a time-of-day server in host 2

The Internet transport layer

- Services are defined by two protocols
  - UDP (connectionless): sends a “datagram”
  - TCP (connection oriented): transfers a reliable bytestream

- Addressing: port numbers
  - Choosing a service during connection establishment: well-known ports

Primitive	Meaning
SOCKET	Create a new communication end point
BIND	Attach a local address to a socket
LISTEN	Announce willingness to accept connections; give queue size
ACCEPT	Block the caller until a connection attempt arrives
CONNECT	Actively attempt to establish a connection
SEND	Send some data over the connection
RECEIVE	Receive some data from the connection
CLOSE	Release the connection

Berkeley sockets:
TCP service primitives

Focus on the Internet

- The Internet is rather important...

- Its transport layer includes many necessary functions
  - developed as “patches” over the years
  - TCP has grown and grown and grown... should be robust against *everything*!
  - some complementary functions <u>inside</u> the net

- The Internet’s design has been criticized a lot
  - especially recently: a lot of funding for “future Internet”
  - but it’s very hard to change it now

Internet terminology

- PDU, SDU, etc.: OSI terminology
  - Internet terminology: datagram, segment, packet
  - they're all the same

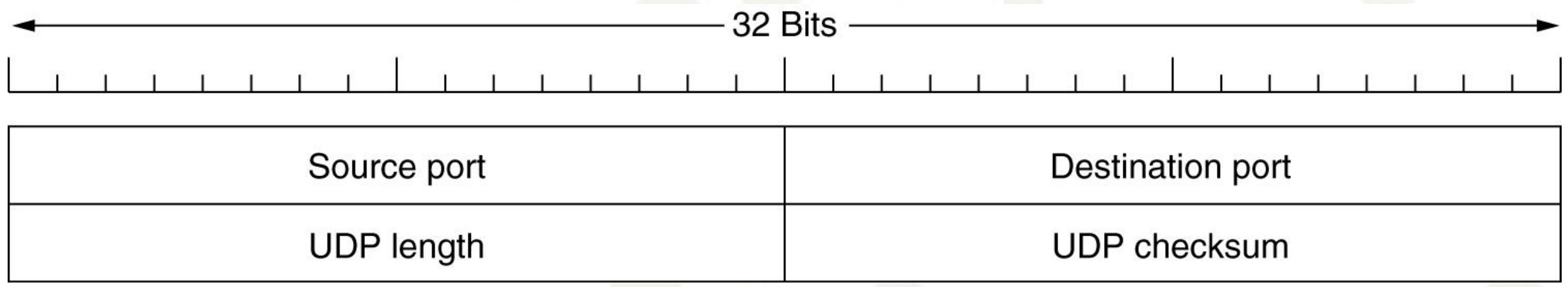
- Theoretically, 1 TCP segment could be split into multiple IP packets
  - hence different words used

- In practice, this is inefficient and not done
  - hence segment = packet

Speaking of packet splitting...

- (IP) fragmentation = inefficient
  - But small packets have large header overhead
- Path MTU Discovery: determine the largest packet that does not get fragmented
  - originally (RFC 1191, 1990): start large, reduce upon reception of ICMP message ➔ black hole problem if ICMP messages are filtered
  - now (RFC 4821, 2007): start small, increase as long as transport layer ACKs arrive ➔ transport protocol dependent
- Network layer function with transport layer dependencies

UDP and UDP Lite



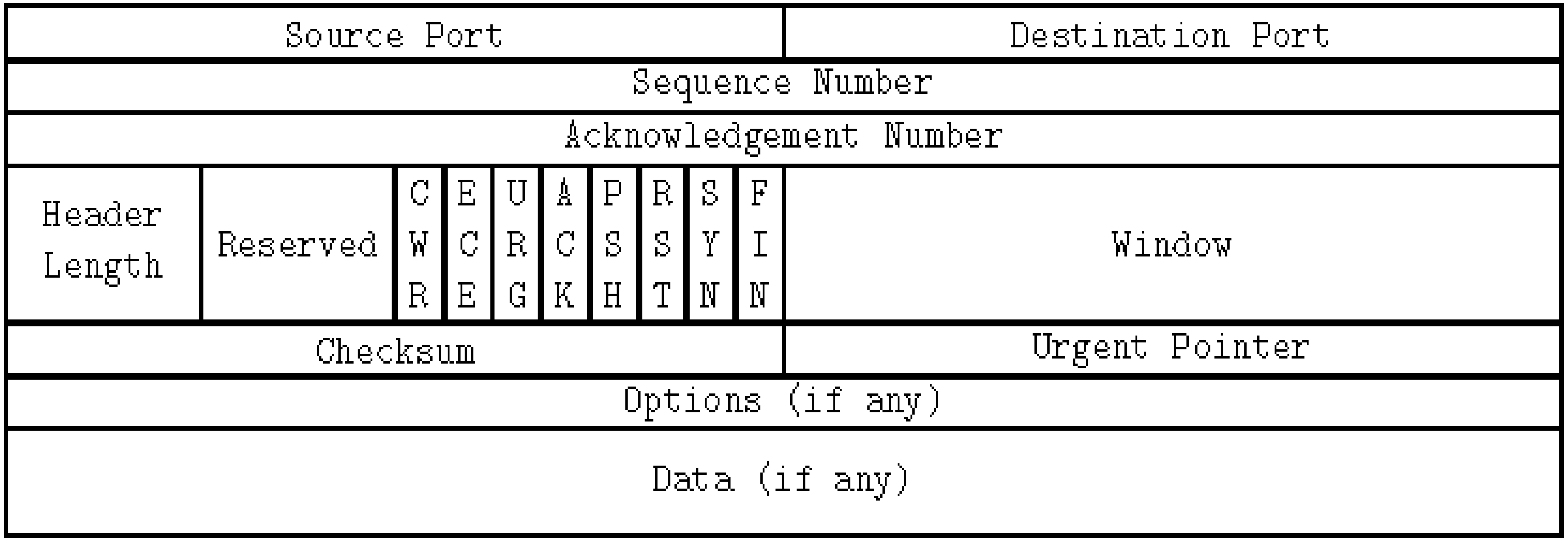
- UDP = IP + 2 features:
  - Ports: identify communicating instances with similar IP address (transport layer)
  - Checksum: Adler-32 covering the whole packet
    - checksum field = 0: no checksum at all! ➔is this useful?
    - ⇒ solution: UDP Lite (length := checksum coverage)
      - advantage: e.g. video codecs can cope with bit errors, but UDP drops whole packet
      - critical: app‘s depending on UDP Lite can depend on lower layers
      - usefulness: often, link layers do not hand over corrupt data

- Usage of UDP: unreliable data transmission (DNS, SNMP, real-time streams, ..)

What TCP does for you (roughly)

- UDP features: multiplexing + protection against corruption
  - ports, checksum
- connection handling
  - explicit establishment + teardown
- stream-based in-order delivery
  - segments are ordered according to sequence numbers
  - only consecutive bytes are delivered
- reliability
  - missing segments are detected (ACK is missing) and retransmitted
- flow control
  - receiver is protected against overload ("sliding window" mechanism)
- congestion control
  - network is protected against overload (window based)
  - protocol tries to fill available capacity
- full-duplex communication
  - e.g., an ACK can be a data segment at the same time (piggybacking)

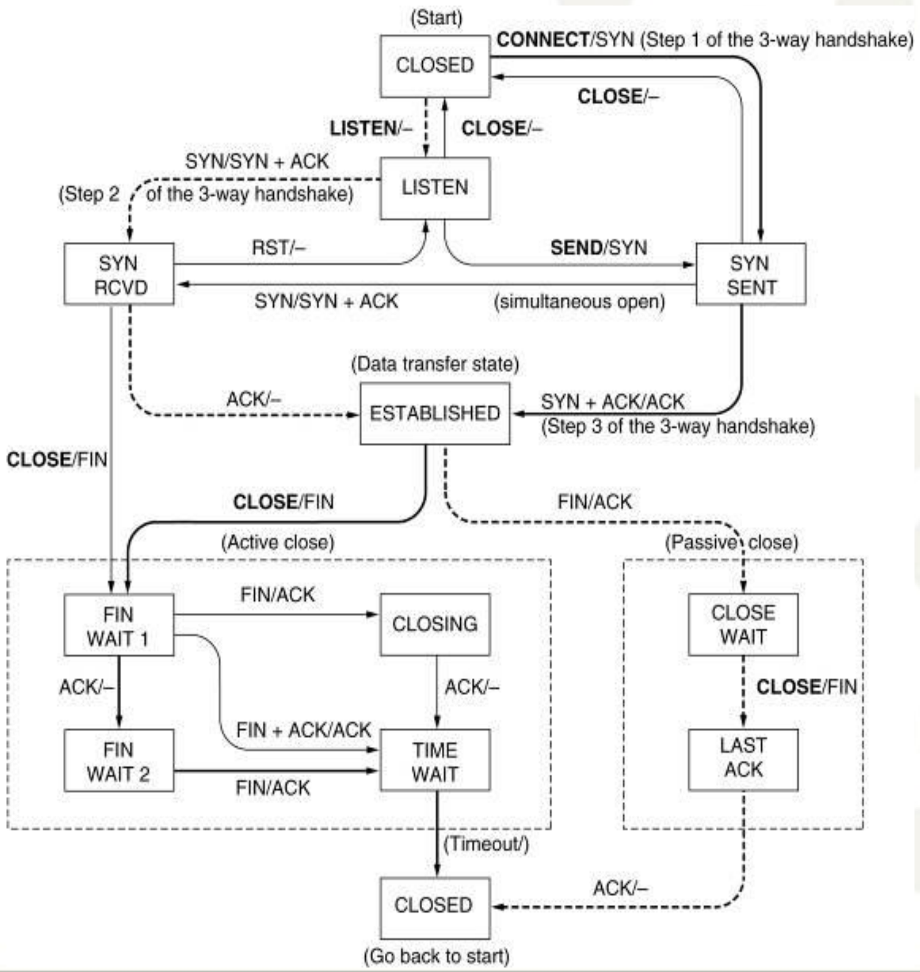
TCP Header



Source Port										Destination Port
Sequence Number										
Acknowledgement Number										
Header Length	Reserved	CWR	ECE	URG	ACK	PSH	RST	SYN	FIN	Window
Checksum										Urgent Pointer
Options (if any)										
Data (if any)										

- Flags indicate connection setup/teardown, ACK, ..
- If no data: packet is just an ACK
- Window = advertised window from receiver (flow control)
  – Field size limits sending rate in today's high speed environments; solution: Window Scaling Option – both sides agree to left-shift the window value by N bit

TCP Connection Management

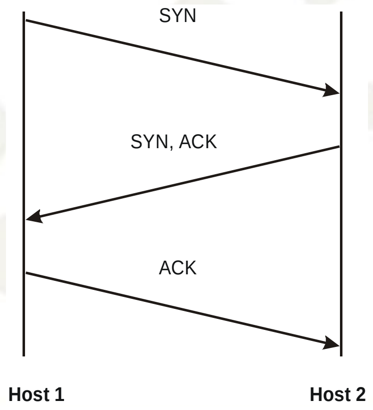


heavy solid line:
normal path for a client

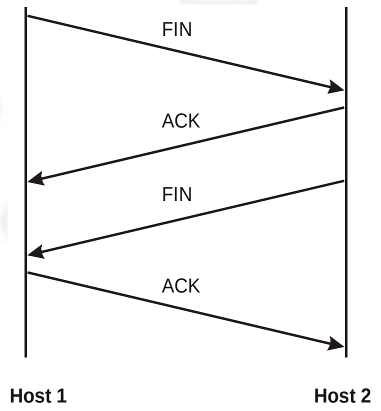
heavy dashed line:
normal path for a server

Light lines:
unusual events

Connection setup



teardown



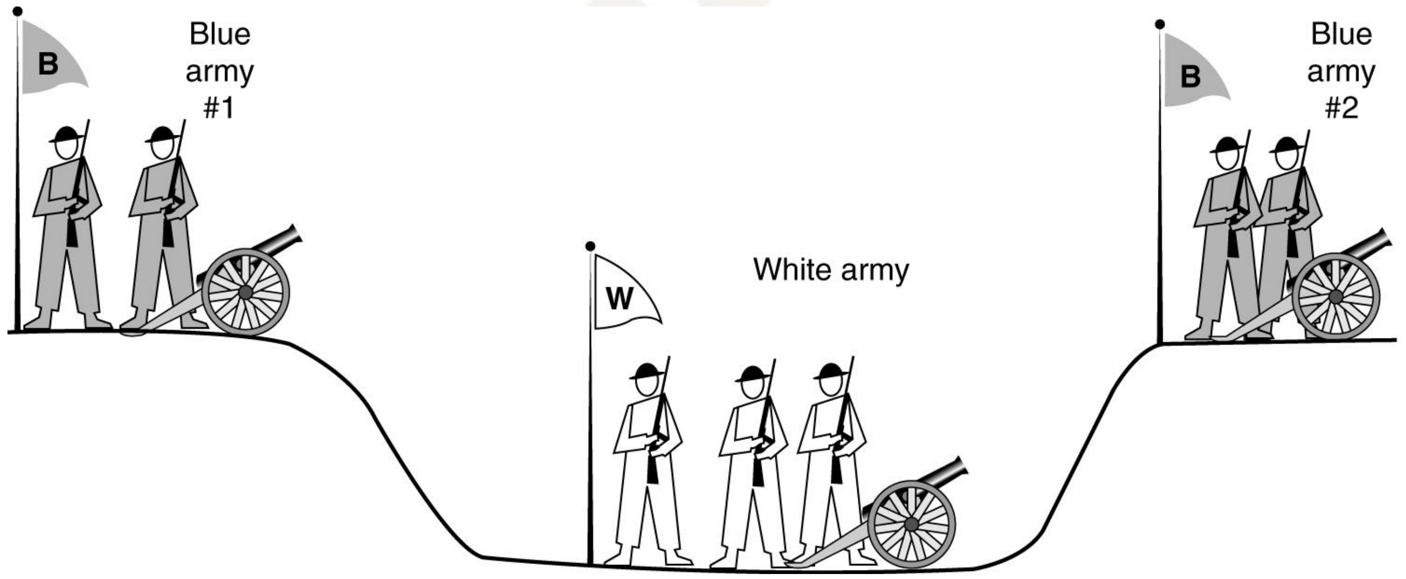
Connection establishment

- Sequence number synchronization ("SYN")
  - avoid mistaking packets that carry the same sequence number but don't belong to the intended connection

- TCP SYN sets up state ("that was the number, I sent a SYN/ACK, now I wait for a response")
  - exploited by SYN flood DoS attack
  - Solution: put state in packets ("cookie")
  - Can be implemented without changing the protocol, by encoding it in sequence numbers

Connection release

- No way to do it without timeouts...



Error control: Acknowledgement

- ACK ("positive" Acknowledgement)

- Purposes:
  - sender: throw away copy of data held for retransmit
  - time-out cancelled
  - msg-number can be re-used

- TCP counts bytes, not segments; ACK carries "next expected byte" (#+1)

- ACKs are cumulative
  - ACK *n* acknowledges all bytes *"last one ACKed"* thru *n-1*

- ACKs should be delayed
  - TCP ACKs are unreliable: dropping one does not cause much harm
  - Enough to send only 1 ACK every 2 segments, or at least 1 ACK every 500 ms (often set to 200 ms)

Error control: Timeout

- Go-Back-N behavior in response to timeout

- Retransmit Timeout (RTO) timer value difficult to determine:
  - too long $\Rightarrow$ bad in case of msg-loss; too short $\Rightarrow$ risk of false alarms
  - General consensus: too short is worse than too long; use conservative estimate

- Calculation: measure RTT (Seg# ... ACK#) , then:
  original suggestion in RFC 793: Exponentially Weighed Moving Average (EWMA)
  - $SRTT = (1-\alpha)\ SRTT + \alpha\ RTT$
  - $RTO = \min(UBOUND, \max(LBOUND, \beta * SRTT))$

- Depending on variation, result may be too small or too large; thus, final algorithm includes variation (approximated via mean deviation)
  - $SRTT = (1-\alpha)\ SRTT + \alpha\ RTT$
  - $\delta = (1 - \beta) * \delta + \beta * [SRTT - RTT]$
  - $RTO = SRTT + 4 * \delta$

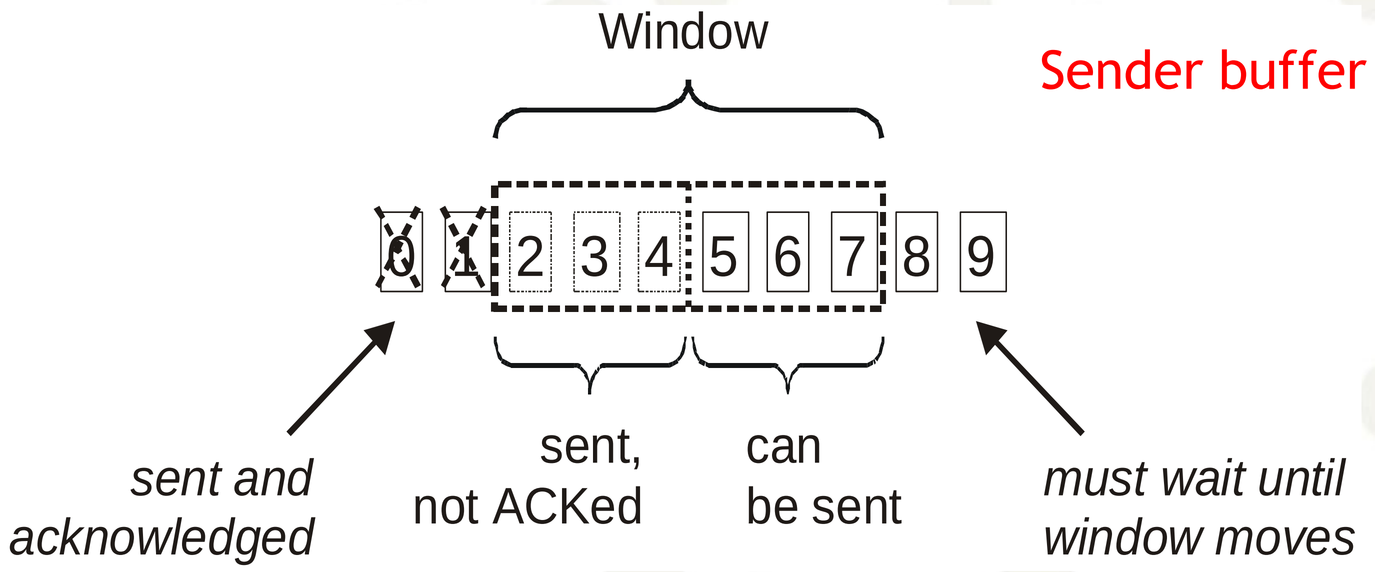
RTO calculation

- Problem: retransmission ambiguity
  - Segment #1 sent, no ACK received → segment #1 retransmitted
  - Incoming ACK #2: cannot distinguish whether original or retransmitted segment #1 was ACKed
  - Thus, cannot reliably calculate RTO!

- Solution 1 [Karn/Partridge]: ignore RTT values from retransmits
  - Problem: RTT calculation especially important when loss occurs; sampling theorem suggests that RTT samples should be taken more often

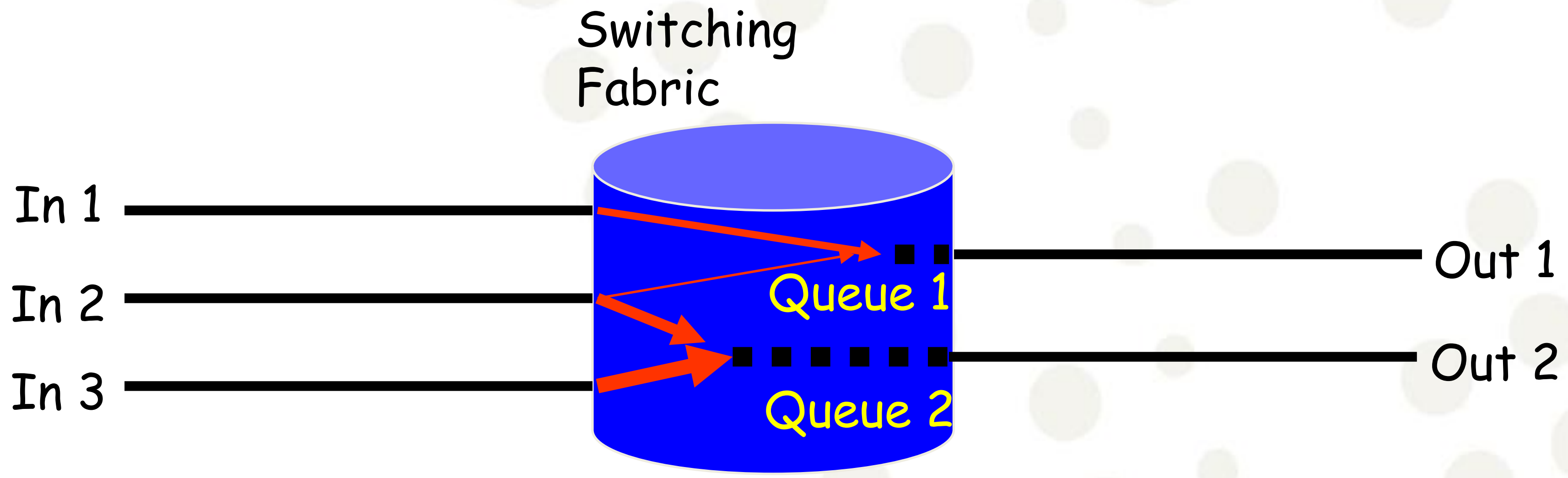
- Solution 2: Timestamps option
  - Sender writes current time into packet header (option)
  - Receiver reflects value
  - At sender, when ACK arrives, RTT = (current time) - (value carried in option)
  - Problems: additional header space; facilitates NAT detection

Window management



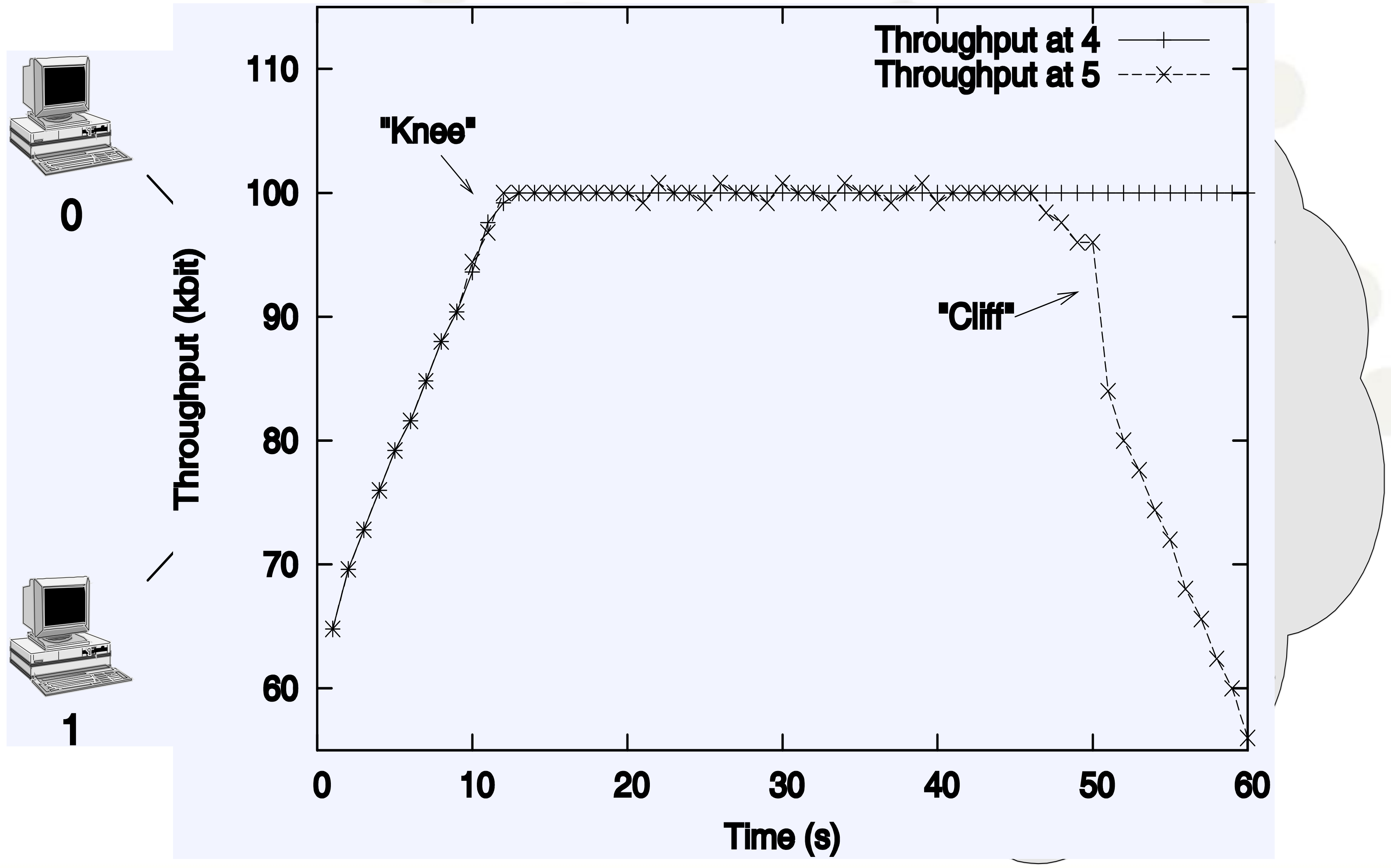
- Receiver “grants“ credit (receiver window, rwnd)
  - sender restricts sent data with window
- Receiver buffer not specified
  - i.e. receiver may buffer reordered segments (i.e. with gaps)

A simple router model



- Switch(ing) fabric forwards a packet (dest. addr.)
  if no special treatment necessary: “fast path“ (hardware)

- Queues grow when traffic bursts arrive
  - low delay = small queues, low jitter = no queue fluctuations
- Packets are dropped when queues overflow (“DropTail queueing“)
  - low loss ratio = small queues


0
1
Throughput (kbit)
Time (s)
Throughput at 4
Throughput at 5
"Knee"
"Cliff"
110
100
90
80
70
60
0
10
20
30
40
50
60

Global congestion collapse

Craig Partridge, Research Director for the Internet Research Department at BBN Technologies:

Bits of the network would fade in and out, but usually only for TCP. You could ping. You could get a UDP packet through. Telnet and FTP would fail after a while. And it depended on where you were going (some hosts were just fine, others flaky) and time of day (I did a lot of work on weekends in the late 1980s and the network was wonderfully free then).

Around 1pm was bad (I was on the East Coast of the US and you could tell when those pesky folks on the West Coast decided to start work...).

Another experience was that things broke in unexpected ways - we spent a lot of time making sure applications were bullet-proof against failures. (..)

Finally, I remember being startled when Van Jacobson first described how truly awful network performance was in parts of the Berkeley campus. It was far worse than I was generally seeing. In some sense, I felt we were lucky that the really bad stuff hit just where Van was there to see it.

Internet congestion control: history

- **around 1986:** first congestion collapse
- **1988:** "Congestion Avoidance and Control" (Jacobson)
  Combined congestion/flow control for TCP
  (also: variation change to RTO calculation algorithm)

- Idea: packet loss = congestion, so throttle the rate; increase otherwise

- Goal: stability - in equilibrum, no packet is sent into the network until an old packet leaves
  - ack clocking, "conservation of packets" principle
  - made possible through window based stop+go - behaviour

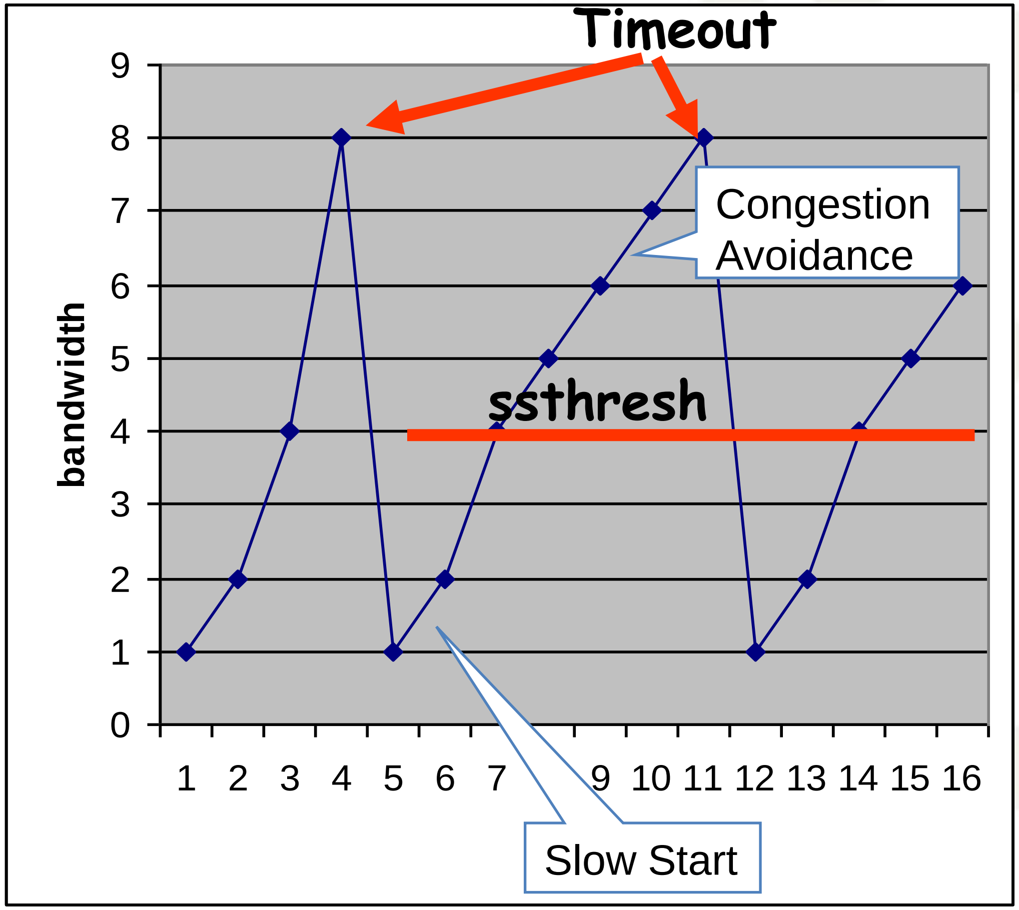
- Superposition of stable systems = stable →
  network based on TCP with congestion control = stable

TCP Congestion Control: Tahoe

- Distinguish:
  - flow control: protect receiver against overload
    (receiver "grants" a certain amount of data ("receiver window" (rwnd)))
  - congestion control: protect network against overload
    ("congestion window" (cwnd) limits the rate: min(cwnd,rwnd) used!)

- Flow/Congestion Control combined in TCP. Two basic algorithms:
  - Slow Start: for each ack received, increase cwnd by 1 packet (exponential growth) until cwnd >= ssthresh
  - Congestion Avoidance: each RTT, increase cwnd by at most 1 packet (linear growth - "additive increase“)

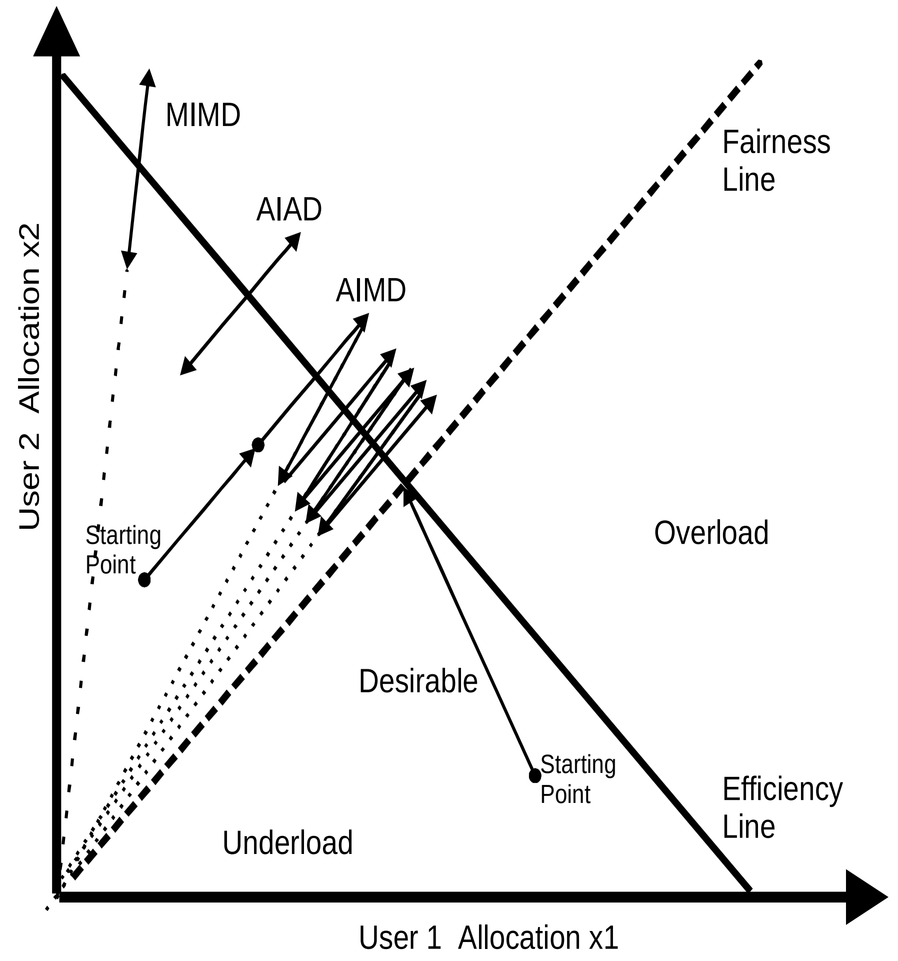
TCP Congestion Control: Tahoe /2



- If a packet or ack is lost (timeout), set cwnd = 1, ssthresh = cwnd / 2 (“multiplicative decrease") - exponential backoff

- *Actually, “Flightsize/2“ instead of cwnd/2 because cwnd might not always be fully used*

Background: AIMD

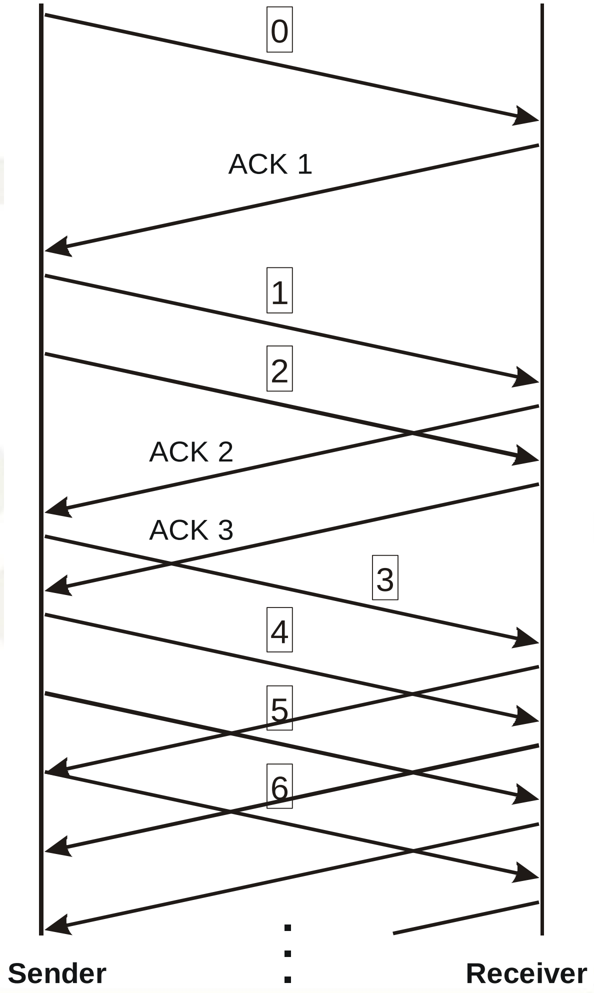


Connection startup

- Slow start: 3 RTTs for 3 packets = inefficient for very short transfers

- Example: HTTP Requests

- Thus, initial window
  IW = min(4*MSS, max(2*MSS, 4380 byte))
  - why these values?
  - worked well a long time ago; increasing them is being discussed right now (March 2010)



Fast Retransmit / Fast Recovery (Reno)

Reasoning: slow start = restart; assume that network is empty

But even similar incoming ACKs indicate that packets arrive at the receiver!

Thus, slow start reaction = too conservative.

1. Upon reception of third duplicate ACK (DupACK): ssthresh = FlightSize/2

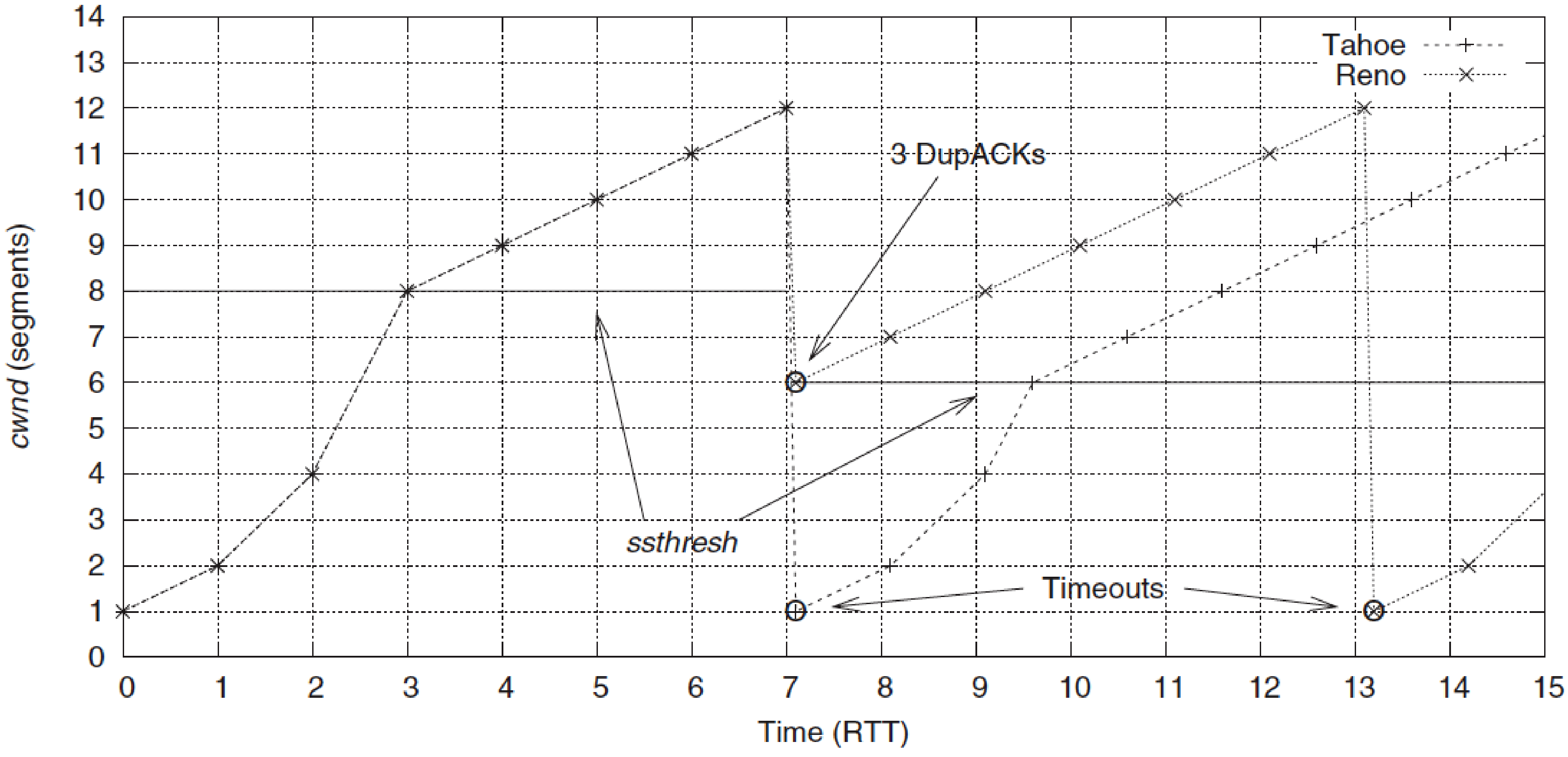
2. Retransmit lost segment (fast retransmit);
   cwnd = ssthresh + 3*SMSS
   ("inflates" cwnd by the number of segments (three) that have left the network and which the receiver has buffered)

3. For each additional DupACK received: cwnd += SMSS
   (inflates cwnd to reflect the additional segment that has left the network)

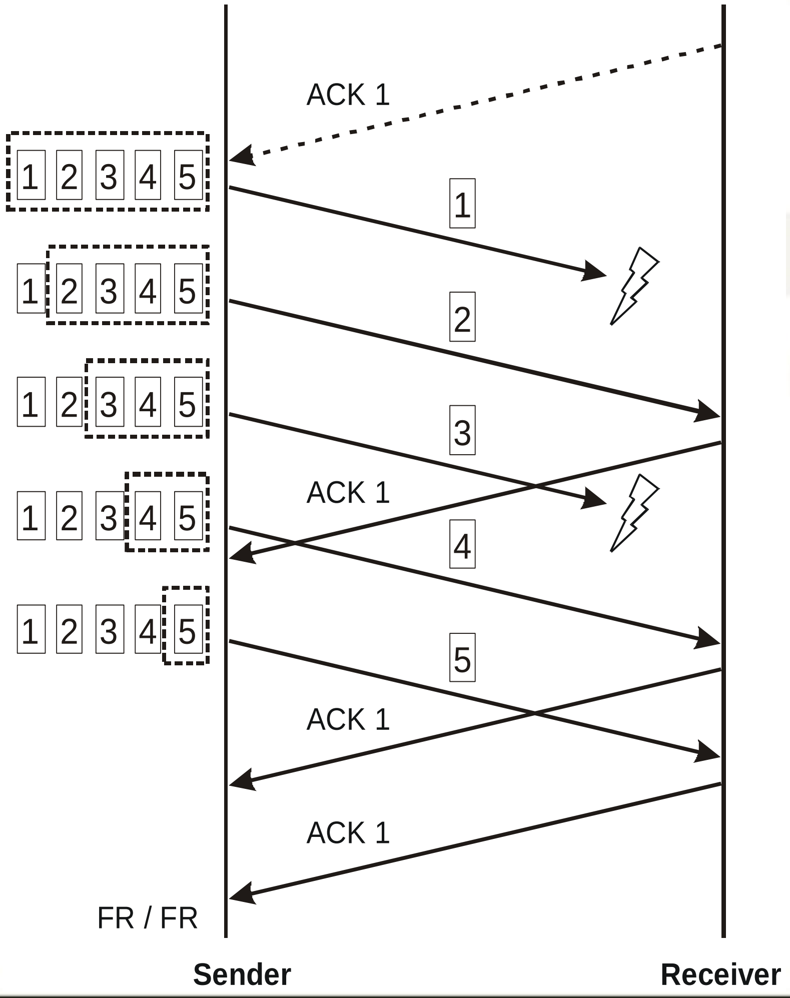
4. Transmit a segment, if allowed by the new value of cwnd and rwnd

5. Upon reception of ACK that acknowledges new data ("full ACK"):
   "deflate" window: cwnd = ssthresh (the value set in step 1)

Tahoe vs. Reno



Multiple dropped segments

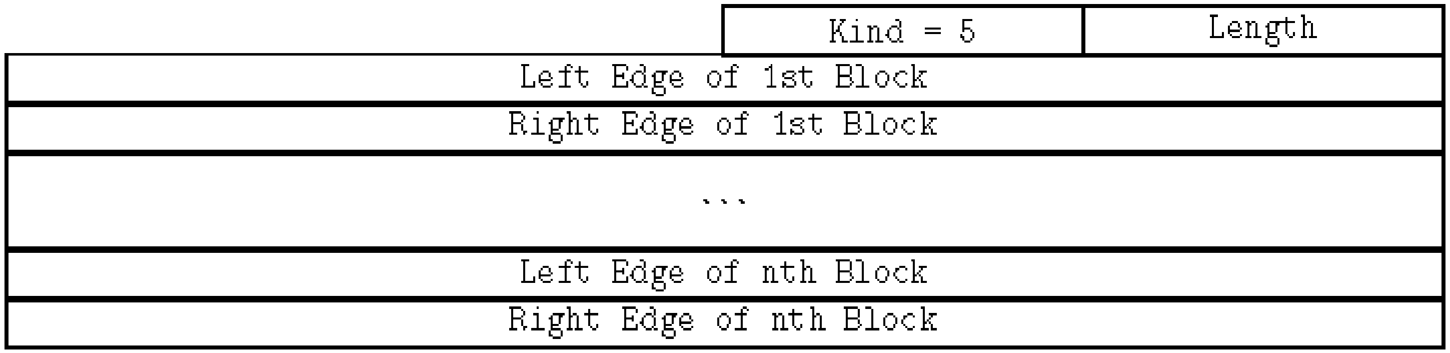


- Sender cannot detect loss of multiple segments from a single window
- Insufficient information in DupACKs
- NewReno:
  - stay in FR/FR when partial ACK arrives after DupACKs
  - retransmit single segment
  - only full ACK ends process
- Important to obtain enough ACKs to avoid timeout
  - Limited transmit: also send new segment for first two DupACKs

Non-Congestion Robustness (NCR)

- Assumption: 3 DupACKs clearly indicate loss
  - Can be incorrect when packets are reordered
- Reordering is not rare
  - And new mechanisms in the network could be devised if TCP was robust against reordering (e.g. consider splitting a flow on multiple paths)
- Approach: Increase the number of DupACKs N to approx. 1 cwnd
- Extended Limited Transmit; 2 variants
  - Careful Limited Transmit: send 1 new packet for every other DupACK until N is reached (halve sending rate, but send new data for a while)
  - Aggressive Limited Transmit: send 1 new packet for every DupACK until N is reached (delay halving sending rate)
  - Full ACK ends process

Selective ACKnowledgements (SACK)



- Example on NewReno slide: send ACK 1, SACK 3, SACK 5 in response to segment #4

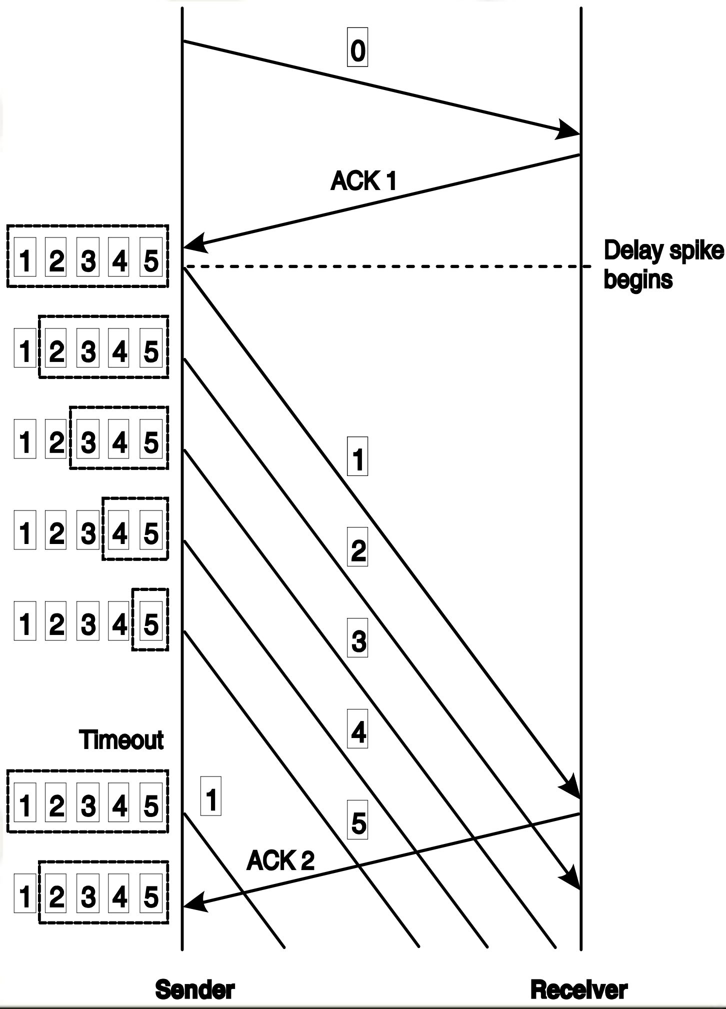
- Better sender reaction possible
  - Reno and NewReno can only retransmit a single segment per window
  - SACK can retransmit more (RFC 3517 – maintain scoreboard, pipe variable)
  - Particularly advantageous when window is large (long fat pipes)

- but: requires receiver code change

- Extension: DSACK informs the sender of duplicate arrivals

Spurious timeouts

- Common occurrence in wireless scenarios (handover): sudden delay spike
- Can lead to timeout
  → slow start
  - But: underlying assumption: “pipe empty“ is wrong! (“spurious timeout“)
  - Old incoming ACK after timeout should be used to undo the error
- Several methods proposed
  Examples:
  - Eifel Algorithm: use timestamps option to check: timestamp in ACK < time of timeout?
  - DSACK: duplicate arrived
  - F-RTO: check for ACKs that shouldn't arrive after Slow Start



Appropriate Byte Counting

- Increasing in Congestion Avoidance mode: common implementation (e.g. Jan'05 FreeBSD code): cwnd += SMSS*SMSS/cwnd for every ACK (same as cwnd += 1/cwnd if we count segments)
  - Problem: e.g. cwnd = 2: 2 + 1/2 + 1/ (2+1/2)) = 2+0.5+0.4 = 2.9
    thus, cannot send a new packet after 1 RTT
  - Worse with delayed ACKs (cwnd = 2.5)
  - Even worse with ACKs for less than 1 segment (consider 1000 1-byte ACKs)
    → too aggressive!

- Solution: Appropriate Byte Counting (ABC)
  - Maintain bytes_acked variable; send segment when threshold exceeded
  - Works in Congestion Avoidance; but what about Slow Start?
    - Here, ABC + delayed ACKs means that the rate increases in 2*SMSS steps
    - If a series of ACKs are dropped, this could be a significant burst ("micro-burstiness"); thus, limit of 2*SMSS per ACK recommended

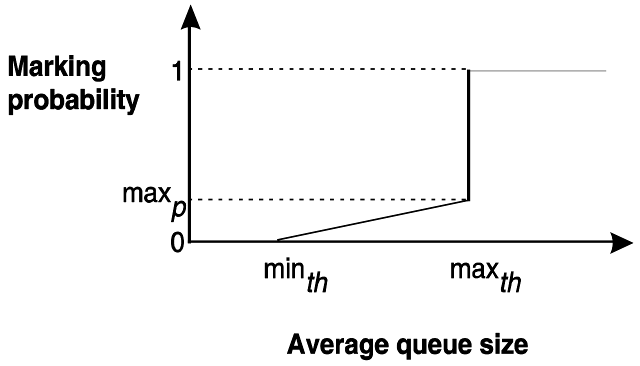
Limited Slow Start

- Slow start problems:
  - initial ssthresh = constant, not related to real network
    this is especially severe when cwnd and ssthresh are very large
    - Proposals to initially adjust ssthresh failed: must be quick <u>and</u> precise
  - Assume: cwnd and ssthresh are large, and avail.bw. = current window + 1 packet?
    - Next updates (cwnd++ for every ACK) will cause many packet drops

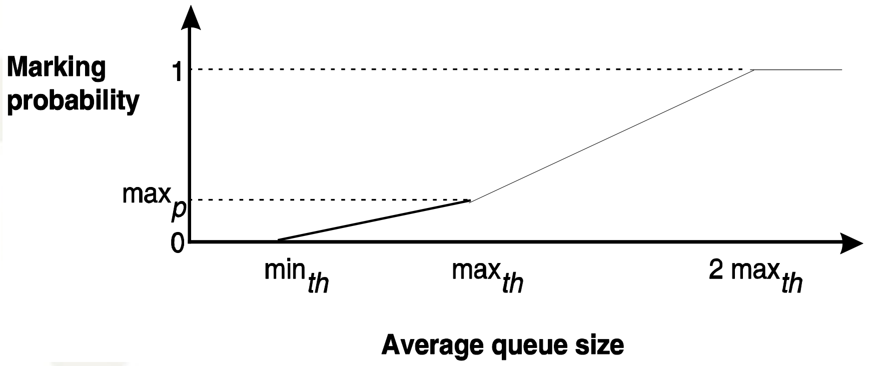
- Solution: Limited Slow Start
  - cwnd <= max_ssthresh: normal operation; recommend. max_ssthresh=100 SMSS
  - else K = int(cwnd/(0.5*max_ssthresh), cwnd += int(MSS/K)
  - More conservative than Slow Start:
    for a while cwnd+=MSS/2, then cwnd+=MSS/3, etc.

Active Queue Management

- Monitor queue, do not only drop upon overflow $\Rightarrow$ more intelligent decisions
  - Qavg = (1 - Wq) x Qavg + Qinst x Wq
    (Qavg = average occupancy, Qinst = instantaneous occupancy,
    Wq = weight - hard to tune, determines how aggressive RED behaves)
- Goals: eliminate phase effects, manage fairness
  ("punish" flows that are too aggressive)
  - Aggressive flows have more packets in the queue; thus, dropping a random one is more likely to affect such flows
  - Also possible to differentiate traffic via drop function(s)



RED



RED in "gentle" mode

Explicit Congestion Notification (ECN)

- Instead of dropping, set a bit

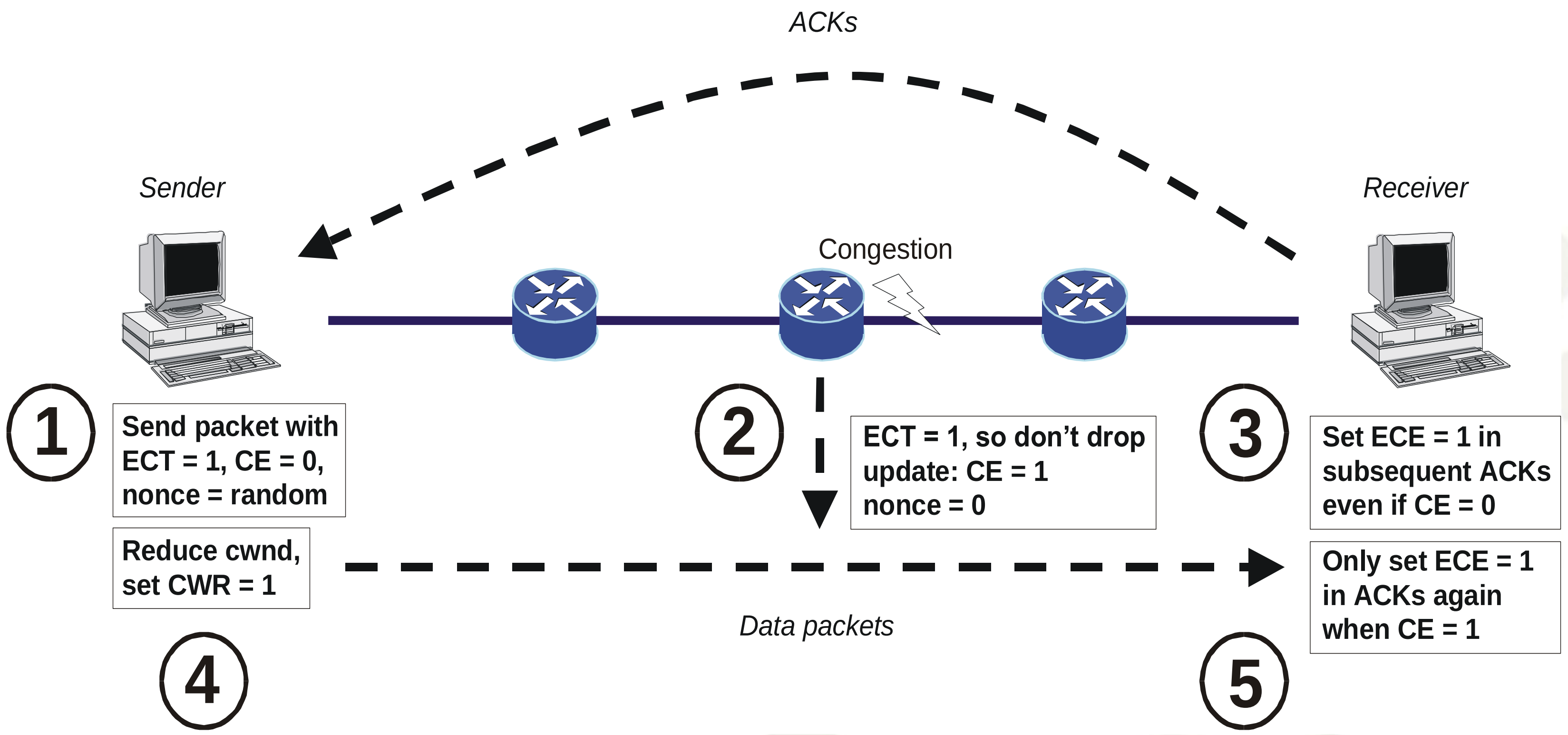
- Receiver informs sender about bit; sender behaves as if a packet was dropped
  $\Rightarrow$ actual communication between end nodes and the network

- Note: ECN = true congestion signal (i.e. clearly not corruption)

- Typical incentives:
  - sender = server; efficiently use connection, fairly distribute bandwidth
    - use ECN as it was designed
  - receiver = client; goal = high throughput, does not care about others
    - ignore ECN flag, do not inform sender about it

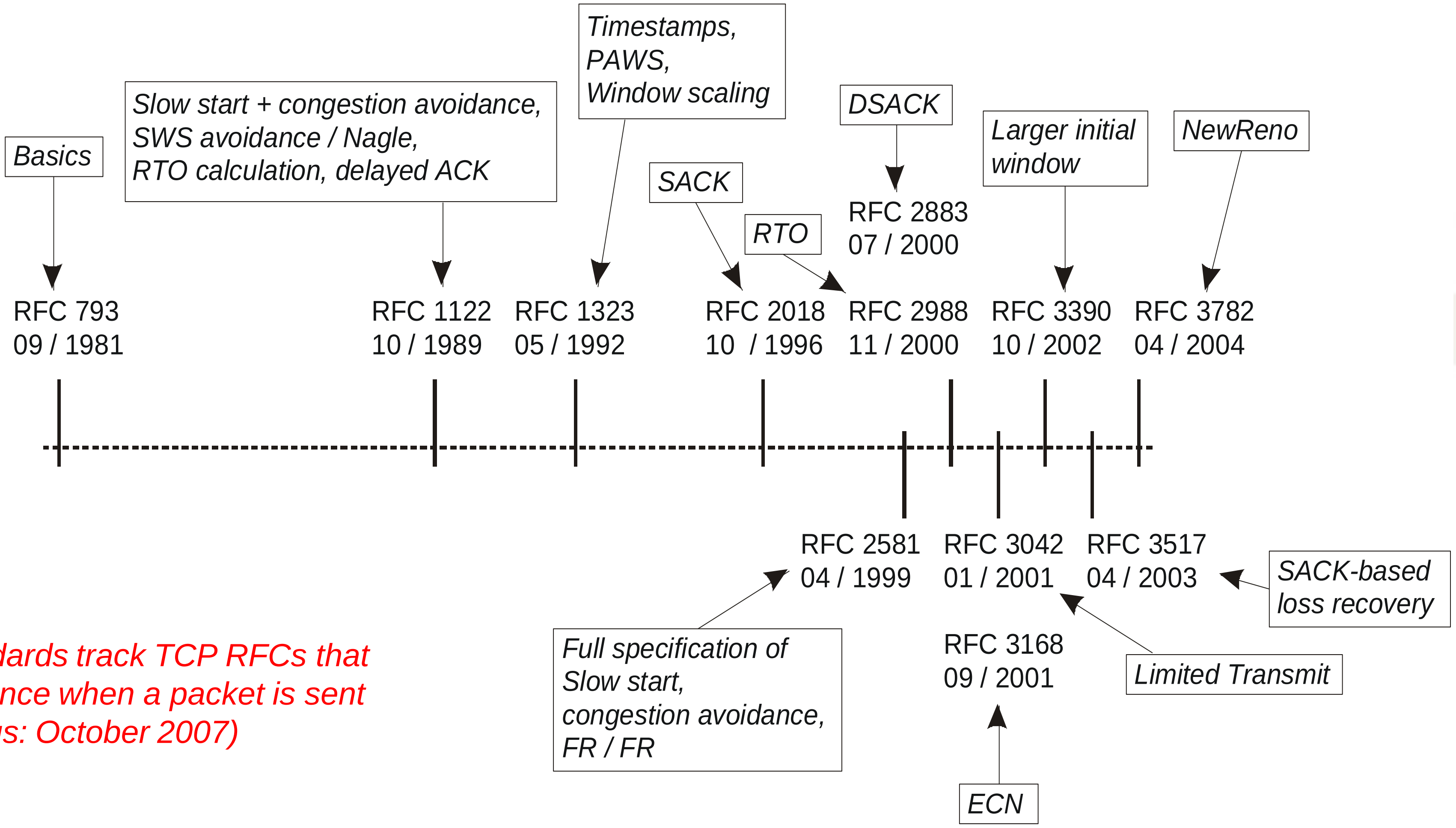
- Need to make it impossible for receiver to lie about ECN flag when it was set!
  - Solution: nonce = random number from sender, deleted by router when setting ECN
  - Sender believes „no congestion“ iff correct nonce is sent back

ECN in action


ACKs
Sender
Receiver
Congestion
1
Send packet with ECT = 1, CE = 0, nonce = random
2
ECT = 1, so don't drop update: CE = 1 nonce = 0
3
Set ECE = 1 in subsequent ACKs even if CE = 0
Reduce cwnd, set CWR = 1
Only set ECE = 1 in ACKs again when CE = 1
Data packets
4
5

- Nonce provided by bit combination:
  - ECT(0): ECT=1, CE=0;   ECT(1): ECT=0, CE=1
- Nonce usage specification still experimental

TCP History



Standards track TCP RFCs that influence when a packet is sent (status: October 2007)

References

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