KHOA CÔNG NGHỆ THÔNG TIN-ĐHKHTN
CSC11004 - MẠNG MÁY TÍNH NÂNG CAO

TCP CONGESTION CONTROL

Lê Ngọc Sơn

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KHOA CÔNG NGHỆ THÔNG TIN
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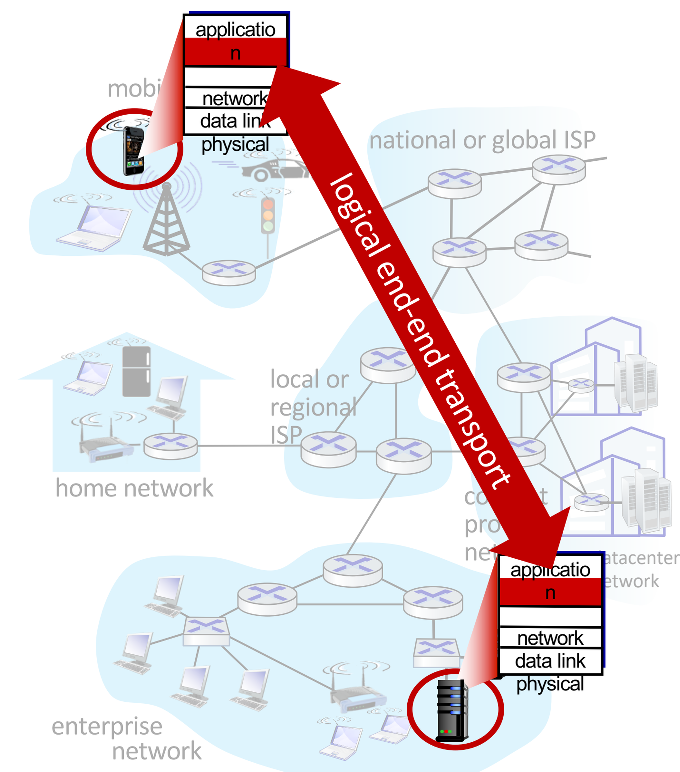
Two principal Internet transport protocols

- **TCP:** Transmission Control Protocol
  - reliable, in-order delivery
  - congestion control
  - flow control
  - connection setup
- **UDP:** User Datagram Protocol
  - unreliable, unordered delivery
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



Agenda

- ☐Transport-layer services
- ☐UDP and TCP
- ☐TCP Flow control
- ☐Principles of congestion control
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UDP: User Datagram Protocol

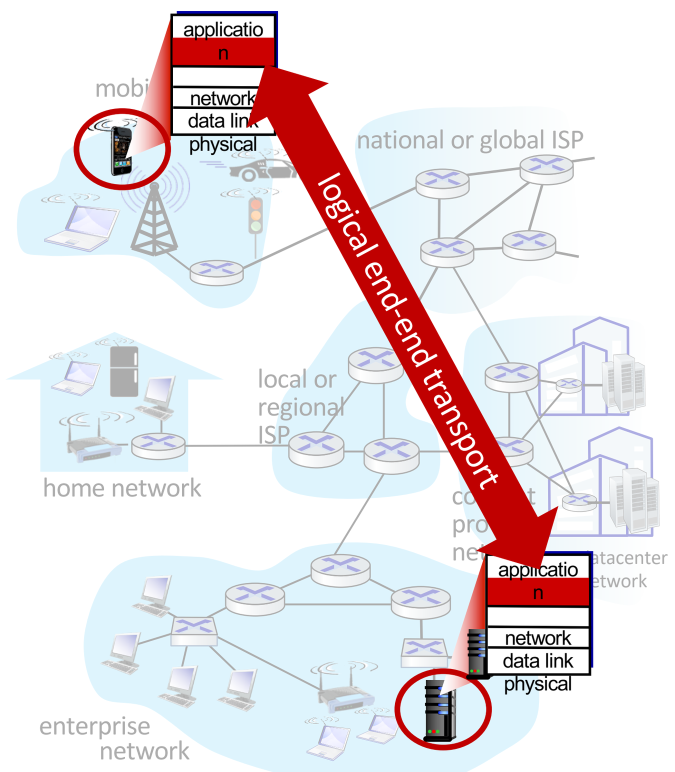
- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out-of-order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

Why is there a UDP?

- no connection establishment (which can add RTT delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control
  - UDP can blast away as fast as desired!
  - can function in the face of congestion

Transport services and protocols

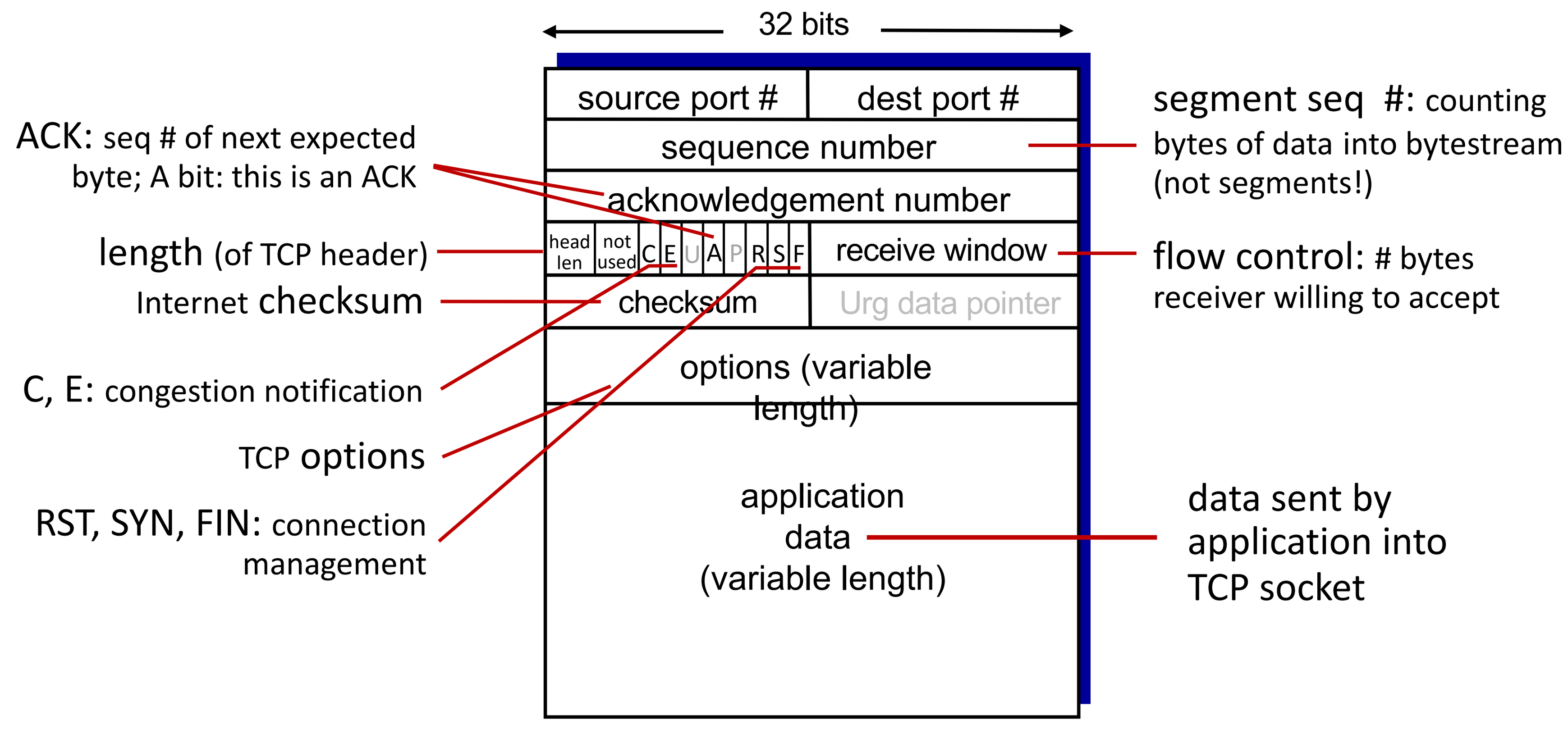
- provide *logical communication* between application processes running on different hosts
- transport protocols actions in end systems:
  - sender: breaks application messages into *segments*, passes to network layer
  - receiver: reassembles segments into messages, passes to application layer
- two transport protocols available to Internet applications
  - TCP, UDP



TCP: overview RFCs: 793,1122, 2018, 5681, 7323

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte steam:*
  - no "message boundaries"
- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- cumulative ACKs
- pipelining:
  - TCP congestion and flow control set window size
- connection-oriented:
  - handshaking (exchange of control messages) initializes sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure



TCP sequence numbers, ACKs

Sequence numbers:

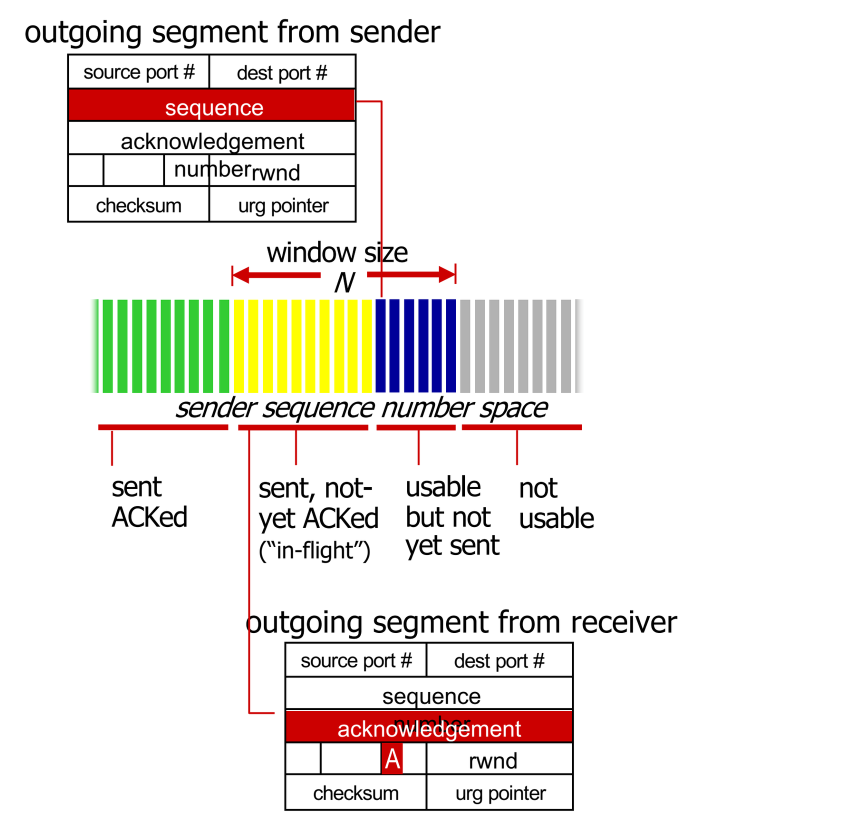
- byte stream "number" of first byte in segment's data

Acknowledgements:

- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

- A: TCP spec doesn't say, - up to implementor



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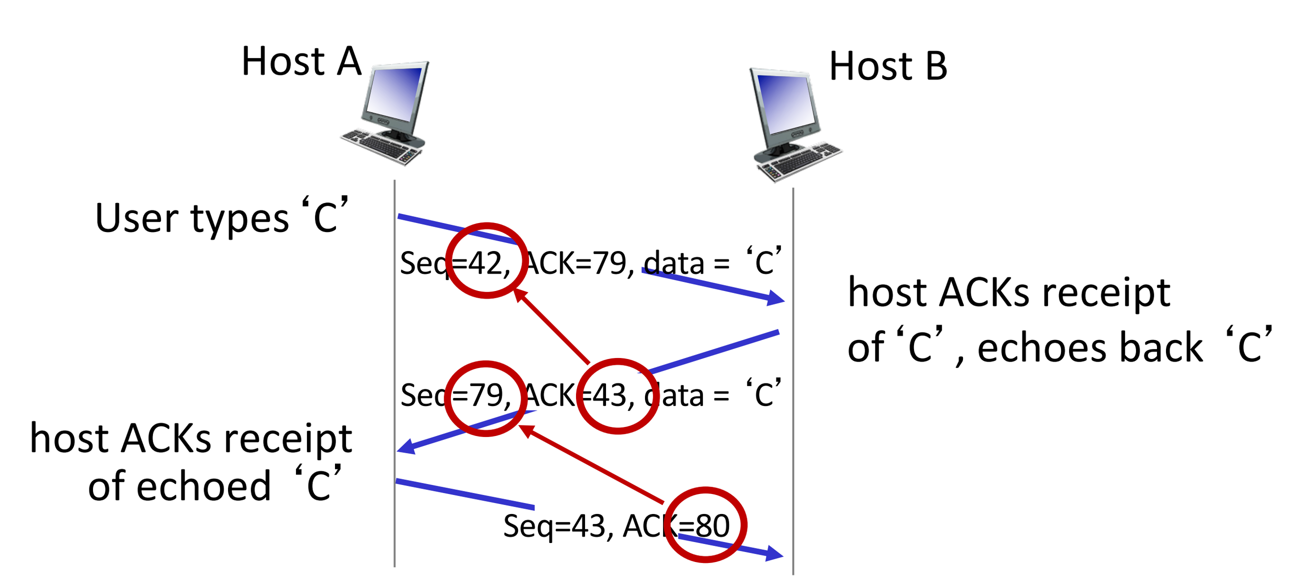


simple telnet scenario

TCP round trip time, timeout

Q: how to set TCP timeout value?

- longer than RTT, but RTT varies!
- *too short:* premature timeout, unnecessary retransmissions
- *too long:* slow reaction to segment loss

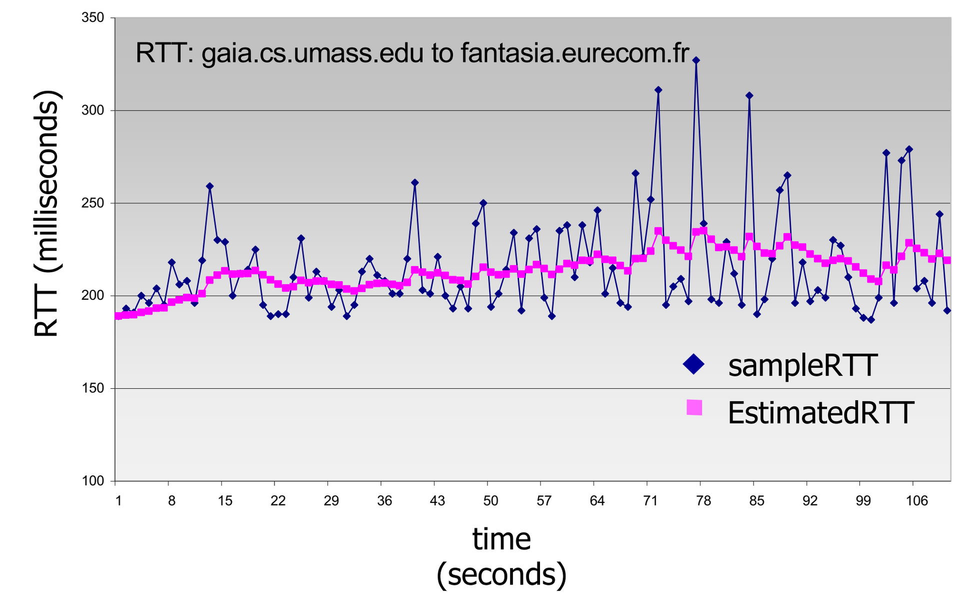
Q: how to estimate RTT?

- `SampleRTT`: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- `SampleRTT` will vary, want estimated RTT "smoother"
  - average several *recent* measurements, not just current `SampleRTT`

TCP round trip time, timeout

```
EstimatedRTT = (1- α)*EstimatedRTT + α*SampleRTT
```

- exponential weighted moving average (EWMA)
- influence of past sample decreases exponentially fast
- typical value: α = 0.125



TCP round trip time, timeout

- timeout interval: **EstimatedRTT** plus "safety margin"
  - large variation in **EstimatedRTT:** want a larger safety margin

```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```

estimated RTT "safety margin"

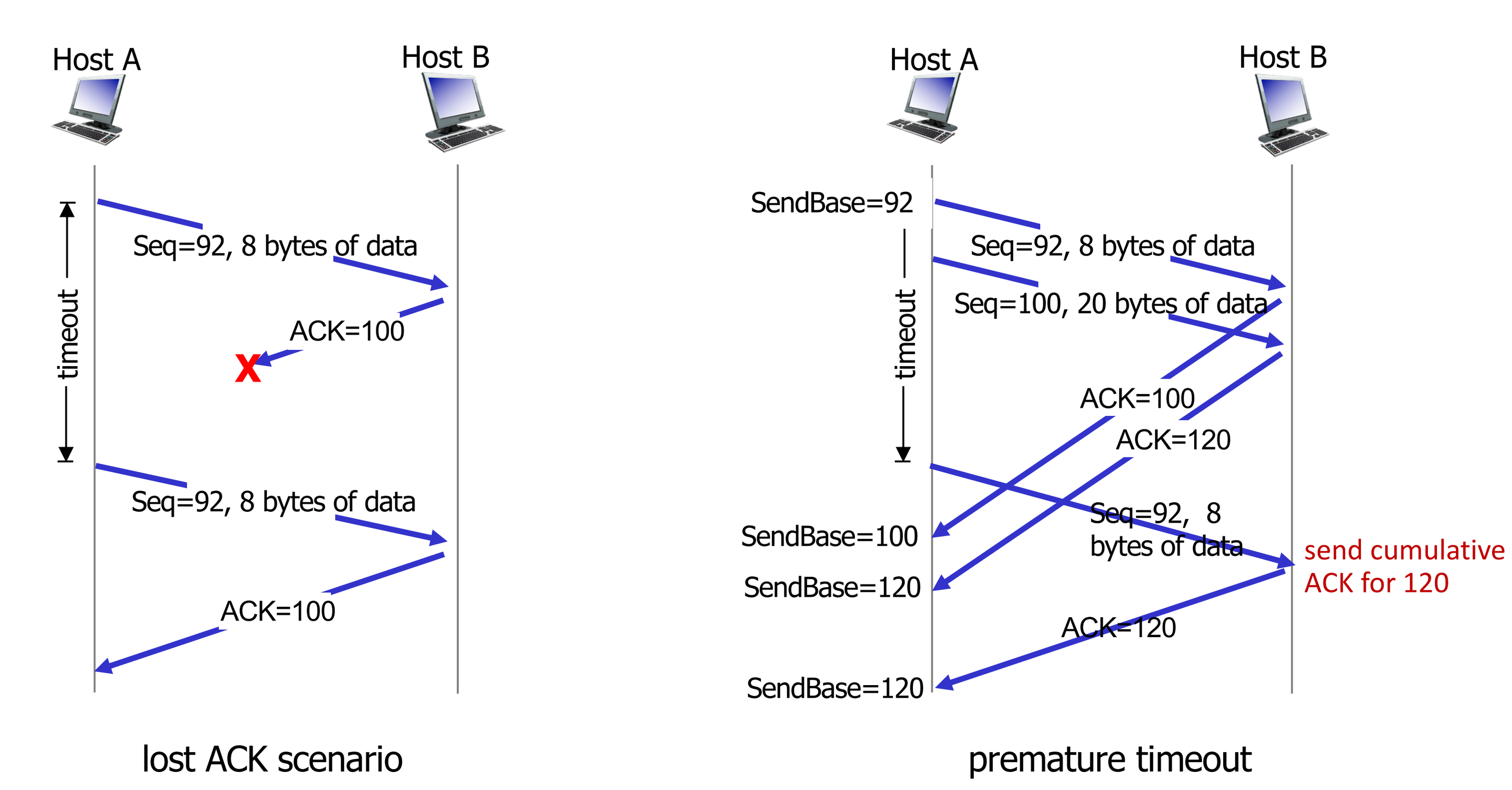
- **DevRTT**: EWMA of **SampleRTT** deviation from **EstimatedRTT**:

```
DevRTT = (1-β)*DevRTT + β*|SampleRTT-EstimatedRTT|
```

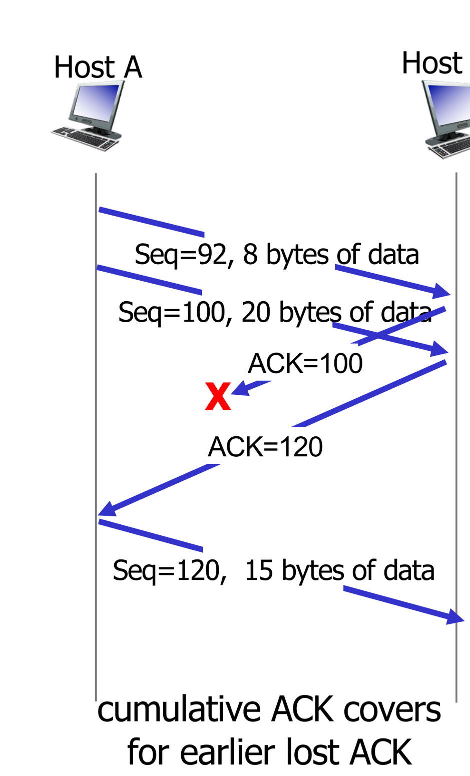
(typically, β = 0.25)

* Check out the online interactive exercises for more examples: http://gaia.cs.umass.edu/kurose_ross/interactive/

TCP: retransmission scenarios



TCP: retransmission scenarios



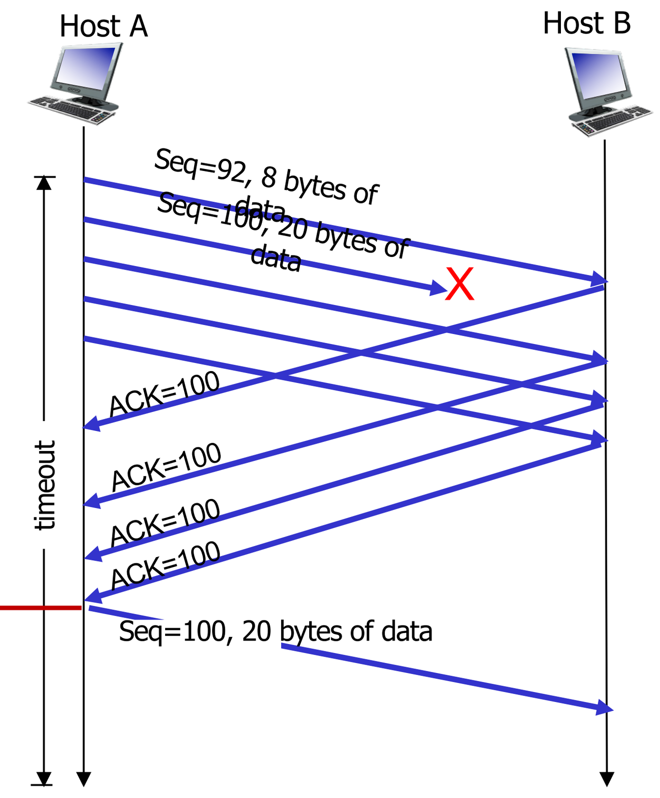
TCP fast retransmit

TCP fast retransmit

if sender receives 3 additional ACKs for same data ("triple duplicate ACKs"), resend unACKed segment with smallest seq #

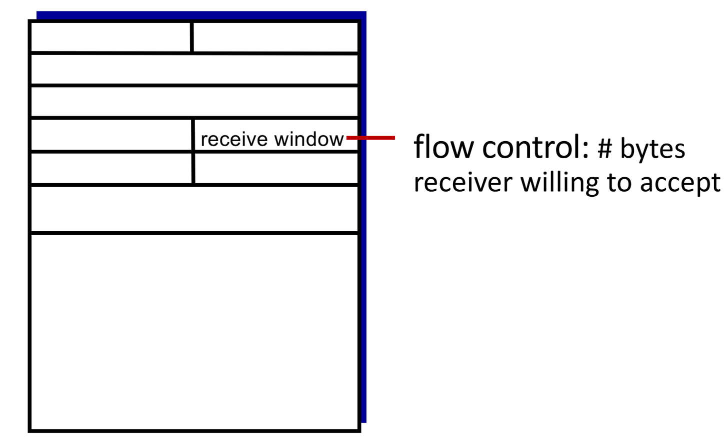
- likely that unACKed segment lost, so don't wait for timeout

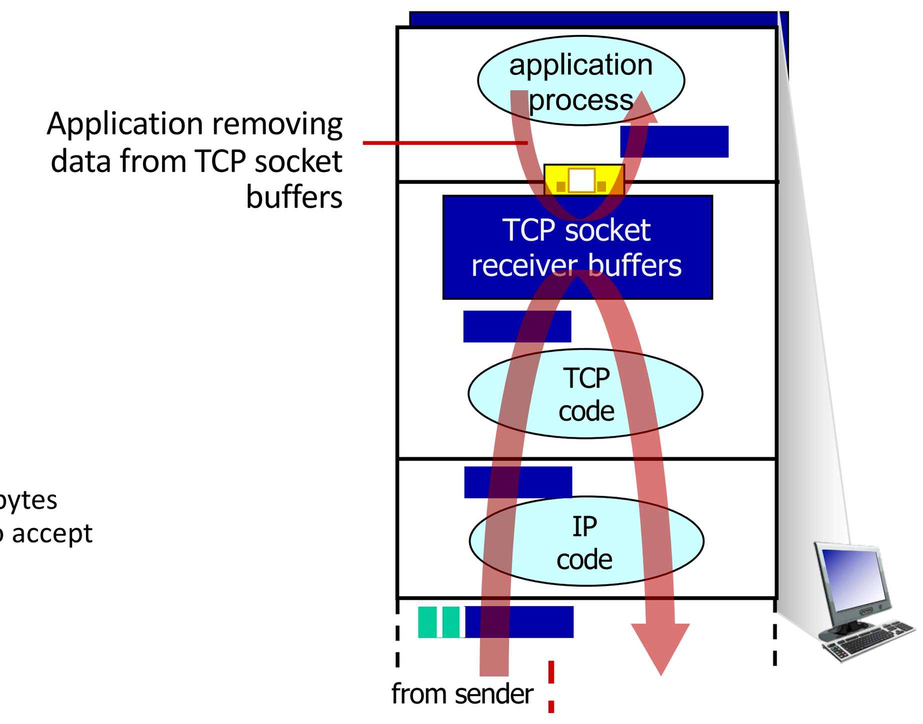
Receipt of three duplicate ACKs indicates 3 segments received after a missing segment – lost segment is likely. So retransmit!



TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?





receiver protocol stack

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TCP flow control

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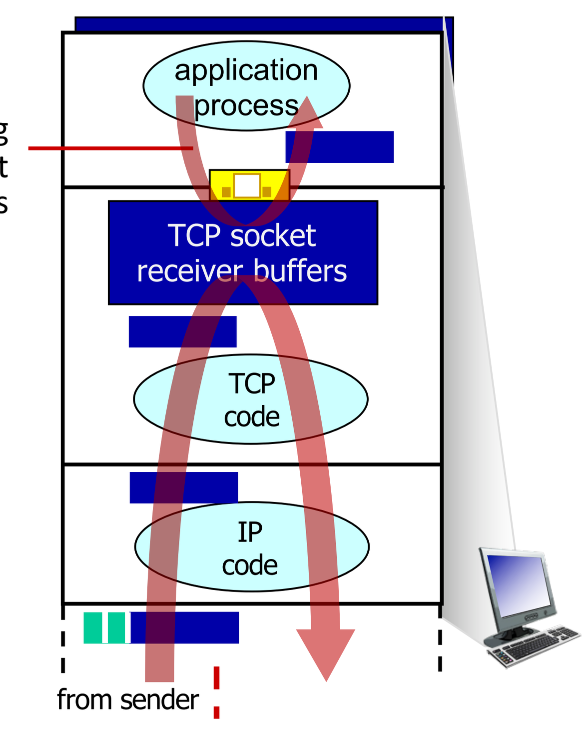
flow control

receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast



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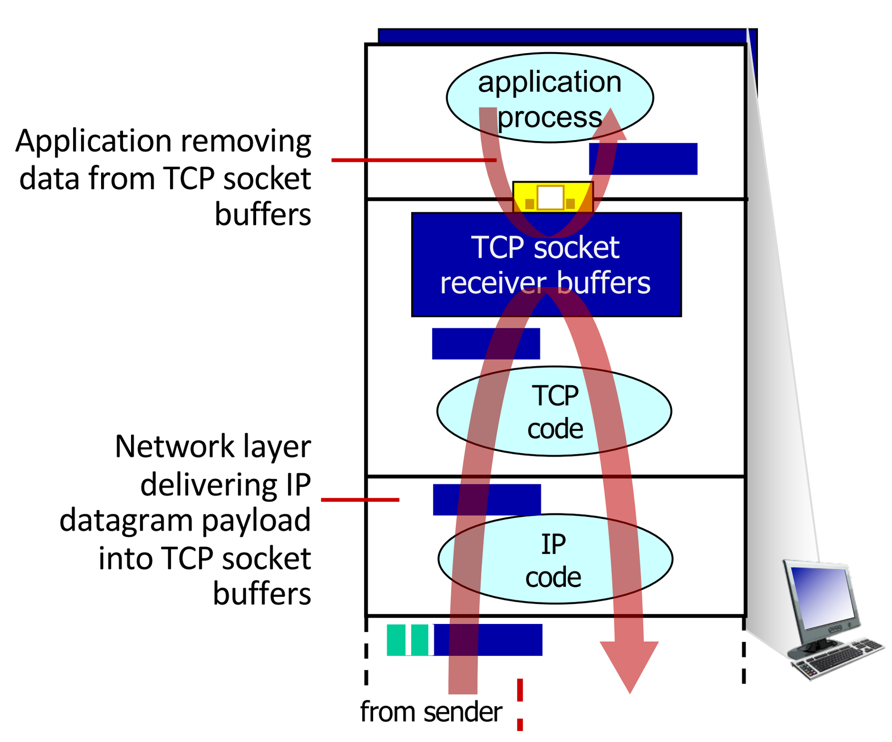
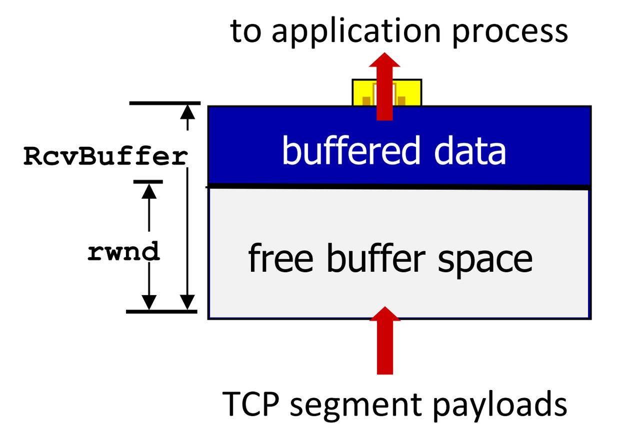


receiver protocol stack

TCP flow control

- TCP receiver "advertises" free buffer space in **rwnd** field in TCP header
  - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **RcvBuffer**
- sender limits amount of unACKed ("in-flight") data to received **rwnd**
- guarantees receive buffer will not overflow



TCP receiver-side buffering

TCP flow control

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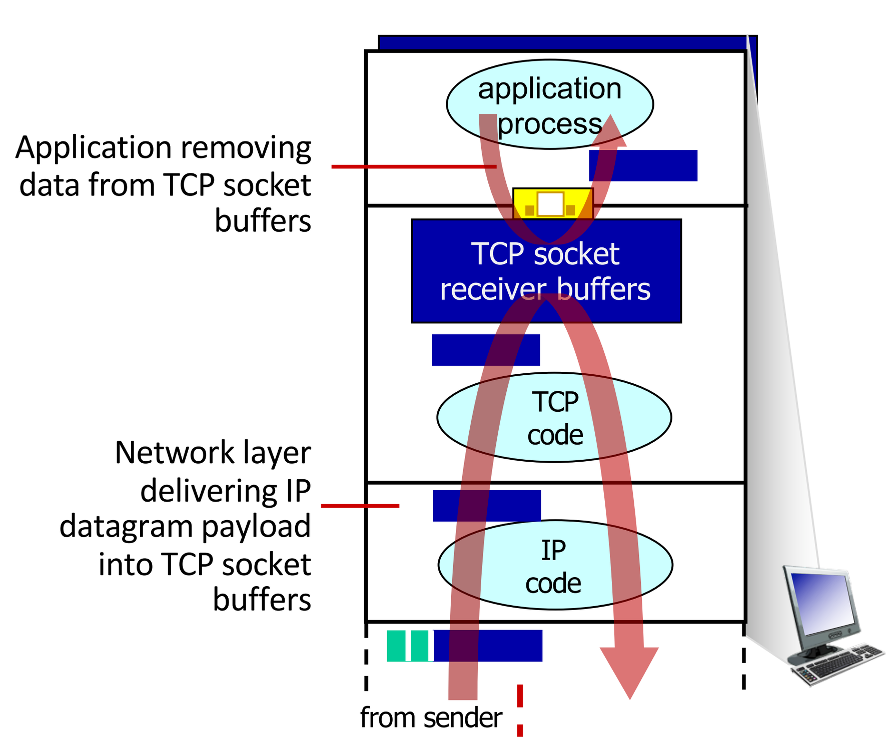

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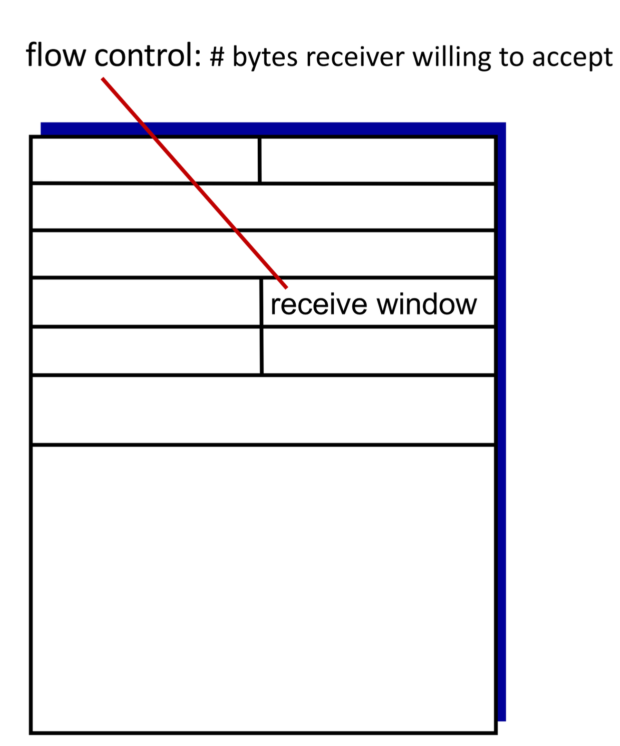

TCP segment format

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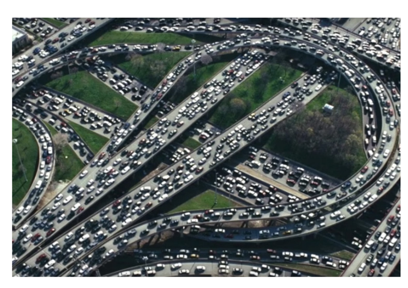
Principles of congestion control

Congestion:

- informally: "too many sources sending too much data too fast for *network* to handle"
- manifestations:
  - long delays (queueing in router buffers)
  - packet loss (buffer overflow at routers)
- different from flow control!
- a top-10 problem!



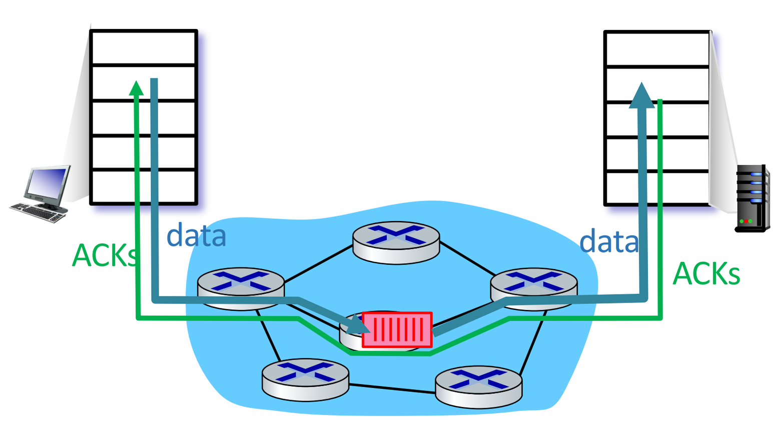
congestion control: too many senders, sending too fast

flow control: one sender too fast for one receiver

Approaches towards congestion control

End-end congestion control:

- ☐no explicit feedback from network
- ☐congestion *inferred* from observed loss, delay
- approach taken by TCP

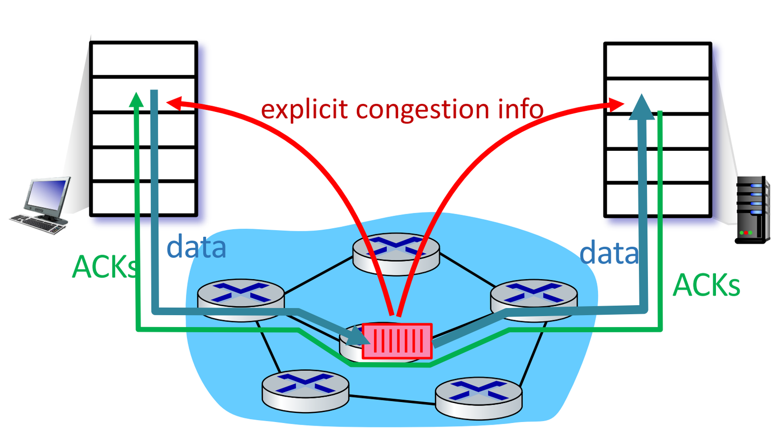


Approaches towards congestion control

Network-assisted congestion control:

- routers provide *direct* feedback to sending/receiving hosts with flows passing through congested router
- may indicate congestion level or explicitly set sending rate

☐TCP ECN, ATM, DECbit protocols



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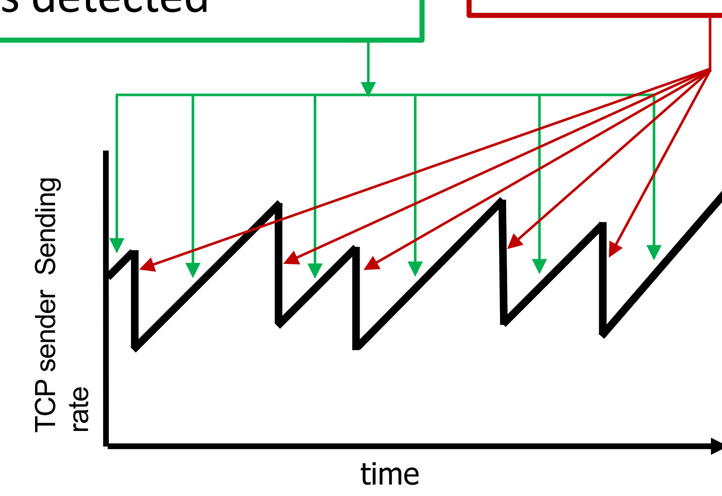
TCP congestion control: AIMD

- *approach:* senders can increase sending rate until packet loss (congestion) occurs, then decrease sending rate on loss event

Additive Increase: increase sending rate by 1 maximum segment size every RTT until loss detected

Multiplicative Decrease: cut sending rate in half at each loss event



AIMD sawtooth behavior: *probing* for bandwidth

TCP AIMD: more

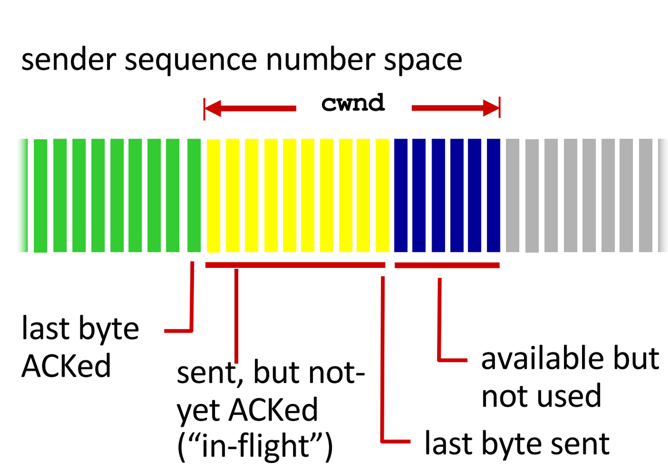
Multiplicative decrease detail: sending rate is

- Cut in half on loss detected by triple duplicate ACK (TCP Reno)
- Cut to 1 MSS (maximum segment size) when loss detected by timeout (TCP Tahoe)

Why AIMD?

- AIMD – a distributed, asynchronous algorithm – has been shown to:
  - optimize congested flow rates network wide!
  - have desirable stability properties

TCP congestion control: details



TCP sending behavior:

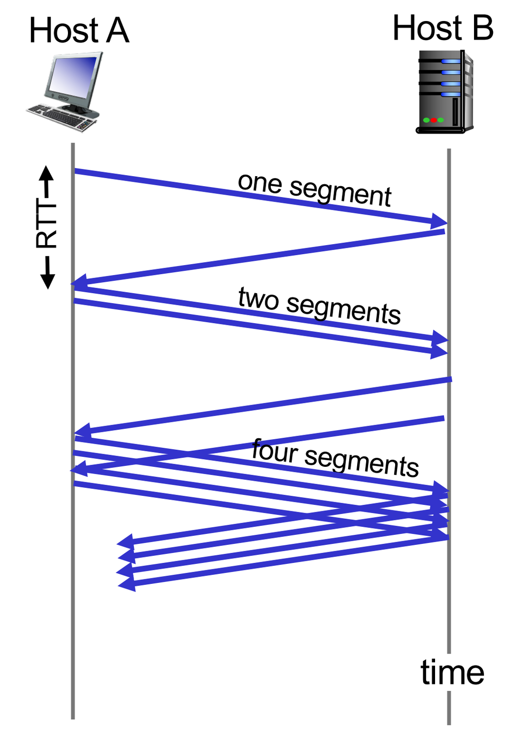
- *roughly:* send cwnd bytes, wait RTT for ACKS, then send more bytes

$$\text{TCP rate} \approx \frac{\text{cwnd}}{\text{RTT}} \text{ bytes/sec}$$

- TCP sender limits transmission: `LastByteSent- LastByteAcked ≤ cwnd`
- `cwnd` is dynamically adjusted in response to observed network congestion (implementing TCP congestion control)

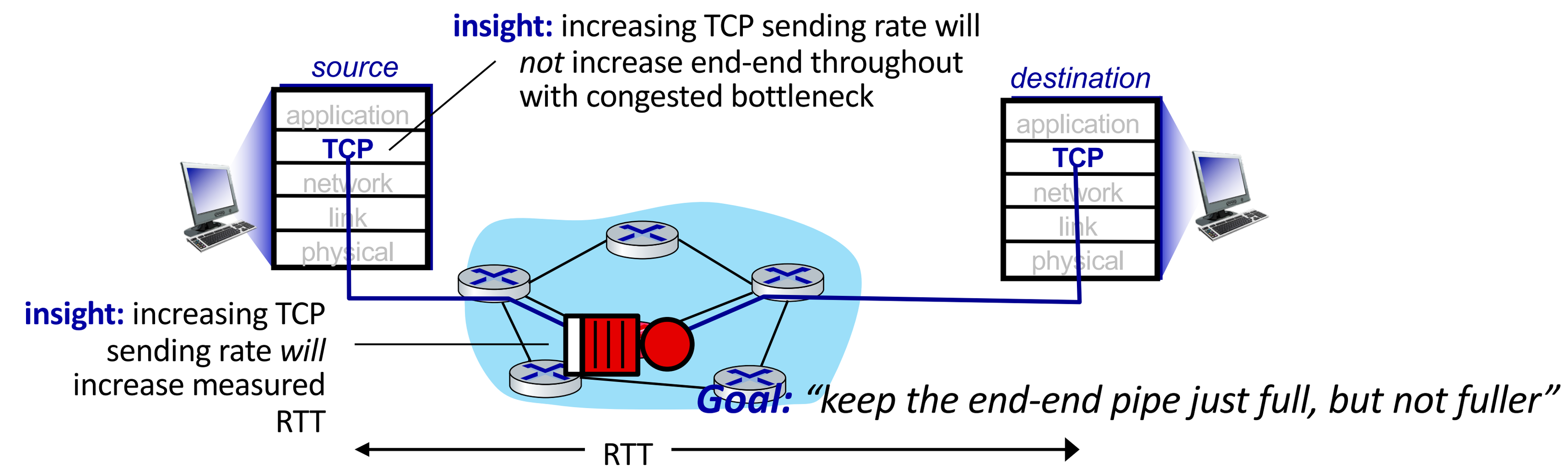
TCP slow start

- when connection begins, increase rate exponentially until first loss event:
  - initially **cwnd** = 1 MSS
  - double **cwnd** every RTT
  - done by incrementing **cwnd** for every ACK received
- *summary:* initial rate is slow, but ramps up exponentially fast



TCP and the congested "bottleneck link"

- TCP increase TCP's sending rate until packet loss occurs at some router's output: the *bottleneck link*
- understanding congestion: useful to focus on congested bottleneck link



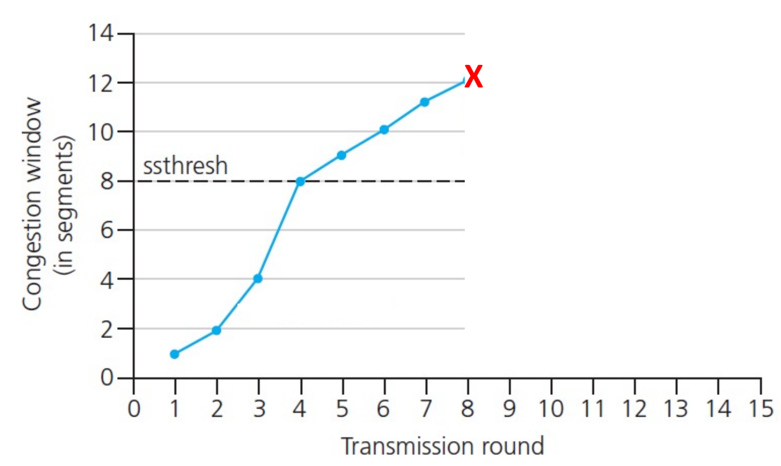
TCP: from slow start to congestion avoidanc

Q: when should the exponential increase switch to linear?

A: when **cwnd** gets to 1/2 of its value before timeout.

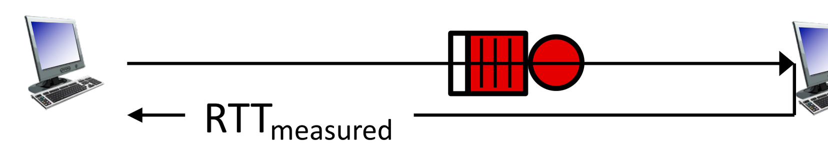
Implementation:

- variable **ssthresh**
- on loss event, **ssthresh** is set to 1/2 of **cwnd** just before loss event



Delay-based TCP congestion control

Keeping sender-to-receiver pipe "just full enough, but no fuller": keep bottleneck link busy transmitting, but avoid high delays/buffering



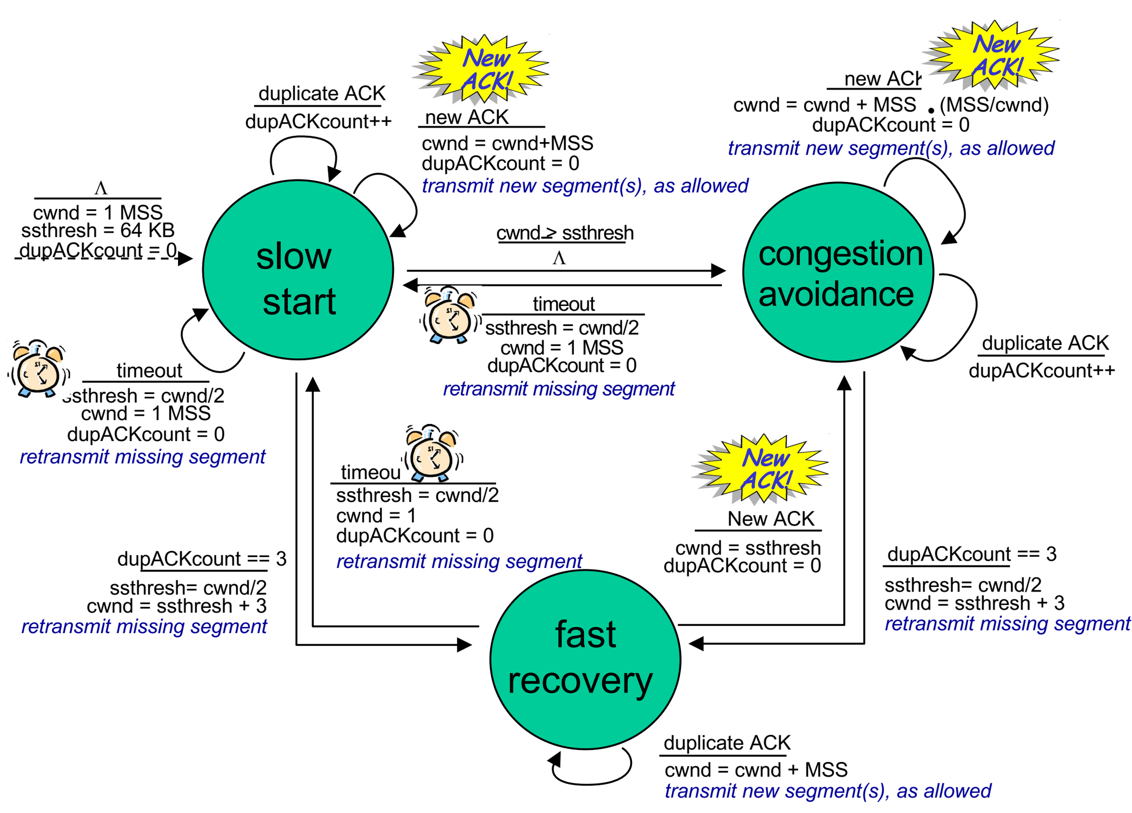
$$\text{measured throughput} = \frac{\text{\# bytes sent in last RTT interval}}{RTT_{measured}}$$

Delay-based approach:

- RTT_{min} - minimum observed RTT (uncongested path)
- uncongested throughput with congestion window `cwnd` is cwnd/RTT_{min}

if measured throughput "very close" to uncongested throughput
 increase `cwnd` linearly /* since path not congested */
else if measured throughput "far below" uncongested throughout
 decrease `cwnd` linearly /* since path is congested */

Summary: TCP congestion control



Delay-based TCP congestion control

- congestion control without inducing/forcing loss
- maximizing throughout ("keeping the just pipe full… ") while keeping delay low ("…but not fuller")
- a number of deployed TCPs take a delay-based approach
  - BBR deployed on Google's (internal) backbone network

TCP and the congested "bottleneck link"

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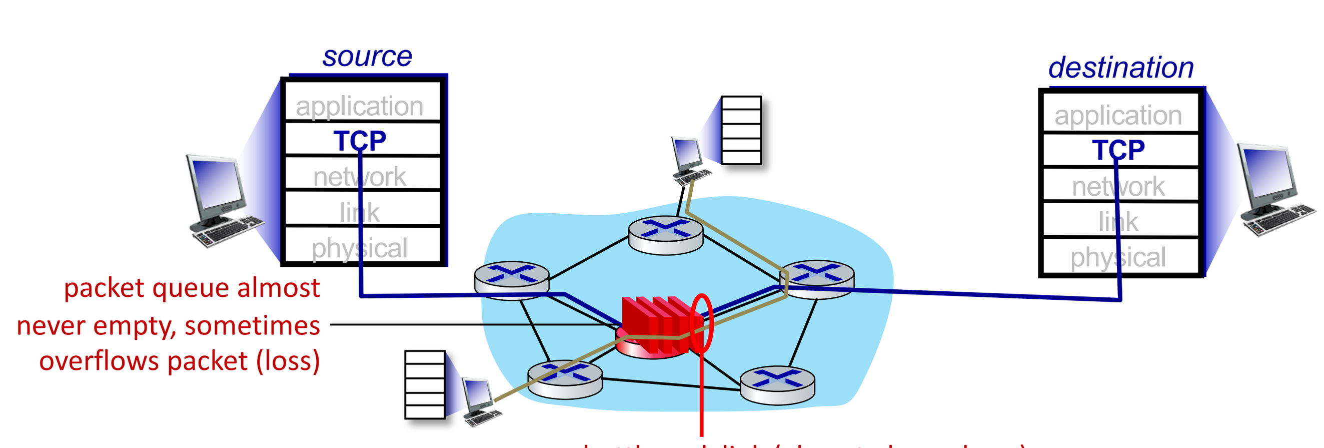
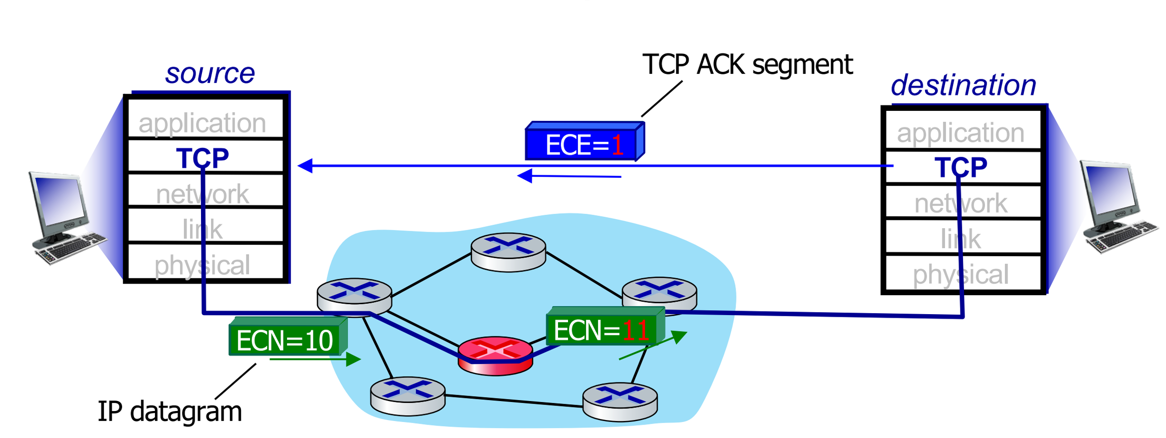


Explicit congestion notification (ECN)

TCP deployments often implement *network-assisted* congestion control:

- two bits in IP header (ToS field) marked *by network router* to indicate congestion
  - *policy* to determine marking chosen by network operator
- congestion indication carried to destination
- destination sets ECE bit on ACK segment to notify sender of congestion
- involves both IP (IP header ECN bit marking) and TCP (TCP header C,E bit marking)



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TCP Congestion Control Today

TCP BIC	TCP CUBIC
• optimize long fat networks • Binary Increase Congestion control • used by default in Linux kernels 2.6.8 through 2.6.18	• Improvement of TCP BIC • used by default in Linux kernels between versions 2.6.19 and 3.2. • MacOS adopted CUBIC by at least the OS X Yosemite release in 2014 • Microsoft adopted it by default in Windows 10.1709 Fall Creators Update (2017), and Windows Server 2016 1709 update

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Other variants

Variant	Feedback	Required changes	Benefits
(New) Reno	Loss	—	—
Vegas	Delay	Sender	Less loss
High Speed	Loss	Sender	High bandwidth
BIC	Loss	Sender	High bandwidth
CUBIC	Loss	Sender	High bandwidth
C2TCP[9][10]	Loss/Delay	Sender	Ultra-low latency and high bandwidth
NATCP[11]	Multi-bit signal	Sender	Near Optimal Performance
Elastic-TCP	Loss/Delay	Sender	High bandwidth/short & long-distance
Agile-TCP	Loss	Sender	High bandwidth/short-distance
H-TCP	Loss	Sender	High bandwidth
FAST	Delay	Sender	High bandwidth
Compound TCP	Loss/Delay	Sender	High bandwidth
Westwood	Loss/Delay	Sender	L
Jersey	Loss/Delay	Sender	L
BBR[12]	Delay	Sender	BLVC, Bufferbloat
CLAMP	Multi-bit signal	Receiver, Router	V
TFRC	Loss	Sender, Receiver	No Retransmission
XCP	Multi-bit signal	Sender, Receiver, Router	BLFC
VCP	2-bit signal	Sender, Receiver, Router	BLF
MaxNet	Multi-bit signal	Sender, Receiver, Router	BLFSC
JetMax	Multi-bit signal	Sender, Receiver, Router	High bandwidth
RED	Loss	Router	Reduced delay
ECN	Single-bit signal	Sender, Receiver, Router	Reduced loss

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Question?