

Goal: predict the RTT      Lecture 17

EWMA: Exponentially weighted Moving Average



Sample average:  $\frac{\sum_{i=1}^n S_i}{n}$

Moving Average:  $\frac{\sum_{i=1}^w S_i}{n}$        $w$  is the size of window

EWMA:  $S_{av}^n = \alpha S_{av}^{n-1} + (1-\alpha) S_n$        $0 < \alpha < 1$

$\uparrow$  history                       $\downarrow$  new sample

$\rightarrow S_{av}^1 = \alpha \cdot S_{av}^I + (1-\alpha) S_1$        $S_{av}^I$ : initial value of the history

$S_{av}^2 = \alpha S_{av}^1 + (1-\alpha) S_2$

$= \alpha [\alpha S_{av}^I + (1-\alpha) S_1] + (1-\alpha) S_2$

$= \alpha^2 S_{av}^I + \alpha(1-\alpha) S_1 + (1-\alpha) S_2$

$S_{av}^3 = \alpha^3 S_{av}^I + \alpha^2(1-\alpha) S_1 + \alpha(1-\alpha) S_2 + (1-\alpha) S_3$

$S_{av}^n = \alpha^n S_{av}^I + \alpha^{n-1}(1-\alpha) S_1 + \dots$

$$0 < \alpha < 1$$

↓  
Contribution of  $S_1$  is  
decaying exponential as  
we get more samples

$$\underline{EWMA} : \alpha^n (1-\alpha) S_1 + \alpha^{n-1} (1-\alpha) S_2 + \dots + (1-\alpha) S_n$$

i) We have samples of RTT

{ timestamp segments

{ Find the time when the ACK is received  
→  $RTT_n$

Smoothed RTT :  $SRTT$

$$SRTT_n = (1-\alpha) SRTT_{n-1} + \alpha RTT_n$$

↗ ↖ ↘ ↙  
history ↖ current sample ↘

$\alpha$  is close to 1  $\Rightarrow$  more importance to the  
current sample

$\alpha$  is close to 0  $\Rightarrow$  more importance to the  
history

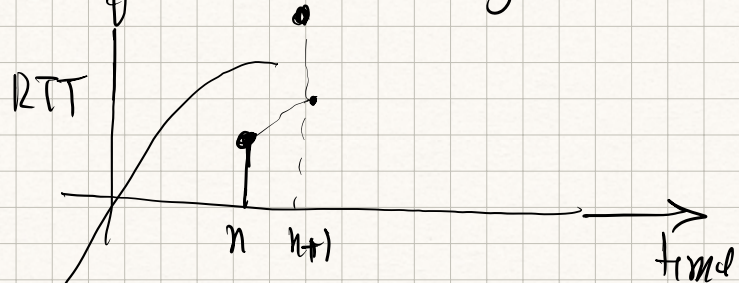
→ Whenever an RTT sample is received  
we calculate the SRTT

⇒ set the timeout proportional to SRTT

In early 1980s this was implemented but  
it did not work

SRTT does not quite track large variations

in RTT



large changes occur due to  
queuing delays

$\text{dev RTT}_n$ : deviation in the RTT  
when the  $n^{\text{th}}$  sample is  
received

$$\text{dev RTT}_n = \underbrace{(1-\beta)}_{\text{Exponential averaging}} \text{dev RTT}_{n-1} + \underbrace{\beta}_{\text{history of deviation}} (\text{RTT}_n - \text{SRTT}_n)_{\text{current sample of the denominator}}$$

Exponential  
averaging

history of deviation

current sample  
of the denominator

Timeout value:  $SRTT_n + 4 \cdot devRRTT_n$   
(RTO)

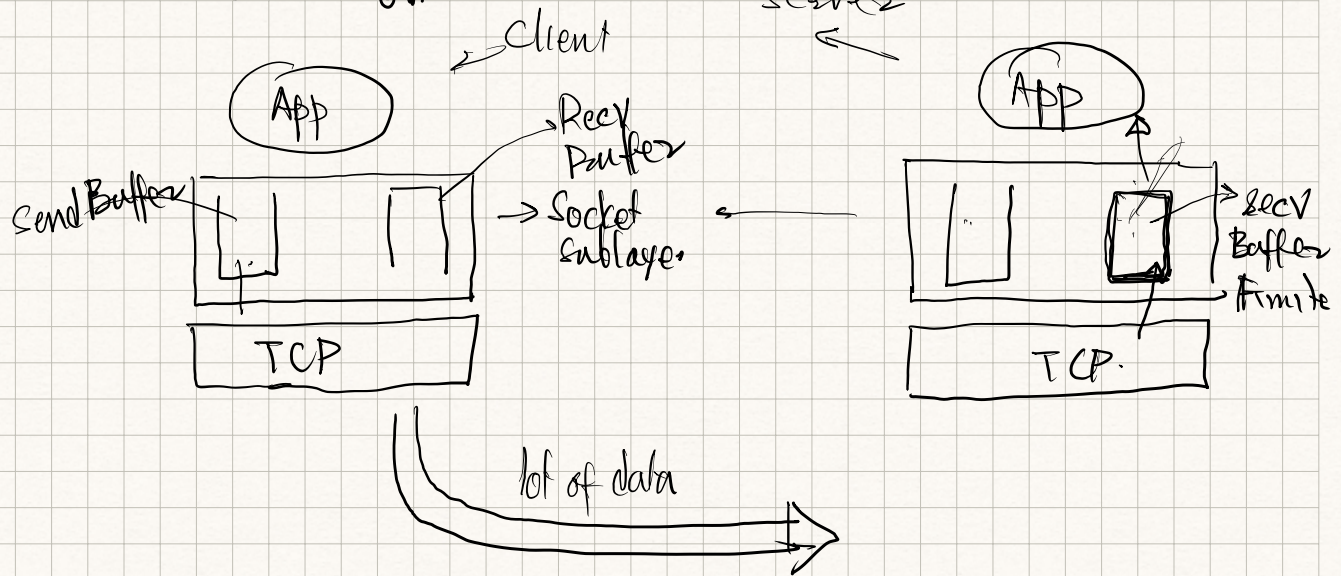
$\alpha$  is small  $\sim 0.125 \frac{1}{8}$

$\beta$  is large  $\sim \boxed{0.875(?)}$

## TCP: Transmission Control Protocol

Congestion Control

Flow Control



Client side is fast

Network is fast

Server side App is slow

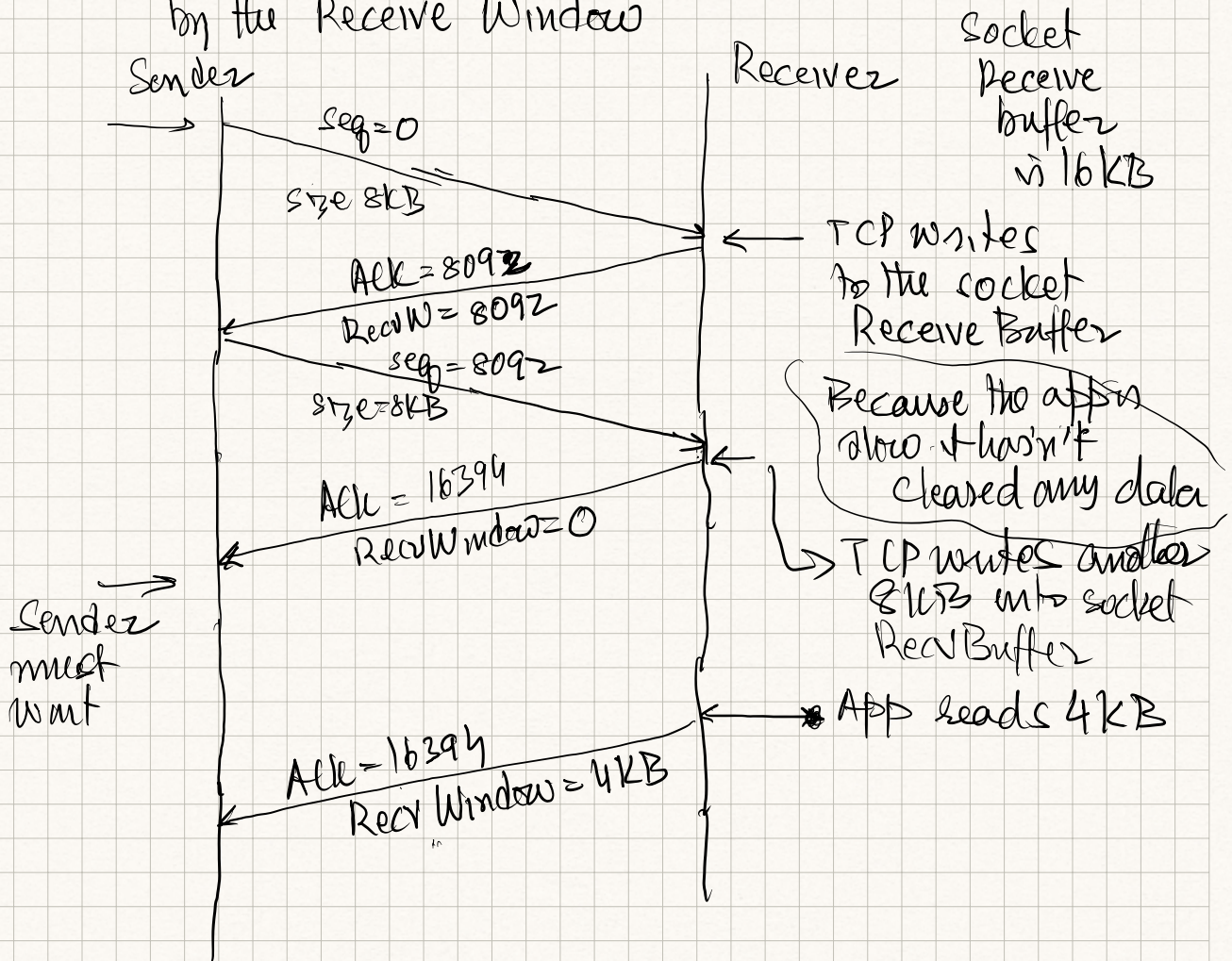
Flow control is a method to pace the sender

$\Rightarrow$  end-to-end between the TCP

sender and TCP receiver

Method: TCP header  
Receive Window

- When the receiver sends ACK it also sends the remaining space in the socket receive buffer to the sender in the Receive Window field
- The sender's window size is capped by the Receive Window



## Congestion Control

- "Flow control" between the sender and the network
- How fast should the sender send?
  - ⇒ Depends on congestion in the network
- ⇒ How is congestion in the network detected?

Internet is a distributed system  
we don't have a central controller  
that can tell each sender what  
rate to send data

Each sender must determine on  
its own

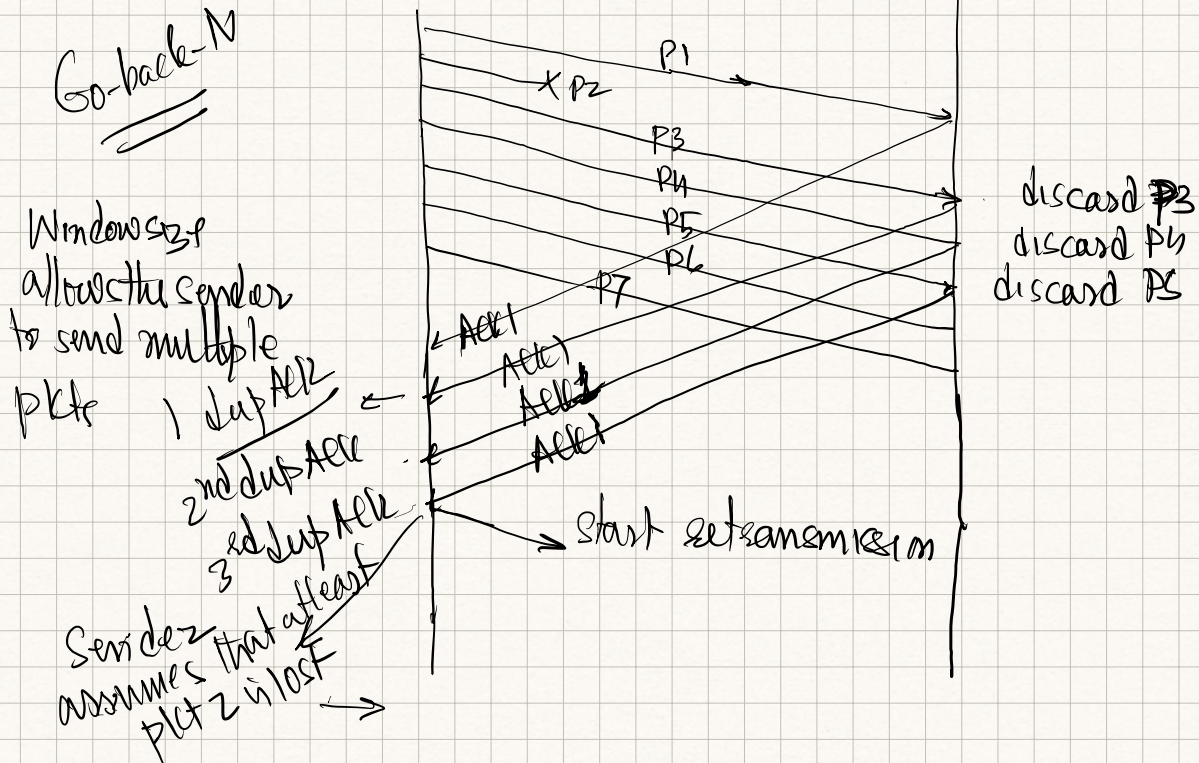
- Ⓐ: How is congestion detected?
- { loss based
  - { delay based

loss based: If a pkt is lost then the sender determines that the network is congested

how do we know that a pkt is lost?

⇒ when we have a timeout

Typically, timeouts are large. And there is another method to determine packet loss



3 dup Ack's is another way to determine packet loss

plot loss is a signal that the network  
is congested

→ delay based signals of congestion

if we know the base RTT

increase beyond the base RTT is

a signal that the network is congested