

Transport Layer 4

ELEC3227/ELEC6255

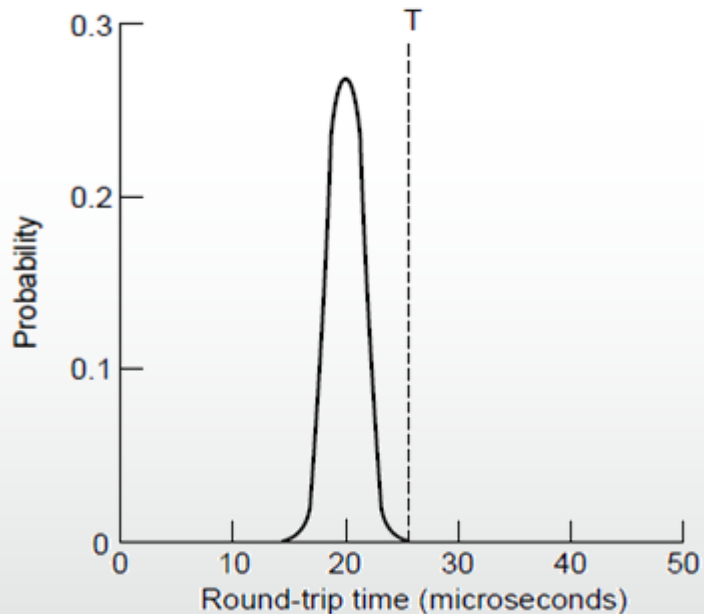
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Overview

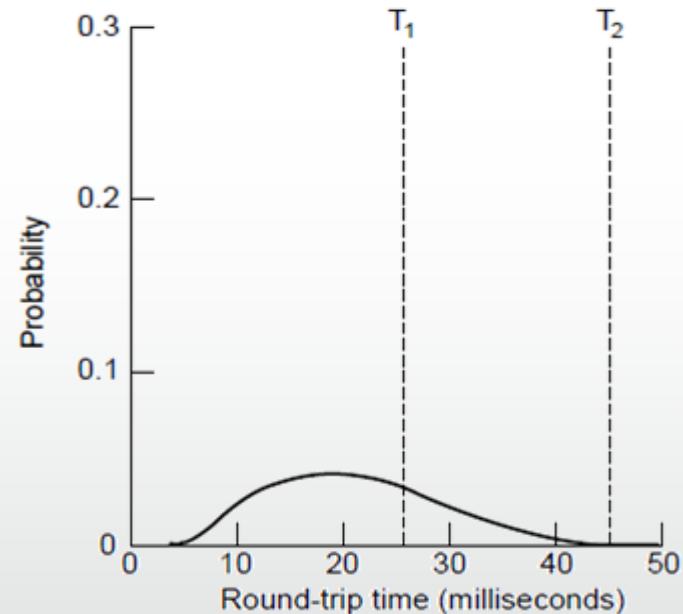
- TCP Congestion Control
- Real-time Transport
- Questions

TCP Timer Management

- TCP estimates retransmit timer from segment RTTs
 - Tracks both average and variance (for Internet case)
 - Timeout is set to average plus 4 x variance



LAN case – small,
regular RTT



Internet case – large,
varied RTT

TCP Congestion Control

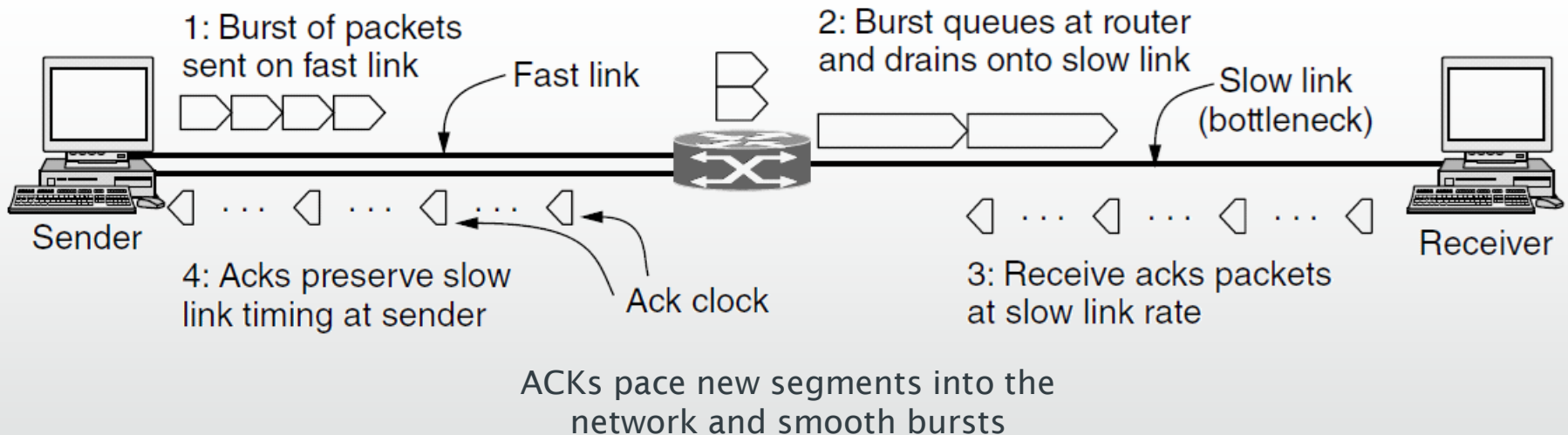
TCP uses AIMD with loss signal to control congestion

- Implemented as a congestion window (cwnd) for the number of segments that may be in the network
- Uses several mechanisms that work together

Name	Mechanism	Purpose
ACK clock	Congestion window (cwnd)	Smooth out packet bursts
Slow-start	Double cwnd each RTT	Rapidly increase send rate to reach roughly the right level
Additive Increase	Increase cwnd by 1 packet each RTT	Slowly increase send rate to probe at about the right level
Fast retransmit / recovery	Resend lost packet after 3 duplicate ACKs; send new packet for each new ACK	Recover from a lost packet without stopping ACK clock

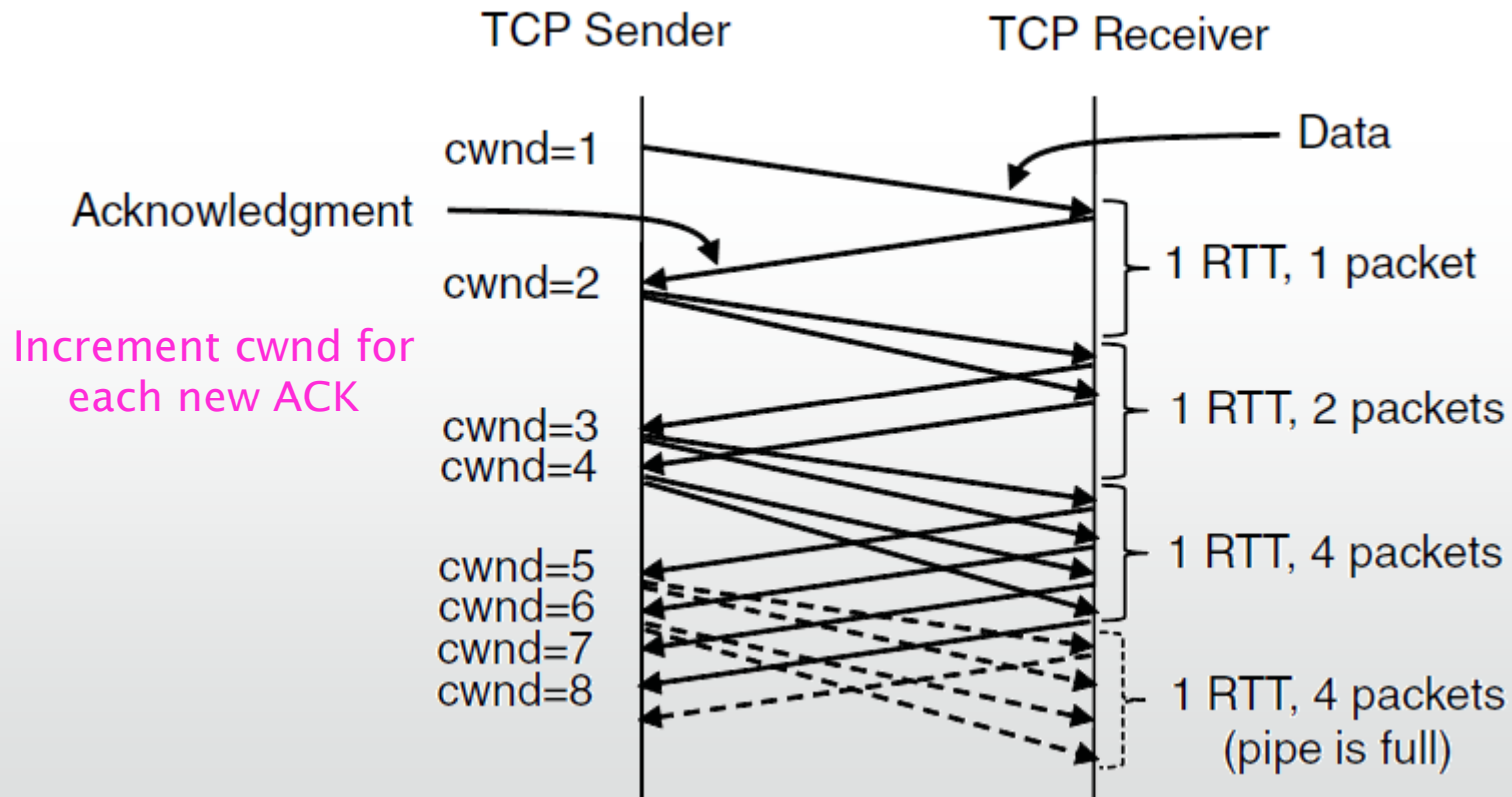
TCP Congestion Control

- Congestion window controls the sending rate
 - Rate is $cwnd / RTT$; window can stop sender quickly
 - ACK clock (regular receipt of ACKs) paces traffic and smoothes out sender bursts



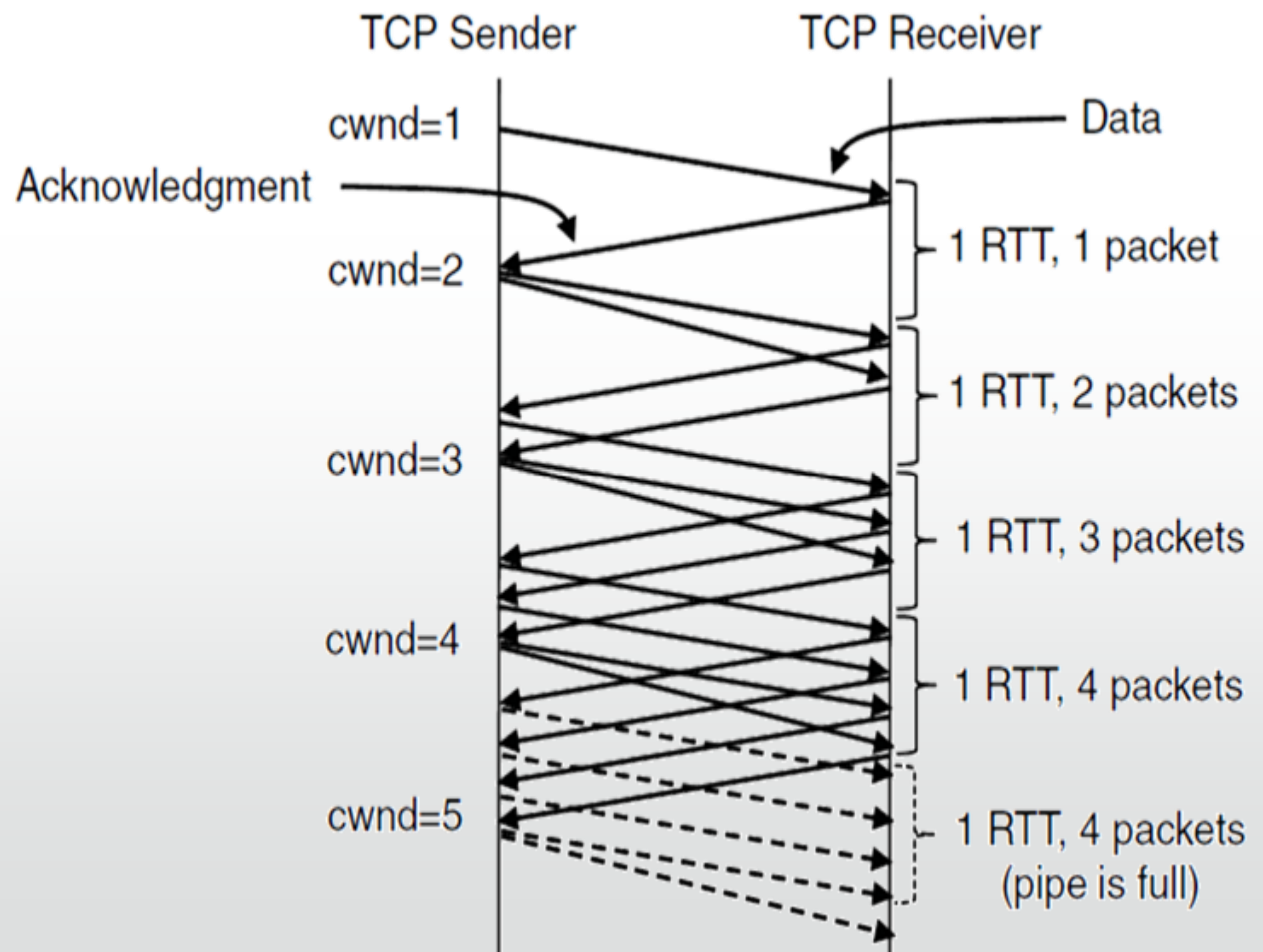
TCP Congestion Control

- Slow start grows congestion window exponentially
 - Doubles every RTT while keeping ACK clock going



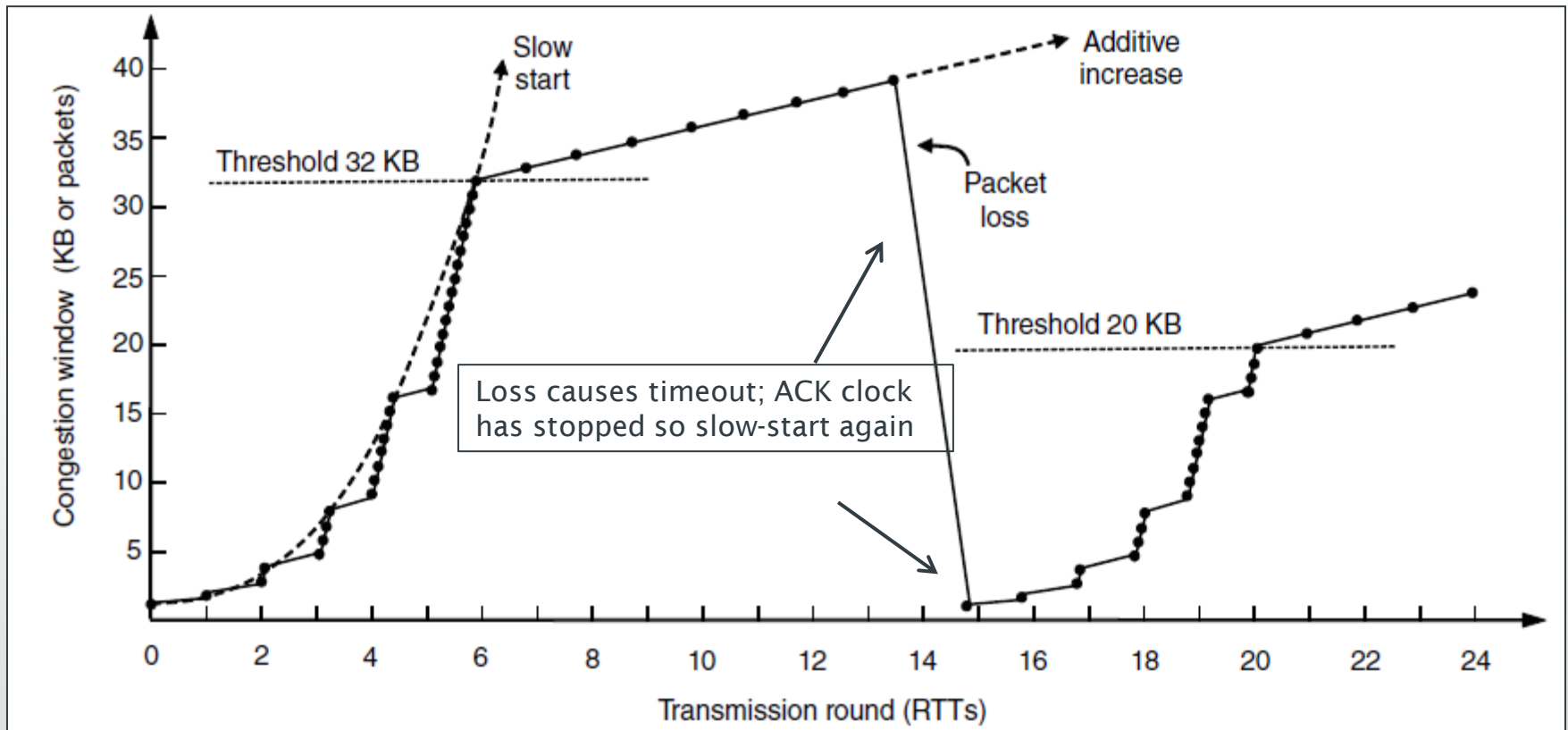
TCP Congestion Control

- Additive increase grows cwnd slowly
 - Adds 1 every RTT
 - Keeps ACK clock



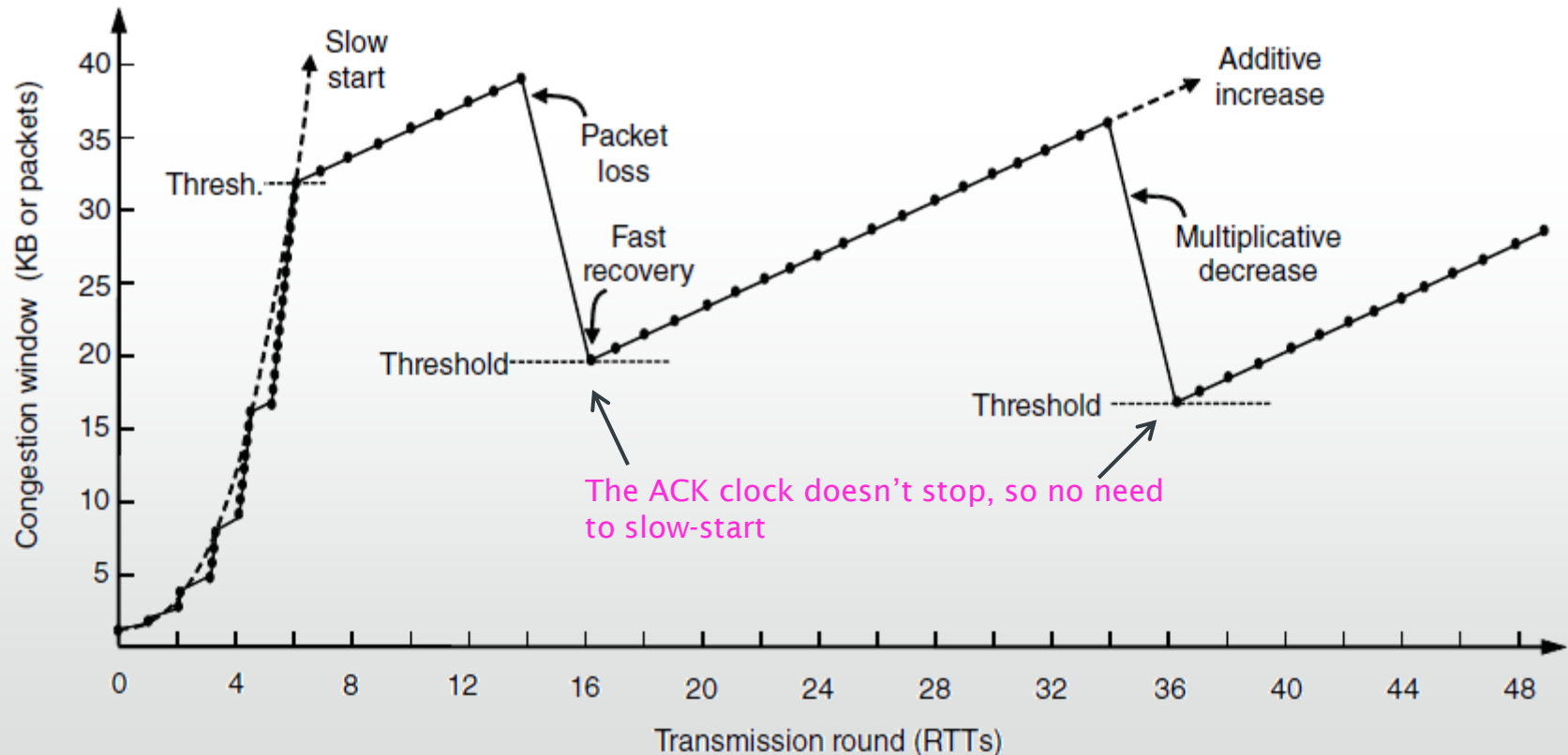
TCP Congestion Control

- Slow start followed by additive increase (TCP Tahoe)
 - Threshold is half of previous loss cwnd



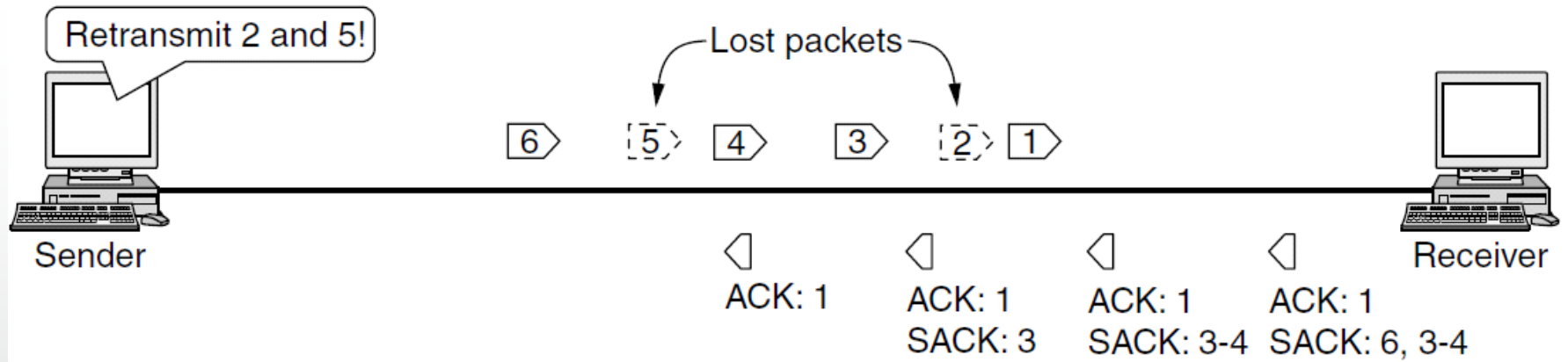
TCP Congestion Control

- With fast recovery, we get the classic sawtooth (TCP Reno)
 - Retransmit lost packet after 3 duplicate ACKs
 - New packet for each dup. ACK until loss is repaired



TCP Congestion Control

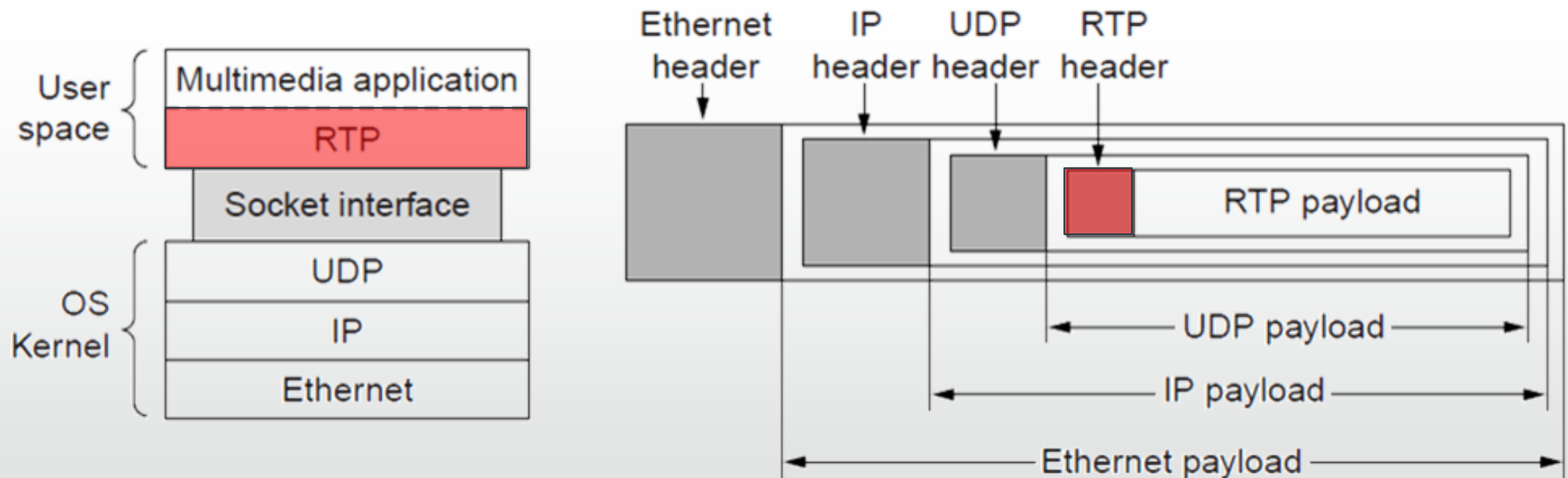
- SACK (Selective ACKs) extend ACKs with a vector to describe received segments and hence losses
 - Allows for more accurate retransmissions / recovery



No way for us to know that 2 and 5
were lost with only ACKs

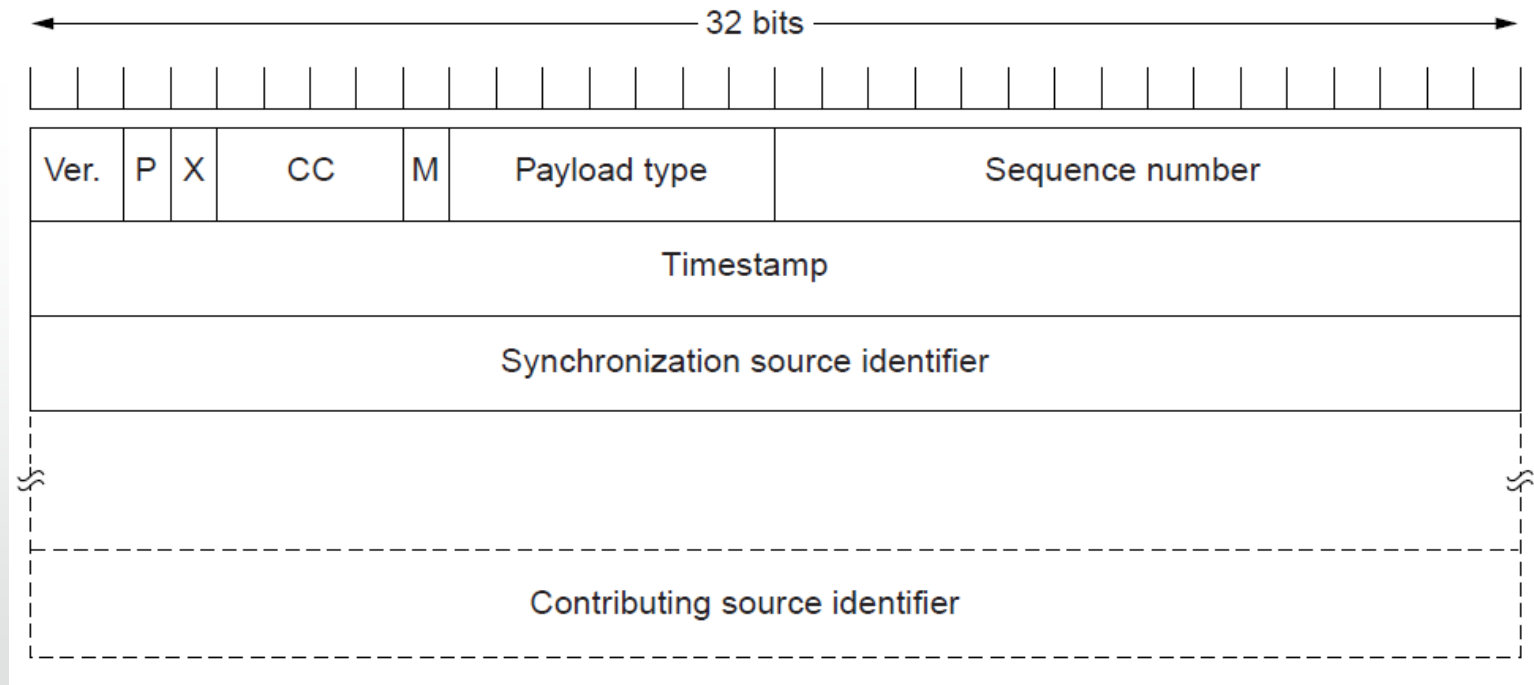
Real-Time Transport

- RTP (Real-time Transport Protocol) provides support for sending real-time media over UDP
 - Often implemented as part of the application



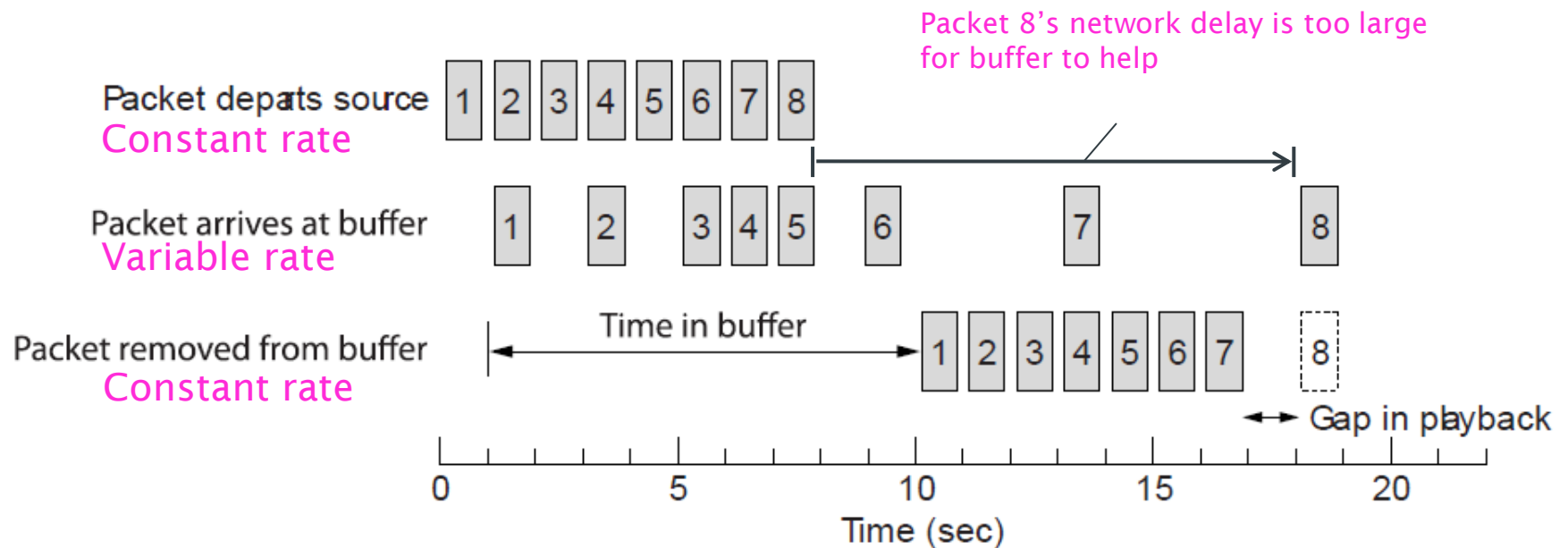
Real-Time Transport

- RTP header contains fields to describe the type of media and synchronize it across multiple streams



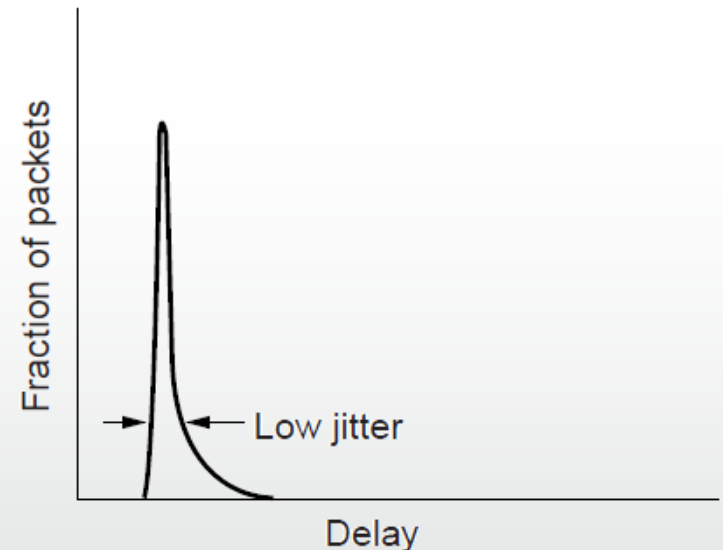
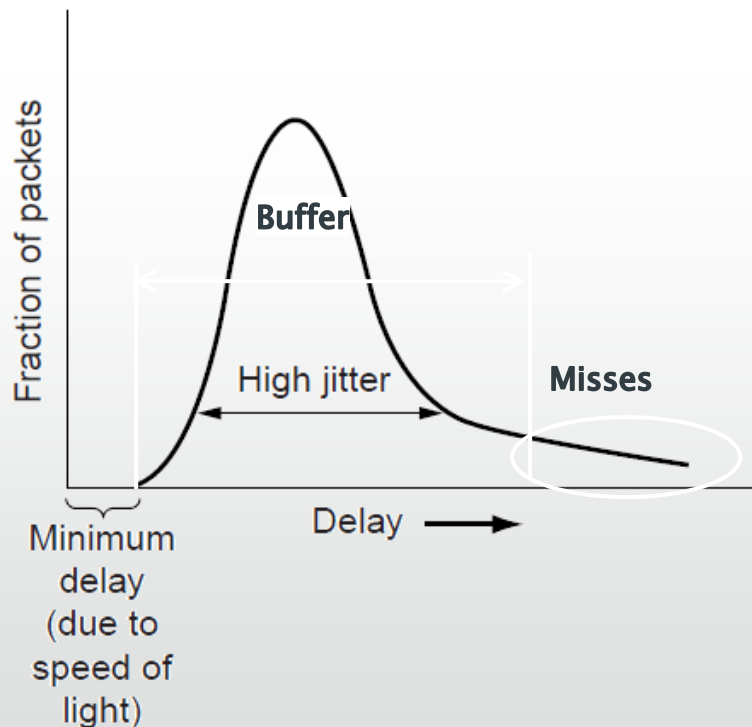
Real-Time Transport

- Buffer at receiver is used to delay packets and absorb jitter so that streaming media is played out smoothly



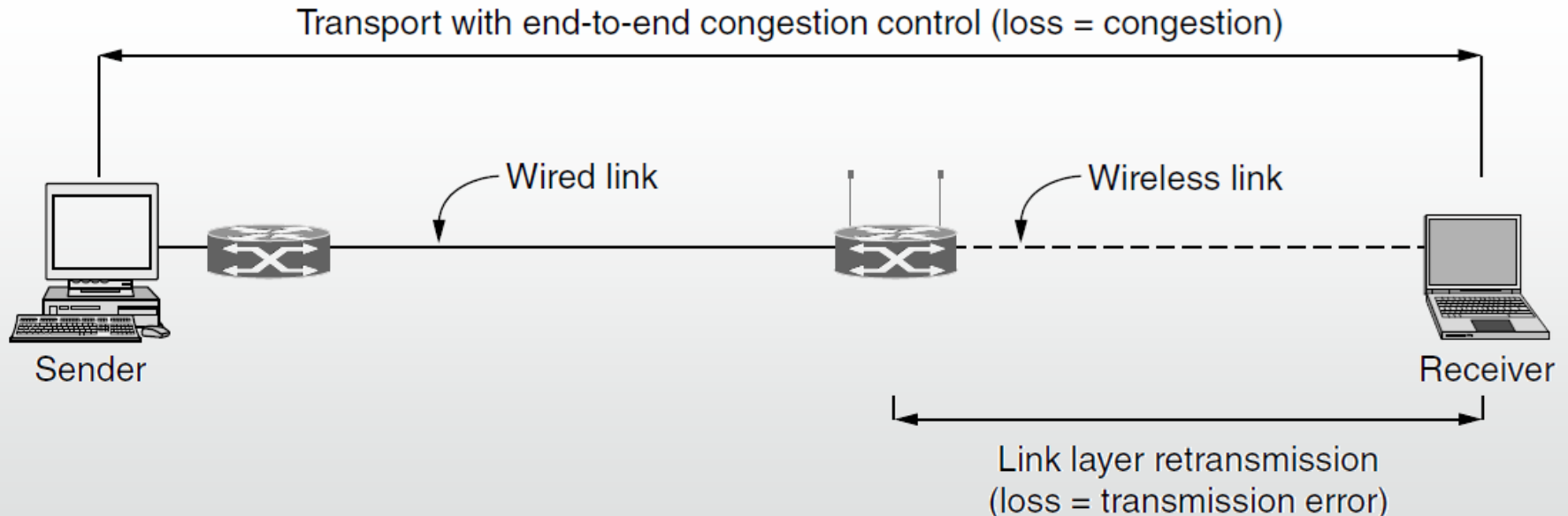
Real-Time Transport

- High jitter, or more variation in delay, requires a larger playout buffer to avoid playout misses
 - Propagation delay does not affect buffer size



Wireless Issues

- Wireless links lose packets due to transmission errors
 - Do not want to confuse this loss with congestion
 - Or connection will run slowly over wireless links!
- Strategy:
 - Wireless links use ARQ (Automatic Repeat Request), which masks errors



Summary

- Regulating the Sending Rate
- TCP Connection State
- TCP Sliding Window
- TCP Congestion Control

Questions

Question 1

In both parts of the Internet File Server example, there is a comment that the value of SERVER PORT must be the same in both client and server.

Why is this so important?

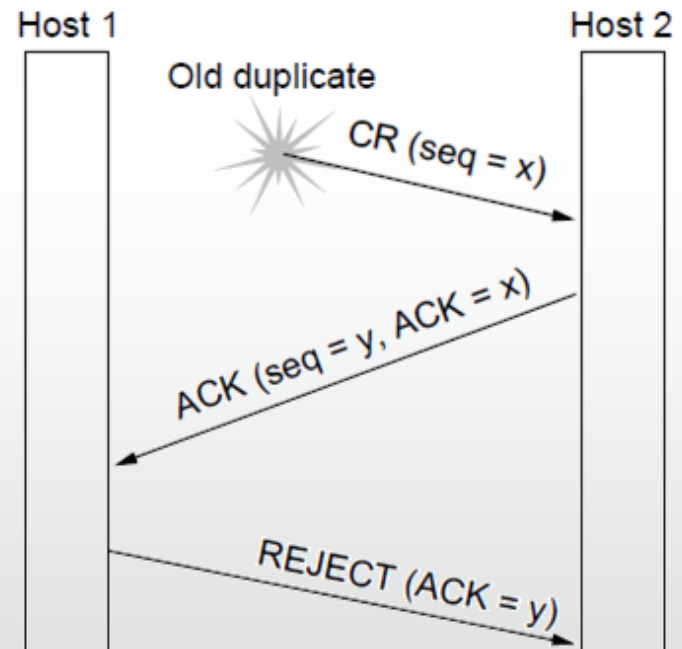
Question 2

Suppose that the clock-driven scheme for generating initial sequence numbers is used with a **15-bit wide clock counter**. The clock **ticks** once every **100 msec**, and the maximum packet lifetime is **60 sec**. How often need resynchronization take place:

- (a) in the worst case?
- (b) when the data consumes 240 sequence numbers/min?

Question 3

Why does the **maximum packet lifetime, T** , have to be large enough to ensure that not only the packet but also its acknowledgements have vanished?



Question 4

Why does **UDP** exist? Would it not have been enough to just let user processes send **raw IP packets**?

Question 5

Both **UDP** and **TCP** use **port numbers** to identify the destination entity when delivering a message. Give two reasons why these protocols invented a **new abstract ID (port numbers), instead of using process IDs**, which already existed when these protocols were designed.

Question 6

A process on host 1 has been assigned port p , and a process on host 2 has been assigned port q .

Is it possible for there to be **two or more TCP connections** between these two ports at the same time?

Question 7

In a network whose **max segment** is **128 bytes**, max **segment lifetime** is **30 sec**, and has **8-bit sequence numbers**, what is the maximum data rate per connection?

Question 8

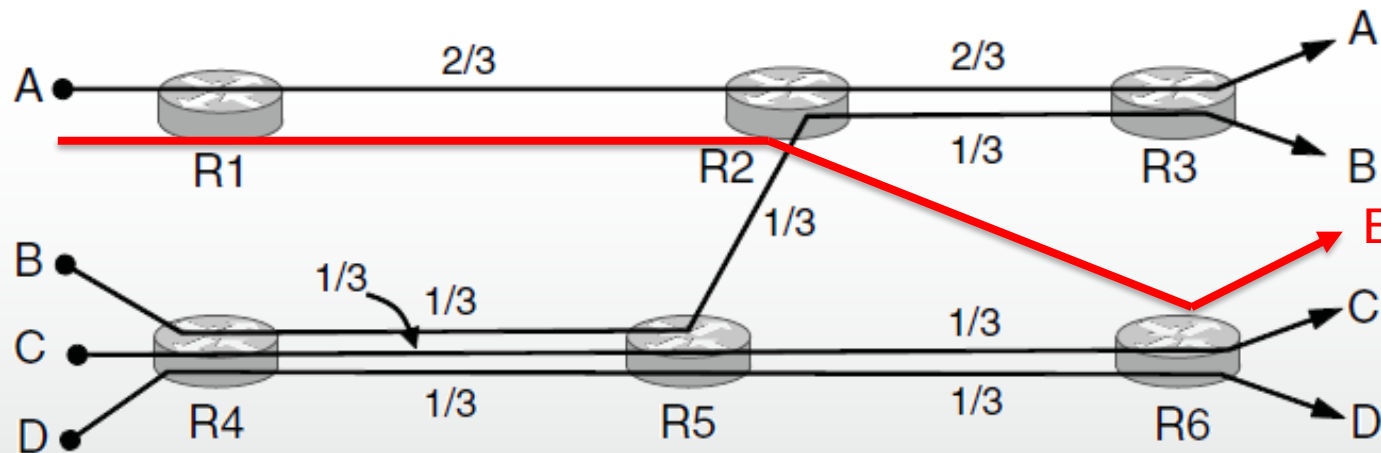
To get around the problem of sequence numbers wrapping around while old packets still exist, one **could use 64-bit sequence numbers**.

However, theoretically, an **optical fiber can run at 75 Tbps**. What **maximum packet lifetime** is required to make sure that future 75-Tbps networks do not have wraparound problems even with 64-bit sequence numbers?

Assume that **each byte has its own sequence number**, as TCP does.

Question 9

Suppose a new flow E is added that takes a path from $R1$ to $R2$ to $R6$. How does the max-min bandwidth allocation change for the five flows?



Question 10

Some policies for fairness in congestion control are:

1. Additive Increase Additive Decrease (AIAD)
2. Multiplicative Increase Additive Decrease (MIAD)
3. Multiplicative Increase Multiplicative Decrease (MIMD)

What are their convergence and stability properties?