Part IV:
Congestion Control

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Outline

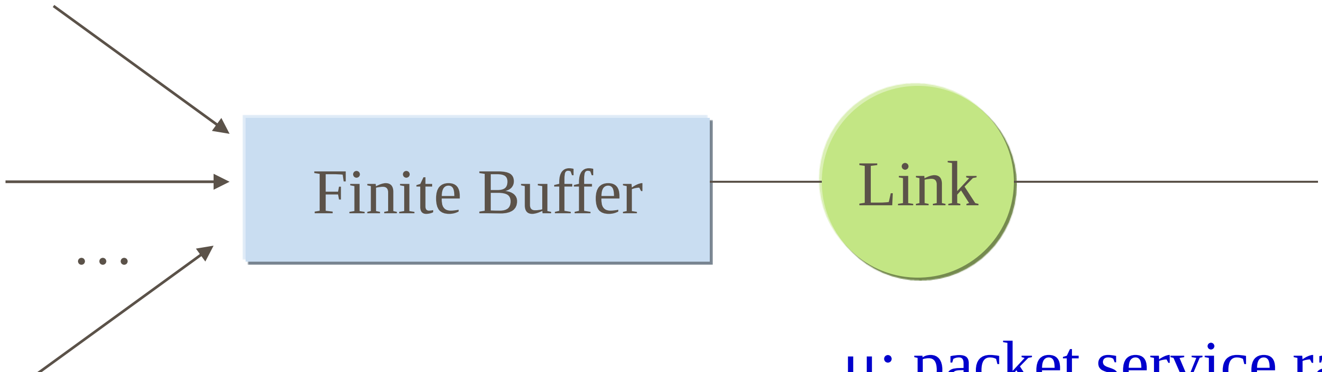
- What is Congestion?
- How to Deal With Congestion?
- Congestion Avoidance (Prevention) Methods
- Congestion Handling (Reactive) Methods
- Summary

What is Congestion?

- Congestion is a ***<u>situation</u>*** (**state**) in which ***performance degrades*** due to the ***saturation*** of network resources such as communication links, processors and memory buffers
- Adverse (unfortunate and harmful) effects include the *long* delays, *inefficient* resource utilization or *waste* and *possible network collapse*
- A critical and significant issue for packet-switched networks

Congestion at a Communication Link

- Packets from *multiple* upstream routers
- Multiple sessions with *different* QoS requirements



μ: packet service rate

λ: packet arrival rate

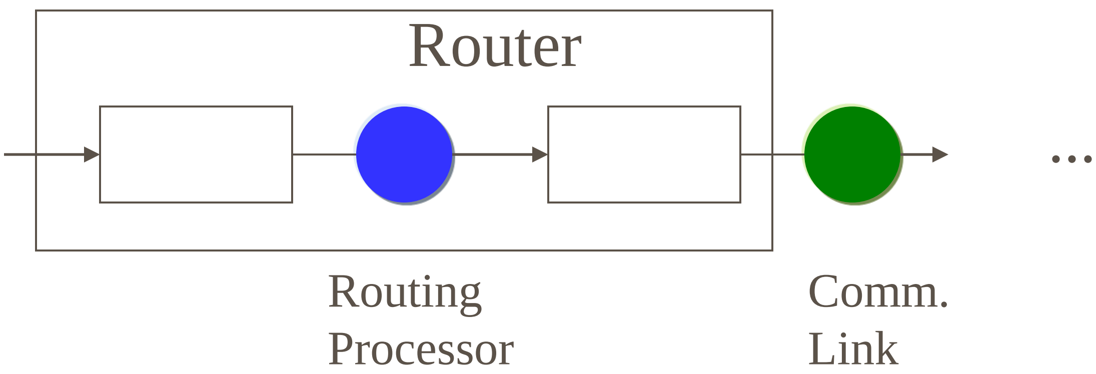
Congestion at a Communication Link

- When ***excessive*** traffic enters a *part* of a network and ***lasts*** for some period of time,
  - Input traffic load ***exceeds*** *link capacity*
  - Excessive traffic is *temporarily* stored in the ***buffer*** with ***finite capacity***
  - When buffer *is full*, arriving packets are being ***dropped***
- ***Bursty*** *traffic* (self-similarity plus long-range dependence) often causes congestion

Congestion at Communication Link (cont'd)

- How about *infinite* buffers?
  - Packets still experience *long delays*
  - Packets may be *timed out* in the protocol layer and duplicates are sent
  - Worsen the situation (more packets arrive)

Transmission Bottleneck



- Slow routing processor(s)
  - e.g., packet buffering, routing table lookup, etc.
- Low-bandwidth links
  - easily becoming the bottleneck
- Capacity upgrade, ***bottleneck shift***

More about Congestion

- Congestion tends to feed upon itself and become worse
  - e.g., TCP window-based congestion control
- Congestion handling is a global issue
  - to ensure the total offered traffic *won't* exceed what the network can handle
- Research on how to accurately measure or assess bottleneck link(s)

Congestion Control Schemes

- Measures for controlling network traffic in order to *prevent*, *avoid*, or *recover* from network congestion
- Two areas of control
  - congestion avoidance
  - congestion recovery

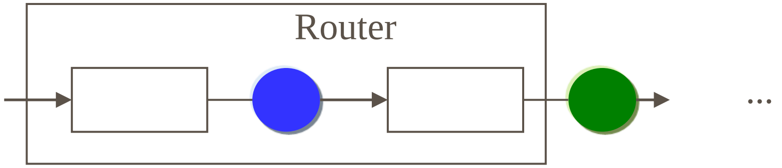
Approaches

- Window-based (buffer) flow control
- Source quench
- Slow start/congestion avoidance in TCP

Flow Control

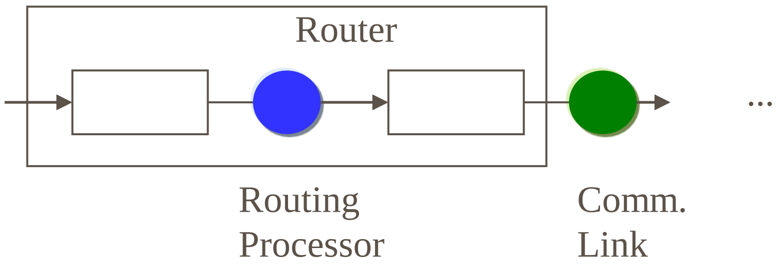
- To control the traffic flow between a sender and a receiver
  - point-to-point (e.g., HDLC) or end-to-end (e.g., TCP)
- To ensure that a faster sender won't overrun a slower receiver
- Usually achieved by performing *direct* feedback from the receiver to tell the sender its receiving capability and status

Congestion Control



- Cause
  - traffic load (temporarily) is *greater* than the resources can handle
- Solutions
  - ***Increase the Resources***
    - e.g., split traffic on multiple routes vs. one best route
    - put backup router on-line in the presence of congestion

Congestion Control (cont'd)



- ***Decrease the Load***
  - Blocking new traffic, denial of service to new users
  - Degrading services to some or all users
  - Having users schedule their demands in a more predictable way.
- For networks support *virtual circuits*, congestion control can be performed at the ***network layer***
- For *datagram* networks, congestion control is often performed at the ***transport layer.***

Congestion Control Model

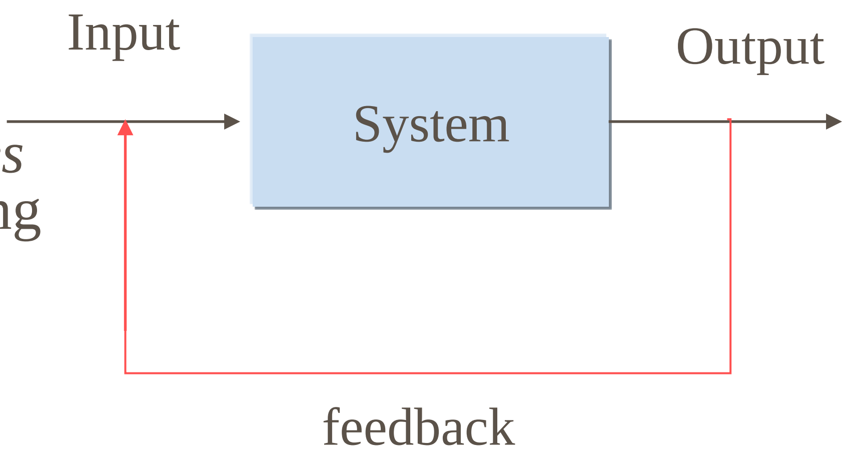
- Control Theory
- Open Loop Solution
  - Act at the source
  - Act at the destination
- Closed Loop Solution
  - Explicit feedback
  - Implicit feedback

Open Loop-based Congestion Control

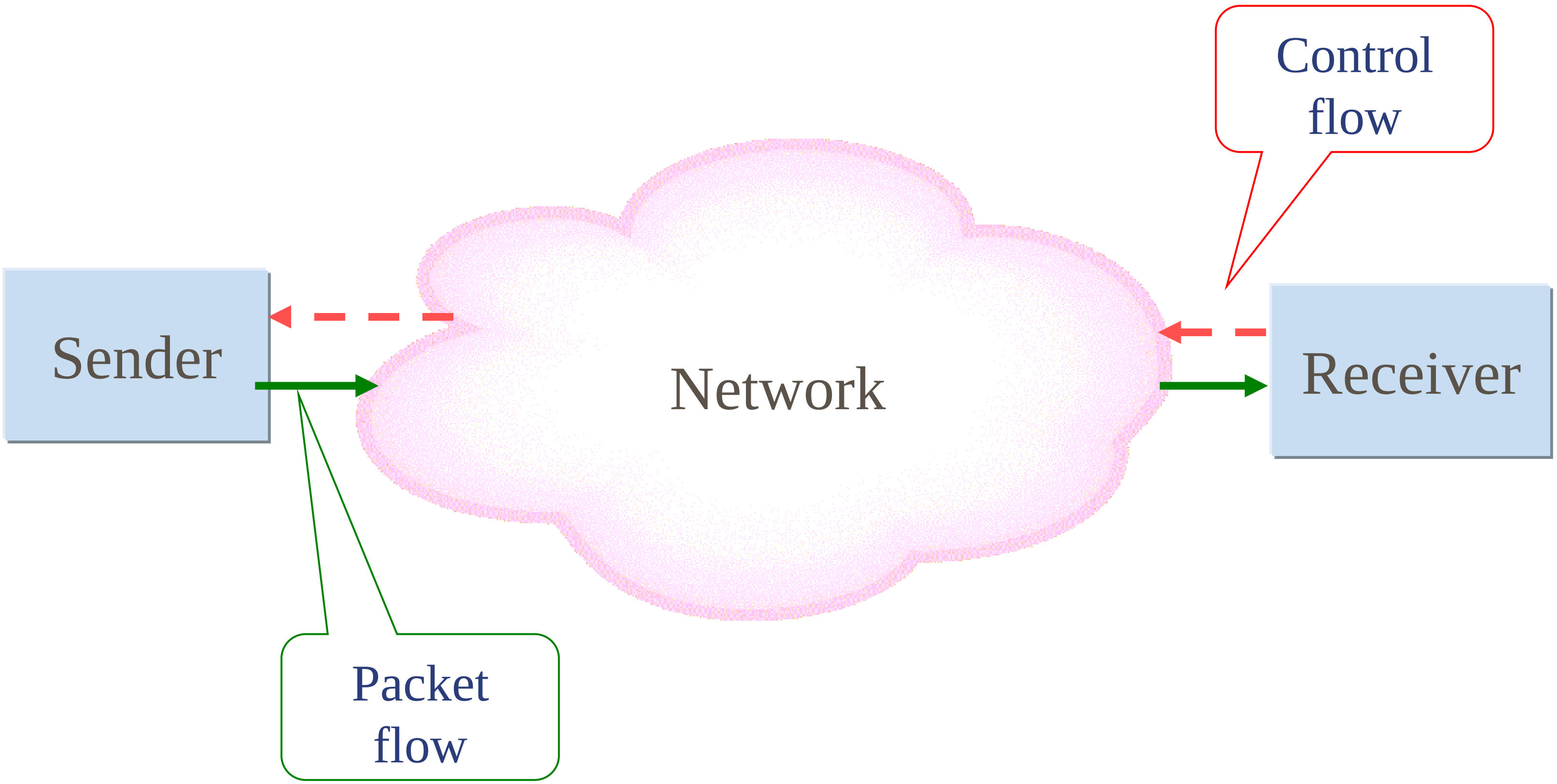
- Attempts to solve the problem by *good designs* so congestion does not occur.
- Approaches
  - schemes to decide when to *accept* new traffic, e.g., *admission control*
  - schemes to decide when to *discard* packets and which ones, e.g., *active queue management* (e.g., RED)
  - appropriate *scheduling* algorithms to decide which packet to transmit (e.g., weighted fair queueing (WFQ))

Closed Loop-based Congestion Control

- Based on the ***feedback** from the network*
- Three parts
  - ***Monitor*** *network resources* (links and routing/switching equipment) to detect *when* and *where* congestion occurs
  - ***Pass*** congestion information to ***places*** where action *can* be taken
  - ***Take* actions** to correct the problem



Congestion Control Model: Closed Loop Solution



Closed Loop-based Congestion Control (cont'd)

- Explicit feedback
  - *Special packets* are sent back from the point of congestion to notify traffic source affected
  - e.g., the "*rate-based congestion control scheme*" for ABR traffic in ATM networks (info)
- Implicit feedback
  - Source deduces the existence of congestion based on its local observations
  - e.g., in TCP

Closed Loop-based Congestion Control (cont'd)

- <u>Performance metrics</u> (measures) for congestion detection, e.g.,
  - percentage of packet drops (due to buffer overflow)
  - buffer queue length (threshold-based)
  - number of timeout packets or retransmissions
  - packet delays (and/or variations)

How to learn about the network state?

- Approach #1 - Probe packets
  - A ***traffic source*** *periodically* sends ***probe packets*** to the network.
  - A network node(s) (routers, switches, etc.) will ***mark*** probe packets when congestion is detected.
- Approach #2 - Congestion Indication
  - Packet header carries a "***congestion indication***" field.

How to learn about the network state? (cont'd)

  - It is marked when an intermediate node is under congestion.
  - Receiver sends a *congestion notification message* back to sender if any packets received have this field marked.
- Approach #3 – Network Notification
  - N*etwork nodes **automatically*** issue congestion notification message to traffic sources.
  - Traffic sources ***reduce*** sending rates.

How to Avoid or Reduce Congestion?

- ***Bursty*** *traffic* often causes network congestion
- Let a traffic source perform ***traffic spacing*** to reduce the degree of burstiness when injecting traffic into the network
  - To lessen *spiky* bursts
  - It <u>cannot</u> avoid congestion
  - But … it *helps* to reduce the probability of congestion
  - Traffic enters network in a more *predictable* rate

Traffic Shaping

- To regulate the average rate and peak rate (burstiness) of data transmission
  - sliding window limits the amount of data in transit, *not* the sending rate
- Used in ATM networks to enforce a traffic source to send data complying with the traffic description in the service contract.
- It helps the network to keep up its promised delivery QoS (more predictable traffic load).

Traffic Shaping and Policing

- A ***source*** can use a leaky bucket (or token bucket) to shape its traffic flow for traffic description ***conformance***
- A network can use a leaky bucket (or token bucket) for the ***conformance check*** of a traffic flow (or a connection or virtual circuit)

Admission Control

- Blocking any new traffic entering the network in the presence of congestion
- Direct traffic flow away from congested spots

Choke Packets

- Similar concepts with rate-based congestion control scheme in ATM networks
- Criterion for congestion condition
  - metrics, e.g., link utilization, queue length, etc.
  - threshold-based

- <u>Scheme</u>
  - *A congested node (router) sends a <u>choke</u> <u>packet</u> to the source host*
  - The original packets are tagged to avoid generate any more choke packets further along the path.
  - When receiving a choke packet, a source host ***reduces*** its data sending rate to the specified destination.
  - When a source does not receive any more choke packets within a period of time, it starts ***increasing*** the sending rate.

Summary

- Congestion control has been one of the major issues since the inception of networks.
- It has become a nightmare of both users and network operators.
- This problem will continue to exist.
- Congestion must be controlled in multiple ways – avoidance (good designs, admission control, traffic control, etc.), handling and recovery.

Active Queue Management: Random Early Detection Gateway for Congestion Avoidance

Sally Floyd and Van Jacobson, ”Random Early Detection Gateways for Congestion Avoidance,” IEEE/ACM Transactions on Networking, Vol. 1, No. 4, pp. 397-413, 1993.

Queue Management

- It is to manage **the length of packet queues** by **dropping** packets when necessary or appropriate.
- *Active* queue management
- The traditional technique for managing router queue lengths is "tail drop".
  - A max length (in terms of packets) is set for each queue.
  - Incoming packets are accepted for the queue until the max length is reached, then drop subsequent incoming packets until the queue decreases.

The need for active queue management

- Two important drawbacks:
  - Lock-out : in some situations tail drop allow a single connection or a few flows to monopolize queue space, preventing other connections from getting room in the queue.
  - However, this does not take into account that packet bursts play in Internet performance.

Motivation

- High-speed networks with *large delay-bandwidth products*
- Gateways should have *large maximum queues* to accommodate transient congestion.
- But … it is undesirable to have large queues which were full much of the time.
  - large average queueing delay, possibly packet drops which may cause TCP congestion avoidance phase to slow start phase
- What desired are mechanisms to keep throughput high but average queue size low.

Congestion Control: various approaches

- Explicit feedback from the network
- If no explicit feedback, it is left to the *transport layer protocols* to infer congestion from various *observations* and *estimates*
  - e.g., packet drops, changes in round-trip time, changes in throughput, etc.

Random Early Detection: an active queue management method

- The focus is on the design of queue management.
- Idea
  - ***monitor*** and *control* the *average queue size* at the router and
  - ***notify*** *the connections causing congestion*.
- Assume FIFO queueing discipline.
  - Scales well and is easy to implement

Global Synchronization Problem

- TCP – one single packet drop (BSD 4.3 TCP Tahoe) makes the connection enter the Slow-Start phase, reducing the congestion window size to one.
- This problem is resulted from many TCP connections *reducing* their windows *at the same time*.
- In the case of a shared queue and tail drop discipline, global synchronization may result in *low utilization* and *throughput*.
- TCP recovering from a burst of packets drops is more difficulty.

Active queue management: Fairness

- Queue management does *not* provide general fairness among flows, e.g.,
  - Two TCP connections may receive different bandwidths because they have different round-trip times.

$$B(p) = \frac{1}{RTT}\sqrt{\frac{3}{2bp}} + o(1/\sqrt{p}) \qquad (20)$$

  - A flow without using congestion control may receive more bandwidth than a flow that does.
- General fairness can be achieved by *adding per-flow scheduling* such as Weighted Fair Queueing.

Note: Between two “triple-duplicate" ACK (TD) loss indications, the sender is in congestion avoidance, and the window increases by 1/b packets per round

Design Ideas

- Randomization – choose connections to notify of congestion
- For *transient congestion* – a temporary increase in the queue
- For *long-lived congestion* – an increase in the average queue size; send congestion to randomly selected connections to decrease their windows.

- Wish to *make the probability that a connection is notified is proportional to that connection's share of the throughput through the gateway*.

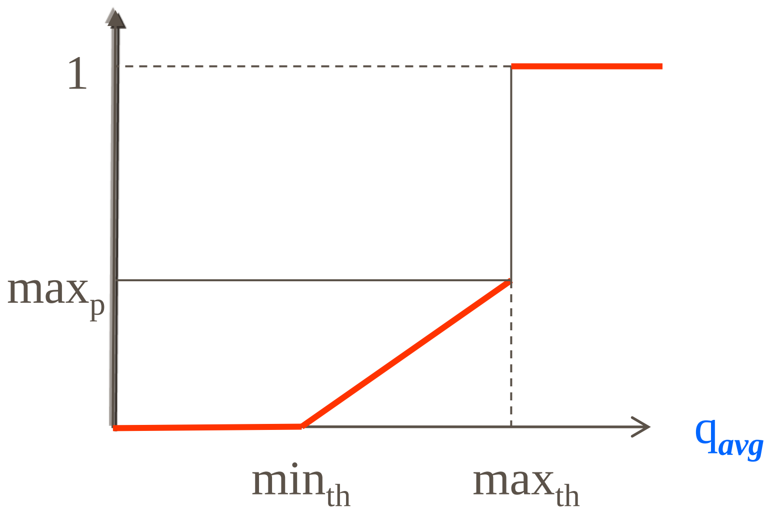
RED – an active queue management algorithm

- Drop arriving packets *probabilistically* when based on the average queue size
- Calculate average queue size (either in units of packets or of bytes)
  - Use a low-pass filter – exponential weighted moving average
- Packet marking (or drop) decision
  - Two parameters, *minimum threshold* (min_{th}) and *maximum threshold* (max_{th}) are used to determine if to mark an incoming packet.

RED - packet marking decision

- Goals
  - to ensure average queue size does not significantly exceed the maximum threshold.
- Method - compare average queue size with min_{th} and max_{th}

For each packet arrival
calculate the average queue size q_{avg}
if $min_{th} <= q_{avg} <= max_{th}$
calculate probability p_a;
with probability p_a mark the arriving packet;
else if $max_{th} <= q_{avg}$
mark the arriving packet;



The RED Algorithm (1/3)

- Notations
  - q_{avg}: average queue size
  - *q_time*: start of the queue idle time
  - *count*: packets since last marked packet
  - w_q: queue weight
  - min_{th}: minimum threshold for queue
  - max_{th}: maximum threshold for queue
  - max_p: maximum value for p_b
  - p_a: current packet-marking probability
  - q: current queue size
  - *time*: current time
  - $f(t)$: a linear function of the time t

The RED Algorithm (2/3)

Initialization:

$q_{avg} \leftarrow 0;$

$count \leftarrow 1;$ // *count*: packets since last marked packet

for each packet arrival:

calculate the new average queue size q_{avg};

if the queue is nonempty

$q_{avg} \leftarrow (1-w_q)q_{avg}+w_q\, q$ // w_q: queue weight

// q: current queue size

else

$m \leftarrow f(time - q\_time);$ // *time*: current time, *q_time*: start of

// the queue idle time, *f(t)*: a linear function of the time *t*

$q_{avg} \leftarrow (1-w_q)^m q_{avg}\ ;$

The RED Algorithm (3/3)

if min_{th} <= q_{avg} < max_{th}
{
increment *count;*

$$p_b \leftarrow \max_p \cdot \left(\frac{q_{avg} - \min_{th}}{\max_{th} - \min_{th}}\right)$$

$p_a \leftarrow p_b / (1 - count * p_b)$;

with probability p_a, mark the arriving packet;
count $\leftarrow$ 0;
}
else if max_{th} <= q_{avg}
mark the arriving packet;

// *count*: packets since last marked packet

- q_{avg} determines the degree of burstiness allowed in the gateway queue.
- avoid biases and global synchronization, and **mark packets sufficiently frequently to control the average queue size.**
- **wish to mark packets at *fairly evenly-spaced intervals*.**

The RED Algorithm: Design Guidelines

- **The design of p_b** // $p_b \leftarrow max_p(q_{avg} - min_{th})/(max_{th} - min_{th})$;
  - It varies linearly from 0 to max_p
  - **max_p: the maximum value of p_b**
- max_p
  - e.g. max_p= 1/50 (on average, roughly one out of 50 arriving packets is dropped.)
  - Packet marking prob. changes slowly as the average queue size changes to avoid oscillations.
  - Experiments show that never set max_p> 0.1
- p_a
  - **Wish to assure p_a increases slowly as the count increases since the last marked packet.**
  - So to ensure it won't wait too long before marking a packet
  - Ensure packets are marked at *fairly evenly-spaced intervals*
  - **(p_a<=1, count <=1/p_b – 1) (<u>when count approaches 1/p_b , with approximate probability 1, the arriving packet is marked.)</u>**

Measure the queue in bytes instead of packets

- A packet is marked *proportional to the packet **size*** in bytes

  $p_b = max_p\ (q_{avg} - min_{th})/(max_{th} - min_{th})$

  $p_b = p_b\ (pkt\_size/max\_pkt\_size)$

  $p_a = p_b\ /(1 - count \times p_b)$

- A large FTP packet is more likely to be marked than a small Telnet packet.

Calculating p_b, p_a - geometric distribution

- Let X be the number of packet arrivals during intermarking time
- We know

  $Prob[X=n]=(1-p_b)^{n-1}p_b$
- X is geometric random variable and $E[X] = 1/p_b$

- But … **if to maintain a constant average queue size, one would need to mark packets at fairly regular intervals.**
- Namely, we do **not** want to have too many marked packets **close together**, **nor too long to mark a packet**. (they may cause global synchronization)

Calculating p_b, p_a - uniform distribution

- So … we wish X to be a **uniform** random variable from $\{1, 2, \ldots, 1/p_b\}$
- To achieve this, we need the marking probability be

  $p_b /(1- \text{count} \times p_b)$

  - Count – the number of unmarked packets that have arrived since last marked packet.
- We have

$$\Pr ob[X = n] = \frac{p_b}{1-(n-1)p_b} \prod_{i=0}^{n-2} \left(1 - \frac{p_b}{1-ip_b}\right)$$

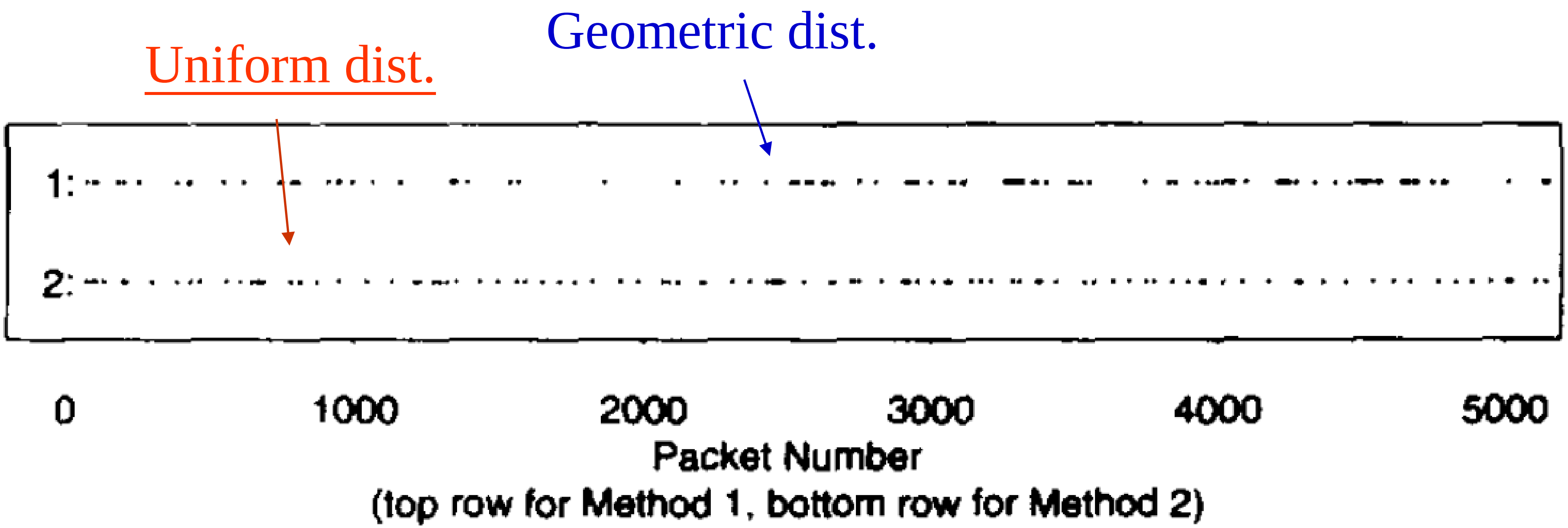
$$= \begin{cases} p_b & \text{for } 1 \le n \le 1/p_b \\ 0 & \text{for } n > 1/p_b \end{cases}$$

$$E[X] = 1/2p_b + 1/2$$

$P_a = P_b / (1 - count * P_b)$: derivation

- The goal is to have uniform distribution:
  $P_{a1} = P_b$
  $P_{a2}(1-P_{a1}) = P_b$
  $P_{a3}(1-P_{a2})(1-P_{a1}) = P_b$
  .
  $P_{an}(1-P_{an-1})\ldots\ldots(1- P_{a1}) = P_b$
- 上下兩式相除, 可以得到
  $P_{a1} = P_b$ ---(1)
  $P_{a2} = P_{a1} / (1 - P_{a1})$ ---(2)
  $P_{a3} = P_{a2} / (1 - P_{a2})$ ---(3)
  .
  $P_{an} = P_{an-1} / (1 - P_{an-1})$ ---(n)
- 第一式代到第二式
  $P_{a2} = P_b / (1 – P_b)$ 代到第三式
  $P_{a3} = P_b / (1 - 2P_b)$ 代到第四式
  $P_{a4} = P_b / (1 - 3P_b)$
  .
  $P_{an} = P_b / (1 – (n-1)P_b)$ 另n-1 = count 則可以得到$P_a = P_b / (1 – count * P_b)$

Comparing packet marking under geometric and uniform probability distributions



- p_a=0.02 for geometric
- p_b=0.01, $p_a=p_b/(1+ip_b)$
- They all mark roughly 100 out of 5000

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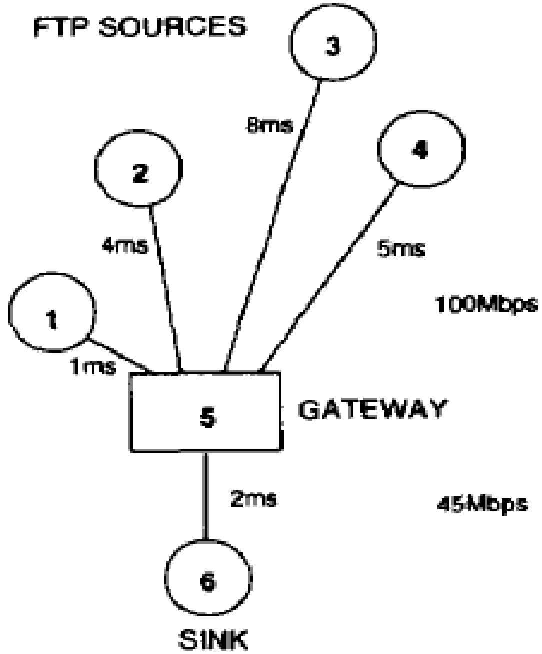


Fig. 4. Simulation network.

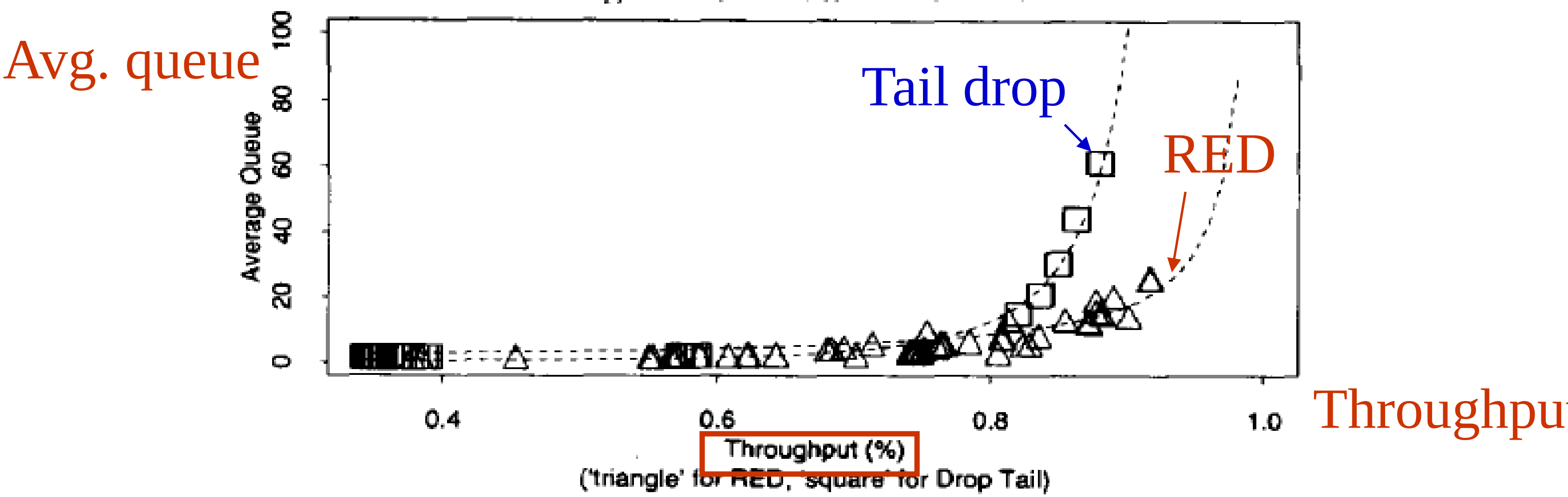
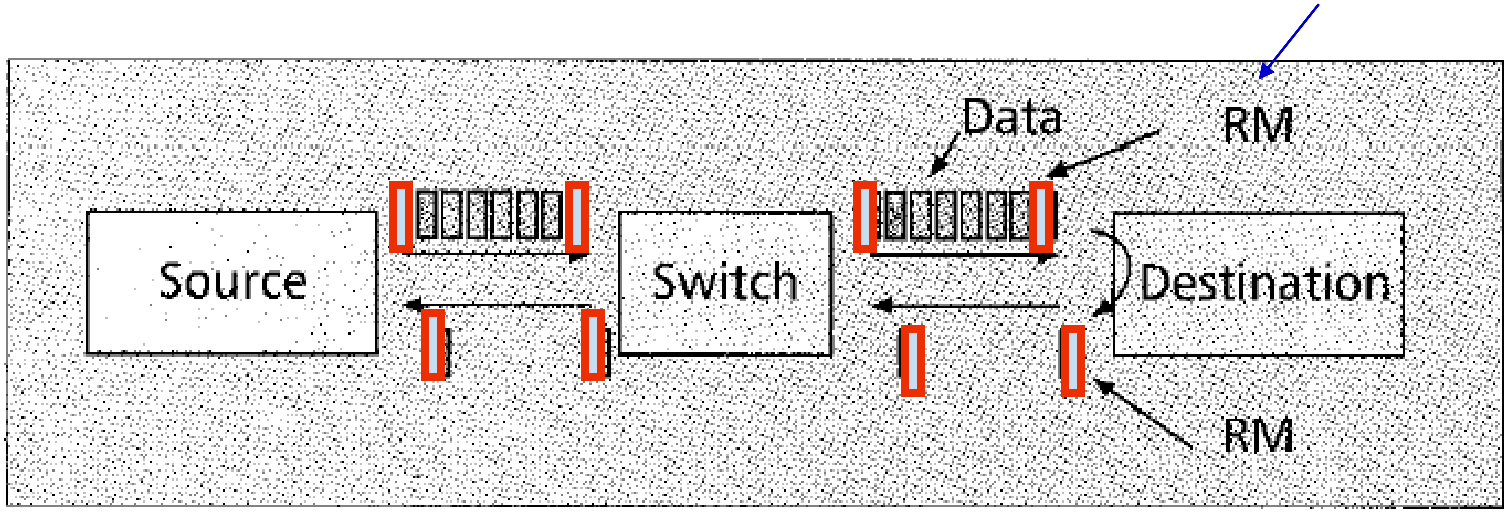


Fig. 5. Comparing Drop Tail and RED gateways.

ATM ABR Congestion Control

ATM ABR Traffic Management Framework

Resource Management Cell



■ **Figure 1.** *ABR traffic management model: source, switch, destination, and resource management cells.*

ABR Congestion Control

- A Rate-based End-to-end Closed-Loop approach
  - *Continuous* feedbacks between network and sources.
  - Resource Management (RM) cells travel from the source to the destination and back to the source.
- Rate-based
  - Sources send data at a specified **rate**.
  - Different from TCP window-based congestion control.
    - Sources limit their transmission to a particular *number* of packets

Ways of Sending Congestion Feedback to Sources

- Explicit Forward Congestion Indication (EFCI)
- Congestion Indication (CI)
- Explicit Rate (ER)

Explicit Forward Congestion Indication (EFCI)

- One bit in *data cell*
- Set by a congested switch
- Destination saves the EFCI state of last data cell.
- If EFCI is set when it turns around an RM cell, it uses the CI bit to give feedback to the source.
- Sender may adjust its sending rate accordingly
- *Binary* or *EFCI* switches.

Congestion Indication (CI)

- RM cells have two bits in the payload:
  - CI bit
  - no increase (NI) bit
- CI bit is set by a congested switch if severe congestion occurs.
- *Relative rate marking* switches.

Explicit Rate (ER)

- A field in the payload of an RM cell
- Can be reduced by congested switch to any desired value.
- *Explicit rate* switches.
- Under certain circumstances, congested switches can generate RM cells and send them immediately to the sources.

TCP: Congestion Control

TCP Congestion Control

- Slow start
- Congestion avoidance

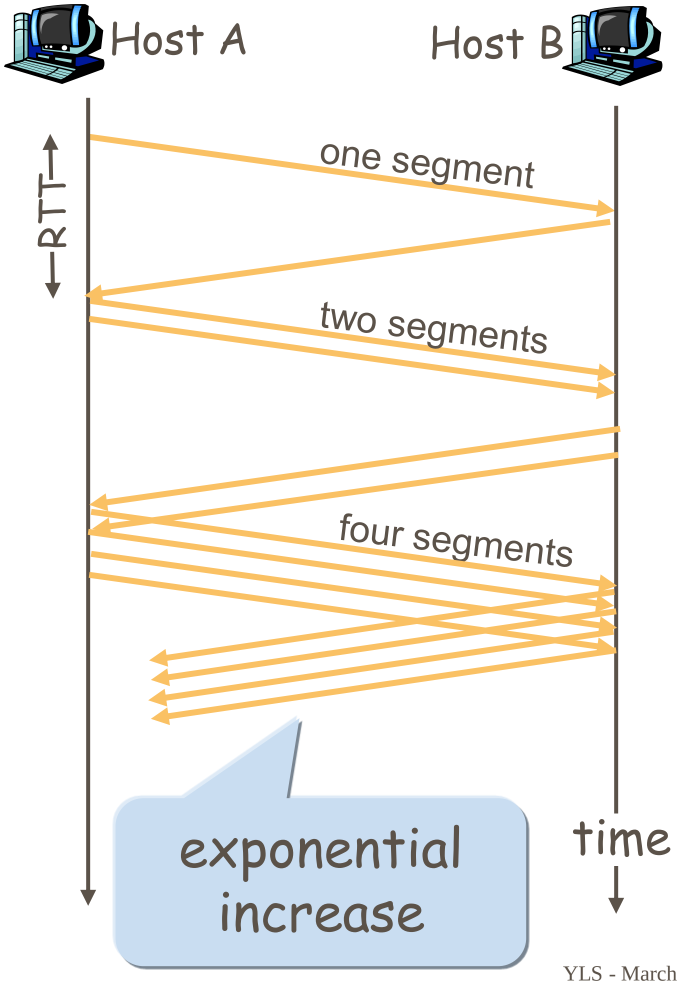
Slow-Start

- To get data flowing there must be acks to clock out packets; but to get acks there must be data flowing.
- Maintain a per connection state variable in the sender – "congestion window" *cwnd*
- "When to enter Slow-Start Phase?"
  - When a connection begins
  - After a timeout

TCP Slow Start

-> PROBE network maximum "throughput"!

- When connection begins, increase rate exponentially until first loss event:
  - double cwnd every RTT
  - done by incrementing cwnd for every ACK received
- Summary: initial rate is slow but ramps up exponentially fast



TCP Slow-Start (2/3)

- To get data flowing there must be acks to clock out packets; but to get acks there must be data flowing.

- Maintain a per connection state variable in the sender – "congestion window" *cwnd*

- "When to enter Slow-Start Phase?"
  - When a connection begins
  - After a timeout

Slow Start (cont'd)

- Algorithm –
  - When starting or restarting after a loss, set cwnd=1 packet.
  - One ack for each new data, i.e. cwnd =cwnd+1.
  - When sending, send the min(receiver's_advtiseWin, cwnd)
- Each time an ACK is received, *cwnd* is incremented by one segment size.
  - *cwnd* is maintained in bytes.
  - The *segment size* is announced by the *receiver*.

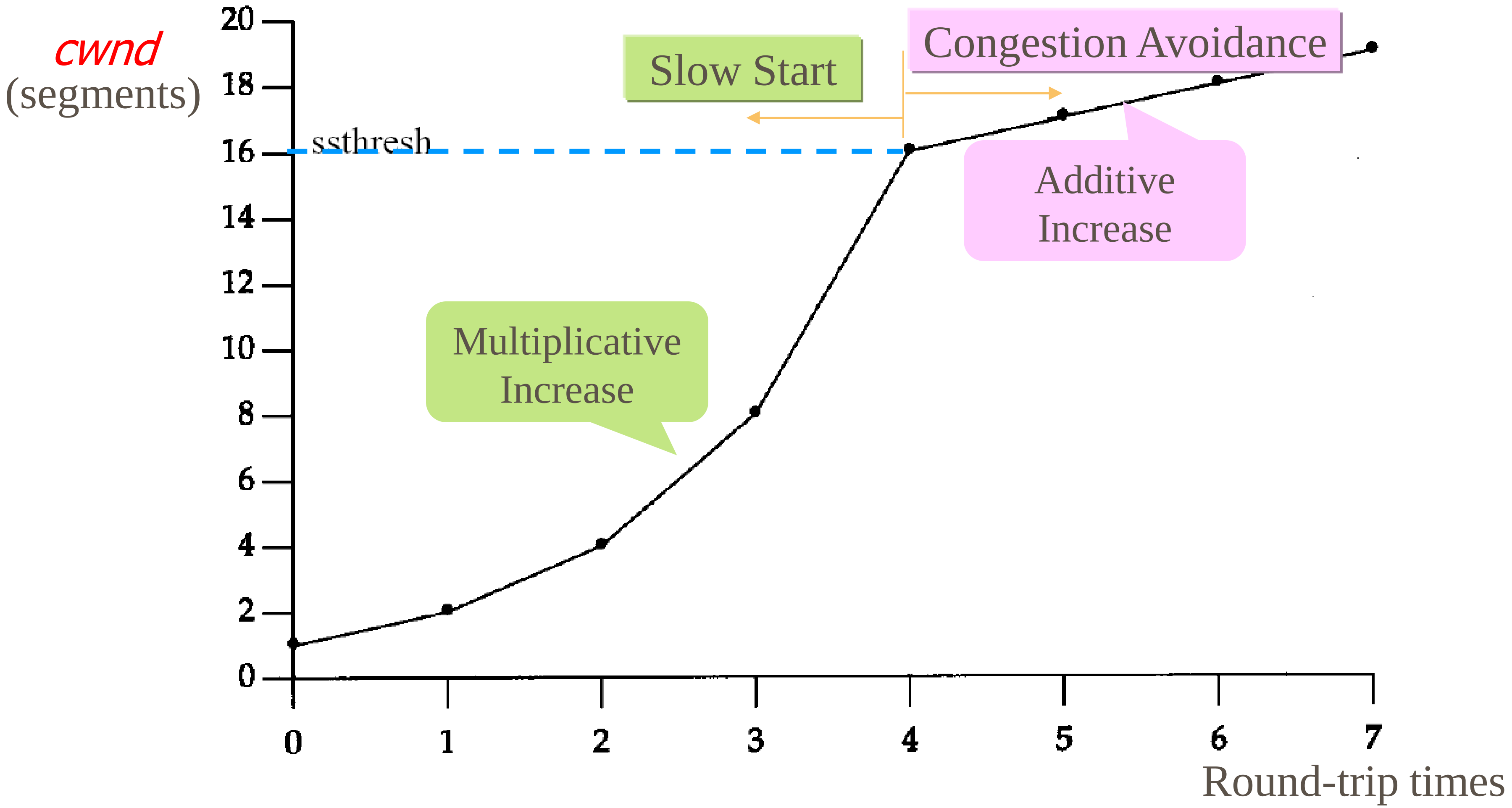
Congestion Avoidance

- Congestion is indicated by a *timeout* or the reception of *three* duplicate ACKs.
- The goal is to avoid increasing the window size too quickly and causing additional congestion.

Congestion Avoidance Algorithm

- Slow start phase
  - When a connection begins: ***cwnd*** is one segment and ***ssthresh*** (slow start threshold) is 65,535 bytes.
  - When congestion occurs, *ssthresh=cwnd/2*, cwnd=1
- Once *cwnd=ssthresh*, the connection enters the congestion avoidance phase.
  - On each ack for new data, cwnd=cwnd+1/cwnd (additive increase)
  - When sending, send the min(receiver's AdvertiseWinow, cwnd)

AIMD: *additive increase, multiplicative decrease*



※Visualization of slow start and congestion avoidance.

Duplicate ACKs

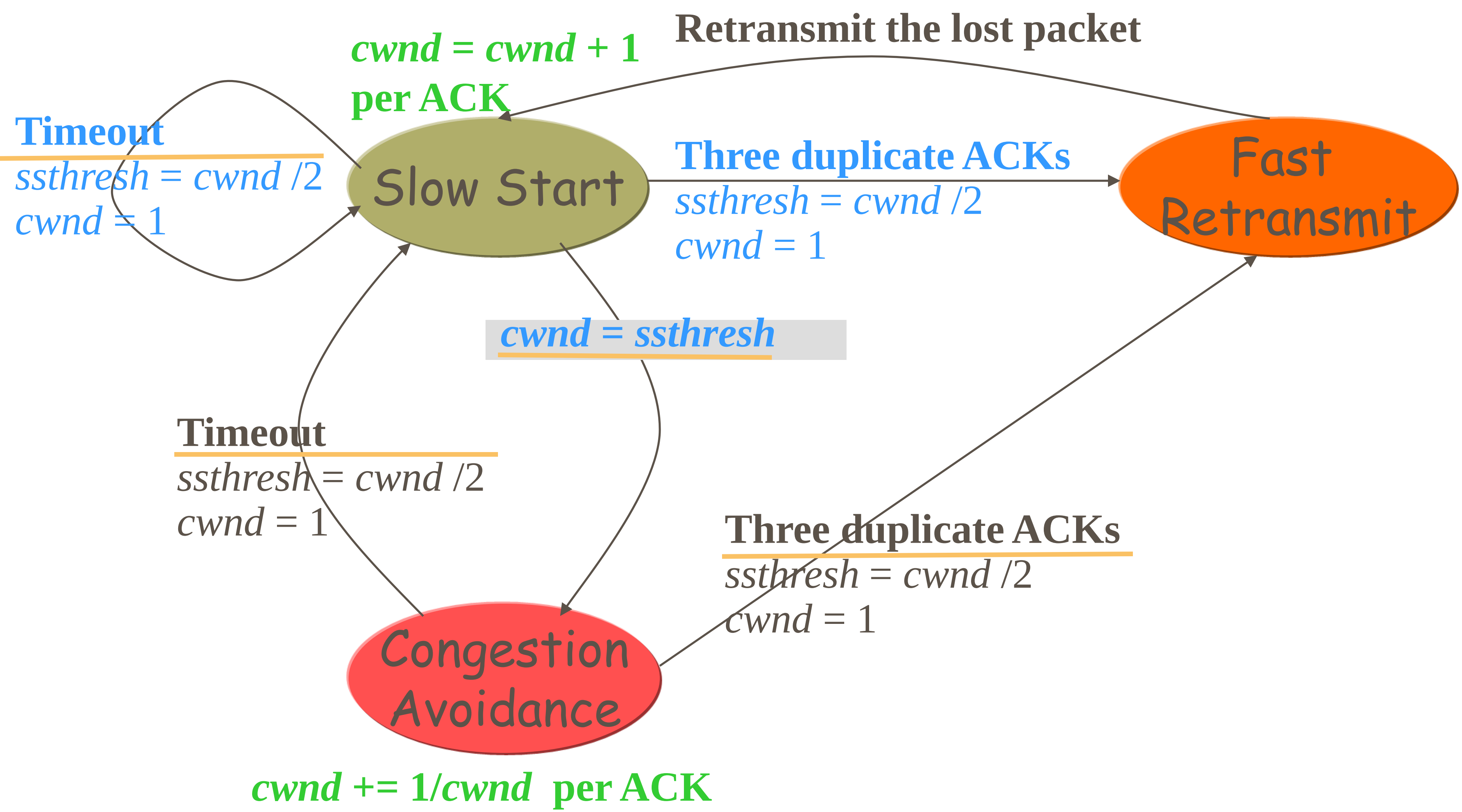
- If there are less than 3 duplicate ACKs, it is assumed that there is just a *reordering* of the segments.
- If 3 or more duplicate ACKs are received in a row, it is a *strong* indication that a segment has been lost.
- <u>Fast Retransmit</u> and <u>Fast Recovery</u>

  -> TCP-tahoe and TCP-reno

Fast Retransmit

- When 3 duplicate ACKs are received, a retransmission is performed *without* waiting for a retransmission timer to expire.
- ssthresh=cwnd/2 and cwnd = 1; (entering Slow Start phase)
- Retransmit the missing segment.

TCP Tahoe



Retransmit the lost packet
cwnd = cwnd + 1
per ACK
Timeout
ssthresh = cwnd /2
cwnd = 1
Slow Start
Three duplicate ACKs
ssthresh = cwnd /2
cwnd = 1
Fast Retransmit
cwnd = ssthresh
Timeout
ssthresh = cwnd /2
cwnd = 1
Three duplicate ACKs
ssthresh = cwnd /2
cwnd = 1
Congestion Avoidance
cwnd += 1/cwnd per ACK

TCP Tahoe

- After fast retransmit, goes to "slow-start" phase to probe the network again.
- To avoid congest the network.

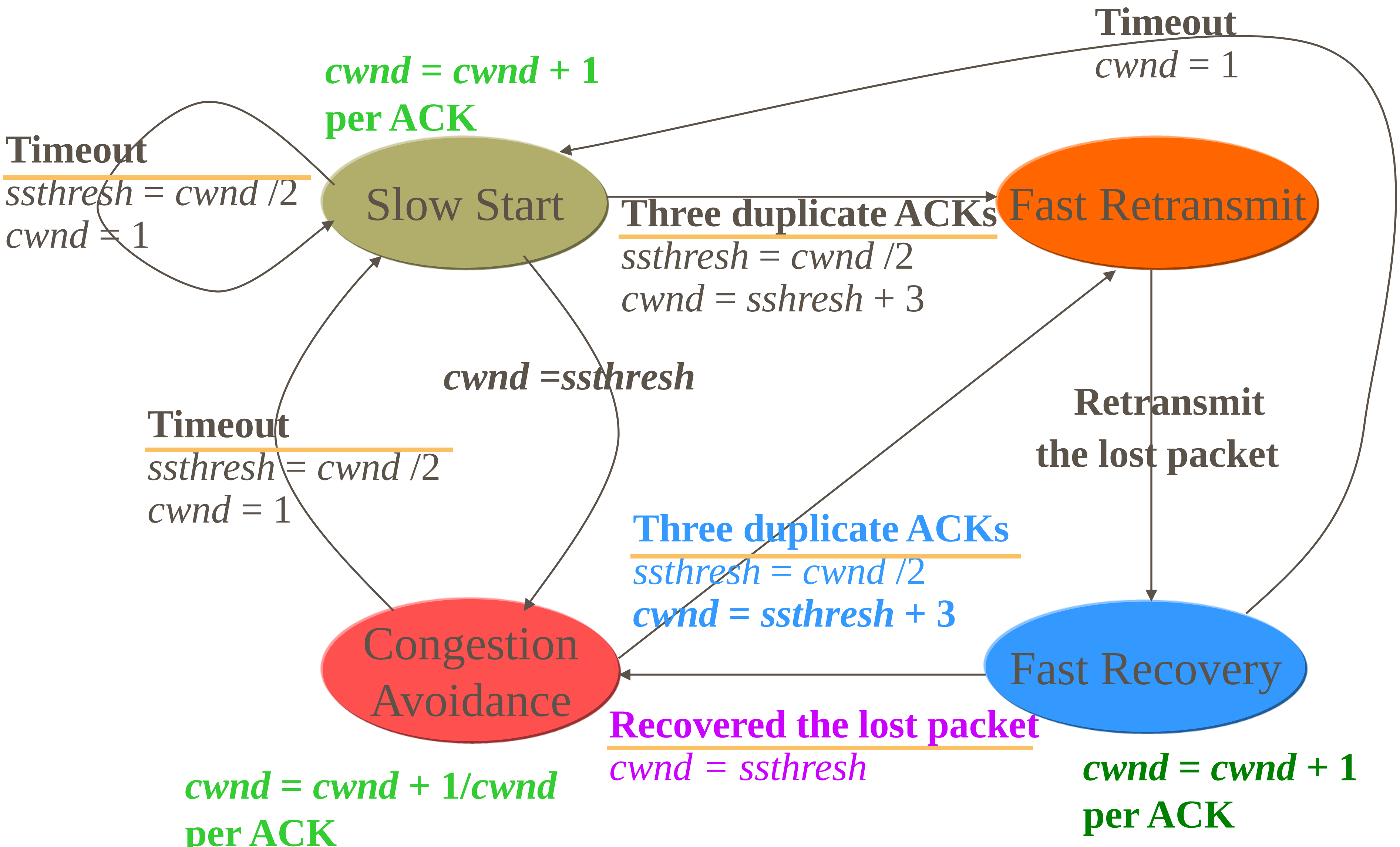
Fast Recovery

- Immediately after fast retransmit, instead of entering slow start, *congestion avoidance* is performed.
- To boot up throughput
- *ssthresh=cwnd/2; cwnd*=*ssthresh* + 3 segments
- Each time an ACK or a duplicate ACK arrives, increment *cwnd* by the segment size *cwnd++;*
- Allow to transmit new packet

Fast Recovery (cont'd)

- When the next ACK arrives that acknowledges the lost data,
  - set *cwnd* to *ssthresh*
  - enter congestion avoidance phase

TCP Reno



Timeout
cwnd = 1
cwnd = cwnd + 1
per ACK
Timeout
ssthresh = cwnd /2
cwnd = 1
Slow Start
Three duplicate ACKs
ssthresh = cwnd /2
cwnd = sshresh + 3
Fast Retransmit
cwnd =ssthresh
Retransmit
the lost packet
Timeout
ssthresh = cwnd /2
cwnd = 1
Three duplicate ACKs
ssthresh = cwnd /2
cwnd = ssthresh + 3
Congestion Avoidance
Fast Recovery
Recovered the lost packet
cwnd = ssthresh
cwnd = cwnd + 1/cwnd
per ACK
cwnd = cwnd + 1
per ACK

TCP-Reno: Congestion Window Size

(1) After every new acknowledgment

Slow start

if (CWND < SSTHRESH)

CWND ← CWND + 1

Congestion avoidance

else

CWND ← CWND + 1/CWND

(2) Upon RTO (retransmission timeout)

Begin of slow start

SSTHRESH ← CWND/2

CWND ← 1

(3) When NDUP (# of duplicate ACKs) exceeds 3

Begin of fast recovery

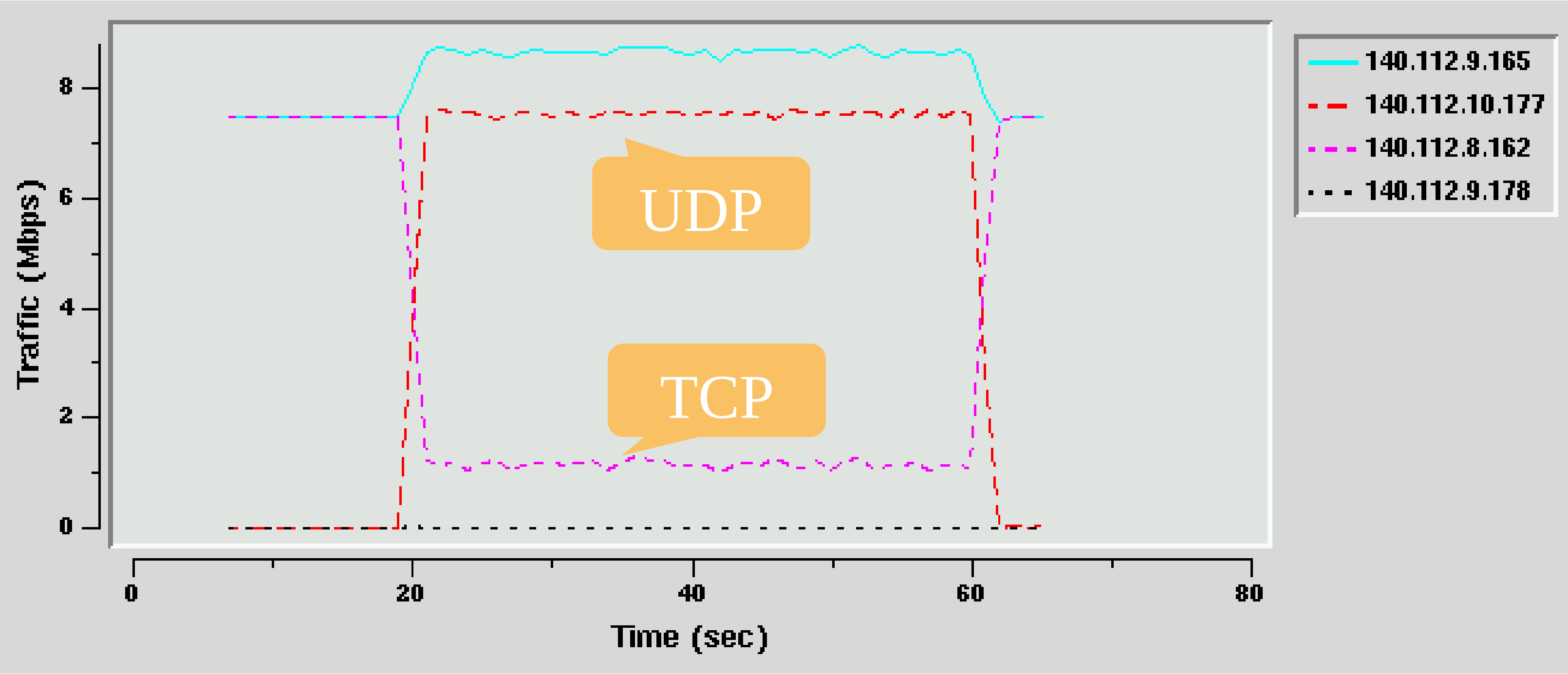
SSTHRESH ← CWND/2

CWND ← CWND/2 + 3

Competition of TCP connection with UDP flow

- Sender 1 (140.112.8.162)先以8 Mbps的速度送出TCP traffic
- 20秒後Sender 2 (140.112.10.177)再以8 Mbps的速度送出UDP traffic
- The buffer space is 100KB for both queues. There is no packet drop.
- After UDP traffic starts, TCP throughput drops to less than 2Mb，UDP has the rest。
- Possible cause: Receiver (140.112.9.165) fails to send ACKs to Sender 1，causing Sender 1以為發生packet loss，因此把window size調降，而使得傳送的速率下降。

Competition of TCP connection with UDP flow (cont'd)



References

- A. S. Tanenbaum, "Computer Networks," 3rd edition, section 5.3, 1996 (except 5.3.2).
- C-Q Yang and A. V. S. Reddy, "A Taxonomy for Congestion Control Algorithms in Packet Switching Networks," IEEE Network, pp. 34-45, July/August 1995.