End-to-End Transport Over Wireless II: Snoop and Explicit Loss Notification

COS 463: Wireless Networks
Lecture 3
Kyle Jamieson

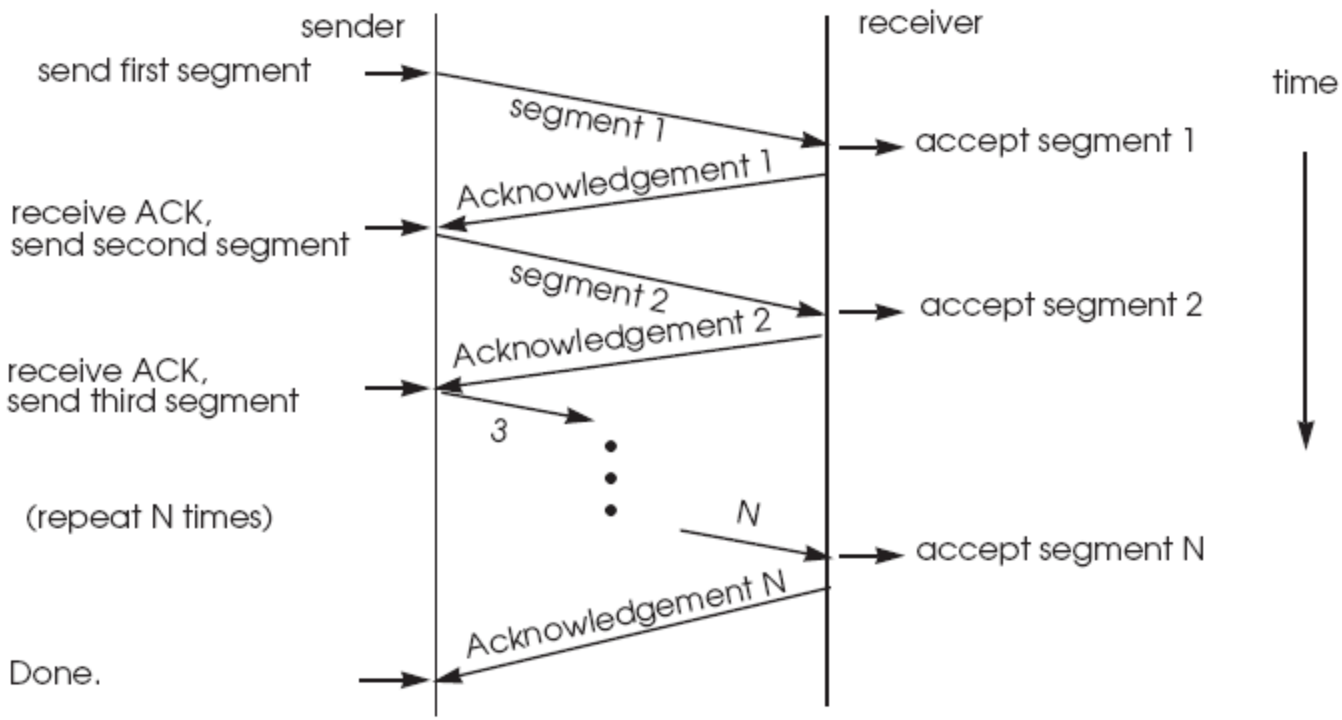
[Various parts adapted from S. Das, B. Karp, N. Vaidya]

Today

1. **Transmission Control Protocol (TCP), Part II**
   - **Window-based flow control**
   - Retransmissions and congestion control

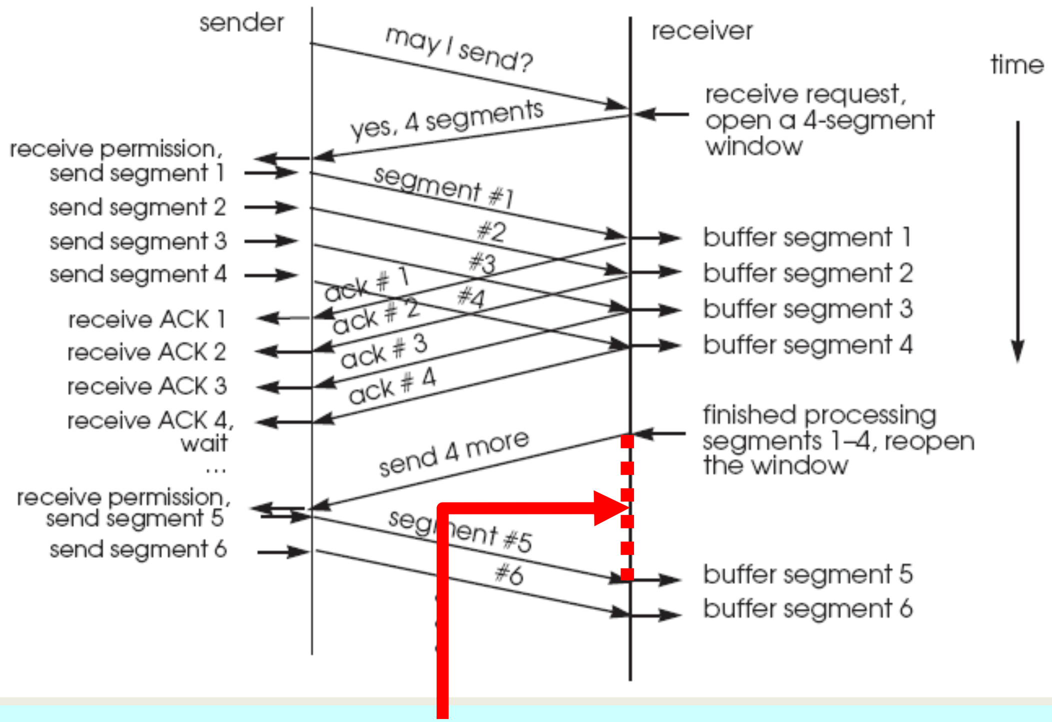
2. TCP over Wireless
   - TCP Snoop
   - Explicit Loss Notification

Window-Based Flow Control: Motivation



- Suppose sender sends one packet, awaits ACK, repeats…
- **Result:** **At most one packet sent, per RTT**
- *e.g.,* 70 ms RTT, 1500-byte packets → **Max t'put: 171 Kbps**

Idea: Pipeline Transmissions
(*Fixed Window-Based Flow Control*)



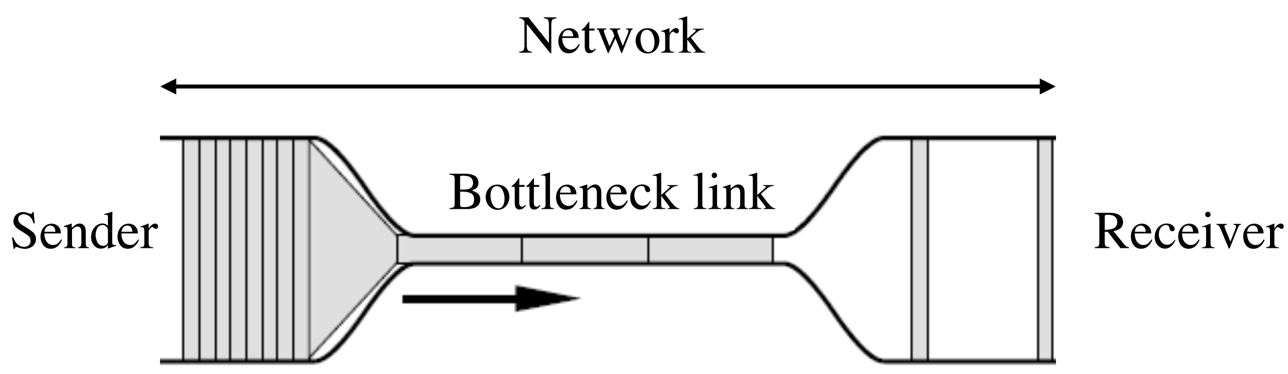
- Di ... CKs
  – ...
  – ... w

But: RTT idle time from grant of new window to data arrival at receiver

Better approach (TCP): sliding window, extends as each ACK returns, so no idle time!

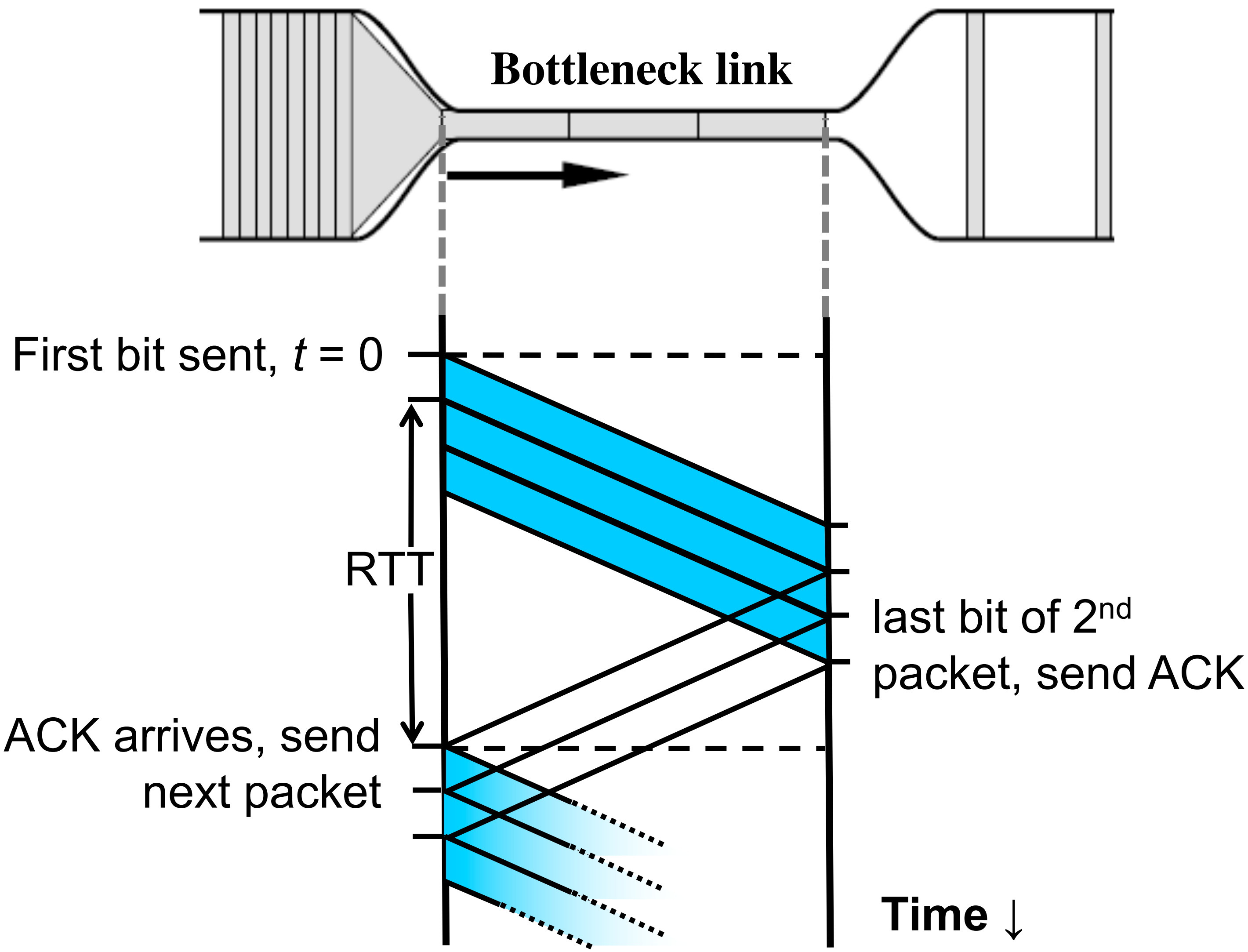
Choosing Window Size

- **Network *bottleneck:*** link of **slowest rate** along path between sender and receiver

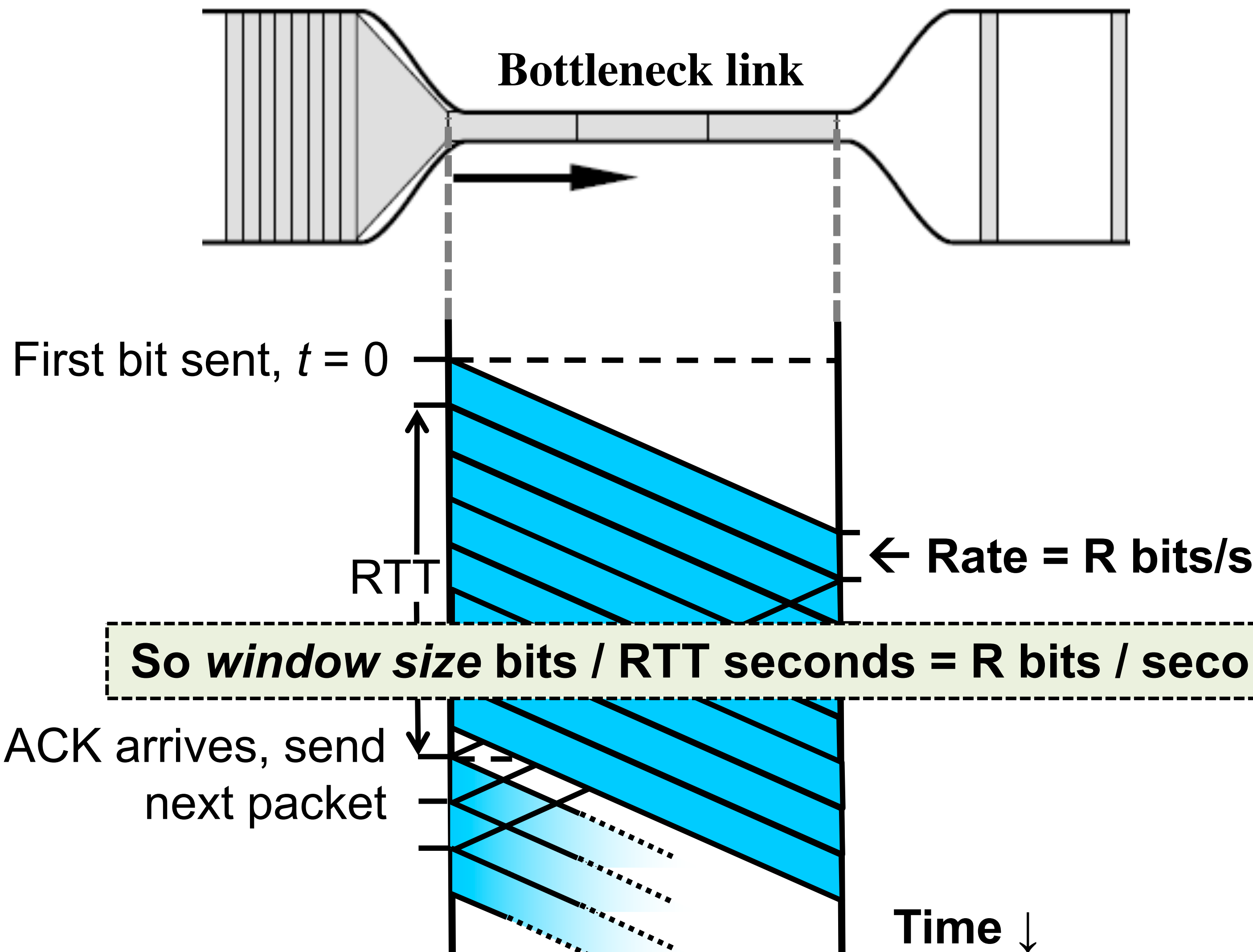


- What size sender window keeps the pipe full?
  - **Window too small: can't fill pipe**
  - **Window too large: unnecessary network load/queuing/loss**

Increasing utilization with pipelining



The Bandwidth-Delay Product



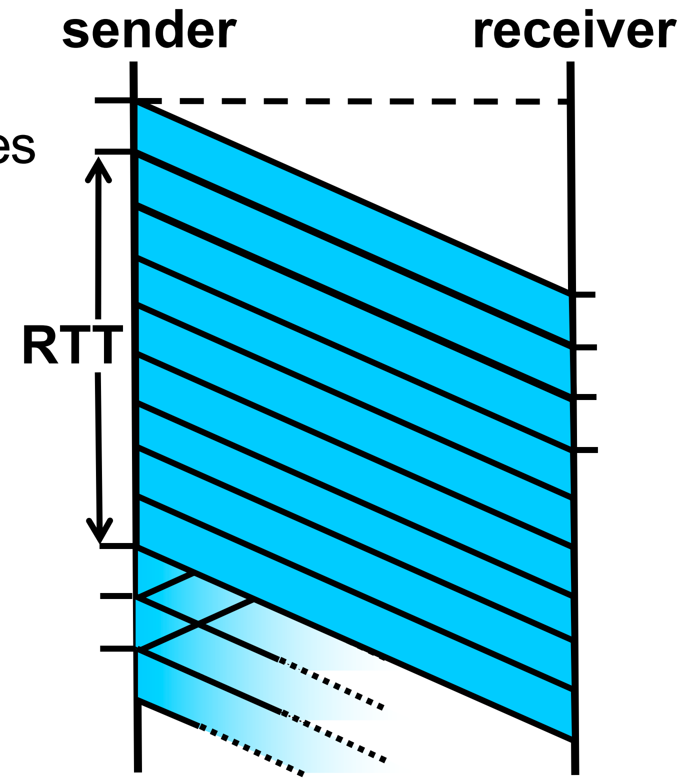
So *window size* bits / RTT seconds = R bits / second

The bandwidth-delay product

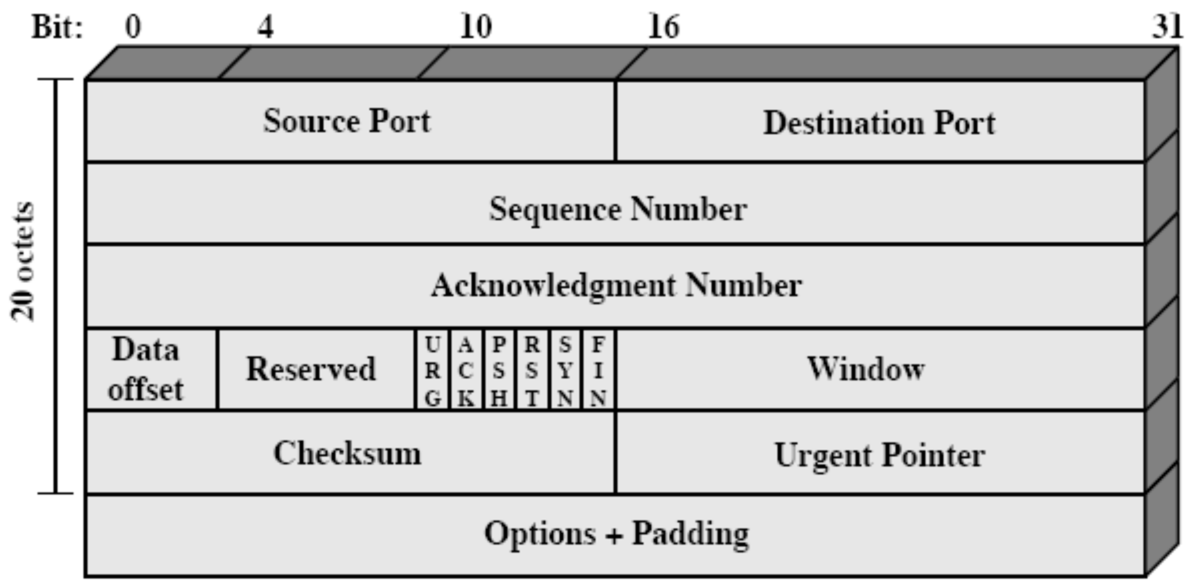
- **Goal: window size = RTT × bottleneck rate**

e.g.: to achieve a bottleneck rate of 1 Mbps across a 70 ms RTT, need a window size:

$$(10^6 \text{ bps} \times .07 \text{ s}) = 70 \text{ Kbits} = 8.75 \text{ Kbytes}$$



TCP Packet Header



- TCP header: 20 bytes long
- Checksum covers TCP packet + **"pseudo header"**
  - IP header source and destination addresses, protocol
  - Length of TCP segment (TCP header + data)

TCP Header Details

- Connections **inherently bidirectional;** all TCP headers carry **both data & ACK sequence numbers**
- 32-bit **sequence numbers** are in units of **bytes**
- Source and destination ***port numbers***
  - Multiplexing of TCP by applications
  - **UNIX:** local ports below 1024 **reserved (only root may use)**
- ***Window*** **field:** advertisement of **number of bytes advertiser willing to accept**

TCP: Data Transmission

- Each byte **numbered sequentially (modulo 2^{32})**

- **Sender buffers data** in case retransmission required
- **Receiver buffers data** for in-order reassembly

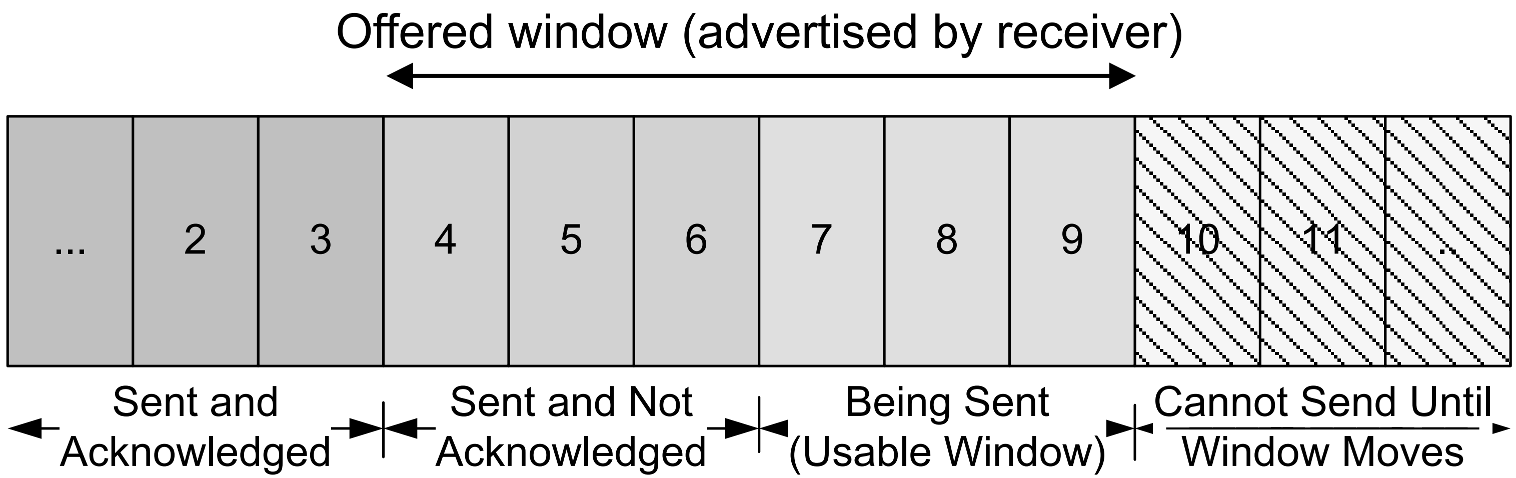
- **Sequence number (seqno) field** in TCP header indicates first user payload byte in packet

TCP: Receiver functionality

- Receiver indicates ***offered window size W*** explicitly to sender in **`window`** field in TCP header
  – Corresponds to **available buffer space** at receiver

- Receiver sends ***cumulative ACKs:***
  – ACK number in TCP header names **highest contiguous byte number received** thus far, +1
  – one ACK per received packet, **or:**
    - ***Delayed ACK:*** receiver batches ACKs, sends one for every pair of data packets (200 ms max delay)

TCP: Sender's Window



- **Usable window** at sender:
  - Left edge advances as packets sent
  - Right edge advances as receive window updates arrive

Today

1. **Transmission Control Protocol (TCP)**
   - Window-based flow control
   - **Retransmissions and congestion control**

2. TCP over Wireless
   - TCP Snoop
   - Explicit Loss Notification

TCP: Retransmit Timeouts

- **Recall:** Sender sets timer for each sent packet
  - Expected time for ACK to return: RTT
  - when ACK returns, timer canceled
  - if timer expires before ACK returns, packet resent

- TCP estimates RTT using measurements mi from timed packet/ACK pairs
  - $RTT_i = ((1 - \alpha) \times RTT_{i-1} + \alpha \times m_i)$

- **Original TCP retransmit timeout: $RTO_i = \beta \times RTT_i$**
  - **original TCP: $\beta = 2$**

Mean and Variance: Jacobson's RTT Estimator

- Above 30% link load at router, $\beta \times RTT_i$ **retransmits too early!**
  - Response to **increasing load:** waste bandwidth on duplicate packets; **result: congestion collapse!**

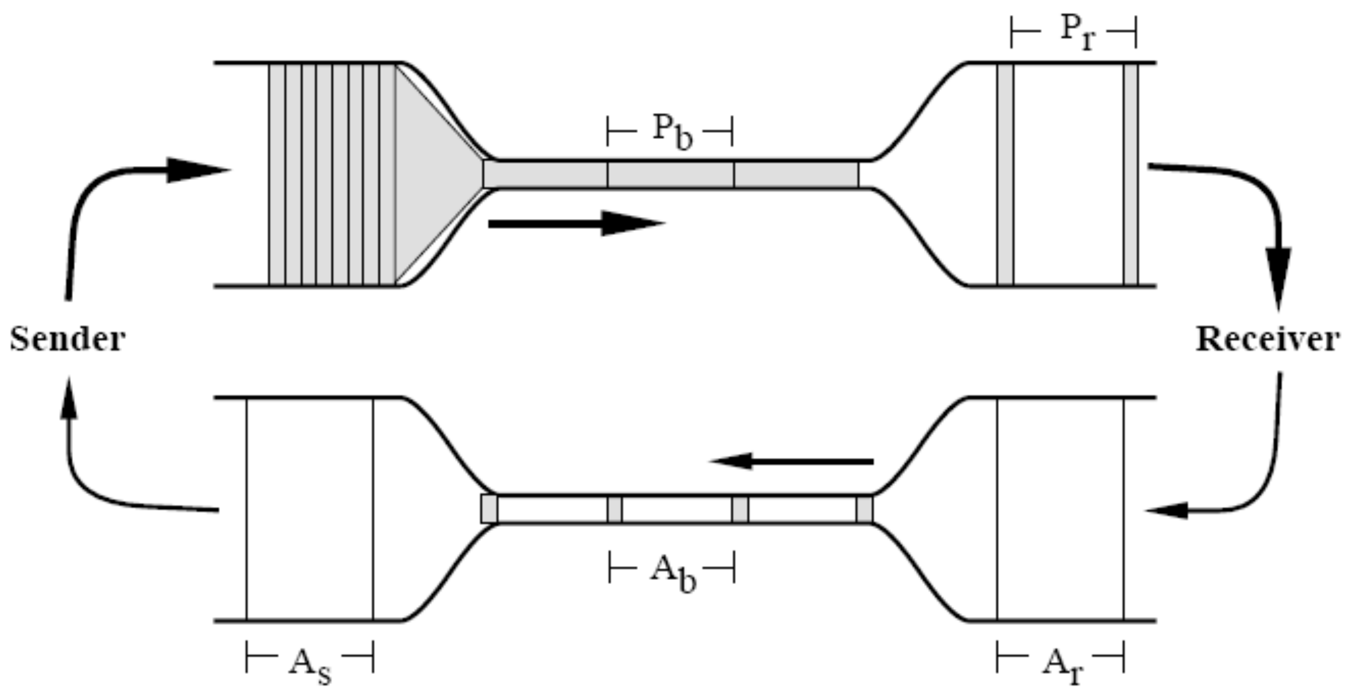
- **Idea** [Jacobson 88]: Estimate ***mean deviation*** **v_i,** (EWMA of $|m_i - RTT_i|$), a stand-in for variance:

  $$v_i = v_{i-1} \times (1-\gamma) + \gamma \times |m_i - RTT_i|$$

  - Then use retransmission timeout **$RTO_i = RTT_i + 4v_i$**

Mean and Variance RTT estimator used by all modern TCPs

Self-Clocking Transmission



- Self-clocking transmission: Conservation of Packets
  - each ACK returns, one data packet sent
  - spacing of returning ACKs: matches spacing of packets in time at **slowest link on path**

Goals in Congestion Control

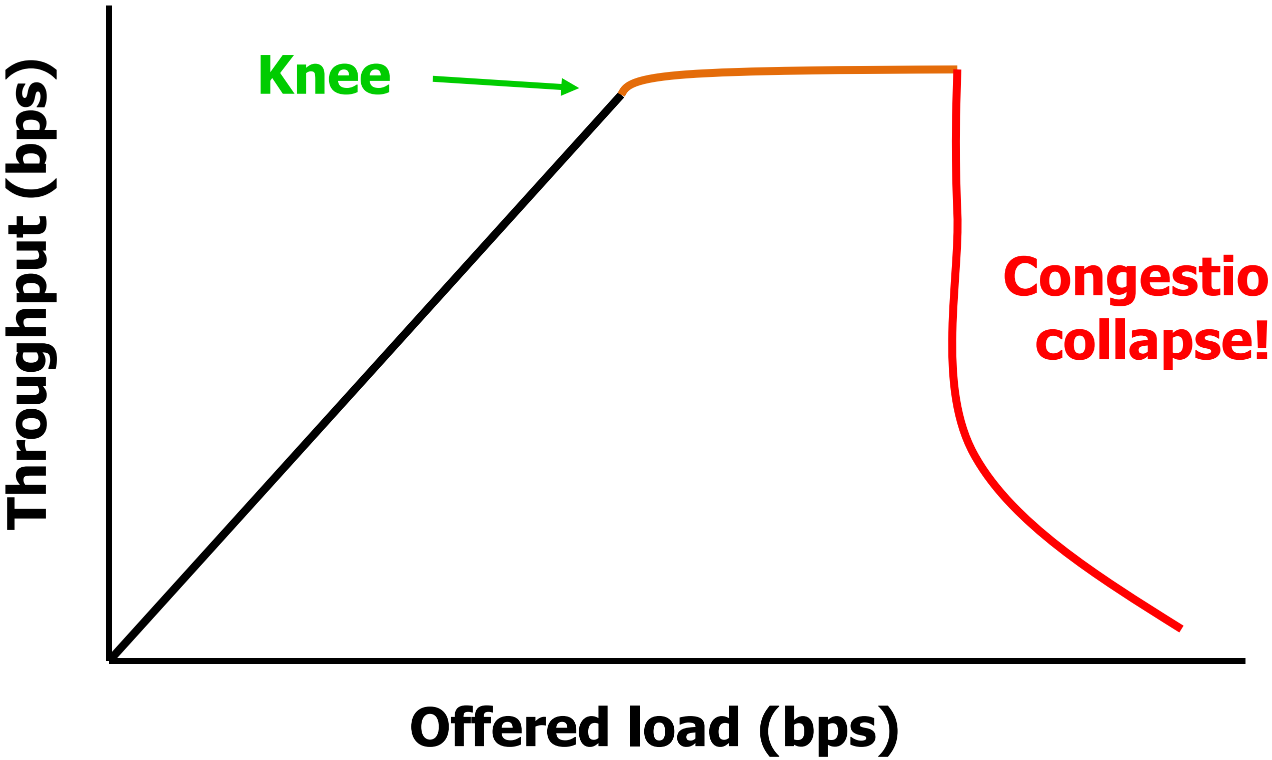
1. Achieve **high utilization** on links; don't waste capacity!

2. Divide bottleneck link capacity **fairly among users**

3. Be **stable:** converge to steady allocation among users

4. Avoid **congestion collapse**

Congestion Collapse



- Cliff behavior observed in [Jacobson 88]

Congestion Requires Slowing Senders

- Bigger buffers cannot prevent congestion: **senders must slow down**
- Absence of ACKs **implicitly indicates congestion**
- TCP sender's window size determines sending rate

- **Recall:** Correct window size is **bottleneck link bandwidth-delay product**

- How can the sender learn this value?
  - **Search** for it, by adapting window size
  - **Feedback** from network: ACKs return (window OK) or do not return (window too big)

Reaching Equilibrium: Slow Start

- At connection start, sender sets ***congestion window size,*** **`cwnd`**, to **`pktSize`** (one packet's worth of bytes), not whole window

- Sender sends up to **min(`cwnd`, `W`)**
  - Upon return of each ACK, increase **`cwnd`** by **`pktSize`** bytes until W reached
  - "Slow" means **exponential window increase!**

- Takes $\log_2(W / \mathbf{pktSize})$ RTTs to reach receiver's advertised window size W

Avoiding Congestion: Multiplicative Decrease

- Recall sender uses window of size **min(cwnd, W)**, where W is receiver's advertised window

- Upon **timeout** for sent packet, sender **presumes packet lost to congestion,** and:
  1. sets ssthresh = cwnd / 2
  2. sets cwnd = pktSize
  3. uses slow start to grow cwnd up to ssthresh

- **End result:** **cwnd = cwnd / 2**, via slow start

Taking Your Fair Share: Additive Increase

- **Drops indicate** sending **more** than fair share of bottleneck
- **No feedback** to indicate using **less** than fair share

- **Solution:** **Speculatively increase window size** as ACKs return
  - **Additive increase:** For each returning ACK,
    $cwnd = cwnd + (pktSize \times pktSize) / cwnd$
  - Increases cwnd by ≈ pktSize bytes per RTT

Combined algorithm: Additive Increase, Multiplicative Decrease (AIMD)

Refinement: Fast Retransmit (I)

- Sender **must wait well over RTT** for timer to expire before loss detected

- TCP's minimum retransmit timeout: **1 second**

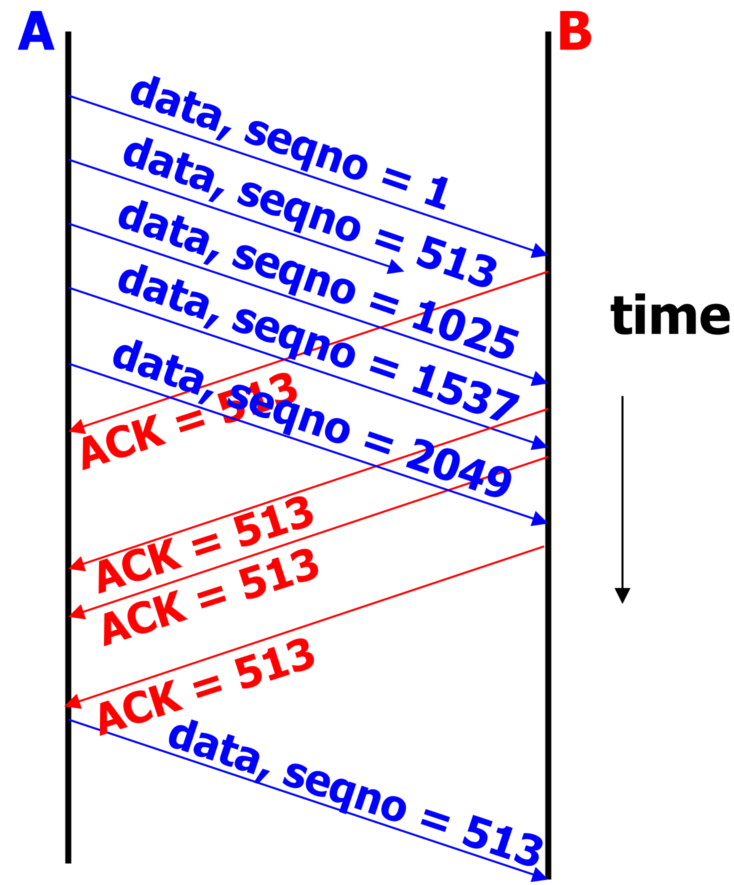
- **Another indicator of loss:**
  - Suppose sender sends: 1, 2, 3, 4, 5 (...but **2 is lost**)
  - Receiver receives: 1, 3, 4, 5
  - Receiver sends cumulative ACKs: 2, 2, 2, 2
    - Loss causes ***duplicate ACKs***

Fast Retransmit (II)

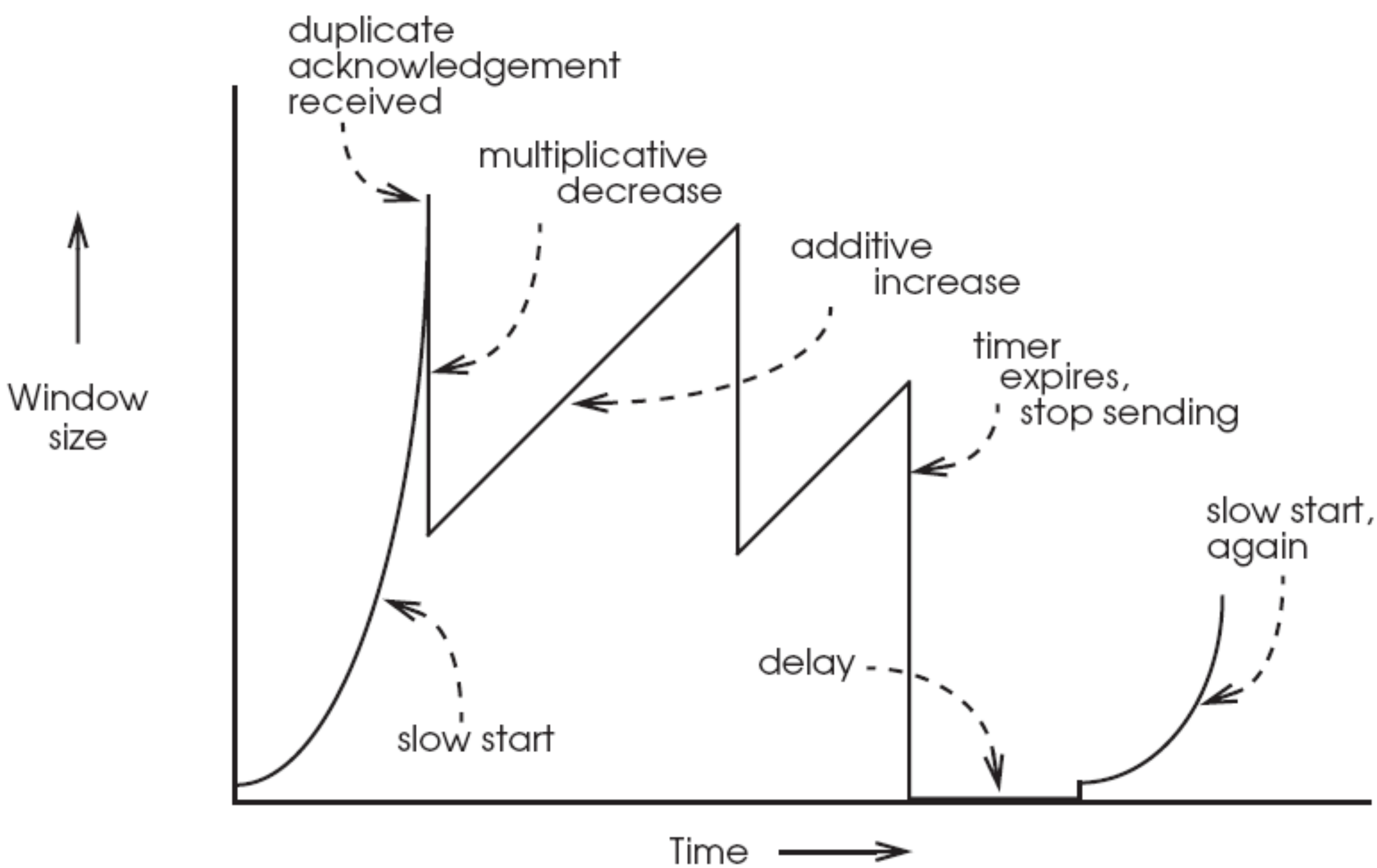
- Upon arrival of **three duplicate ACKs,** sender:

1. **sets** cwnd = cwnd / 2
2. **retransmits** "missing" packet
3. **no slow start**

- Not only loss causes dup ACKs
  - **Packet reordering, too**



AIMD in Action



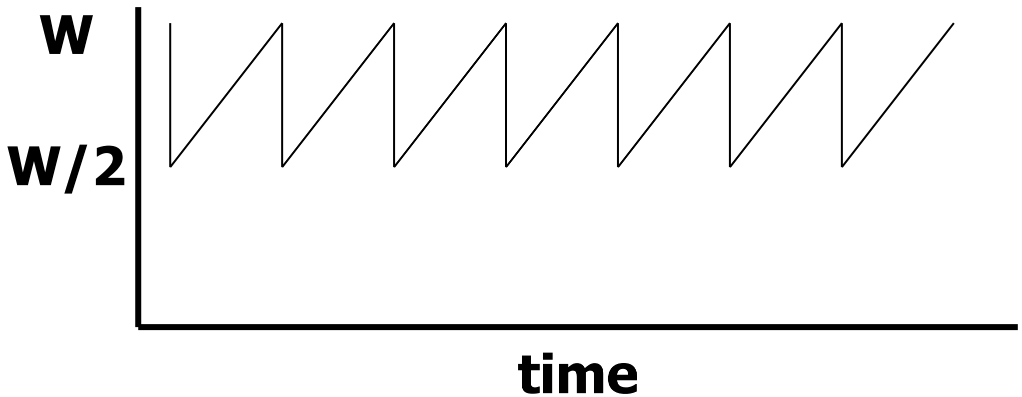
Modeling Throughput, Loss, and RTT

- *How do **packet loss rate** and **RTT** affect **throughput** TCP achieves?*

- **Assume:**

1. **Only fast retransmits**
2. **No timeouts (so no slow starts in steady-state)**

Evolution of Window Over Time



- **Average window size: ¾W**
- One window of packets is sent per RTT
- Bandwidth:
  - **¾W** packets per RTT
  - **(¾W x packet size) / RTT bytes per second**
  - W depends on loss rate…

Window Size Versus Loss

- Assume no delayed ACKs, fixed RTT
- `cwnd` grows by one packet per RTT
  - So it takes **W/2 RTTs** to go from window size W/2 to window size W; this period is one ***cycle***
- How many packets sent in total, in a cycle?
  - (¾W packets / RTT) x (W/2 RTTs) = $3W^2/8$ packets
- One loss per cycle (as window reaches W)
  - So, the **packet loss rate *p*** = $8/3W^2$
  - $W = \sqrt{(8/3p)}$

Throughput, Loss, and RTT Model

- $W = \sqrt{(8/3p)} = (4/3) \times \sqrt{(3/2p)}$
- Recall, **bandwidth *B*** = (3W/4 x packet size) / RTT

$$B = \text{packet size} / (RTT \times \sqrt{(2p/3)})$$

- **Consequences:**

1. Increased **loss quickly reduces throughput**
2. At same bottleneck, flow with **longer RTT** achieves **less throughput** than flow with shorter RTT!

Stretch Break, Q&A

- Suppose a Princeton Plasma Physics Lab experiment is generating scientific data at 5 Gbit/sec
- Want to send this data across the Internet to CERN, Switzerland for analysis with a 5 Gbit/s backbone link
  - Arrange for 10 Gbit/s links from PPPL to Internet Service Provider (ISP), and ISP to CERN
  - PPPL-CERN round trip time is 200 milliseconds

How large would the TCP sender's send window need to be at a minimum in order to achieve a throughput of 5 Gbit/s?

Today

1. Transmission Control Protocol (TCP) primer, cont'd

2. **TCP over Wireless**
   - **TCP Snoop**
   - Explicit Loss Notification

Review: TCP on Wireless Links

- TCP interprets any **packet loss** as a **sign of congestion**
  - TCP sender **reduces congestion window**

- On **wireless links**, packet loss can also occur due to random channel errors, or interference
  - Temporary loss **not due to congestion**
  - Reducing window **may be too conservative**
  - Leads to **poor throughput**

Review: Two Broad Approaches

1. **Mask wireless losses from TCP sender**
   – Then TCP sender will not reduce congestion window

   – Split Connection Approach
   – **TCP Snoop**

2. Explicitly notify TCP sender about cause of packet loss

TCP Snoop: Introduction

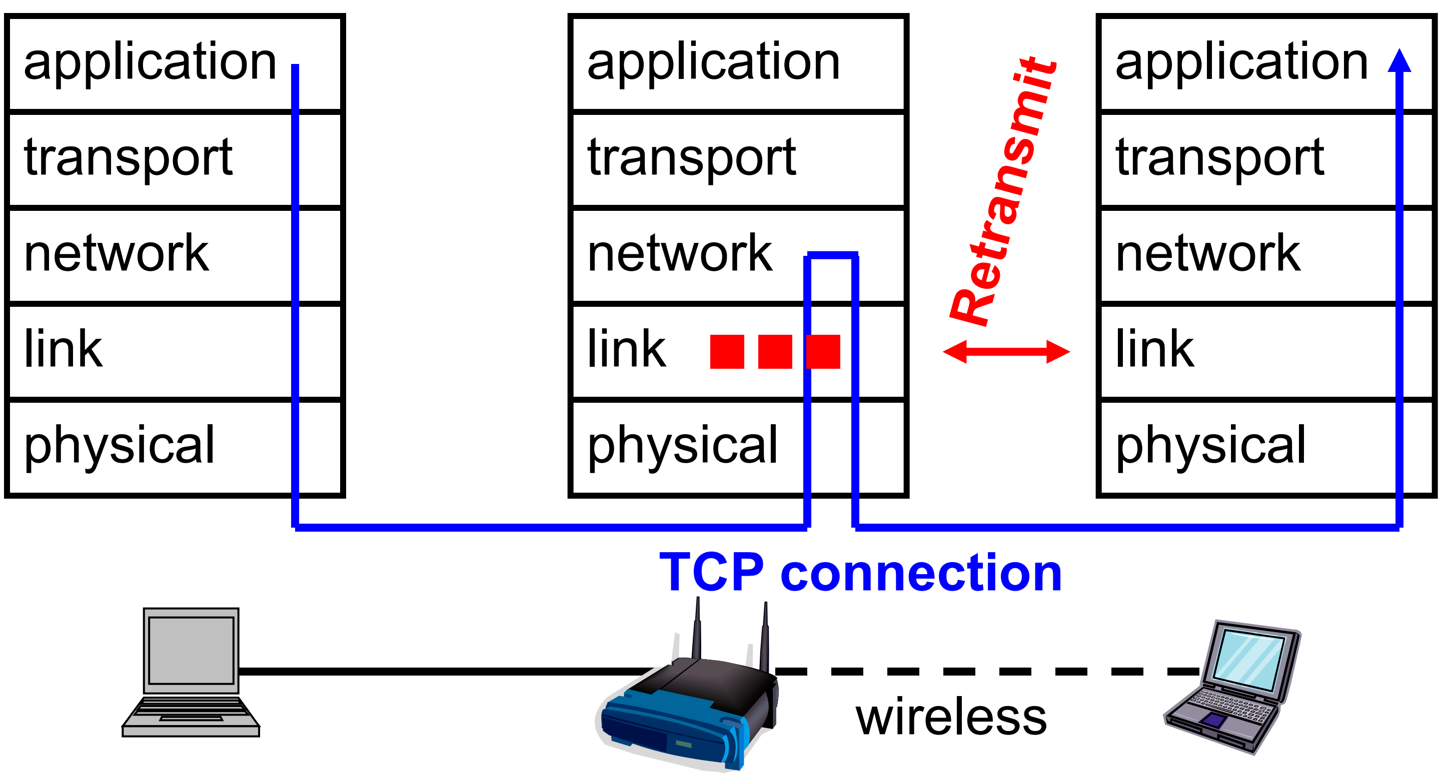
- Removes most significant problem of split connection: **breaking end-to-end semantics**
  - No more split connection
  - **Single end-to-end connection** like regular TCP

- TCP Snoop only modifies the AP

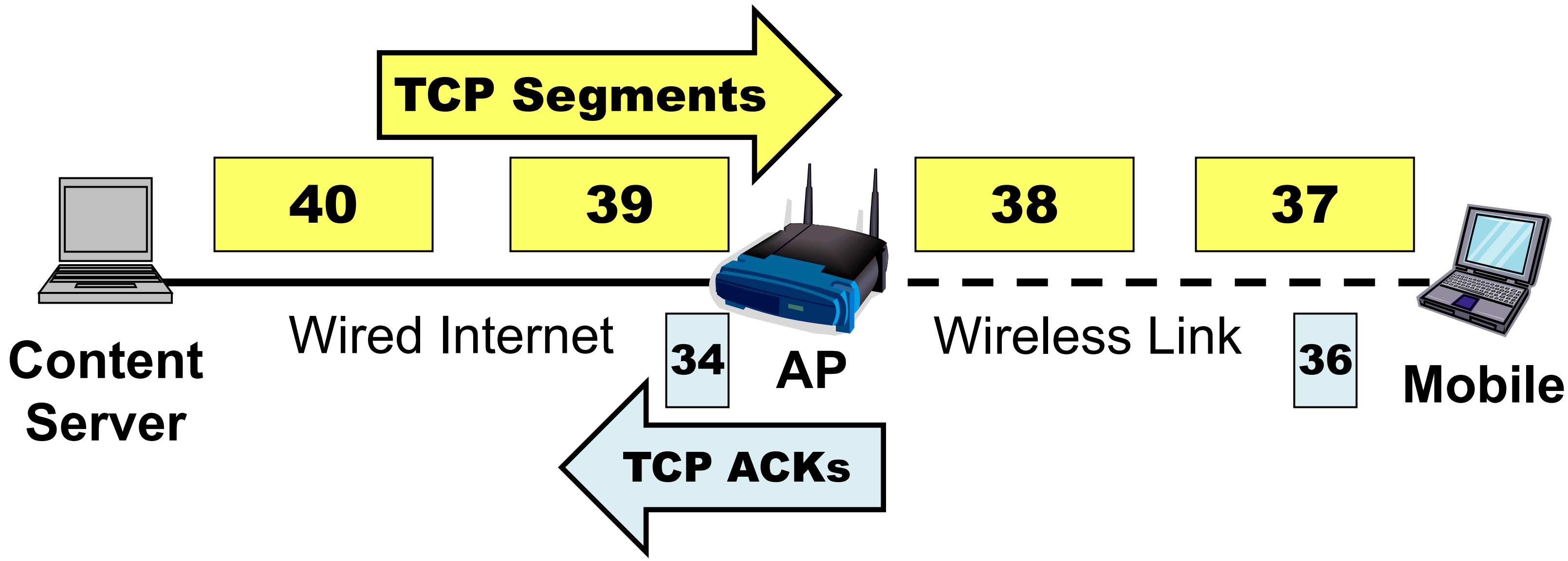
- **Basic Idea (Downlink traffic):**
  - AP **"snoops" on TCP traffic** to and from the mobile
    - **Quickly retransmits** packets it thinks may be lost over the wireless link

Snoop Protocol: High-level View

■ Per TCP-connection state



TCP Snoop: Downlink traffic case



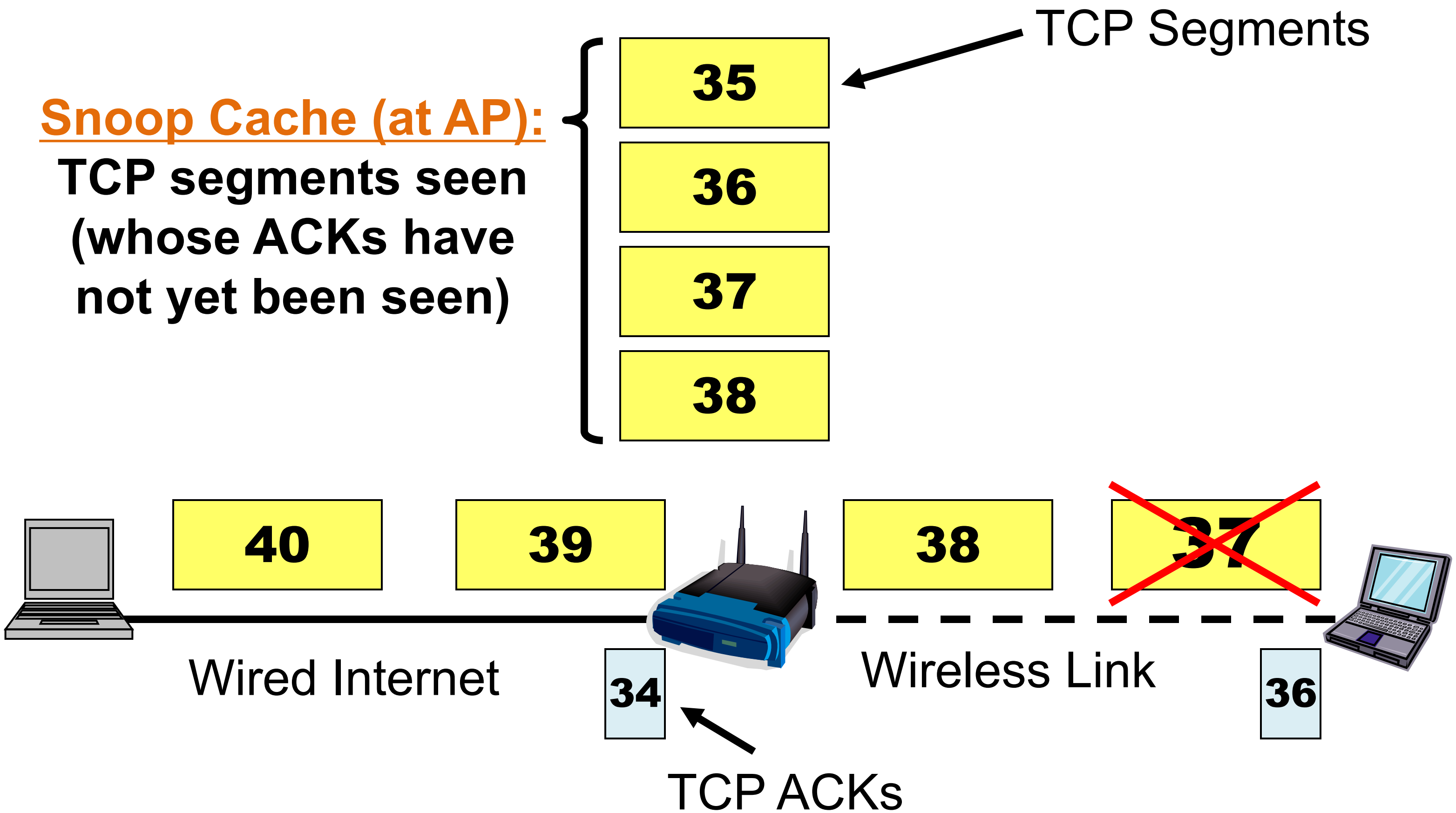
- AP **buffers downlink** TCP segments
  - Until it receives **corresponding ACK** from mobile
- AP **snoops on uplink** TCP ACKs
  - Detects downlink wireless TCP segment loss via duplicate ACKs or time-out

TCP Snoop Goal: Recover wireless downlink loss

- When AP detects a lost TCP segment:
  - **Locally, quickly retransmit** that segment over the wireless link
  - **Minimize duplicate ACKs** flowing back to server

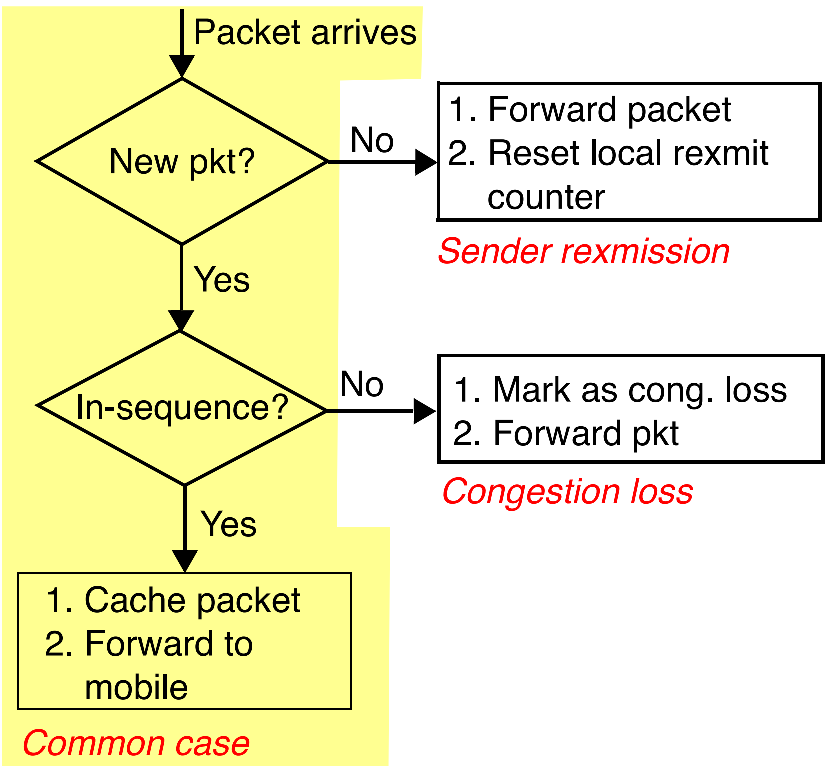
- **Goal:** Content server **unaware of wireless loss and retransmission**
  - **No reduction in `cwnd`**

TCP Snoop: Downlink Example

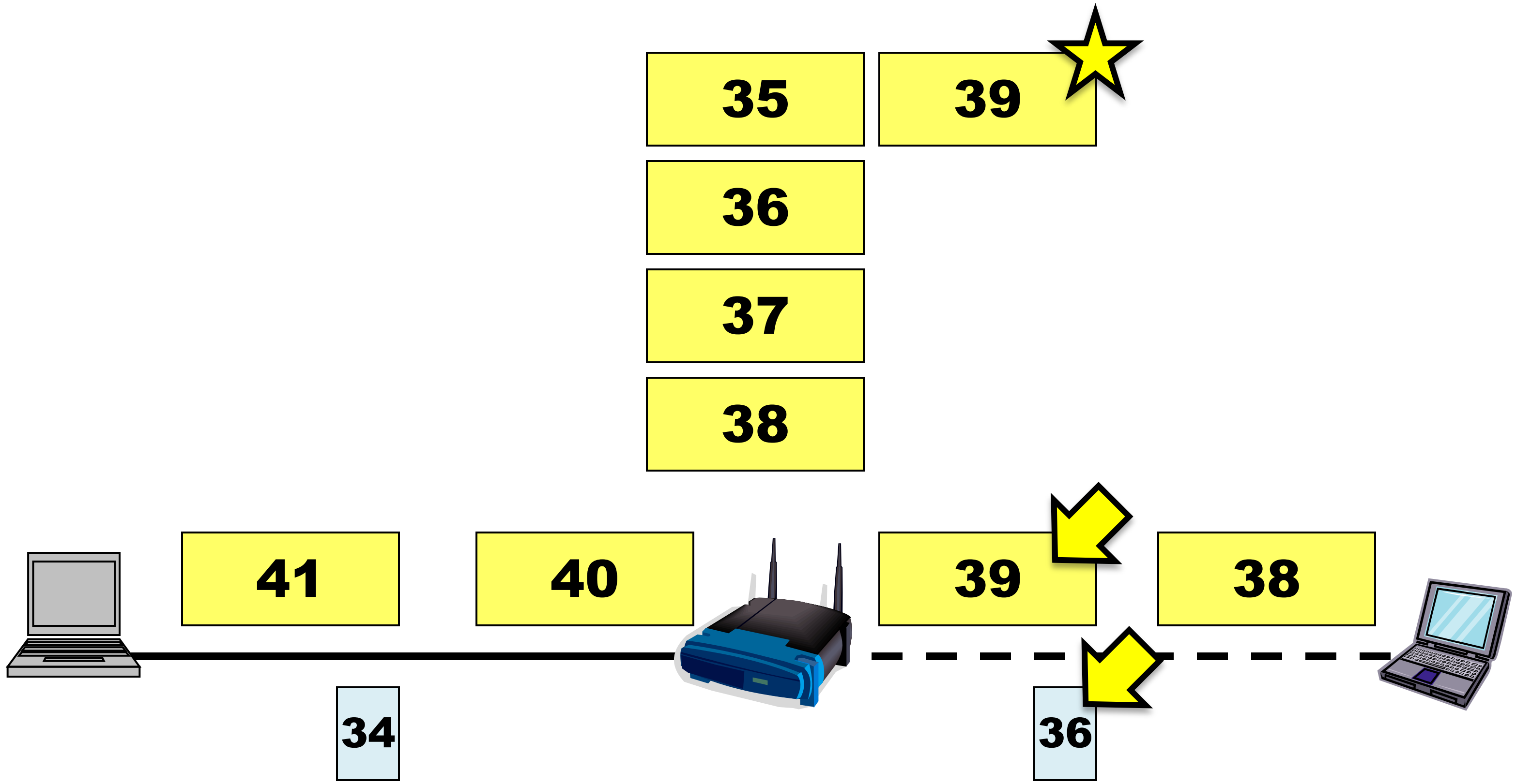


Downlink traffic operation, at Snoop AP

Downlink TCP segments:



TCP Snoop: Downlink example



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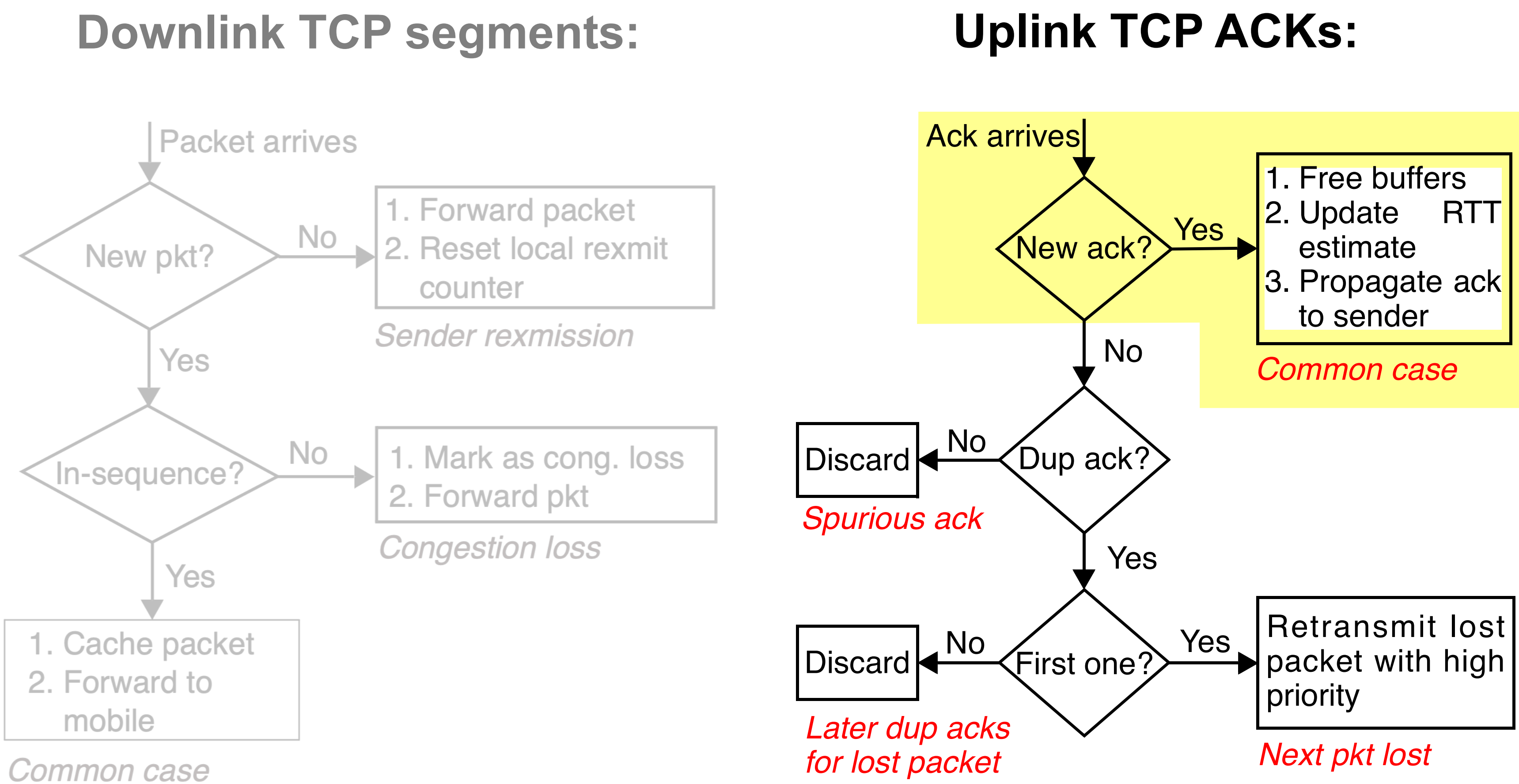
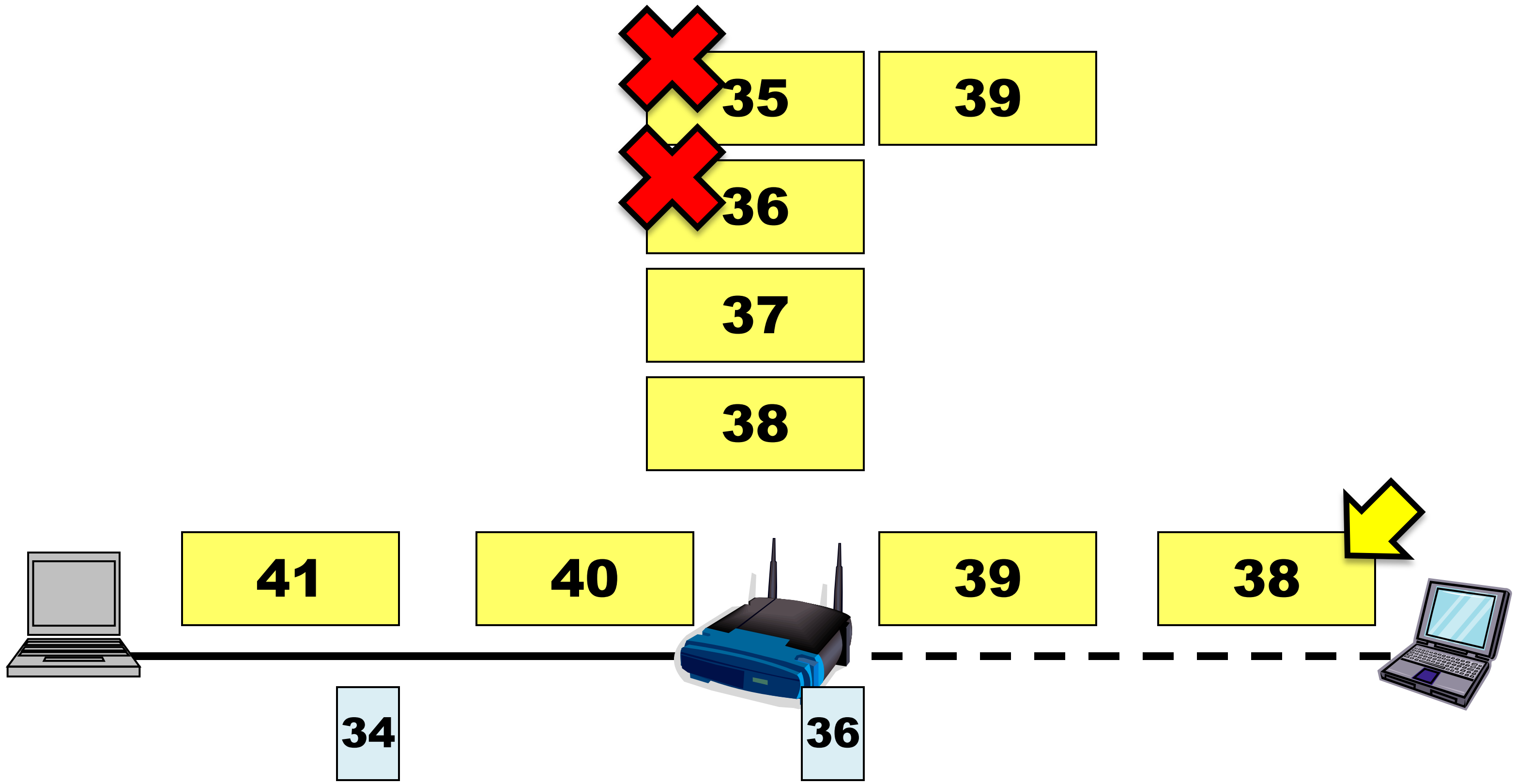
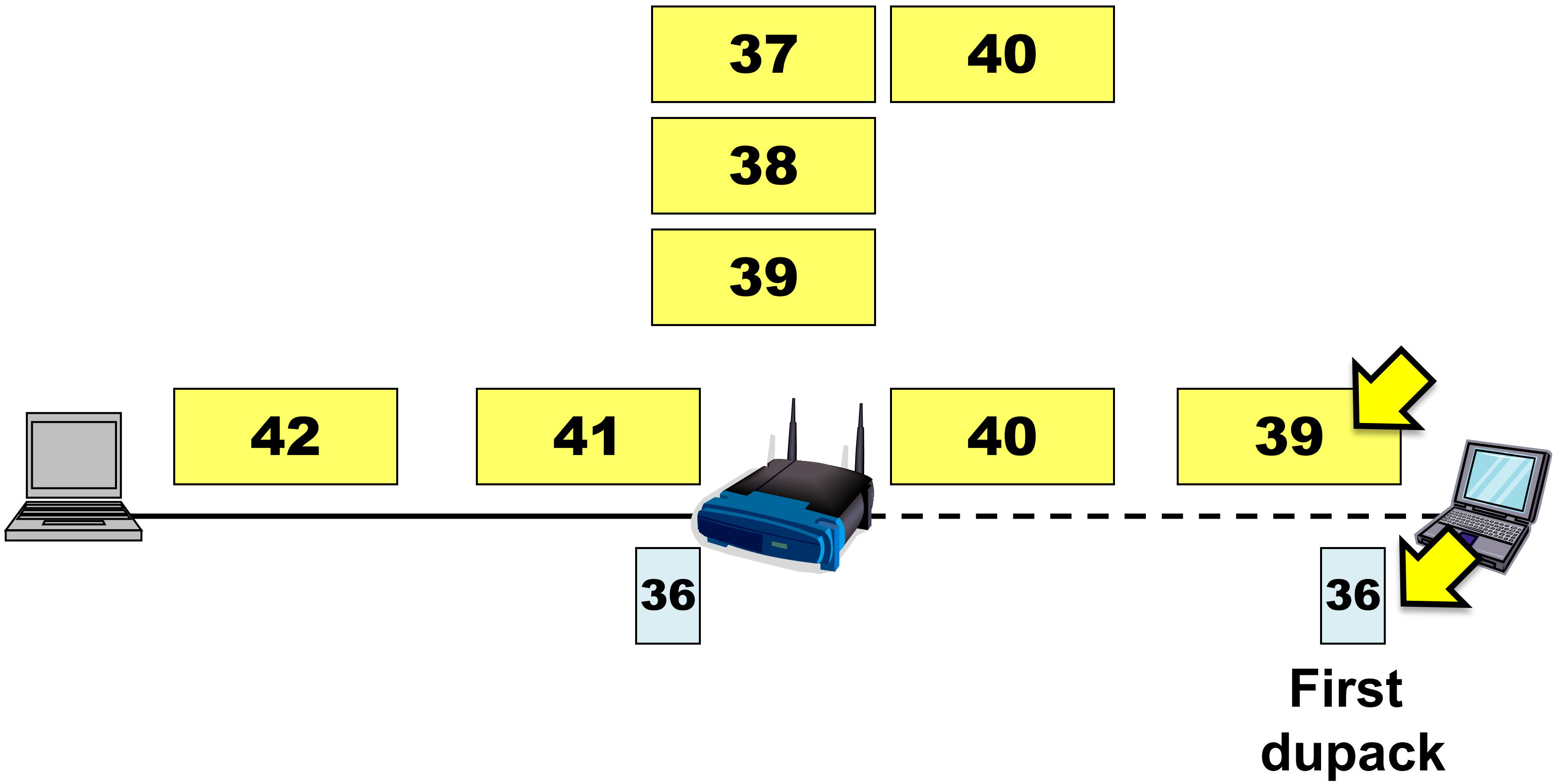


TCP Snoop: Downlink example

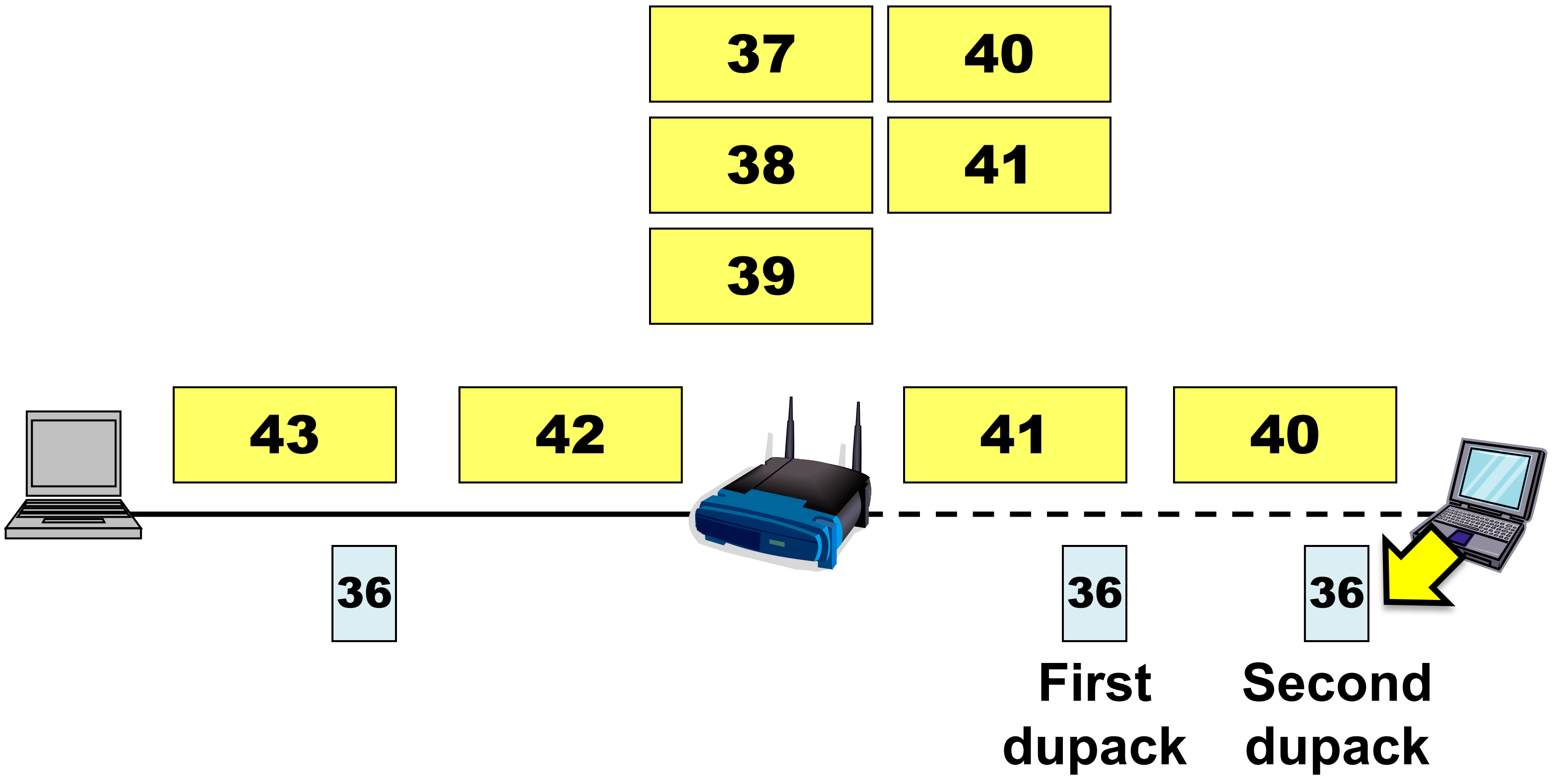


TCP Snoop: Downlink example

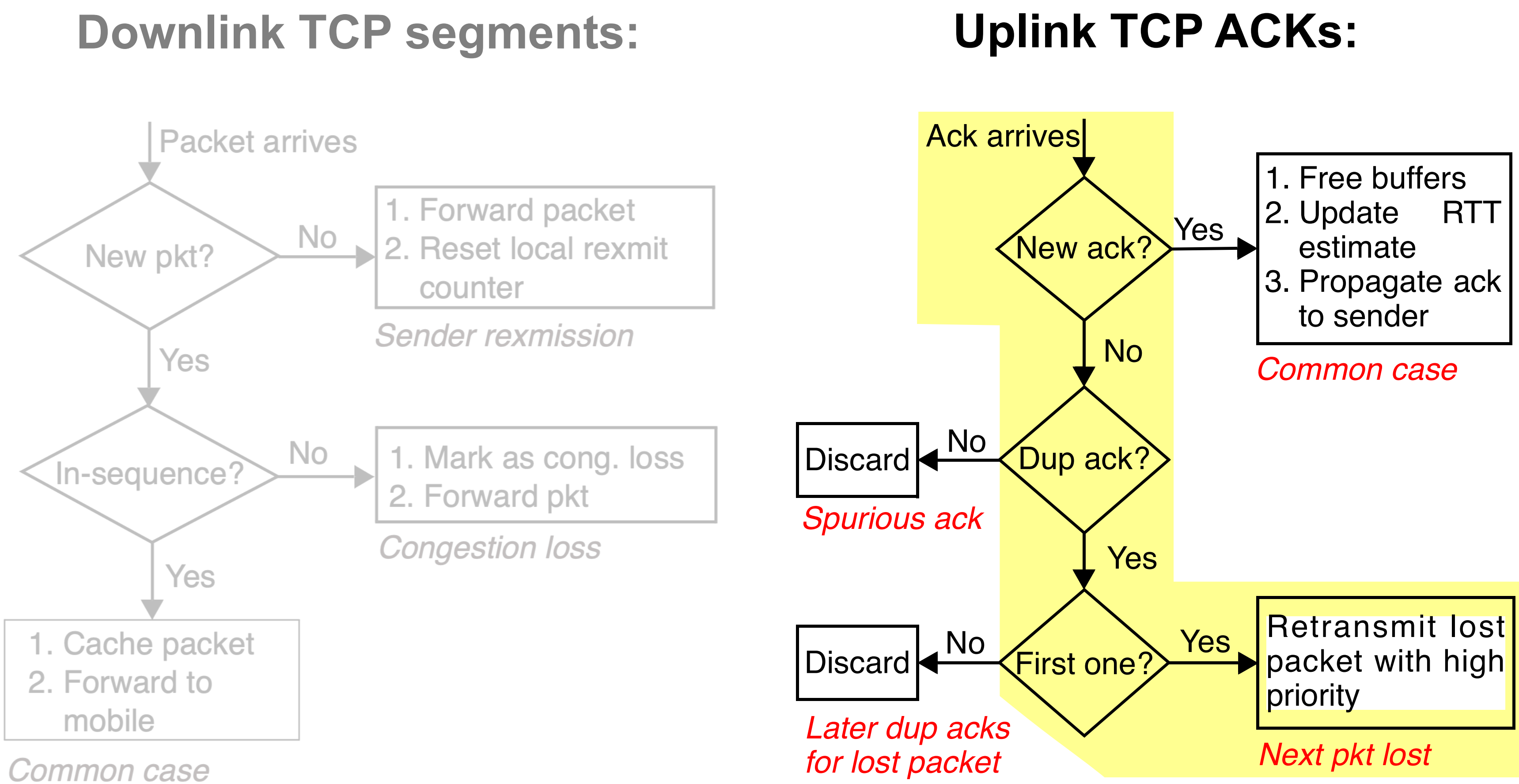


- TCP receiver does not delay duplicate ACKs (*dupacks*)

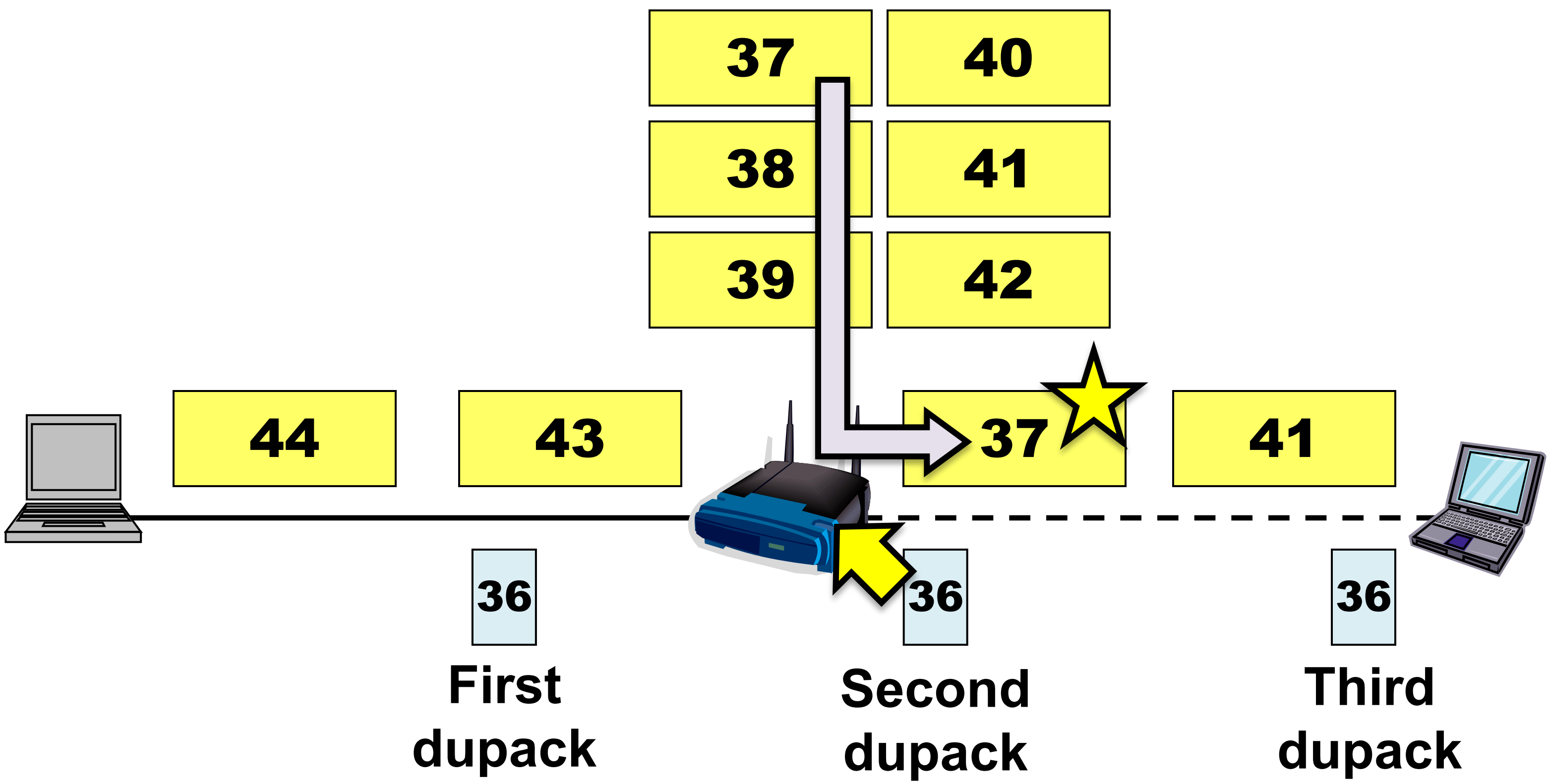
TCP Snoop: Downlink example



Downlink traffic operation, at Snoop AP



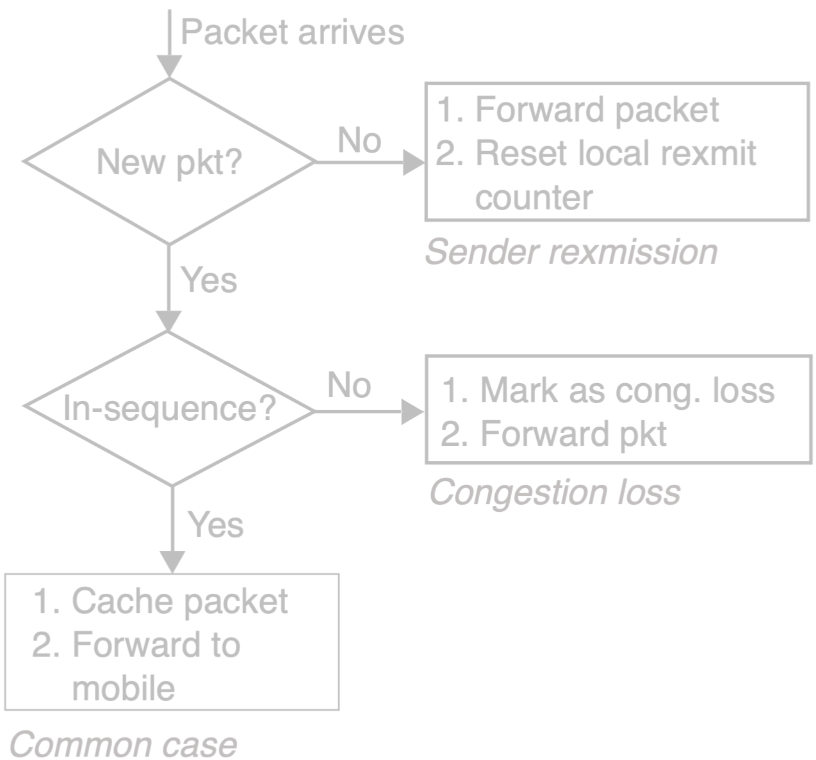
TCP Snoop: Downlink example



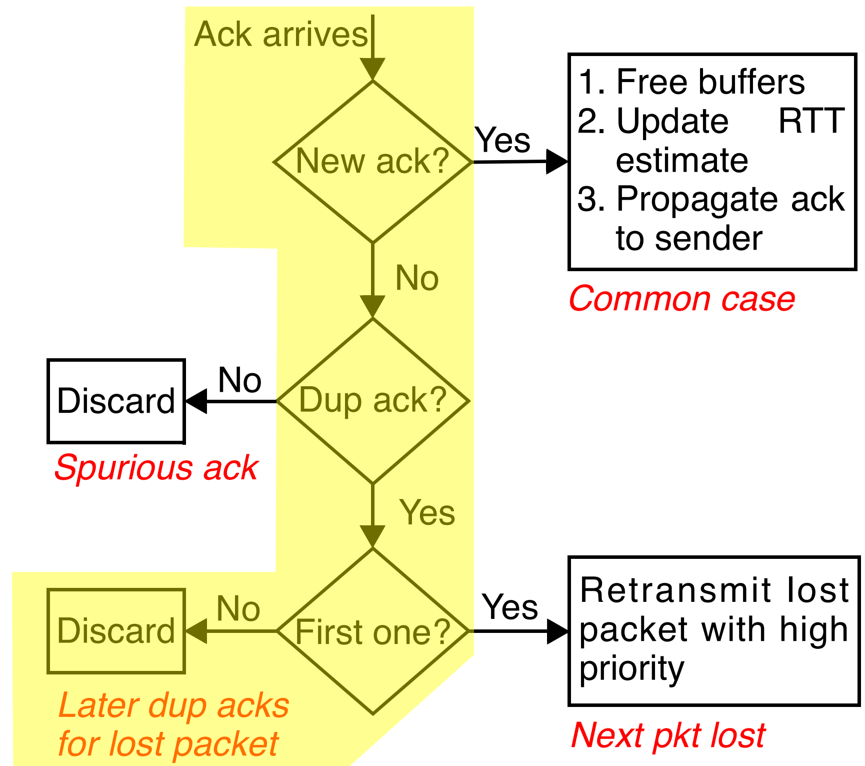
- Dupack triggers retransmission of packet 37 from AP

Downlink traffic operation, at Snoop AP

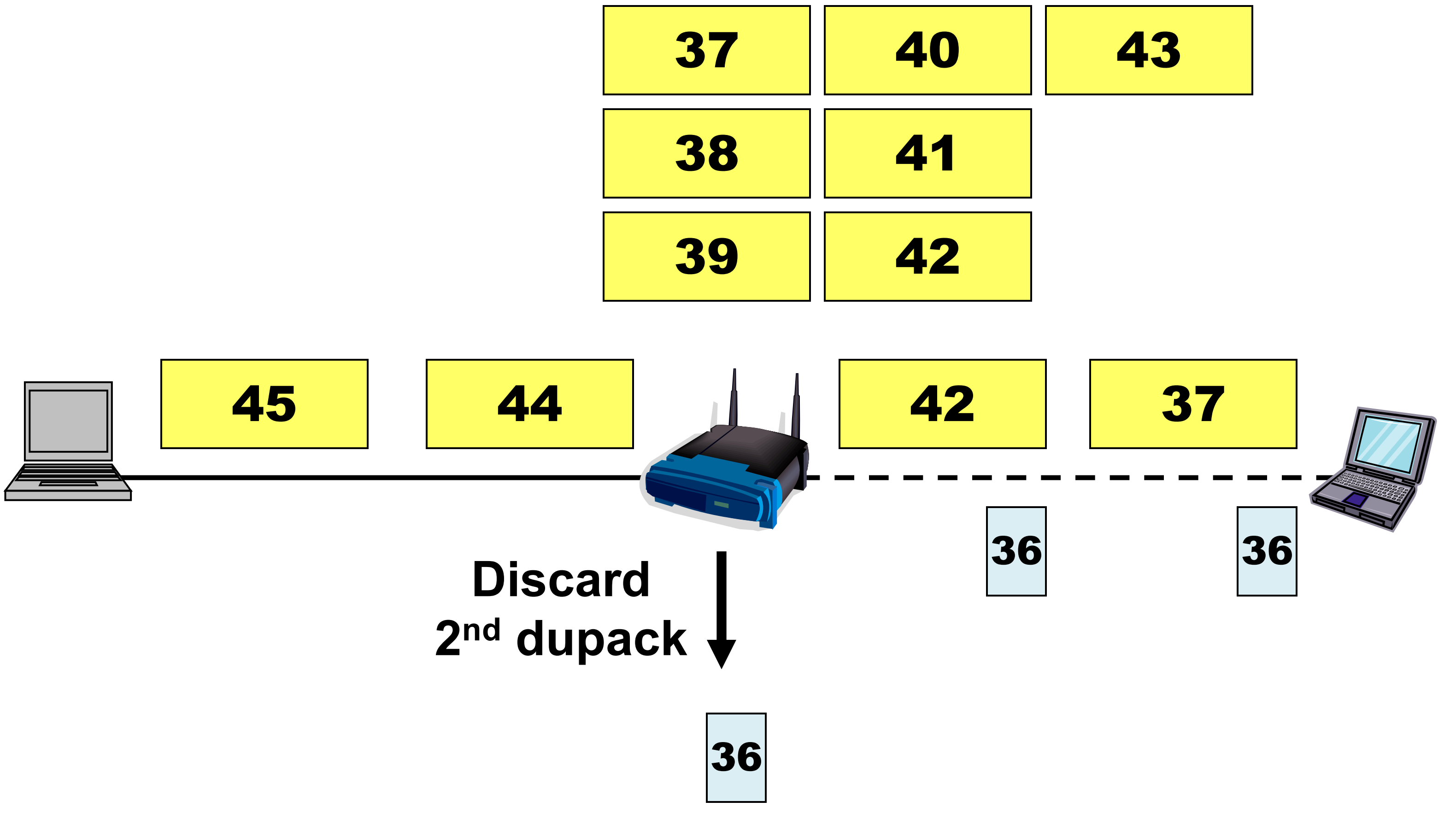
Downlink TCP segments:



Uplink TCP ACKs:



TCP Snoop: Downlink example



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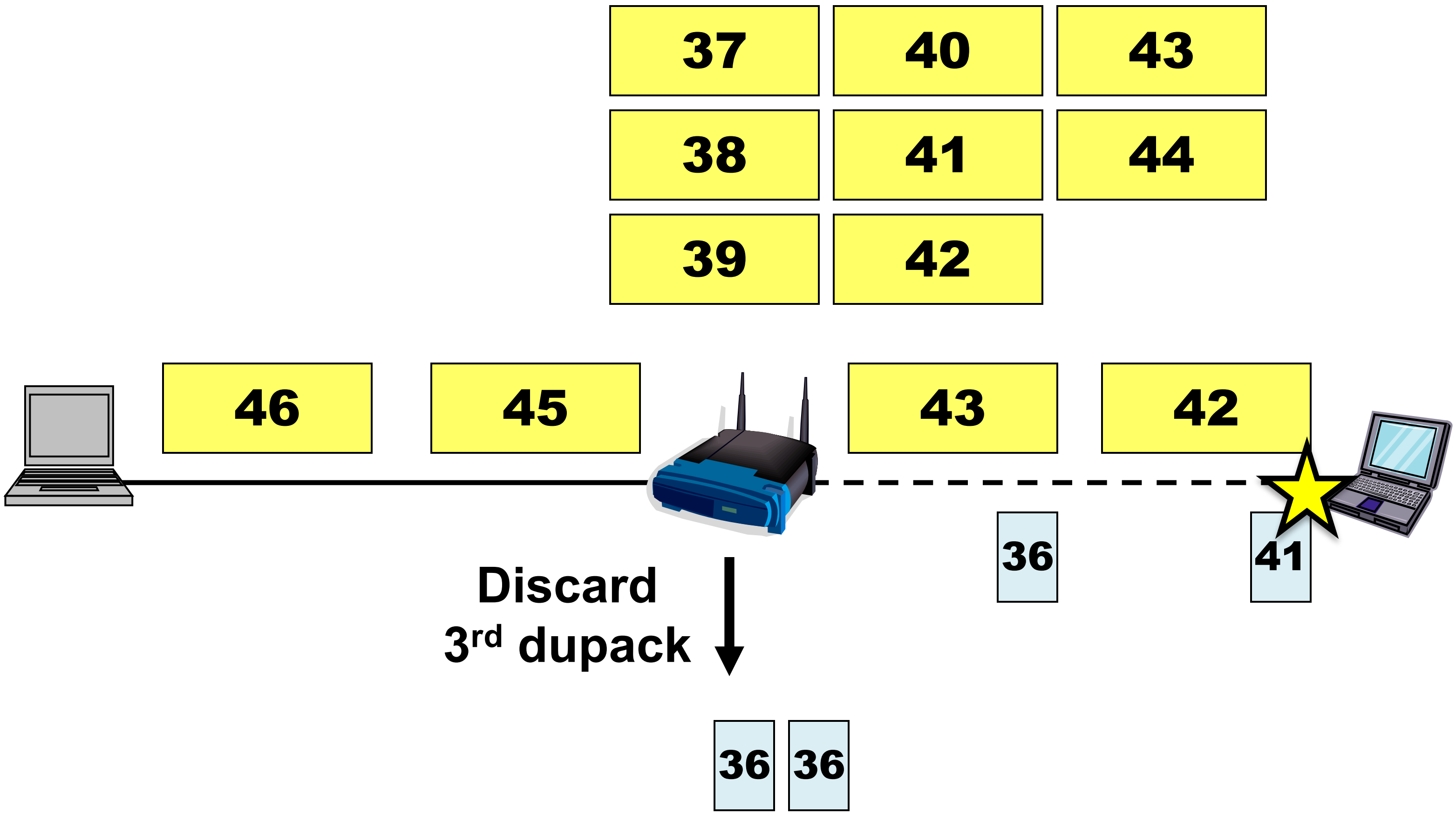

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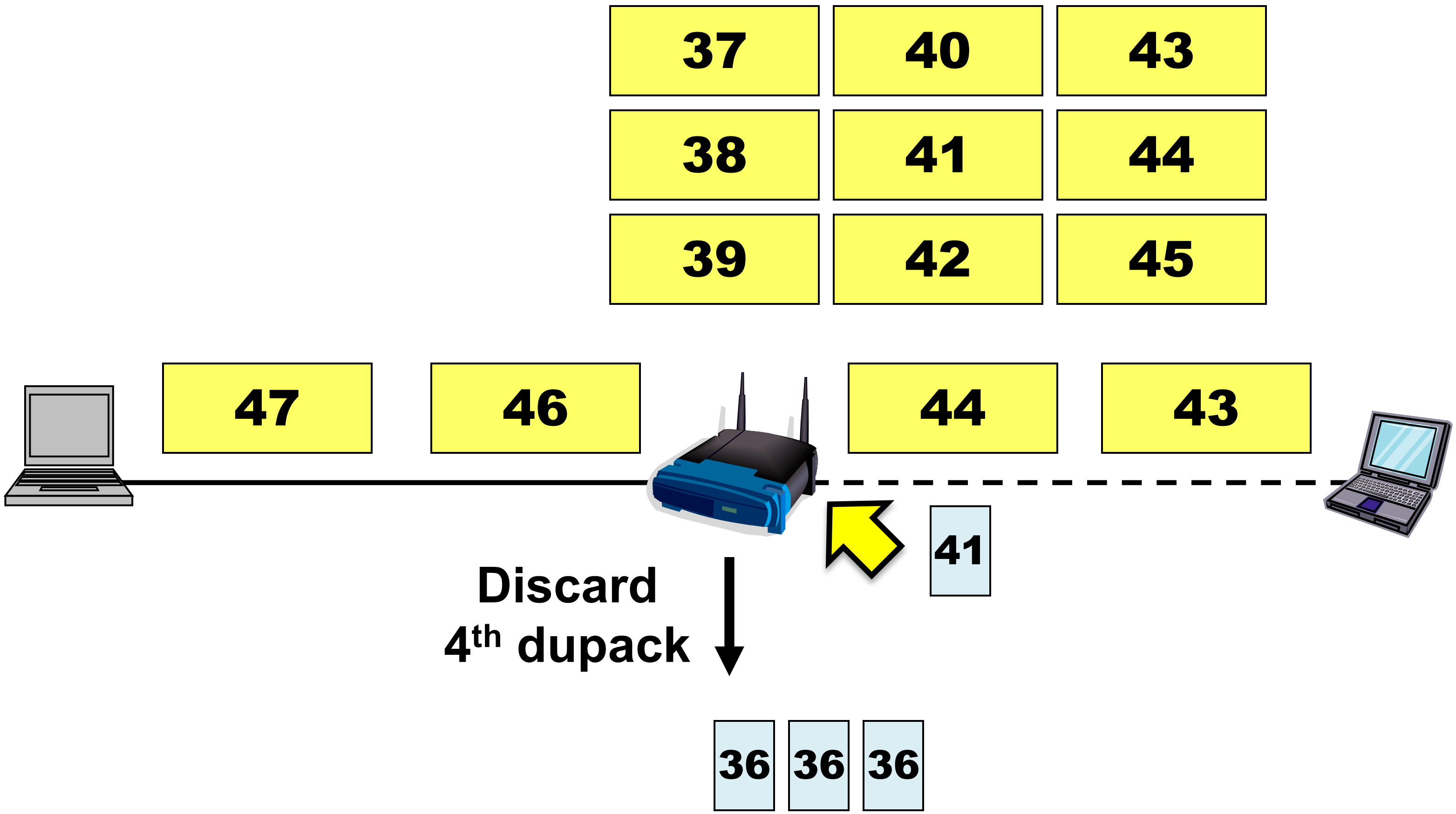
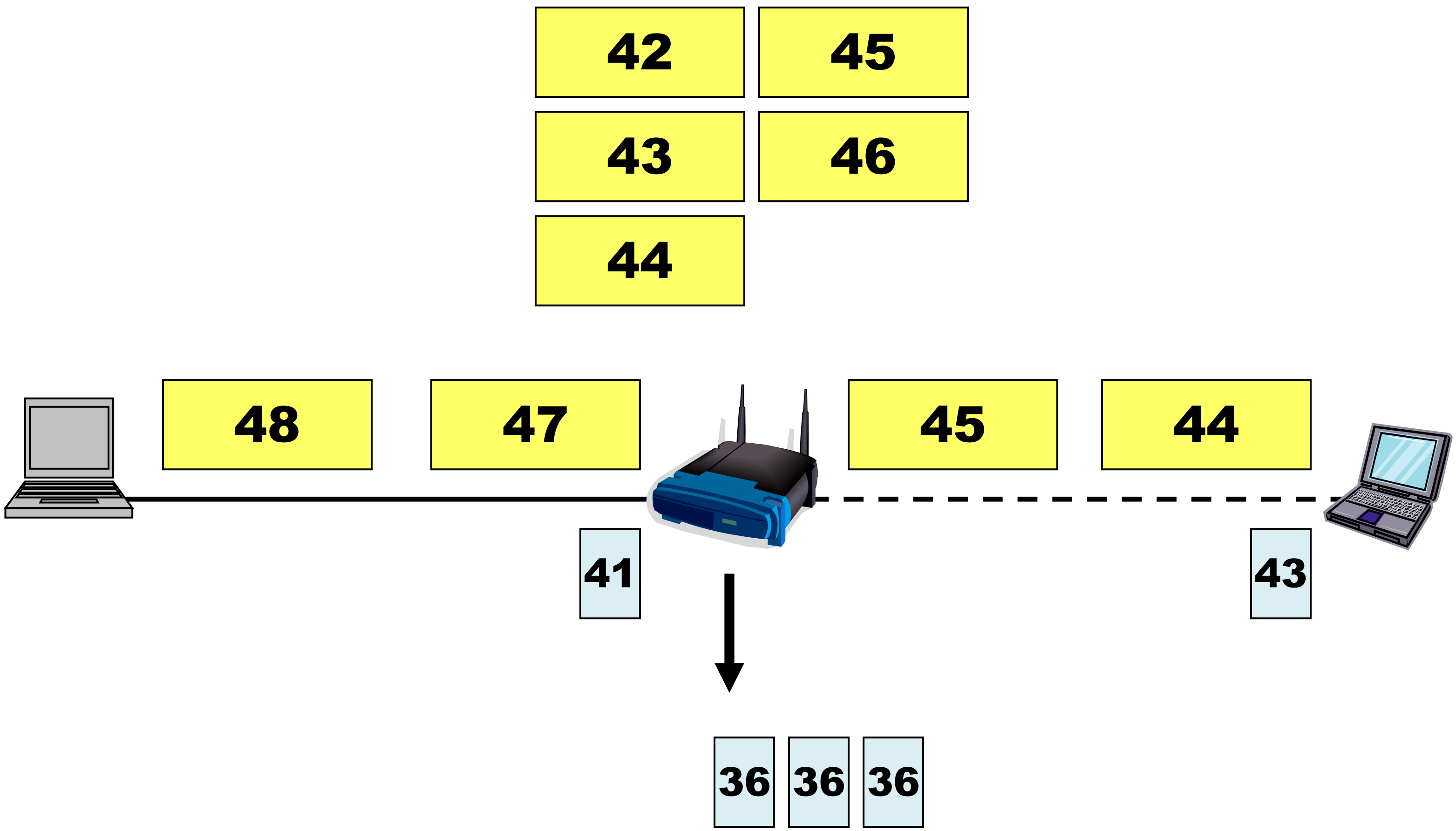
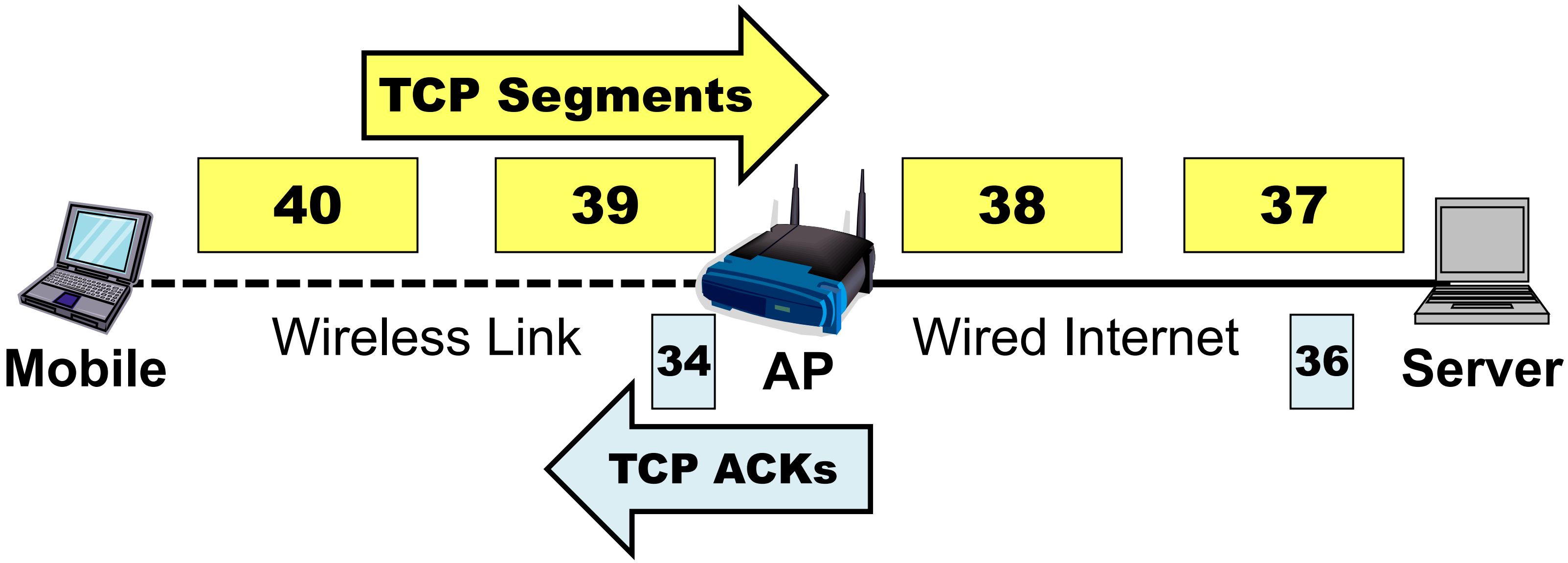


- TCP sender does not fast retransmit

TCP Snoop: Downlink example



Uplink traffic case



- **Less-common** case but **becoming more prevalent**
- Buffer & retransmit TCP segments at AP? **Not likely useful**
- Run Snoop agent on the Mobile? **Not likely useful**

Snoop TCP: Advantages

- Works **without modification** to mobile or server
- Preserves end-to-end semantics. Crash does not affect correctness, only performance.
- **After an AP handoff:** New AP needn't Snoop TCP
  - Can **automatically fall back** to regular TCP operation
  - No state need be migrated (but if done, can improve performance)
  - Note such "state" is called ***soft state***
    - Good if available, but correct functionality otherwise

Two Broad Approaches

1. Mask wireless losses from TCP sender
   – Then TCP sender will not reduce congestion window

   – Split Connection Approach
   – TCP Snoop

2. **Explicitly notify TCP sender about cause of packet loss**

Explicit Loss Notification (ELN)

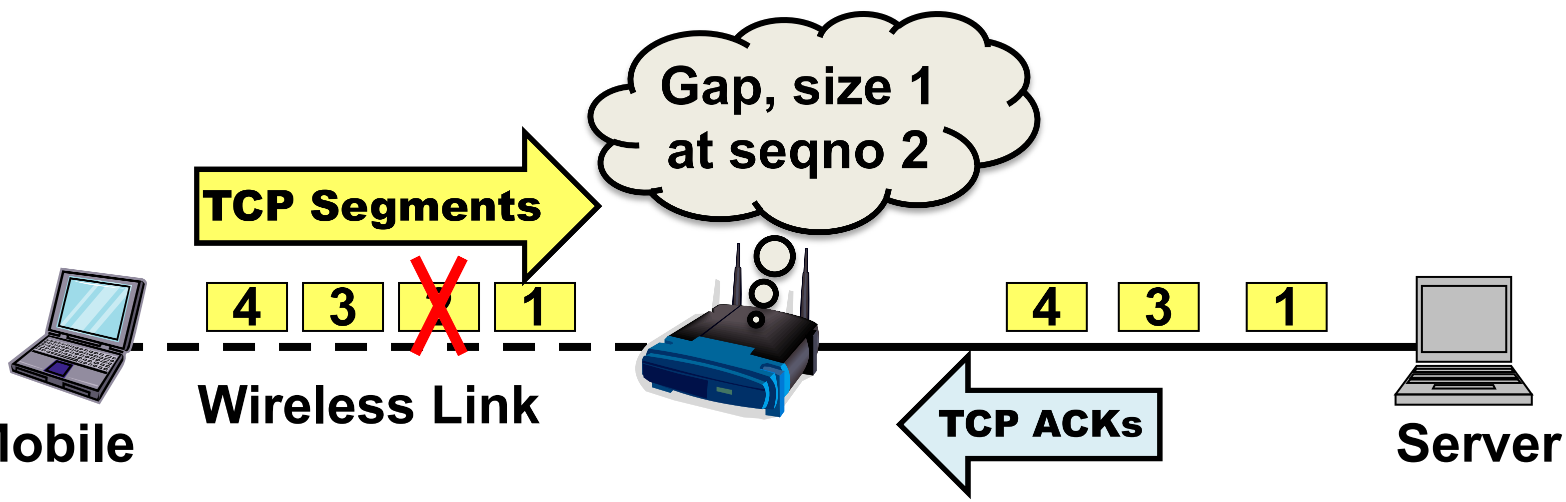
- Notify the TCP sender that a **wireless link** (not congestion) caused a certain packet loss

- Upon notification, TCP sender retransmits packet, but **doesn't reduce congestion window**

- Many **design options:**
  - **Who** sends notification? **How** is notification sent? How is notification **interpreted** at sender?
    - We'll discuss one example approach

ELN for uplink TCP traffic

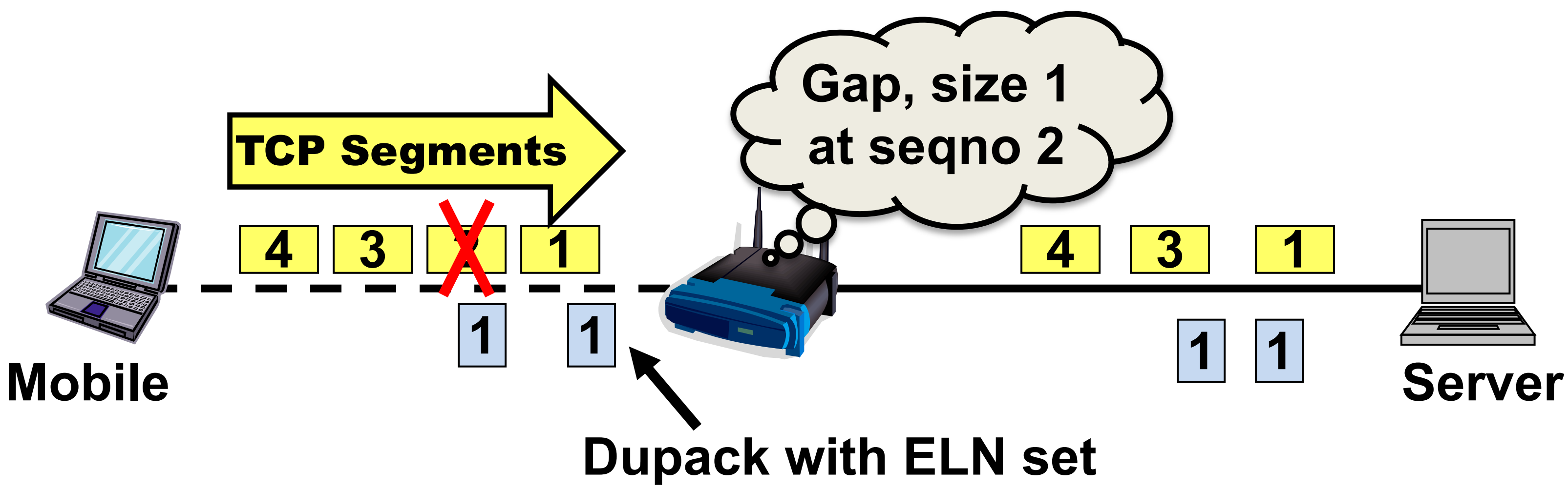
- AP keeps track of **gaps** in the TCP packet sequence received from the mobile sender



ELN for uplink TCP traffic

- When **AP** sees a **dupack:**
  - AP **compares** dupack seqno with its recorded gaps
    - **If match:** AP **sets ELN bit** in dupack and forwards it

- When **mobile** receives **dupack with ELN bit set:**
  - Resends packet, but **doesn't reduce** congestion window



Thursday Topic:

Link Layer I: Time, Frequency, and Code Division

Precepts Next Week:

Introduction to Lab 1