## EN.601.414/614 Computer Networks

## Transport Layer

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Fall 2020 (TuTh 1:30-2:45pm on Zoom)



#### Recap: Web components

#### Infrastructure:

- **≻**Clients
- >Servers (DNS, CDN, Datacenters)

#### Content:

➤ URL: naming content

➤ HTML: formatting content

Protocol for exchanging information: HTTP

#### Recap: HTTP

- Client-server architecture
  - ➤ Server is "always on" and "well known"
  - ➤ Clients initiate contact to server
- Synchronous request/reply protocol
  - ➤ Run over TCP, Port 80
- Stateless
- ASCII format
  - ➤ Before HTTP/2

#### Recap: CDN

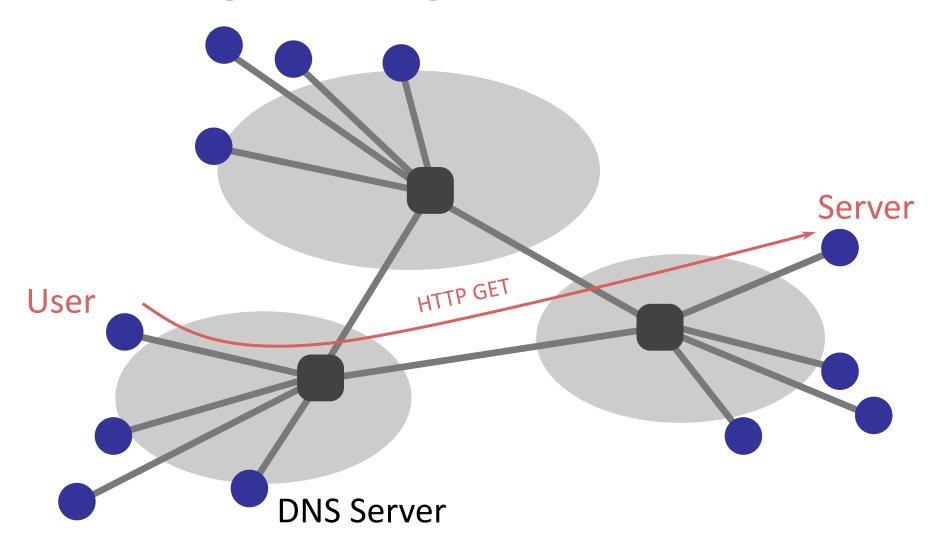
- Caching and replication as a service
- Large-scale distributed storage infrastructure (usually) administered by one entity
  - ➤ e.g., Akamai has servers in 20,000+ locations
- Combination of caching and replication
  - >Pull: Direct result of clients' requests (caching)
  - > Push: Expectation of high access rate (replication)
- Can do some processing to handle dynamic webpage content

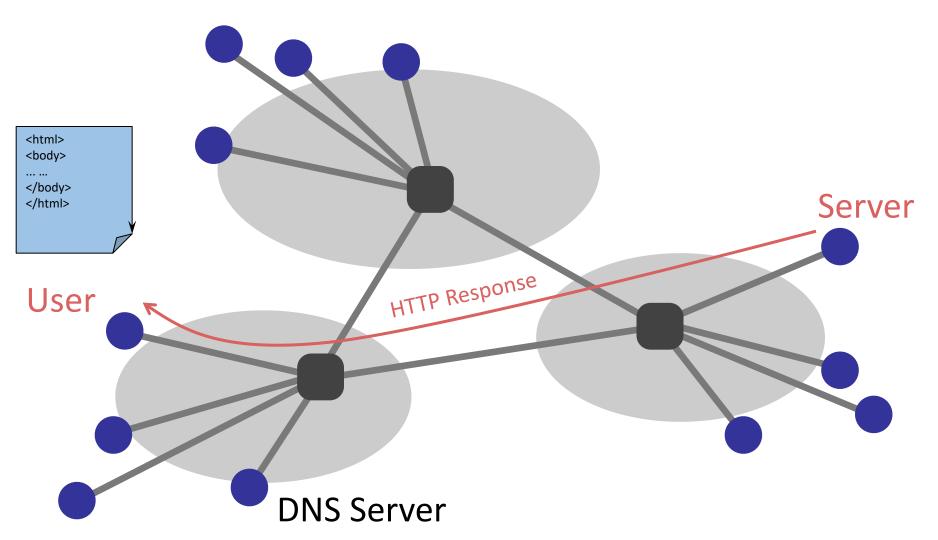
#### Recap: DNS

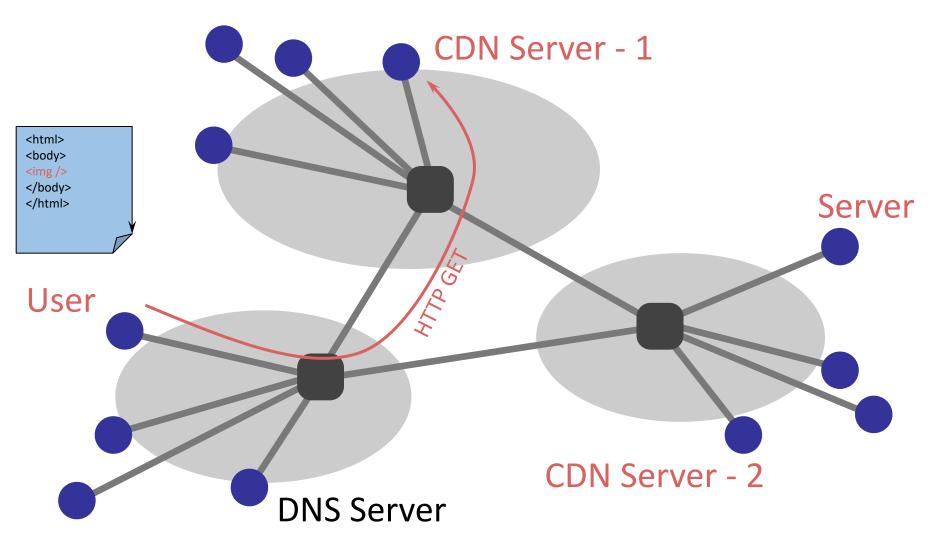
#### Resolve names to addresses

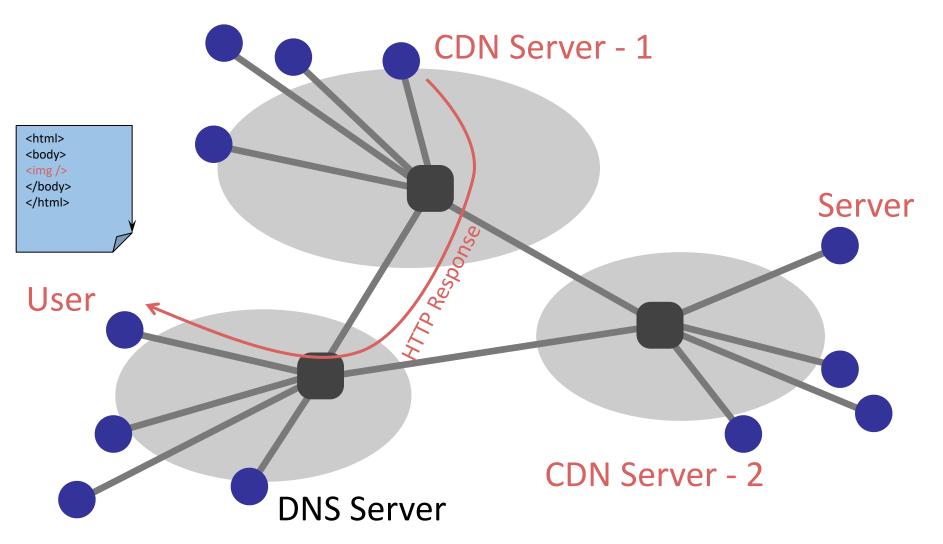
- ➤ Uniqueness: no naming conflicts
- ➤ Scalable: many names and frequent updates
- ➤ Distributed, autonomous administration
  - Ability to update my own (machines') names
  - Don't have to track everybody's updates
- ➤ Highly available
- ➤ Lookups are fast

#### Level of indirection







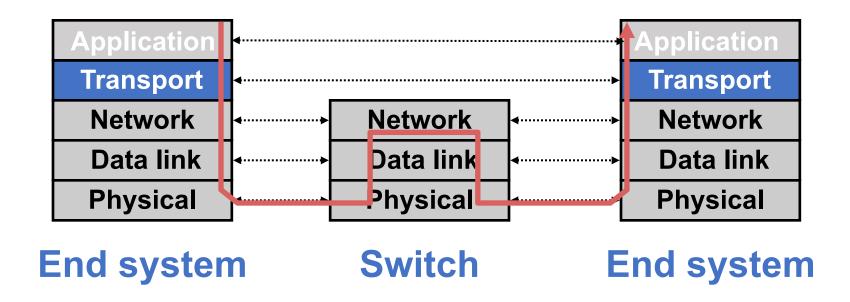


## Agenda

- Transport layer basics
- UDP
- Designing a reliable transport protocol

#### Transport layer

 Layer at end hosts, between the application and network layer



#### Why a transport layer?

- IP packets are addressed to a host but end-to-end communication is between application processes at hosts
  - ➤ Need a way to decide which packets go to which applications (multiplexing/demultiplexing)
- IP provides a weak service model (best-effort)
  - ➤ Packets can be corrupted, delayed, dropped, reordered, duplicated
  - ➤ No guidance on how much traffic to send and when
  - Dealing with this is tedious for application developers

## Multiplexing & demultiplexing

#### Multiplexing (Mux)

➤ Gather and combining data chunks at the source host from different applications and delivering to the network layer

#### Demultiplexing (Demux)

➤ Delivering correct data to corresponding sockets from a multiplexed stream

- Communication between processes
  - ➤ Mux and demux from/to application processes
  - >Implemented using *ports*

- Communication between processes
- Provide common end-to-end services for app layer [optional]
  - ➤ Reliable, in-order data delivery
  - ➤ Well-paced data delivery
    - Too fast may overwhelm the network
    - Too slow is not efficient

- Communication between processes
- Provide common end-to-end services for app layer [optional]
- TCP and UDP are the common transport protocols
  - ➤ Also SCTP, MPTCP, SST, RDP, DCCP, ...

- Communication between processes
- Provide common end-to-end services for app layer [optional]
- TCP and UDP are the common transport protocols
- UDP is a minimalist transport protocol
  - ➤ Only provides mux/demux capabilities

- Communication between processes
- Provide common end-to-end services for app layer [optional]
- TCP and UDP are the common transport protocols
- UDP is a minimalist transport protocol
- TCP offers a reliable, in-order, byte stream abstraction
  - ➤ With congestion control, but without performance guarantees (delay, bandwidth, etc.)

#### Applications and sockets

- Socket: software abstraction for an application process to exchange network messages with the (transport layer in the) operating system
- Two important types of sockets
  - ➤ UDP socket: TYPE is SOCK DGRAM
  - ➤TCP socket: TYPE is SOCK STREAM

#### **Ports**

- 16-bit numbers that help distinguishing apps
  - > Packets carry src/dst port number in transport header
  - ➤ Well-known (0-1023) and ephemeral ports
- OS stores mapping between sockets and ports
  - ➤ Port in packets and sockets in OS
  - For UDP ports (SOCK\_DGRAM)
    - OS stores (local port, local IP address) ←→ socket
  - For TCP ports (SOCK\_STREAM)
    - OS stores (local port, local IP, remote port, remote IP) ← → socket

#### **UDP: User Datagram Protocol**

- Lightweight communication between processes
  - >Avoid overhead and delays of order & reliability
- UDP described in RFC 768 (1980!)

➤ Destination IP address and port to support demultiplexing

SRC port	DST port
Length	Checksum
DATA	

## UDP (cont'd)

- Optional error checking on the packet contents
  - >(checksum field = 0 means "don't verify checksum")
  - See textbook on how checksums are calculated
- Source port is also optional
  - ➤ Useful to respond back to the sender in some cases

#### Why a transport layer?

- IP packets are addressed to a host but end-to-end communication is between application processes at hosts
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## Reliable transport

In a perfect world, reliable transport is easy

# 

#### Reliable transport

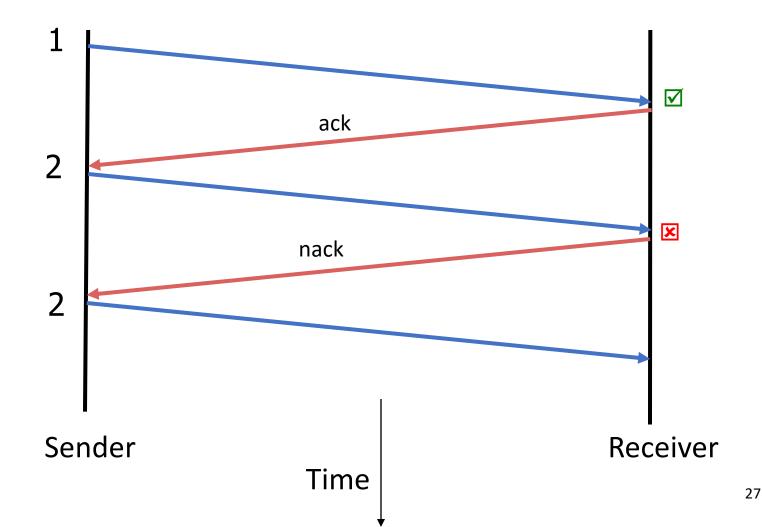
- In a perfect world, reliable transport is easy
- All the bad things best-effort can do
  - >A packet is corrupted (why?)
  - >A packet is lost (why?)
  - >A packet is delayed (why?)
  - ➤ Packets are reordered (why?)
  - >A packet is duplicated (why?)

#### Reliable transport

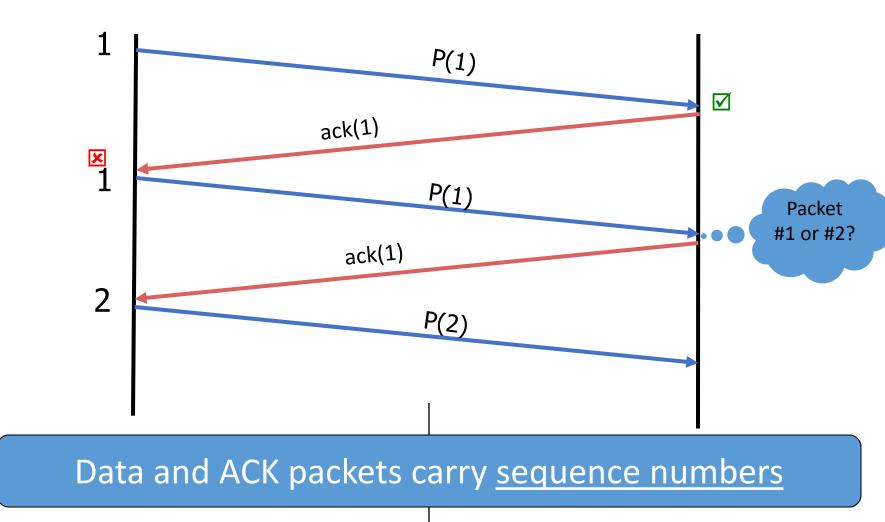
#### Mechanisms for coping with bad events

- ➤ Checksums: to detect corruption
- > ACKs: receiver tells sender that it received packet
- NACK: receiver tells sender it did not receive packet
- > Sequence numbers: a way to identify packets
- > Retransmissions: sender resends packets
- >Timeouts: a way of deciding when to resend packets
- Forward error correction: a way to mask errors without retransmission
- Network coding: an efficient way to repair errors

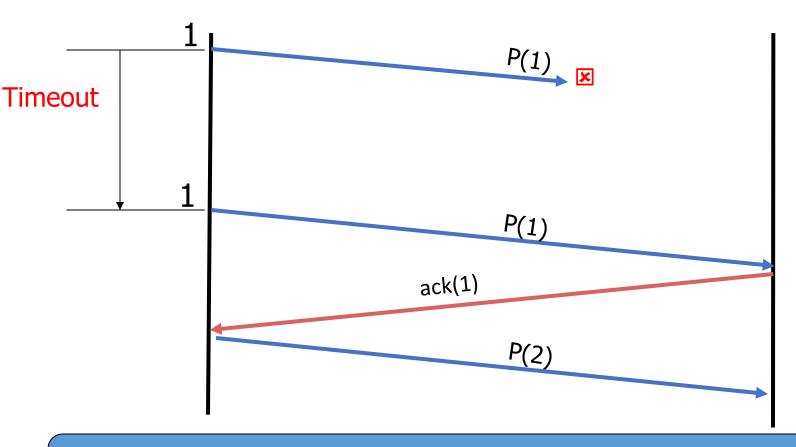
## Dealing with packet corruption



#### Dealing with packet corruption

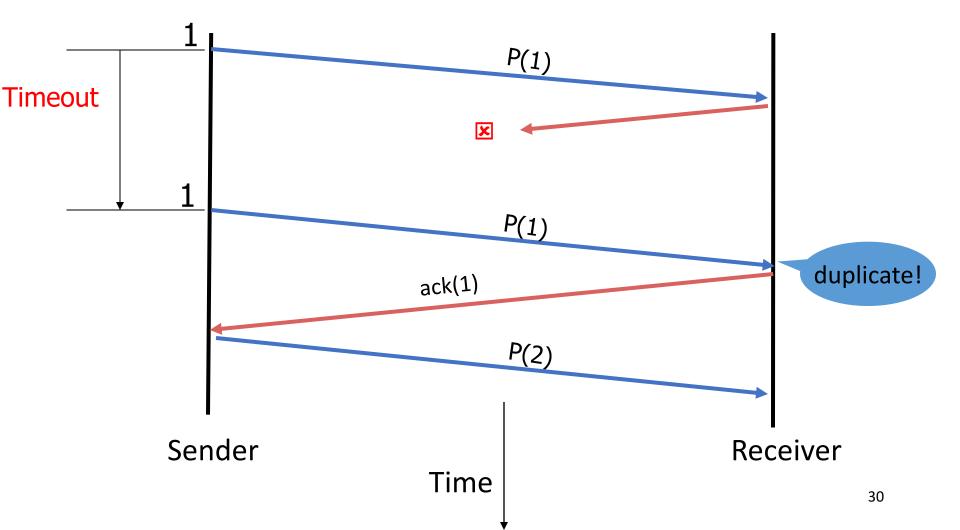


## Dealing with packet loss

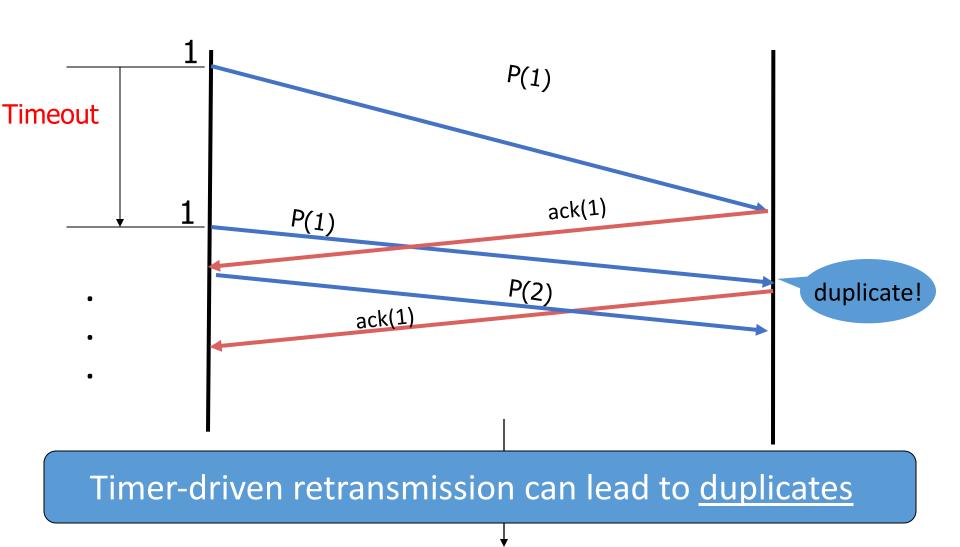


<u>Timer-driven loss detection</u>
Set timer when packet is sent; retransmit on timeout

## Dealing with packet loss (of ack)



#### Dealing with packet duplicate



#### Components of a solution

- Checksums (to detect bit errors)
- Timers (to detect loss)
- Acknowledgements (positive or negative)
- Sequence numbers (to deal with duplicates)

# Designing a reliable transport

#### A Solution: "Stop and Wait"

#### @Sender

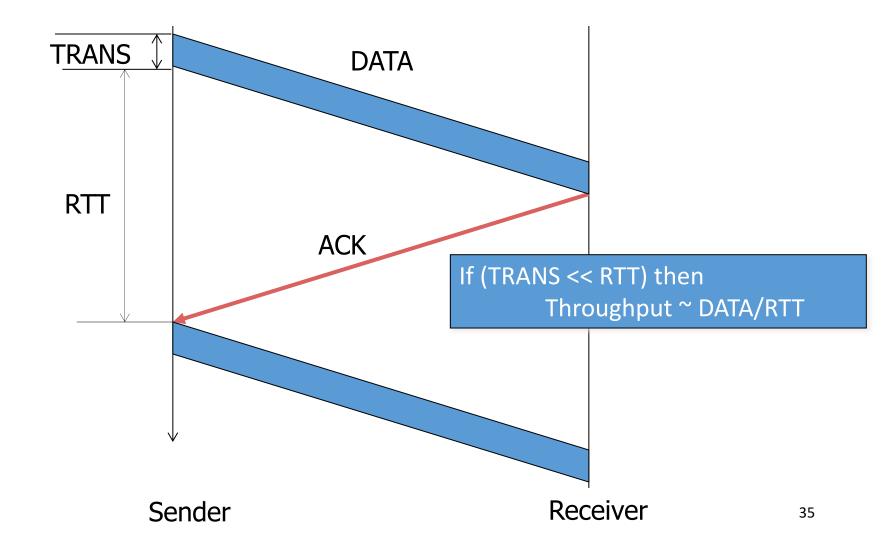
- Send packet(I); (re)set timer; wait for ack
- If (ACK)
  - I++; repeat
- If (NACK or TIMEOUT)
  - repeat

#### @Receiver

- Wait for packet
- If packet is OK, send ACK
- Else, send NACK
- Repeat

 A correct reliable transport protocol, but an extremely inefficient one

## Stop & Wait is inefficient



#### Orders of magnitude

- Transmission time for 10Gbps link:
  - >~ microsecond for 1500 byte packet
- RTT:
  - ≥1,000 kilometers ~ O(10) milliseconds

#### Three design decisions

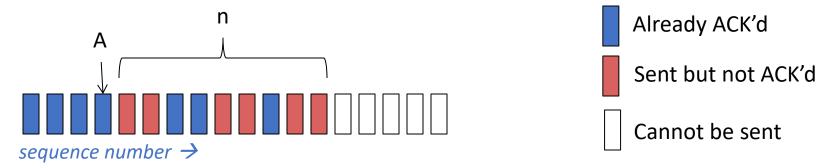
- Which packets can sender send?
  - ➤ Sliding window
- How does receiver ack packets?
  - **≻**Cumulative
  - **≻**Selective
- Which packets does sender resend?
  - ➤ Go-Back N (GBN)
  - ➤ Selective Repeat (SR)

#### Sliding window

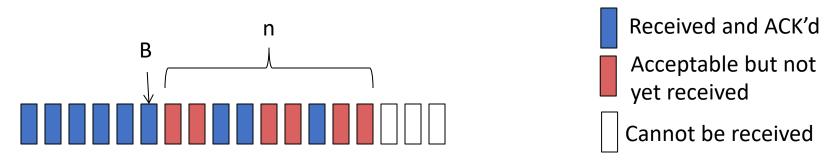
- Window = set of adjacent sequence numbers
  - The size of the set is the window size
  - ➤ Assume window size is n
- General idea: send up to n packets at a time
  - Sender can send packets in its window
  - > Receiver can accept packets in its window
  - ➤ Window of acceptable packets "slides" on successful reception/acknowledgement
  - ➤ Window contains all packets that might still be in transit
- Sliding window often called "packets in flight"

#### Sliding window

 Let A be the last ack'd packet of sender without gap; then window of sender = {A+1, A+2, ..., A+n}



 Let B be the last received packet without gap by receiver, then window of receiver = {B+1,..., B+n}



#### Throughput of sliding window

- If window size is n, then throughput is roughly
   ➤MIN(n\*DATA/RTT, Link Bandwidth)
- Compare to Stop and Wait: Data/RTT

What happens when n gets too large?

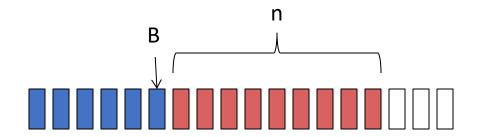
## Acknowledgements w/ sliding window

#### Two common options

Cumulative ACKs: ACK carries next in-order sequence number that the receiver expects

#### Cumulative acknowledgements

At receiver



- Received and ACK'd

  Acceptable but not yet received
- Cannot be received

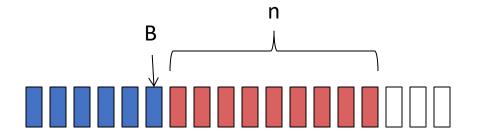
• After receiving B+1, B+2

$$B_{new} = B+2$$

Receiver sends ACK(B+3) = ACK(B<sub>new</sub>+1)

# Cumulative acknowledgements (cont'd)

At receiver

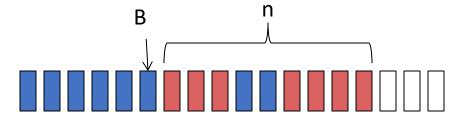


Received and ACK'd

Acceptable but not yet received

Cannot be received

• After receiving B+4, B+5



Receiver sends ACK(B+1)

## Acknowledgements w/ sliding window

- Two common options
  - Cumulative ACKs: ACK carries next in-order sequence number the receiver expects
  - Selective ACKs: ACK individually acknowledges correctly received packets
- Selective ACKs offer more precise information but require more complicated book-keeping

#### Sliding window protocols

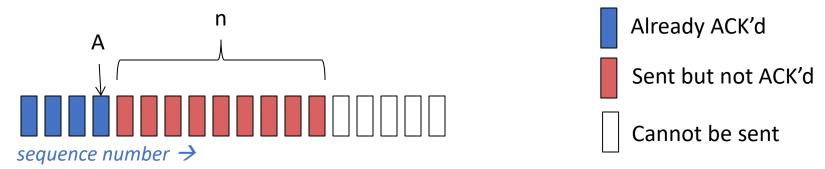
- Resending packets: two canonical approaches
  - ➤ Go-Back-N
  - ➤ Selective Repeat
- Many variants that differ in implementation details

#### Go-Back-N (GBN)

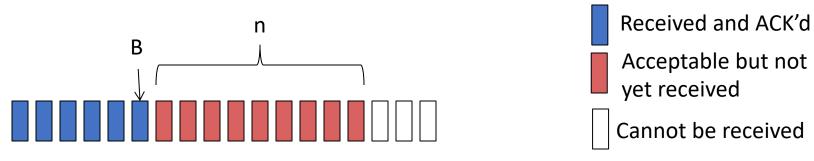
- Sender transmits up to n unacknowledged packets
- Receiver only accepts packets in order
  - Discards out-of-order packets (i.e., packets other than B+1)
- Receiver uses cumulative acknowledgements
  - ➤i.e., sequence# in ACK = next expected in-order sequence#
- Sender sets timer for 1st outstanding ack (A+1)
- If timeout, retransmit A+1, ..., A+n

#### Sliding window with GBN

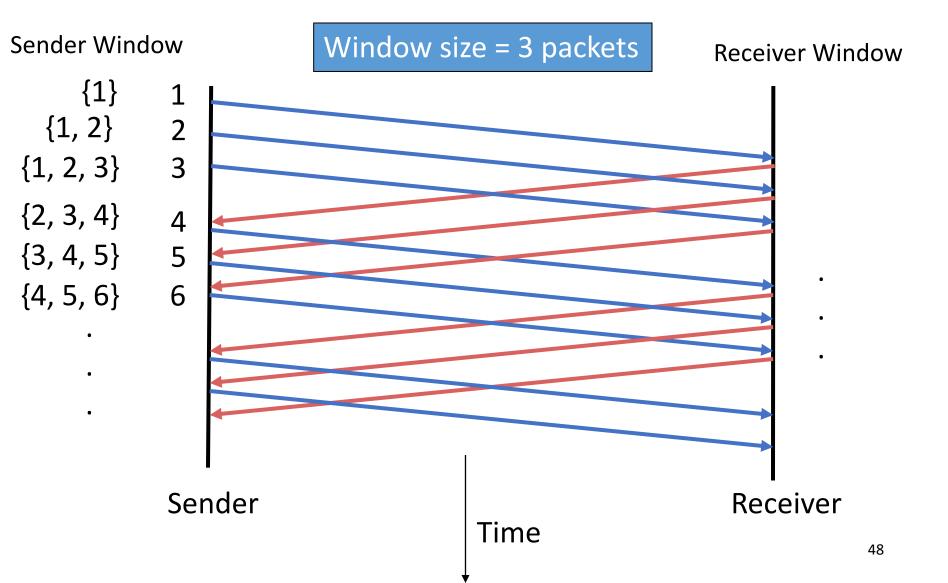
 Let A be the last ack'd packet of sender without gap; then window of sender = {A+1, A+2, ..., A+n}



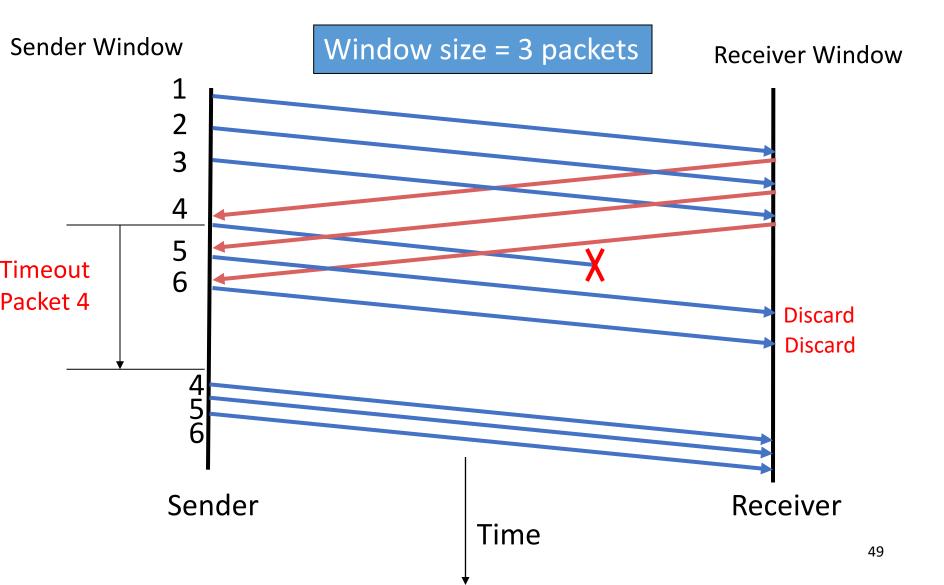
 Let B be the last received packet without gap by receiver, then window of receiver = {B+1,..., B+n}



### GBN example w/o errors



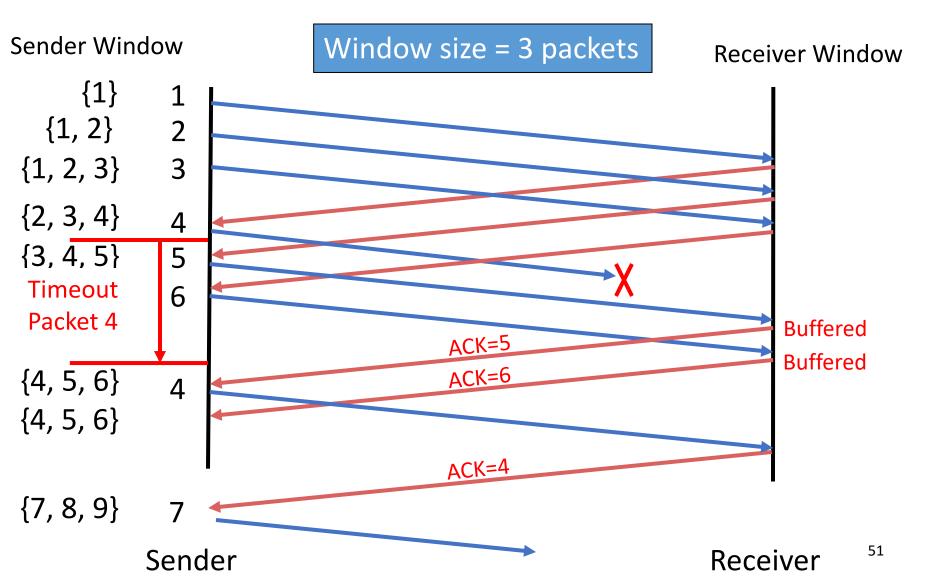
### GBN example with errors



#### Selective Repeat (SR)

- Sender: transmit up to n unacknowledged packets
- Assume packet k is lost, k+1 is not
  - ➤ Receiver: indicate packet k+1 correctly received
  - Sender: retransmit only packet k on timeout
- Efficient in retransmissions but complex bookkeeping
  - ➤ Need a timer/flag per packet

### SR example with errors



#### Observations

- With sliding windows, it is possible to fully utilize a link, provided the window size is large enough.
- Sender has to buffer all unacknowledged packets, because they may require retransmission
- Receiver may be able to accept out-of-order packets, but only up to its buffer limits
- Implementation complexity depends on protocol details (GBN vs. SR)

#### Components of a solution

- Checksums (for error detection)
- Timers (for loss detection)
- Acknowledgments
  - **≻**Cumulative
  - **>** Selective
- Sequence numbers (duplicates, windows)
- Sliding windows (for efficiency)
- Reliability protocols use the above to decide when and what to retransmit or acknowledge

#### **Group Discussion**

- Topic: sliding window for a reliable transport
  - ➤ Compare the pros and cons of Go-Back-N (GBN) and Selective Repeat (SR)
  - Choose a specific scenario (e.g., communication over a lossy wireless network). Should we use GBN or SR in this scenario? Why?
- Discuss in groups, and each group chooses a leader to summarize the discussion
  - In your group discussion, please do not dominate the discussion, and give everyone a chance to speak
  - Turn on your video if you can

#### GBN vs. Selective Repeat

- When would GBN be better?
  - ➤ When error rate is low; wastes bandwidth otherwise

- When would SR be better?
  - ➤ When error rate is high; otherwise, too complex

#### Summary

- Transport layer allows applications to communicate with each other
- Provides unreliable and reliable mechanisms
- Possible to build reliable transport over unreliable medium

- Assignment 2: hands-on experience on building a reliable transport
- Next lecture: TCP

# Thanks! Q&A