CS/ECE 439: Wireless Networking

Transport Layer – dealing with errors and unreliability
Hello!

My computer’s name is Alice.
Hello! My Computer’s name is Alice.
Suppose error protection identifies valid and invalid packets

- How?

- Can we make the channel appear reliable?
  - Insure packet delivery
  - Maintain packet order
  - Provide reliability at full link capacity
Reliable Transmission Outline

- Fundamentals of Automatic Repeat reQuest (ARQ) algorithms
  - A family of algorithms that provide reliability through retransmission

- ARQ algorithms (simple to complex)
  - stop-and-wait
  - sliding window
    - go-back-n
    - selective repeat
Terminology

- **Acknowledgement (ACK)**
  - Receiver tells the sender when a frame is received
    - Selective acknowledgement (SACK)
      - Specifies set of frames received
    - Cumulative acknowledgement (ACK)
      - Have received specified frame and all previous

- **Timeout (TO)**
  - Sender decides the frame (or ACK) was lost
  - Sender can try again
Stop-and-Wait

Basic idea

1. Send a frame
2. Wait for an ACK or TO
3. If TO, go to 1
4. If ACK, get new frame, go to 1
Stop-and-Wait: Success

How long should the timeout be?

What can go wrong? How will it affect our protocol?
Stop-and-Wait: Lost Frame

Sender

Receiver

Timeout

Frame

ACK

Timeout

Frame

RTT
Stop-and-Wait: Lost ACK

Sender

Timeout

Receiver

Frame

Timeout

Frame

ACK

RTT

ACK

Timeout

Time
Stop-and-Wait: Delayed Frame

How can receiver distinguish between two frames?

How many bits do you need for sequence numbers?
Stop-and-Wait

Goal
- Guaranteed, at-most-once delivery

Protocol Challenges
- Dropped frame/ACK
- Duplicate frame/ACK

Requirements
- 1-bit sequence numbers (if physical network maintains order)
  - sender tracks frame ID to send
  - receiver tracks next frame ID expected
Stop-and-Wait

- We have achieved
  - Frames delivered reliably and in order
  - Is that enough?

- Problem
  - Only allows one outstanding frame
    - Does not keep the pipe full

- Example
  - 100ms RTT
  - One frame per RTT = 1KB
  - 1024x8x10 = 81920 kbps
  - Regardless of link bandwidth!
Stop-and-Wait

\[ U_{\text{sender}} = \frac{L / R}{RTT + L / R} \]
Keeping the Pipe Full

- Last bit transmitted, \( t = \frac{L}{R} \)
- First packet bit arrives
- Last packet bit arrives, send ACK
- ACK arrives, send next packet, \( t = RTT + \frac{L}{R} \)
- Last bit of 2nd packet arrives, send ACK
- Last bit of 3rd packet arrives, send ACK

Increase utilization by a factor of 3!

\[
U_{\text{sender}} = \frac{3 \times \frac{L}{R}}{RTT + \frac{L}{R}}
\]
Concepts

- Consider an ordered stream of data frames
- Stop-and-Wait
  - Window of one frame
  - Slides along stream over time
Concepts

- Sliding Window Protocol
  - Multiple-frame send window
  - Multiple frame receive window
Sliding Window

- **Send Window**
  - Fixed length
  - Starts at earliest unacknowledged frame
  - Only frames in window are active
Sliding Window

- **Receive Window**
  - Fixed length (unrelated to send window)
  - Starts at earliest frame not received
  - Only frames in window accepted
Sliding Window Terminology

- **Sender Parameters**
  - Send Window Size (SWS)
  - Last Acknowledgement Received (LAR)
  - Last Frame Sent (LFS)

SWS = 4
LAR = 14
LFS = 18
Sliding Window Terminology

- **Receiver Parameters**
  - Receive Window Size (**RWS**)
  - Next Frame Expected (**NFE**)
  - Last Frame Acceptable (**LFA**)

RWS = 6
NFE = 4   LFA = 9
Sliding Window Details

- Sender Tasks
  - Assign sequence numbers
  - On ACK Arrival
    - Advance LAR
    - Slide window

Time

SWS = 4  SWS = 4

LAR = 14  LFS = 18

Receive ACK 16
Sliding Window Details

- **Receiver Tasks**
  - **On Frame Arrival (N)**
    - Silently discard if outside of window
      - $N < NFE$ (NACK possible, too)
      - $N \geq NFE + RWS$
    - Send cumulative ACK if within window

**Diagram:**
- RWS = 6
- NFE = 4
- LFA = 9
- Time
  - Receive Frame 4
  - Send ACK 7
Sliding Window Details

- **Receiver Tasks**
  - **On Frame Arrival (N)**
    - Silently discard if outside of window
      - N < NFE (NACK possible, too)
      - N ≥ NFE + RWS
    - Send cumulative ACK if within window

- **Graphical Representation**
  - RWS = 6
  - NFE = 8
  - LFA = 13
  - Time
Sliding Window Details

- Sequence number space
  - Finite number, so wrap around
  - Need space larger than SWS (outstanding frames)
    - In fact, need twice as large
Window Sizes

- How big should we make SWS?
  - Compute from delay x bandwidth

- How big should we make RWS?
  - Depends on buffer capacity of receiver
Delay x Bandwidth Product - Revisited

- **Amount of data in “pipe”**
  - channel = pipe
  - delay = length
  - bandwidth = area of a cross section
  - bandwidth x delay product = volume
Delay x Bandwidth Product

- **Pipe**
  - Half of data that must be buffered before sender responds to slowdown request
Delay x Bandwidth Product

- **Bandwidth x delay product**
  - How many bits the sender must transmit before the first bit arrives at the receiver if the sender keeps the pipe full
  - Takes another one-way latency to receive a response from the receiver
ARQ Algorithm Classification

- Three Types:
  - Stop-and-Wait: $\text{SWS} = 1$, $\text{RWS} = 1$
  - Go-Back-N: $\text{SWS} = N$, $\text{RWS} = 1$
  - Selective Repeat: $\text{SWS} = N$, $\text{RWS} = M$
    - Usually $M = N$

![Diagram showing the classification of ARQ algorithms]

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Sliding Window Variations: Go-Back-N

- SWS = N, RWS = 1
- Receiver only buffers one frame
- If a frame is lost, the sender may need to retransmit up to N frames
  - i.e., sender “goes back” N frames

Variations

- How long is the frame timeout?
- Does receiver send NACK for out-of-sequence frame?
Go-Back-N: Cumulative ACKs

Packets 2, 3, 4, 5 are retransmitted.

Timeout for Packet 2

loss
Sliding Window Variations: Selective Repeat

- \( SWS = \mathbf{N}, \ RWS = \mathbf{M} \)
- Receiver individually acknowledges all correctly received frames
  - Buffers up to \( \mathbf{M} \) frames, as needed, for eventual in-order delivery to upper layer
- If a frame is lost, sender must only resend
  - Frames lost within the receive window
- Variations
  - How long is the frame timeout?
  - Use cumulative or per-frame ACK?
  - Does protocol adapt timeouts?
  - Does protocol adapt SWS and/or RWS?
Selective Repeat

Packet 2 is retransmitted

Packet loss
Roles of a Sliding Window Protocol

- Reliable delivery on an unreliable link
  - Core function
- Preserve delivery order
  - Controlled by the receiver
- Flow control
  - Allow receiver to throttle sender

- Separation of Concerns
  - Must be able to distinguish between different functions that are sometimes rolled into one mechanism
TCP Data Transport

- Data broken into segments
  - Limited by maximum segment size (MSS)
  - Defaults to 352 bytes
  - Negotiable during connection setup
  - Typically set to
    - MTU of directly connected network – size of TCP and IP headers

- Three events cause a segment to be sent
  - ≥ MSS bytes of data ready to be sent
  - Explicit PUSH operation by application
  - Periodic timeout
TCP Byte Stream

Application process

Write bytes

TCP
Send buffer

TCP Segment
TCP Segment
TCP Segment

Application process

Read bytes

TCP
Recv buffer
ACKing and Sequence Numbers

- **Sender sends packet**
  - Data starts with sequence number $X$
  - Packet contains $B$ bytes
    - $X, X+1, X+2, \ldots, X+B-1$

![Diagram showing packet structure with sequence numbers and bytes][1]

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ACKing and Sequence Numbers

- Upon receipt of packet, receiver sends an ACK
  - If all data prior to $X$ already received:
    - ACK acknowledges $X+B$ (because that is next expected byte)
ACKing and Sequence Numbers

- Upon receipt of packet, receiver sends an ACK
  - If highest byte already received is some smaller value Y
    - ACK acknowledges Y+1
    - Even if this has been ACKed before
TCP Sliding Window Protocol

- Sequence numbers
  - Indices into byte stream
- ACK sequence number
  - Actually next byte expected as opposed to last byte received
TCP Sliding Window Protocol

- Advertised window
  - Enables dynamic receive window size

- Receive buffers
  - Data ready for delivery to application until requested
  - Out-of-order data to maximum buffer capacity

- Sender buffers
  - Unacknowledged data
  - Unsent data out to maximum buffer capacity
TCP Sliding Window Protocol – Sender Side

- LastByteAcked \(\leq\) LastByteSent
- LastByteSent \(\leq\) LastByteWritten
- Buffer bytes between LastByteAcked and LastByteWritten

**Diagram:**
- Maximum buffer size
- Advertised window
- First unacknowledged byte
- Last byte sent
- Data available, but outside window
TCP Sliding Window Protocol – Receiver Side

- LastByteRead < NextByteExpected
- NextByteExpected ≤ LastByteRcvd + 1
- Buffer bytes between NextByteRead and LastByteRcvd
Flow Control vs. Congestion Control

- **Flow control**
  - Preventing senders from overrunning the capacity of the receivers

- **Congestion control**
  - Preventing too much data from being injected into the network, causing switches or links to become overloaded

- Which one does TCP provide?
  - TCP provides both
    - Flow control based on advertised window
    - Congestion control discussed later in class
Advertised Window Limits Rate

- $W = \text{window size}$
  - Sender can send no faster than $W/\text{RTT}$ bytes/sec
  - Receiver implicitly limits sender to rate that receiver can sustain
  - If sender is going too fast, window advertisements get smaller & smaller
Reasons for Retransmission

- Packet Timeout
- Packet lost

- Packet ACK Timeout
- ACK lost
- Duplicate Packet

- Packet ACK Timeout
- ACK lost
- Duplicate Packet

- Packet ACK Timeout
- Early timeout
- Duplicate Packets
How Long Should Sender Wait?

- Sender sets a timeout to wait for an ACK
  - Too short
    - wasted retransmissions
  - Too long
    - excessive delays when packet lost
TCP Round Trip Time and Timeout

How should TCP set its timeout value?
- Longer than RTT
  - But RTT varies
- Too short
  - Premature timeout
  - Unnecessary retransmissions
- Too long
  - Slow reaction to segment loss

Estimating RTT
- SampleRTT
  - Measured time from segment transmission until ACK receipt
  - Will vary
  - Want smoother estimated RTT
- Average several recent measurements
  - Not just current SampleRTT
TCP Congestion Control

- **Idea**
  - Assumes best-effort network
  - Each source determines network capacity for itself
  - Implicit feedback
  - ACKs pace transmission (self-clocking)

- **Challenge**
  - Determining initial available capacity
  - Adjusting to changes in capacity in a timely manner
TCP Congestion Control

- Basic idea
  - Add notion of congestion window
  - Effective window is smaller of
    - Advertised window (flow control)
    - Congestion window (congestion control)
  - Changes in congestion window size
    - Slow increases to absorb new bandwidth
    - Quick decreases to eliminate congestion
TCP Congestion Control

- **Specific strategy**
  - **Self-clocking**
    - Send data only when outstanding data ACK’d
    - Equivalent to send window limitation mentioned
TCP Congestion Control

Specific strategy

Self-clocking
- Send data only when outstanding data ACK’d
- Equivalent to send window limitation mentioned

Growth
- Add one maximum segment size (MSS) per congestion window of data ACK’d
- It’s really done this way, at least in Linux:
  - see tcp_cong_avoid in tcp_input.c.
  - Actually, every ack for new data is treated as an MSS ACK’d
- Known as additive increase
TCP Congestion Control

Specific strategy (continued)

- Decrease
  - Cut window in half when timeout occurs
  - In practice, set window = window / 2
  - Known as multiplicative decrease

- Additive increase, multiplicative decrease (AIMD)
Additive Increase/ Multiplicative Decrease

- **Tools**
  - React to observance of congestion
  - Probe channel to detect more resources

- **Observation**
  - On notice of congestion
    - Decreasing too slowly will not be reactive enough
  - On probe of network
    - Increasing too quickly will overshoot limits
Additive Increase/ Multiplicative Decrease

- New TCP state variable
  - `CongestionWindow`
    - Similar to `AdvertisedWindow` for flow control
    - Limits how much data source can have in transit
      - \( \text{MaxWin} = \text{MIN}(\text{CongestionWindow}, \text{AdvertisedWindow}) \)
      - \( \text{EffWin} = \text{MaxWin} - (\text{LastByteSent} - \text{LastByteAcked}) \)
      - TCP can send no faster than the slowest component, network or destination

- Idea
  - Increase `CongestionWindow` when congestion goes down
  - Decrease `CongestionWindow` when congestion goes up
Additive Increase/ Multiplicative Decrease

Question

How does the source determine whether or not the network is congested?

Answer

Timeout signals packet loss
Packet loss is rarely due to transmission error (on wired lines)
Lost packet implies congestion!
AIMD – Sawtooth Trace

- Packet loss is seen as sign of congestion and results in a multiplicative rate decrease
  - Factor of 2
- TCP periodically probes for available bandwidth by increasing its rate

![Graph showing AIMD Sawtooth Trace]

Loss halved
TCP Start Up Behavior

- How should TCP start sending data?
  - AIMD is good for channels operating at capacity
  - AIMD can take a long time to ramp up to full capacity from scratch

It could take a long time to get started!
TCP Start Up Behavior

- How should TCP start sending data?
  - AIMD is good for channels operating at capacity
  - AIMD can take a long time to ramp up to full capacity from scratch
  - Use Slow Start to increase window rapidly from a cold start
TCP Start Up Behavior: Slow Start

- Initialization of the congestion window
  - Congestion window should start small
    - Avoid congestion due to new connections
  - Start at 1 MSS,
    - Initially, Cwnd is 1 MSS
    - Initial sending rate is MSS/RTT
  - Reset to 1 MSS with each timeout
    - timeouts are coarse-grained, ~1/2 sec
TCP Start Up Behavior: Slow Start

- Growth of the congestion window
- Linear growth could be pretty wasteful
  - Might be much less than the actual bandwidth
  - Linear increase takes a long time to accelerate
- Start slow but then grow fast
  - Sender starts at a slow rate
  - Increase the rate exponentially
  - Until the first loss event
Slow Start Example

CWD size:

Src

Dest

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Slow Start

- **Used**
  - When first starting connection
  - When connection times out

- **Why is it called slow-start?**
  - Because TCP originally had no congestion control mechanism
  - The source would just start by sending a whole window’s worth of data
TCP Congestion Control

- Maintain threshold window size
  - Threshold value
    - Initially set to maximum window size
    - Set to 1/2 of current window on timeout
  - Use multiplicative increase
    - When congestion window smaller than threshold
    - Double window for each window ACK’d

- In practice
  - Increase congestion window by one MSS for each ACK of new data (or N bytes for N bytes)
Slow Start

- How long should the exponential increase from slow start continue?
  - Use \texttt{CongestionThreshold} as target window size
  - Estimates network capacity
  - When \texttt{CongestionWindow} reaches \texttt{CongestionThreshold} switch to additive increase

![Diagram showing exponential and linear probing](image-url)
Slow Start

- **Initial values**
  - $\text{CongestionThreshold} = 8$
  - $\text{CongestionWindow} = 1$

- **Loss after transmission 7**
  - $\text{CongestionWindow currently} = 12$
  - Set $\text{CongestionThreshold} = \text{CongestionWindow}/2$
  - Set $\text{CongestionWindow} = 1$
Slow Start

- Example trace of CongestionWindow

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- Problem
  - Have to wait for timeout
  - Can lose half CongestionWindow of data
Fast Retransmit and Fast Recovery

- **Problem**
  - Coarse-grain TCP timeouts lead to idle periods

- **Solution**
  - Fast retransmit: use duplicate ACKs to trigger retransmission
Fast Retransmit and Fast Recovery

- Send ACK for each segment received
- When duplicate ACK’s received
  - Resend lost segment immediately
  - Do not wait for timeout
  - In practice, retransmit on 3rd duplicate

- Fast recovery
  - When fast retransmission occurs, skip slow start
  - Congestion window becomes 1/2 previous
  - Start additive increase immediately
Fast Retransmit and Fast Recovery

Results

- Fast Recovery
  - Bypass slow start phase
  - Increase immediately to one half last successful CongestionWindow ($ssthresh$)
TCP Congestion Window Trace

- Threshold
- Congestion window
- Fast retransmission
- Additive increase
- Slow start period
- Timeouts

Time

Congestion Window