CS4450

Computer Networks:
Architecture and Protocols

Lecture 18
Reliable Transport

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Goal of Today's Lecture

- **Understanding reliable transport conceptually**
    - What are the fundamental aspects of reliable transport
- **Back to architectural principles for one lecture**
- The goal is not to understand a particular protocol (e.g., TCP)
    - TCP involves lots of detailed mechanisms, covered later
- Ground rules for discussion
    - No mention of TCP
    - No mention of detailed practical issues
    - Focus only on "ideal" world of packets and links

You must think for yourself

- **Today's lecture requires you to engage**
  - How would I design a reliable service
- **I will ask a lot of questions today, want you to think about them**
  - Be that kid!

Decisions and Their Principles

- How to break system into modules?
  - Dictated by layering
- Where are modules implemented?
  - Dictated by End-to-End Principle
- Where state is stored?
  - Dictated by fate-sharing

Today We Design Reliable Delivery

- The end-to-end principle tells us?
  - Put reliability in the end-host, not the network
- Layering dictates putting reliability in what layer?
  - Above network layer
  - L4 focusses on process-to-process delivery (“flow”)
- Fate sharing tells us?
  - Keep all reliability state in ends, not in network

Best Effort Service (L3)

- Packets can be lost
- Packets can be corrupted
- Packets can be reordered
- Packets can be delayed
- Packets can be duplicated
- …

How can you possible make anything work with such a service model?

Making Best Effort Service Work

- Engineer network so that average case is decent
  - You can’t make guarantees, but the operator must try…
- Engineer apps so they can tolerate the worst cast
  - They don’t have to thrive, they just can’t die
- A classical case of **architecting for flexibility**
  - And then **engineering for performance**

Reliable Transport is Necessary

- Some app semantics involve reliable transport
    - E.g., file transfer

- **Layer 3 and below provide only unreliable packet delivery**

- Today's question:
    - How can we build a reliable transport service on top of arbitrary unreliable packet delivery?

- A central challenge in bridging the gap between
    - **The abstractions application designers want**
    - **The abstractions networks can easily support**

Important Distinctions

- For functionality implemented in network:
  - Keep minimal (easy to build, broadly applicable)
- For functionality implemented in the application:
  - Keep minimal (easy to write)
  - Restricted to application-specific functionality
- Functionality implemented in **“network stack”**
  - The shared networking code on the host
  - This relieves burden from both application and network
  - **This is where reliability belongs**

Two Different Statements

- **Some applications need reliable service**
    - This means that application writers should be able to assume this, to make their job easier
- **The network <u>must</u> provide reliable service**
    - This contends that applications cannot implement this functionality, so the network must provide it
- Today we're making the first statement and refuting the second…
    - And this simple observation is what advocates of reliable networks (as in telephony) never understood

Challenge For Today

- Building a stack that supports reliable transfer
  - So that individual applications don't need to deal with packet losses, etc.

- What mechanisms can we put in the transport layer to provide reliability?

- Reliability is focused on single "flow"
  - Flow: stream of packets between two **processes**
  - **Usually defined using the 5-tuple:**
    - (sourceIP, destIP, sourcePort, destPort, protocol)

Four Goals for Reliable Transfer

- **Correctness**
  - To be defined
- **“Fairness”**
  - Every flow must get a fair share of network resources
- **Flow Performance**
  - Latency, jitter, etc.
- **Utilization**
  - Would like to maximize bandwidth utilization
  - If network has bandwidth available, flows should be able to use it!

Start With Transfer of a Single Packet

- We can later worry about larger files, but in the beginning it is cleaner to focus on this simple case

Correctness Condition

- Routing had a clean correctness condition
- **We want same kind of "if and only if" characterization of "correct" reliable transport designs**
- This condition is **for the design to be correct**, not the best performant
- One obvious requirement:
  - Transport never claims to have delivered data that wasn't delivered…
- But we need more than that. What?

Correctness Condition?

- How about: “Packet is always delivered to receiver”?
- i.e., Transport is reliable if and only if packets are always delivered to the receiver…
- Isn’t that simple?

WRONG!

- **What if network is partitioned?**
  - Partitioned means that the network is broken into two or more disconnected components…
- We can’t claim a transport design is incorrect if it doesn’t work in a partitioned network!
  - After all, there is no way to reach the destination!

Correctness Condition?

- Packet is delivered to receiver if and only if its possible to deliver packet

WRONG!

- If the network is only available at one instant of time, only an Oracle would know when to send
- **We can't claim a transport design is incorrect if it doesn't know the unknowable…**
- So we need to focus on what the transport design is trying to do, not what it actually accomplishes

Correctness Condition?

- Resend packet if and only if the previous transmission was lost or corrupted
- This is better because it refers to:
  - what the design does (which it can control)
  - not whether it always succeeds (which it can't)

WRONG!

- Impossible
  - “Coordinated Attack” over an unreliable network
- Consider two cases:
  - Packet delivered; all packets from receiver are dropped
  - Packet dropped; all packets from receiver are dropped
- They are indistinguishable to sender
  - In both cases, packet was sent, and no feedback at all
  - **Does it resend, or not?**

Correctness Condition?

- Packet is always resent if the previous transmission was lost or corrupted
- Packet **may** be resent at other times
- Note:
  - This invariant gives us a simple criterion for deciding if an implementation is correct
  - Efficiency and simplicity are separate criteria

Almost Right!

- What's wrong with it?
- An implementation that never sent the packet at all is reliable according to the definition.

Complete Correctness Condition

- **A transport mechanism is "reliable" if and only if**

(a) It resends all dropped or corrupted packets

(b) It attempts to make progress

- Making progress means:
  - **If there is data to send,** transport **eventually attempts** to send data
    - Very important: "eventually attempts"!
    - It should not be blocked for ever
    - And, it may not succeed, but it must attempt
  - Example: If there are ten packets to send, transport can't just send the first five and then stop **for ever**

Complete Correctness Condition

- A transport mechanism is “reliable” if and only if

(a) It resends all dropped or corrupted packets

(b) It attempts to make progress

- **Sufficient (“if”):** transport algorithm will keep trying to deliver packets that have not yet reached the destination

- **Necessary (“only if”):** if it ever lets a packet go undelivered without trying again, or never tries to send a packet when all others have been delivered, it isn’t reliable

Note!

- A transport mechanism can “give up”, but must announce this to application
- If the transport mechanism has tried for some period to deliver the data, and has not succeeded:
  - It might decide that it is better to give up
  - And applications can reinitiate data transfer
  - That is allowed…
- **But it can never falsely claim to have delivered a packet**

We have the correctness condition

- How do we achieve it?
- Focus on single-packet solutions

Solution v1

- **Send every packet as often and fast as possible…**
- Is it correct
    - No.
    - Why?
    - The "if" condition is not satisfied:

      (a) Transport must **attempt to make progress**
    - No way to check whether the packet was dropped or corrupted
        - So, must continue sending the same packet

What's missing?

- Feedback from receiver!
- If receiver does not respond, no way for sender to tell when to stop resending
    - Cannot achieve correctness without feedback

Forms of Feedback

- ACK: Yes, I got a packet
- NACK: No, I did not get the packet
- When is NACK a natural idea?
    - Packet Corruption (I got packet#5 but it was corrupted)
- Ignore NACKs for rest of the lecture…

Solution v2

- Resend packet until you get an ACK
    - And receiver sends per-packet ACKs until data finally stops
- Correct?
    - Yes:
        - All dropped/corrupted packets will be retransmitted
        - The transport will attempt to make progress
- Fair?
    - Over long-term, yes:
        - all sources will get an equal chance to use network resources
- Flow performance?
    - Good but not necessarily optimal
        - Some packets may be retransmitted unnecessarily
- Efficiency:
    - suboptimal; packets retransmitted unnecessarily

Solution v3

- Send packet
    - But now, **set a timer**
- When receiver gets packet, sends ACK
- If sender receives ACK, done
- If no ACK when timer expires, resend

    - Still **correct**, and **fair**
    - **Performance** would argue for **small timeout**
    - **Utilization** would argue for **larger timeout**
        - May want to increase timer each time you try
        - May want to cap the number of retries
        - Problems with this design?

Have “Solved” the Single Packet Case

- Send packet
    - Set a timer
- If no ACK when timer goes off, resend packet
    - And reset timer

- Tradeoff between performance and utilization in selection of timeout:
    - Too small: unnecessary retransmissions (underutilization)
    - Too large: waiting unnecessarily (poor performance)

Multiple Packets

- Service model: reliable stream of packets
  - Hand up contiguous block of packets to application
- Why not use single-packet solution?
  - Send the next packet once the first one has been delivered
  - Problem: Only one packet in flight at a time
    - **Low Effective throughput: Packet Size / RTT**
- Use window based approach
  - Allow for a window of W packets in-flight at any time (unack'ed)
  - Slide the window as packets are ack'ed
  - Sliding window implies W packets are continuous

Window-based Algorithms

- Very simple concept
  - Send W packets
  - When one gets ACK’ed send the next packet in line
  - It really is that simple (until we got to TCP)
- Will consider several variations…
  - But first…

How Big Should the Window be?

- **Windows serve three purposes**
    - Taking advantage of the bandwidth of the links
    - Limiting bandwidth used by a flow (congestion control)
    - Limiting the amount of buffering needed at the receiver
        - Why do receivers need to buffer packets?
            - Answer: packet re-ordering (discussed later)

- If we ignore all but the first goal, then we want to keep the sender always sending (in the ideal case)
    - RTT: from sending first packet until received first ACK

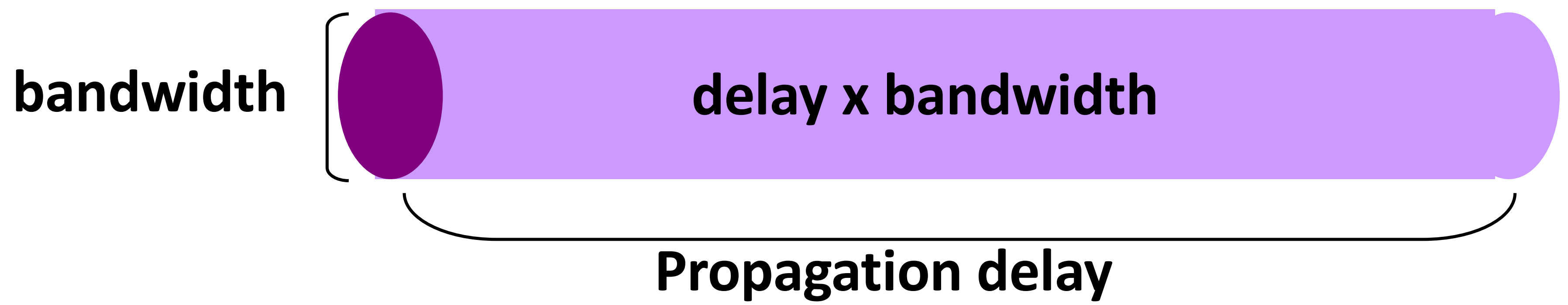
- **Condition:**
    - RTT x B ~ W x Packet Size

What does this mean?

- B is the minimum link bandwidth along the path
    - Obviously shouldn't send faster than that
    - Don't want to send slower than that (for first goal)
- **We want to set W such that:**
    - if I am sending at rate B, then
    - the ACK of the first packet arrives
    - exactly when I just finish sending the last of my W packets
- **Lets me send as fast as the path can deliver…**

RTT x B ~ W x Packet Size

- Recall that **Bandwidth Delay Product**
  - BDP = bandwidth x propagation delay



- **B x RTT is merely 2x BDP**

- Window sizing rule:
  - Total bits in flight is roughly the amount of data that fits into forward and reverse "pipes"
    - Here pipe is complete path, not single link…
    - **This is not "detail", this is a fundamental concept…**

Where Are We?

- **Figured out correctness condition:**
  - Always resend lost/corrupted packets
  - Always try to make progress (but can give up entirely)
- **Figured out single packet case:**
  - Send packet, set timer, resend if no ACK when timer expires
- **Some progress towards multiple packet case:**
  - Allow many packets (W) in flight at once
  - And know what the ideal window size is
    - RTT x B / Packet size
- What's left to design?

Three Design Considerations

- Nature of feedback
  - What should ACKs tell us when we have many packets in flight
- Detection of loss
- Response to loss

Possible Feedback From Receiver

- Ideas?

ACK Individual Packets

- Strengths
    - Know fate of each packet
    - Reordering not a problem
    - Simple window algorithm
        - W independent single packet algorithms
        - When one finishes grab next packet
- Weaknesses?
    - Loss of ACK packet requires a retransmission

Full Information Feedback

- **List all packets that have been received**
  - Give highest cumulative ACK plus any additional packets
  - If packets 1, 2, 3, 5, 6 received: send ACK(3, 5, 6)
- Strengths?
  - As much information as you could hope for
  - Resilient form of individual ACKs
- Weaknesses?
  - Could require sizable overhead in bad cases
  - Feasible if only small holes
    - If packets 1, 5, 6, …., 100 received: ACK(1, 5, 6, …,100)

Cumulative ACK

- ACK the highest sequence number for which all previous packets have been received
  - Implementations often send back “next expected packet”, but that’s just a detail
- Strengths?
  - Resilient to lost ACKs
- Weaknesses?
  - Confused by reordering
  - Incomplete information about which packets have arrived

Detecting Loss

- If packet times out, assume it is lost…
- How else can you detect loss?

Loss With Individual ACKs

- Assume that packet 5 is lost, but no others

- Stream of ACKs will be
    - 1
    - 2
    - 3
    - 4
    - 6
    - 7
    - 8
    - …

Loss With Individual ACKs

- Could resend packet when k “subsequent packets” are received
- Response to loss
    - Resend missing packet
    - Continue window based protocol

Loss With Full Information

- Same story, except that the “hole” is explicit in each ACK
- Stream of ACKs will be
    - Up to 1
    - Up to 2
    - Up to 3
    - Up to 4
    - Up to 4, plus 6
    - Up to 4, plus 6,7
    - Up to 4, plus 6,7,8
    - …

Loss With Full Information

- Could resend packet when k “subsequent packets” are received
- Response to loss
  - Resend missing packet
  - Continue window-based protocol

Loss With Cumulative ACKs

- Assume packet 5 is lost, but no others
- Stream of ACKs will be
    - 1
    - 2
    - 3
    - 4
    - 4 (Sent when packet 6 arrives)
    - 4 (Sent when packet 7 arrives)
    - 4 (Sent when packet 8 arrives)
    - …

Loss With Cumulative ACKs (cont'd)

- Duplicate ACKs are a sign of an isolated loss
  - The lack of ACK progress means 5 hasn't been delivered
  - Stream of duplicate ACKs means some packets are being delivered (one for each subsequent packet)
- Therefore could trigger resend upon receiving k duplicate ACKs
- But response to loss is trickier...

Loss With Cumulative ACKs (cont'd 2)

- Two choices
  - Send missing packet and optimistically assume that subsequent packets have arrived
    - i.e., increase W by the number of duplicate ACKs
  - Send missing packet, wait for ACK
- Timeout-detected losses also problematic
  - If packet 5 times out, packet 6 is about to timeout also
  - Do you resend both?
  - Do you resend 5 and wait?
  - …

Cumulative ACKs

- They make no sense, except as a cheap alternative to full information
    - Less state than full information
    - More resilient than individual ACKs
- But ambiguity in feedback leads to many problems
    - Have other packets arrived?
- Makes retransmission and congestion window management hard
- Will deal with these issues when we come to TCP

All The Bad Things Best Effort Can Do

- Packets can be lost
- Packets can be corrupted
- Packets can be **reordered**
- Packets can be **delayed**
- Packets can be **duplicated**

Effect of Reordering?

- For all designs this looks like "subsequent ACKs"
- This can be mistaken for packet loss
- Hard to realize the difference between these packet arrival patterns:
    - 1, 2, 3, 4, 6, 7, 8, 9,…
    - 1, 2, 3, 4, 6, 7, 8, 9, 5, 10,…

Effect of Long Delays?

- Possible timeouts (for all designs)

Effect of Duplication

- Produce duplicate ACKs
  - Could be confused for loss with cumulative ACKs
  - But duplication is rare…

Possible Design For Reliable Transport

- Full information ACKs
- Window based, with retransmissions after
  - Timeout
  - K subsequent ACKs
- This is correct, high-performant and high-utilization
- How about fairness?

Fairness? (Come back to later)

- Adjust W based on losses…
- In a way that flows receive same shares
- Short version:
  - Loss: cut W by 2
  - Successful receipt of window: W increased by 1

Overview of Reliable Transport

- Window based self control separate concerns
    - Size of W
    - Nature of feedback
    - Response to loss
- Can design each aspect relatively independently
- Can be correct, fair, high-performant and high-utilization

Many Implementation Choices

- Feedback from receiver: ACKs vs NACKs
  - Can NACKs alone achieve correctness
  - Can ACKs alone achieve correctness
- Variations on ACKs
  - Full information
  - Individual packets
  - Cumulative
- When to resend
  - Timeout
  - Duplicate ACKs
  - NACKs

Implementation Choices

- These implementation choices affect:
  - Performance
  - Utilization
  - Fairness
  - ..
- These are important concerns
  - **But correctness is more fundamental**
- Design **must** start with correctness
  - Can then “engineer” its performance with various hacks
  - These shacks can be “fun”, but don’t let them distract you

What Have We Done Today?

- Gone from first principles
    - Correctness condition for reliable transport
- … to design for single packets
- … to design for multiple packets
    - Very close to modern TCP
- … to radically different designs
    - Which could replace TCP
- All done by **you**, in 75 minutes

We tried to understand:

Why is TCP designed the way it is designed!

Why is almost always the most important question!!!