Layer Optimization: Congestion Control
CS 118
Computer Network Fundamentals
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We can lose packets for many reasons

- Corruption
- Not delivered to receiver
- Poor flow control receiver gets overloaded
- But also because of overall network conditions
- network congestion
  If there's too much traffic in the net, not all packets can be delivered
  - Can happen locally at one link or one part of network

Congestion control

network in the middle is too busy

- Receiver might be ready, but is the net?
  - Don't want to overwhelm the network
- We have some windows
  - Send = how much info *can* be outstanding
  - Recv = how much info *can* be reordered
- *Can* isn't the same as *should*

How much SHOULD be outstanding?

combined action of many different senders and receivers

A network problem

- Congestion control is not directly about the sender and receiver
- It’s about the network path they use
  - And share with others
- The shared paths can only handle so much traffic
- A given sender might send less
- But all the senders using the path in combination might overwhelm it
  - Perhaps just part of it

it's not just you or me causing it, it's a global effect

How to address the congestion control problem?

- A global problem, so perhaps a global solution?
- But who is in charge of the problem?
- And how does that party enforce its dictates?
- Instead, if everyone cooperates, maybe we can solve it without global control
- Everyone does his part to solve the problem, leading to a better global solution

But what can I do?

- You can only change your own behavior
- But if everyone does, that will reduce the congestion
- And life becomes better for everyone
- OK, so how do I change my behavior to help?
- And how much should I change it?

one can only control one's own traffic
sender: choose to send less data if there are congestion in the network

Recall the two windows

- Receiver window
  - Reorder out-of-order arrivals
  - Buffer messages until receiver catches up

    how reordering works

- Send window
  - Hold for possible retry until ACKed
  - Emulate how the channel delays/stores messages in a pipeline until ACKed

Send window maximum

- Round-trip to the receiver
  – "BW * delay" product
  – Really "fill the pipe until you get an ACK", presuming there isn't any loss

receiver is ready for more data; you can handle receiver's changes

- Once you fill the pipe, send at the rate you get ACKs
  – ACK clocking
  – Forces sender to pace to the receiver

TCP addresses congestion control at the transport level

TCP and congestion control

TCP is coorperative for congestion control

- TCP is one protocol that addresses congestion control
- Probably the most important congestion control factor in the Internet
- Essentially a cooperative approach
- When congestion occurs, all TCP senders slow down

coorporative to resolve this; we assume that we are going to do something and our combined effort to send things out; if everyone was to run; less traffic at congestion link

TCP's CWND

- Another window used by TCP
- Not the same as the send window
- Not intended to handle flow control
- Rather, to handle congestion control

cwnd - another window used by TCP;
handles congestion control

max segment size

TCP MSS and RTT

MSS - run some link leverl protocol - says biggest message is X bytes

- Two important parameters for TCP use
- MSS – Maximum Segment Size
  - Biggest TCP payload you can fit into one IP packet
  - By default, 536 “octets” (essentially bytes)
  - Find it by trial and error
- RTT – Round Trip Time round trip time
  - Time to send a TCP packet and receive an ACK

time stamp packet

this is important, so we know how much time it takes

Adjusting the congestion window

start congestion window at low value

- TCP CWND management
  - CWND is the send window max
    - Starts at 1, 4, 10K, or 10 packets
- Additive Increase
  - Until you see loss, increase CWND by a constant amount for every ACK

increase cwnd by a constant amount for every ack that you ned

- Multiplicative decrease
  - When you see loss, halve CWND

halve cns if you won'y need ti be the mroe additive increase / multiplicative decrease - feedback control algorithm for TCP congestion avoidance: linear growth of window, and xponential decrease of window

AIMD feedback

- A conservative approach
- Grow slowly by probing
- Backoff faster than you grow if there's signs of trouble

The slow start phase

- New TCP connection starts in a slow start phase
  - Until CWND reaches SSTRESH
    - A parameter of TCP
- CWND grows by 1 for each ACK
  - I.e., CWND doubles* each RTT

keep increasing window - sooner or later hit the slot start threshold
cwnd + 1 for each ACK

cwnd doubles; start with 10 packets; if there are no congestion; everything is okay, back comes first ack, all the way up to 10th ack; seder will get back 10 acks when he sends out 10 messages

for every ack he gets , increase his window

Why's that exponential?

- Sender sends out some number of packets N
  - Without waiting for an ACK
- If all goes well, N ACKs come back quickly
- You add one to CWND for each ACK
- So the next time, you send out 2*N packets
- And expect back 2*N ACKS
- In which case, you add 2*N to CWND
  - Getting 4*N
- That's exponential

slow start is done when you are done
ssthresh is for slow start

Why does it stop?

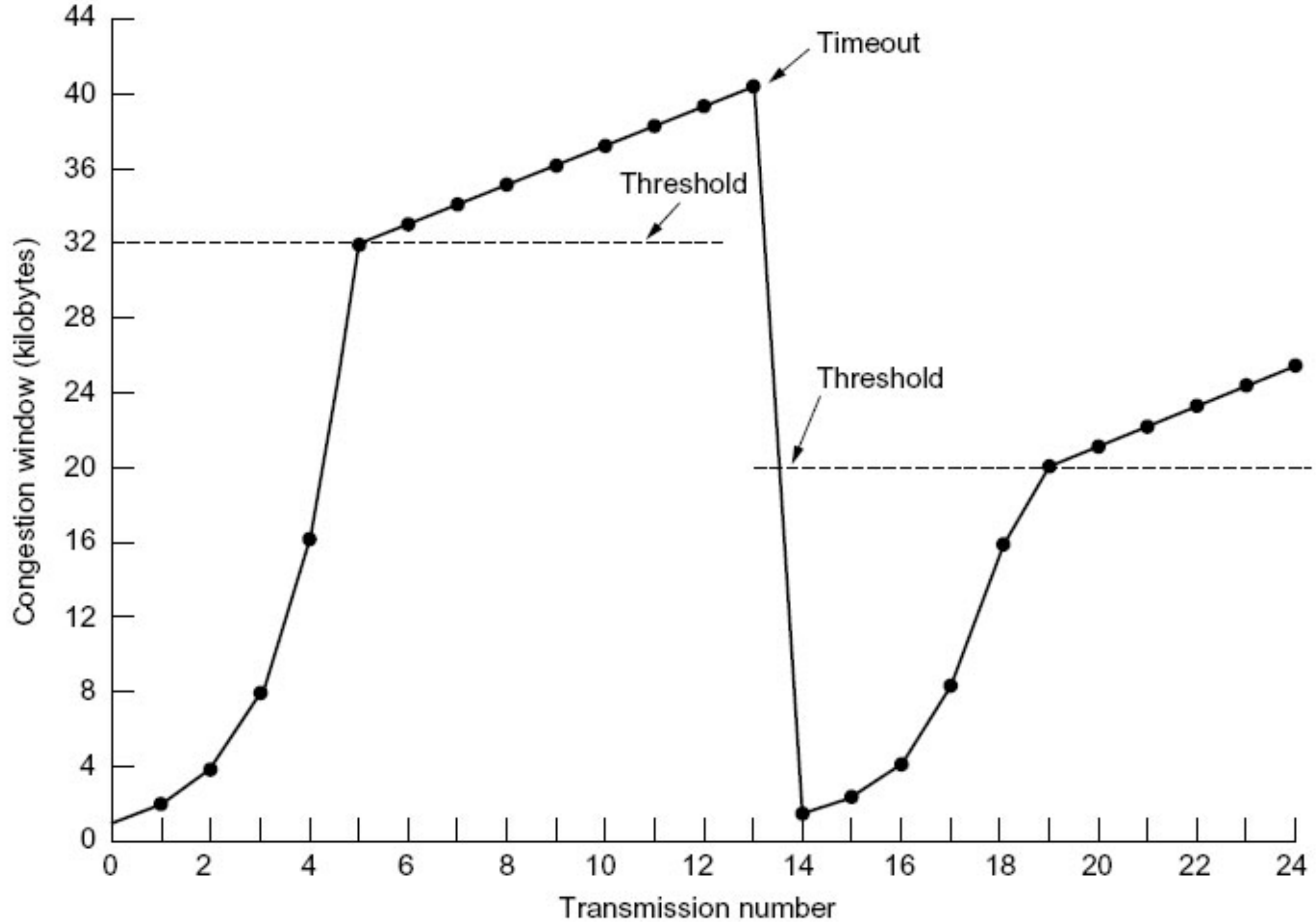
- Either you hit the limit to change TCP congestion control behavior
  – Your CWND reaches SSTHRESH
- Or you time out waiting for an ACK
  – Assuming that the packet is lost
  – Due to congestion
  – Will that assumption always be true . . . ?
- In latter case, also halve SSTHRESH
  – Depending on TCP variant

ssthresh - good for congestion control

Congestion avoidance phase

- Happens once SSTHRESH is reached
- Assumption is that there is no congestion so far
- Inch up a bit further to see if more can be sent
  - Until you reach MAX
- CWND grows by 1 for each RTT
  - NOT each ACK received

Visualization



Details

- CWND doesn't double per RTT in slow start
  – Because receiver doesn't ACK every segment
  – It ACKs every other ("ACK compression")
  – CWND increases by 50% each RTT in slow start
- This is one TCP variant
  – There are dozens, and they keep changing!

TCP's biggest assumption

measure roundtrip time - comparing time stamp needed

- TCP only knows:
  - What arrived
  - A timeout happened

- TCP measures:
  - RTT directly (timestamps)
    - Based on sent packets and ACKs
  - Max receive window (window)
  - ***Network congestion (via timeout!)***

What does a loss mean?

- Corruption
  - Should send more, i.e., send another copy
- Congestion
  - Should send less
- TCP assumes loss implies congestion
  - I.e., the more conservative interpretation

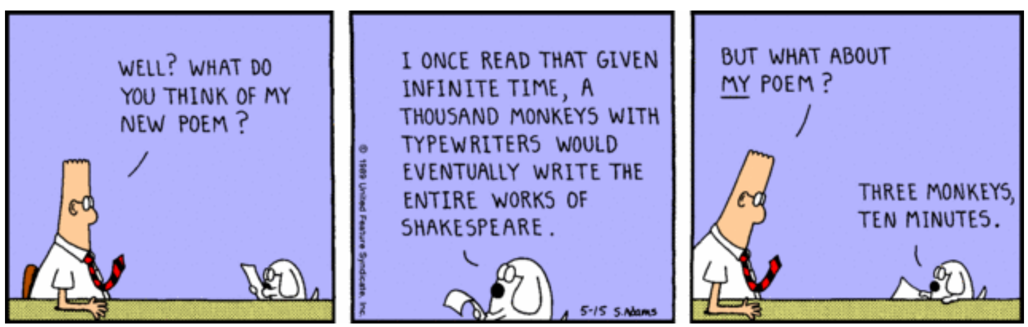
know the difference between corruption and congestion

Impact of loss=congestion

- TCP works poorly when corruption is high
  - I.e., wireless networks
  - When corruption is not due to load
- TCP is aggressive
  - It keeps sending more until something is lost
  - Two TCP flows always fight each other
- But TCP loses to cheaters
  - TCP backs off
  - Others might not

Congestion control algorithms

- Many of them
  - Lots of variations
  - Lots of incremental tweaks
  - Many based on fluid flow, feedback theory
  - Many based on whomever types it in…



Latency management

- Networks have buffers
  - Buffers adjust for bursts

tail drop: have a tail pointer that keeps track of the buffer

- Most networks "tail drop"
  - I.e., keep as many messages as the buffer can hold, and drop ones that arrive once full
- Tail drop favors keeping buffers full
  - Full buffers mean high delays

Solutions to latency management

- Explicit network congestion signals
  – Routers tell endpoints when buffers are filling
- Progressive loss
  – Drop probability increases as buffer grows
  – Don't just wait for "full" and drop all
  – "Random Early Drop" and variants

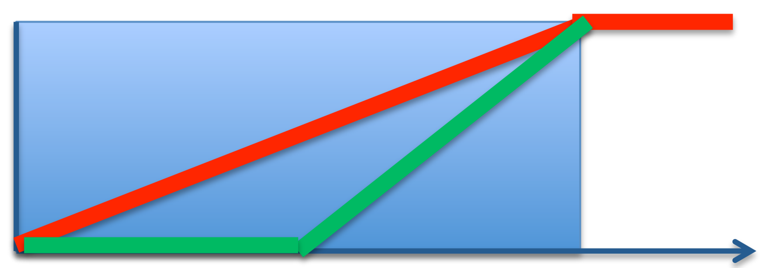
Explicit congestion notification (ECN)

- ECN routers (relays) indicate congestion
  - Mark instead of drop
  - Implies space to hold marked packets
  - So really more like "mark before drop"
  - E.g., mark packets arriving when queue is more than half full
- Endpoints react to ECN flags as if congestion was noticed
  - For TCP, ECN makes the CWND smaller
  - TCP can react to congestion without losing packets

What if ECN isn't available?

- Tail-drop queue
  - Do not drop if there's room
  - Drop if queue is full

- Random Early Detection
  - Drop probability increases as queue grows
  - Various curves



Better buffering

- Relays can cause problems
  - Connections compete one packet at a time
  - Maybe separate buffering by connections is better
    - "Fair queuing"
  - Need better use of buffers
    - Memory is cheap, but has a cost

Space

- Compression
- Caching

Compression

lossless compression: compress send receive decompress
you end up with the same thing as you started with

- Translate a set of long messages into a set of short ones
  - Take a set of messages
  - Represent frequent ones with fewer bits, longer ones with more bits

- Translate a long message into a short one
  - Take a set of groups of symbols in a message
  - Represent frequent groups with fewer bits, longer ones with more bits

Compression examples

- Web traffic
- E-mail
- TCP/IP headers

Web traffic

- HTTP 1.1
  - Compress content of responses
  - E.g., zip images, large text areas
  - Inside Google Chrome browser
- HTTP 2.0
  - Compress headers

E-mail

- By the program
  - Postscript, Word
- By the user in advance
  - Zip folders
- By the email system
  - Compress attachments

TCP/IP headers

- Compress the TCP and IP headers
  - 40 bytes down to 16
  - Most of the header is predictable within a single connection
- Typical for PPP and SLIP (dial-up lines)
  - I.e., over path that doesn't examine the header

TCP/IP compression

- When is it useful?
  - What benefit?
    - For 40B ACK packets, saves 60%
    - For 512B payload data, saves 4%
    - For 1500B segments (Ethernet), saves 1.6%
  - Where useful?
    - ACK-only, BW-limited returns
    - For 2400bps modems (1990), saves 87ms

Required compression information

- Patterns and frequencies of those patterns
  - Usually from a set of previous messages
    - E.g., Morse code
  - Or from previous use on this channel
    - E.g., LZW, used in GIFs
  - Or just obvious patterns
    - Run-length encoding, used for faxes and JPEG

Compression trade-offs

- Trade (consume)
  - Effort
    - CPU works harder
  - Energy
    - CPU burns power
  - Time
    - Encode/decode needs to delay the stream
    - Encode/decode operation takes time
- Gain (produce)
  - Space
    - Smaller message takes up less memory
  - Capacity
    - Smaller message uses less bandwidth
  - Time
    - Smaller message takes less time to transfer

Compression caveats

- Works once
  - Compression removes patterns
  - Works at only ONE layer or over ONE hop
- Obscures information
  - Can’t modify or easily read until undone
  - Uncompress/recompress is expensive
- Small returns if used on only part of large messages
  - HTTP/2 header compression is controversial

Caching

- Save via reuse
  - Over time within one stream
    - If you have the answer from before, use it again
  - Across a set of streams
    - Don’t ask if your friends know the answer

Caching examples

- Inside a protocol
  - TCP control block sharing, TCP/IP compression
- Content
  - ARP, DNS, Web

TCP control block sharing

- New connections start from “zero”
  - Why?
- New connections can reuse
  - From past (reuse CWND, RTT, MSS)
  - From peers (reuse RTT, MSS, split CWND)

Why reuse?

- Change is unlikely
  - Path (routing) tends to be stable
  - Endpoints tend to be stable
  - Aggregate traffic patterns tend to be stable
  - So RTT, MSS tend to be stable
- Why infer when you can share?
  - Endpoints within the same machine can share
  - No need to have CWNDs fight and balance; can just "split at start"

Net effect of TCP sharing

- Less blind probing
  - No need to send large segments to find MSS
  - No need to use RTT over-estimates
- No need to compete via loss
  - Shared info can “rebalance” CWND
- Safe
  - Tries to anticipate transients – only at connection start/end
  - Tries to jump closer to convergence, then lets existing feedback take over

More complex sharing

- Endpoints within a LAN
  - Can share their experience
  - Can explicitly coordinate rather than compete
- Inherently harder
  - No longer just sharing information on a single computer
  - Which means it must be communicated

Information delineation

- Boundaries
- Flows

Boundaries

- Message vs. packet alternatives
  - *Span*: messages longer than a packet

    data data   throw the data into the network

  - *Preserve*: message matches packet
  - *Pack*: packet carries multiple messages
  - *None*: no boundary support (e.g., TCP)

Adding markers is easy…

- Length indicator
  - E-mail attachments, IP packets, HTTP chunks
  - Efficient (rapid jump), but fixed max
- Special symbols ("escape" sequences)
  - Not used for data
  - Arbitrary chunk size, but need to scan

look in packet and say where the packets are

special bit pattern that indicates the start of next message

Deciding marker use is hard

- Costs
  - Gathering small chunks can cause delays
  - Picking the wrong size increases overheads
  - Cost to split/merge or merge/split

    additional cost for packet split or merge
- Risks
  - Lack of fate-sharing
    - Different chunks via different paths

      risk: multiple packets take different paths to destination
      they have different fates

Marker examples

- Length

  attachements are created as a single message altogether

  - *Pack*: HTTP, e-mail, SCTP
  - *Preserve*: UDP, DCCP
  - *Span*: ATM AAL5, IP frag., multipart MIME
- Special symbols ("escape" sequences)
  - ATM, Ethernet preamble

Flows

- Like a channel…
  - Information shared between parties
- …with multiple viewpoints simultaneously
  - One channel
  - Several separate channels

Examples of multiple flows

- Multiplexing
- Striping (inverse multiplexing)
- Partitioning

Multiplexing

- Using one flow to emulate many
  - HTTP chunking and muxing
  - Allows one TCP connection to support concurrent web transfers
- Hazards
  - “Fair sharing”
  - Head-of-line blocking

Fair-sharing

multiplex multiple flows into one

- Merging multiple flows onto one
  - Who goes next? who can have bytes put into packet?

whoever has smallest piece of info
largest piece of info
proportional, round robin

- Various strategies
  - Shortest-first, largest-first, round-robin, proportional
- How is "fair" defined? define fair is important
  - Each according to their needs?
  - Each gets the same?
- How is "each" defined?
  - Per human? Per endpoint? Per application?

Head-of-line blocking

- Consider lines at a market
  – Large basket arrives before 2-item
  – BLOCKS system

starvation

- Avoiding HOL blocking?
  – Limit chunksize
    - E.g., everyone pays 10 items at a time
    - Leaves when done paying for entire basket
  – Use separate connections
    - E.g., multiple TCP connections for web clients

everyone has different lines, so less blocking
if one line is blocked, maybe the other can proceed

Striping

- Making multiple channels appear as one
  - Increased bandwidth
  - Increased reliability
- Examples
  - Multipath TCP
  - SCTP
  - Various datacenter optimizations

Partitioning

- Split one info stream into separate ones
  - To avoid HOL blocking
  - To manage differently (loss vs. recovery)

make audio get better path
we don't want to drop random bits and pieces of both

- Examples
  - Teleconference audio vs. video
  - FTP control vs. content

teleconference - video and voice
bundle together all bits for video, all bits for voice
put in one flow - video is much more important than audio or vice versa
split them up - send video down one stream, audio down another

Translation

- Formats
- Conversion
- Marshalling

Recall encodings

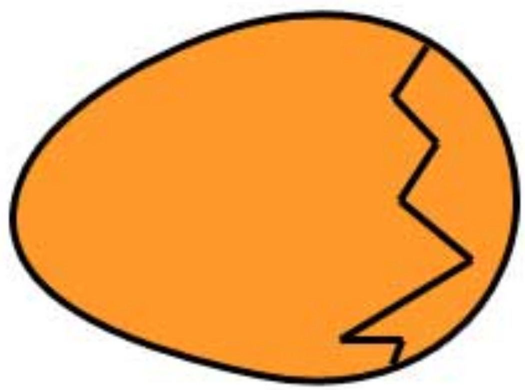
- Represent information with symbols
  - Various strategies
  - Earlier lectures focused on physical, error
- More encoding issues
  - More encoding variants
  - Coordinating the endpoints

Bit order and formats

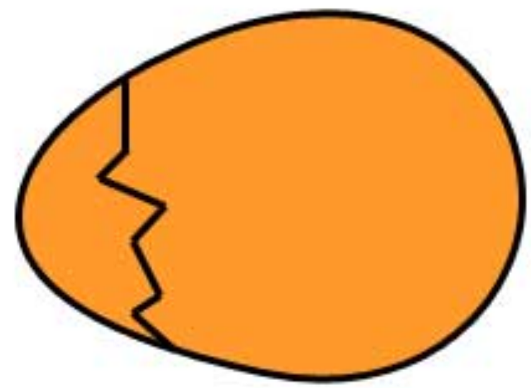
- Many channels exchange bit sequences
  - Upper layers exchange bytes, words, etc.
  - What order?
- LSB vs. MSB
  - LSB-first: enables serial arithmetic
    - Ethernet, Token bus
  - MSB-first:
    - Token ring

On holy wars and plea for peace

- Gulliver's Travels



BIG ENDIAN - The way people always broke their eggs in the Lilliput land



LITTLE ENDIAN - The way the king then ordered the people to break their eggs

Endianess

- Big-endian: ABCD stored as A, B, C, D
  - The Internet
  - Motorola 68000, RISC (PowerPC, SPARC)
  - Telephone numbers
- Little-endian: ABCD stored as D, C, B, A
  - Intel and AMD processors
- Both (configurable)
  - ARM

Conversion

- Host to net, net to host
  - Long, short, etc.
  - Converts from Internet (big-endian) to local

Marshalling

- Packing and unpacking
  - Format conversion
  - Sequencing
  - Labeling
- All for what?
  - Same as for a function call
  - A way to know the meaning of shared bits

Why is marshalling hard?

- Expensive
  - Conversion takes time
- Tedious
  - Many steps to mess up
- Exacting
  - All the steps have to match to work

Summary

- Lots more optimizations and features
  – The details depend on the implementation
- Details matter and they don't
  – Parties must agree on details to communicate
  – Detail differences affect performance
  – But particulars of details not always otherwise critical
  – Things can be done many ways